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April 28, 1989

Dear Industry Engineer:

The theme of this year's NAB Convention is "On-the-Air: Proud Tradition, Dynamic Future." Could there be a more appropriate theme for engineers? We are proud that our radio and television stations stay on the air. And who can doubt that our future is dynamic?

Studying these technical papers will help keep radio and television stations "On-the-Air" and, at the same time, prepare you for the many industry changes to come. These Proceedings contain plenty of helpful and important technical information. We are living in an exciting time.

My best wishes for a productive and enjoyable 1989 NAB Engineering Conference. On behalf of NAB's Science & Technology department staff, I am pleased to present these 1989 Engineering Conference Proceedings.

Best regards,

Michael C. Rau

TABLE OF CONTENTS

BROADCAST AUXILIARY & SATELLITE SYSTEMS

SAFETY, INTERFERENCE AND ENVIRONMENTAL CONCERNS

NEW COMMUNICATIONS TECHNOLOGY

DIGITAL AUDIO INTERCONNECTION

Skip Pizzi
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ABSTRACT

As more of a broadcast station's audio chain is converted to digital audio equipment, less remains analog, and eventually only the wiring between the hardware is left in the analog domain. At that point, the limiting factor to quality becomes the interconnection itself, and the digital-to-analog and analog-to-digital conversions at each device's outputs and inputs can unnecessarily degrade quality. Therefore a standardized form of interconnection that maintains audio signals in the digital domain is now desirable, and will soon be an absolute requirement.

THE CHANGING AUDIO CHAIN

Figure 1 shows the basic block diagram of a typical radio station. The shaded blocks indicate where digital audio has become a significant factor so far. Note that no two contiguous blocks have as yet "gone digital", but once that happens, it would be unsound engineering to continue to interconnect these via analog paths. Until such time, the full advantages of digital audio will not be realized.

Digital Advantages

In addition to the high audio quality that digital audio provides in the amplitude, frequency and time domains, keeping signals in their digital form engenders further benefits. These include the ability to manipulate audio with minimal degradation, relative immunity from EMI radiation, the inclusion of certain control parameters embedded in the digital audio datastream, the potential reduction in the number of cables required, and eventually some advantages relating to high-speed upload and download of audio programming, or application of global processing to a "sound file" in less-than-realtime.

New Studio Wiring

The former trinity of analog audio studio wiring needs -- AC power, audio signal and remote control -- are replaced by a longer list in the digital studio.

AC power of course still remains, but with heavier power handling capability and increased surge protection required. Proper studio grounding is even more essential, since audio noise floors which may have formerly masked minor faults will now be lower and thereby more revealing. The results of any grounding faults may also be more devastating than just a bit of hum or buzz, namely total audio dropouts or circuit damage from inadequately drained static charges, etc.

Analog audio lines will also remain necessary for some time to come, but again grounding schemes must be carefully implemented, for the same reasons as with AC power. RF shielding is increasingly important for these lines as well, to protect analog signals from the often intense RF fields generated by some digital equipment. The use of "star quad"-type cable is highly recommended.

Digital audio lines will also be required, of course. Depending on the format used, wiring may be of the same type used for analog paths, or coaxial cable may be required (more on this below). Future systems may even include some limited-length multi-pair or ribbon cable for parallel busing. Even if the cable type is identical to the that used for analog wiring, these circuits will need to be kept separate, for reasons of channel formatting, connectors required, and patchbay termination differences, along with concern for capacitive or inductive coupling of radiated RF fields between the two formats.

Digital audio **optical** paths may also be useful in the near future, especially for facility-to-facility interconnection. Optical cabling is of little value for busing applications, due to losses incurred at each junction; for permanently installed longer-length serial interconnections, however, they are ideal. Their wideband capability, low loss and total immunity to

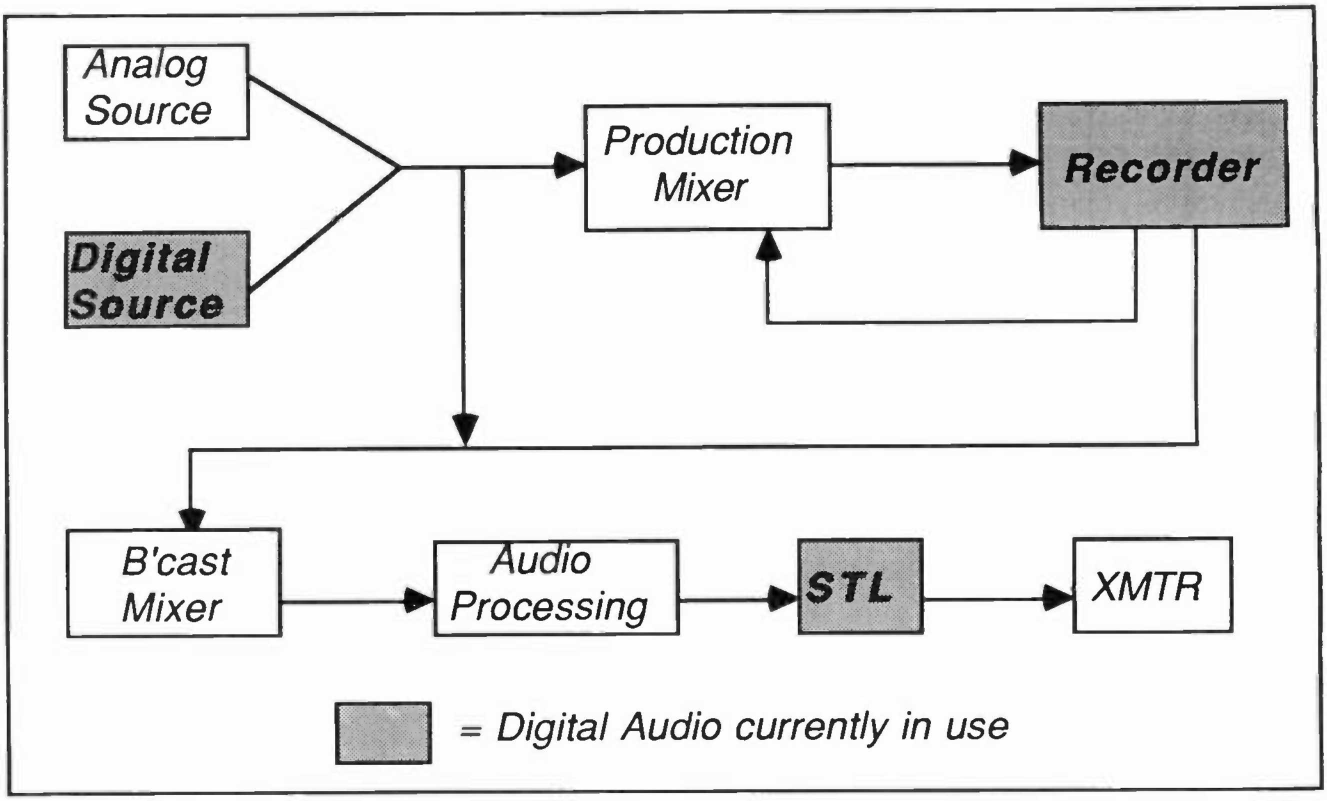

Figure 1: *Typical broadcast station audio flow.*

ground loops and EMI/RFI generation or reception make them a welcome addition to any facility.

Control lines in the digital studio take on a higher level of meaning than simple machine remote control of former times. Included in the new control milieu are machine diagnostics and setup, automation of hardware and software functions, and various forms of synchronization, along with traditional machine control. RS-232 is the typical protocol used for these lines. Studio-to-maintenance shop paths should be installed in this category, to permit remote assessment of hardware operation from the bench when feasible.

MIDI (Musical Instrument Digital Interface) lines are also recommended for production studios, allowing this popular control format to be used with audio processing gear, samplers, sequencers, synthesizers and other analog and digital equipment.

Interconnection Standards

Figure 2 shows a list of current digital audio interconnection formats, both in use and proposed. Since most are proprietary to a particular manufacturer, they are not of universal value. The notable exceptions to this are the AES/EBU (Audio Engineering Society/European Broadcast Union) formats, and the MADI format.

AES/EBU professional is a modified RS-422 protocol, using standard audio cabling terminated on standard 3-pin "XLR" connectors. The format is a balanced, self-synchronizing, two-channel standard, thus allowing professional digital **stereo** audio signals to be routed on a **single**, familiar cable and connector. The AES/EBU consumer format is essentially compatible, but it uses an unbalanced configuration on RCA-type phono plugs. Adapting one to the other can be accomplished by simply wiring XLR-to-phono with pin 2 hot, pins 1 and 3 to ground.

Fortunately, the AES/EBU pro format seems to have established itself as the de facto standard in the industry. Most current professional digital recording equipment is equipped with this DIO format (some CD players as well), along with a manufacturer's format. Among the latter, the Sony SDIF-2 is perhaps the most used, having recently appeared on another manufacturer's digital reverberator, as well as all Sony digital recording

Format	Connectors	Comments
• Sony PCM F-1	----	"internal" only/modifications
• Sony PCM 1610/1630	DB-25 & others	CD editing, mastering
• Sony PCM 3324	Sony 50-pin	balanced version of 1610
• Sony SDIF-2	1-BNC/ch.+ word clock	TTL compat., on coax
• S/PDIF	RCA phono*	consumer equipment
• Melco	DB-25 & others	32 ch. balanced, similar to 3324
• AES/EBU professional	XLR-3*	"pro standard", modified RS-422
• AES/EBU consumer	RCA phono*	consumer, unbalanced
• MADI	BNC	up to 56 ch. on single coax or fiber
• Parallel proposals	var.	cable length limits; new "smart" buses
• Optical proposals	var.	wideband, for longer paths (mux parallel to serial)

*Stereo audio on single connector.

Figure 2: *Current and proposed digital audio interconnection standards.*

products. This is a TTL-compatible format, requiring a separate coaxial cable for each channel, and a third for synchronization, terminated on BNC connectors, so it is not nearly so advantageous as the AES/EBU approach. Other manufacturer formats are typically limited to one piece or class of hardware, as noted in figure 2, although the Sony 1610/1630 format is the industry standard for CD mastering equipment.

MADI is a recently proposed multichannel incarnation of the AES/EBU pro format, allowing up to 56 channels of high-resolution digital audio to be carried on a single coaxial cable or fiber link. The coax application is limited to 150 ft. in length. An external synchronization line is required. Several major manufacturers have endorsed this standard, and it is expected to begin showing up on multitrack recorders and mixing consoles soon.

LONGER PATHS

STL, TSL, and RPU pathways have often been problematic for broadcasters in terms of audio quality and reliability. Some in the industry have taken ad hoc steps to adapt and implement existing digital recording formats to appropriate RF hardware or telco lines, in an attempt to gain digital audio's improvements in this area. Others have developed (or are developing) purpose-built systems for this application. Figure 3 summarizes the industry's current status here.

Note that the general classes of digital links in use are either "pseudovideo" (Sony PCM F-1 or dbx 700 formats) or T-1, the telecommunication industry's 1.544 Mb/s format, both of which require wideband RF channels. This necessitates the use of 18 or 23 GHz microwave bands for the links, increasing their cost and maintenance needs dramatically over standard aural STL equipment. Note also that the pseudovideo and VAMP T-1 (from Graham-Patten Systems, Inc.) formats use about 5 MHz or more for just two one-way audio channels, but telco's T-1 can accommodate more channels of flexible bandwidths, bidirectionally. Resolution and sampling frequencies for high-quality program channels on telco's T-1 are not as high as on the others at present, however, and costs of the non-RF segments are generally higher for the telco-type gear.

The near future holds great promise in this regard, since work on bit-rate reduction with sensitivity to human aural perception limits is currently underway at

Digital System	Processor Cost[1]	Audio Channels	RF Hardware Requirement	Occupied RF Bandwidth (FM)	RF costs[5]	Telco Requirement[9]	Comments
dbx 700	$ 9000[2]	2	23 GHz STL	4 to 10 MHz	$ 12,000+[6]	video ckt.	A
Sony F-1 *	3500[2]	2	23 GHz STL	4 to 10 MHz	12,000+[6]	video ckt.	A
Graham-Patten VAMP-2	5000	2	video STL subcarrier	1 MHz	——[7]	subcarrier of video ckt.	B, C
Graham-Patten VAMP-3	5000	2	23 GHz STL	5 MHz	10,000+[6]	T-1 ckt.	B, D
Telco T-1 rack	7 to 10,000	var.	23 GHz STL	5 MHz+	15,000+[8]	T-1 ckt.	B, E
Dolby Model 500 Series	4000[3]	2 + 1 aux	950 MHz STL	250 KHz or less[4]	<10,000	384 Kb/s data ckt.	B, F

COMMENTS:

A. Approximately 10 ms delay of these systems can bother announcers monitoring air signal while speaking.
B. Insignificant delay through system.
C. Easy and cheap if video STL is already in place.
D. Cost-effective, *one-way* T-1 system.
E. "Soft" configuration, allowing easy expansion and revisions.
F. Very spectrum efficient; could use aural STL frequencies.

NOTES:

1) Cost for both ends. Total system prices include encoder and decoder, but no spares. Some systems use differently priced hardware at each end. All prices approximate.
2) Availability on these units is low at present. Only current model in Sony format is PCM-601ES; requires outboard interface for pro-audio operation (+4 dBm/balanced), included in price shown.
3) Tentative projected price. Availability date is unknown.
4) Using QPSK or other non-FM process.
5) Total system cost for both ends, including typical antennas, but excluding installation and tower construction costs. Assumes single hop.
6) Can include up to three 15 KHz analog audio subcarriers.
7) Assuming video STL already exists, and 1.0 MHz-wide subcarrier is available, no additional RF costs are incurred with this system. Subcarrier generator included in unit.
8) Duplex STL required for full use of bidirectional T-1 system. Can be used one-way for approx. $5,000 less. Processor costs may also decrease somewhat for one-way operation.
9) Corresponding costs vary. Check with your local Telco for rates.

Figure 3: *An overview of current and proposed digital audio STL systems.*

Dolby Laboratories and elsewhere. The high audio quality and robustness with reduced data rates that may result from this research can be coupled with more efficient forms of modulation than the current FM used on most STL's, yielding a digital STL on 250 kHz or less of bandwidth, in which case traditional aural STL allocations could be employed. (Other digital technologies are making these higher-order forms of modulation more cost-effective as well. See reference 4.)

Another difficulty with the pseudo-video systems (and some others) is their use of RAM buffers which introduce 10 ms or more of delay through the system. Announcers monitoring off-air can be disturbed by this while they are speaking. Research conducted by the BBC shows that as little as 7 ms of delay can be disconcerting, even to highly experienced talent. Engineers at the BBC have therefore installed relays to switch headphone monitor selectors automatically to program audio whenever the announcer's mic key is activated in facilities using such STL's.

OTHER ISSUES

In its eventual full implementation, a digital audio system will provide mixing capability in the digital domain. All digital sources fed to such a mixer will of course need to be of the same or compatible formats, and synchronized to a common reference, just as video systems are today. Outboard format and sampling frequency converters are available today as a sort of niche market, where they are used to convert one digital **recording** format to another without resorting to use of an analog path between the two decks. Such devices will be no such luxury when digital mixing of varied sources becomes a reality, but rather an absolute requirement. Implementation for a total "in-house" production will not be difficult, but what of the radio broadcast that includes both in-house and live remote elements, all to be maintained in digital? And what of master control switchers' ability to handle a variety of formats? Obviously, versatile and format-agile devices will be required for the all-digital broadcast facility, relegating such conversions to simple and routine tasks. Again, the BBC is out in front on this, currently developing synchronizers capable of combining local and remote elements in the digital domain.

Finally, the coming of the digital workstation simplifies the issue of digital interconnection a bit, since by its nature, it combines several of the items shown as separate in figure 1 above, thus obviating the need for their external interfacing. The workstation's integration also provides short path lengths between its internal or peripheral elements, such that the new generation of fast and "smart" data buses can be used.

Conclusion

Although much of the above may seem a long way off or not pertinent to the broadcaster, it behooves us all to become conversant with this material as it develops; we thereby gain the familiarization required to be comfortable with such divergent technology when it does actually arrive in our facilities. This may be sooner rather than later, since competitiveness and increased cost-effectiveness will often provide accelerated motivation to the broadcaster, and many digital techniques fit those criteria nicely.

Watch the diagram shown in figure 1 change with time. As soon as two consecutive blocks become predominantly digital, the floodgates to the world of digital interconnection will open. It is your challenge to be ready.

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TESTING DIGITAL AUDIO DEVICES IN THE DIGITAL DOMAIN

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A measurement system is described which allows testing of digital audio devices in the digital domain, analog domain or a combination of the two domains. By generating and analyzing the test signals digitally, the need for a reference standard A/D or D/A converter is eliminated. By combining the digital generation and analysis hardware with analog generation and analysis hardware the system is capable of measuring A/D converters, D/A converters or conversion sections of complete products. The entire system is controlled by a personal computer, allowing fast and thorough testing for engineering or production applications. The techniques employed and the hardware used to implement the system will be described. Measurements of commercial A/D and D/A converters will be used to illustrate the measurement techniques.

Introduction

Digital audio is a rapidly growing field with substantial engineering and development effort spent all around the world. All aspects of digital audio systems, A/D conversion, D/A conversion, digital signal processing, etc. are being improved every day. As with other engineering efforts, there is a constant need for performance assessment to determine relative merits of any design being considered (Cabot 1988). Many parameters are measured. These include amplitude-frequency response, phase-frequency response, gain or loss, harmonic distortion, intermodulation distortion, signal to noise ratio, alias rejection, etc.

The equipment being tested spans a broad spectrum, including digital tape recorders, mixing consoles, disc players, delay units, limiters, reverberators, equalizers and digital communications links. These have been built using analog technology since the beginnings of the audio business. Now all are available in digital implementations. Since they perform functions previously available in analog form they may be tested as black box replacements for the corresponding analog device. This will work fine if the digital version has analog inputs and outputs and the purpose of the testing is equipment acceptance. However if the device only interfaces in the digital domain, or if the purpose of testing is to provide more detailed information for design feedback, this approach will not be adequate.

The Audio Precision System One provides all commonly used analog audio test and measurement functions. The new DSP-1 module described here expands these functions into the digital audio domain, as well as enhancing the operation of the system for analog measurements. The digital signal processing can be used in conjunction with the System One analog notch filter to measure analog signals to a much wider dynamic range. For example harmonics may be measured individually, without any noise floor limitation.

Conventional Approach to Digital Audio Measurements

The conventional approach to characterizing digital audio devices uses an A/D and/or D/A to translate signals between the analog and digital domains. Measurements can then be made with existing analog test equipment and correlated with measurements on competitive analog devices. This technique requires, or takes on faith, that the conversion system used to interface to the device under test is of higher performance. When building cost sensitive, or lower performance, digital equipment this may be the case. When building professional grade equipment the designer generally doesn't have any higher performance conversion systems available to translate between domains.

Digital gain can be introduced to help separate the effects of A/D and D/A conversion. Digital gain is achieved by shifting the bits of the digital word to the left or right by N bits. This produces a gain or loss of 6N dB respectively.

Digital gain emphasizes low level nonlinearities in A/D conversion since small signals from the A/D are reproduced using the full dynamic range of the D/A. Nonlinearities in the conversion are assumed to be near the zero crossing of the transfer characteristics, making those of the A/D a larger percentage of the signal than those of the D/A. If this assumption is not valid for the D/A the measurements will be corrupted, and the A/D will appear to be worse than it actually is. If this assumption is violated by the A/D (its worst nonlinearities are at high levels), this low level testing will make the A/D look better than it actually is.

Digital attenuation emphasizes nonlinearities in D/A conversion since signals are converted into the digital domain with the full dynamic range of the A/D. The most significant bits of the A/D output are fed to the lower bits of the D/A under test, while the D/A's upper bits are zeroed. Again, the nonlinearities are assumed to be near zero on both converters, making those of the D/A a larger percentage of the signal than those of the A/D. If this assumption is not valid for the A/D the digital data will be corrupted, and the D/A will appear to be worse than it actually is. If this assumption is violated by the D/A (its worst nonlinearities are at high levels), this low level testing will make it look better than it actually is.

Any application of digital gain or attenuation in the system under test will limit the dynamic range of the measurements. Gain after the A/D converter of X dB prevents the input signal from exceeding X dB below full scale or the digital word will wrap around. Similarly, attenuation after the A/D of X dB prevents testing the D/A within the top X dB of its range. Since noise floors of the converters remain unchanged with changes in the digital gain, the residual floor of distortion measurements will be reduced by X dB in either case.

Gain and offset stability problems anywhere in the system can prevent accurate measurements at the lsb level. A typical 16 bit digital audio system with a 2V full scale requires a DC stability of approximately 43 microvolts to allow signals to be placed near zero within 1/2 lsb.

The anti-alias and reconstruction filters each introduce response irregularities which may mask the response of the other. The errors in both amplitude and phase response may be subtracted out if they are repeatable enough. It is only necessary to characterize the response of the filters individually using the analog test equipment and use this data to correct the measurements on the complete system. If single or multiple sinewave test signals are being used, where alias products are not a concern, the A/D may be operated without an anti-alias filter. However a reconstruction filter is always needed. In addition to the linear errors these filters also introduce nonlinearities and noise. These may not be compensated or subtracted and must be eliminated in their design. Fortunately, since they may be built completely analog, their measurement and performance verification is straightforward.

The conventional approach also suffers from other inaccuracies of conversion; sample and hold problems, de-glitcher problems, clock jitter, etc. A detailed description of these problems is beyond the scope of this discussion. If the appropriate measurements could be made entirely in the digital domain these problems could be avoided. One of the goals of this project was therefore to develop hardware which could perform all of the standard audio measurements on digital signals without ever processing them in the analog domain.

Another problem is that some parameters of interest to digital designers are difficult to measure in analog domain. These include the standard converter measures of differential nonlinearity, integral linearity and missing codes. They also include digital data processing measurements such as error rate, error correction efficiency and interpolation accuracy.

Typical digital audio device

Figure 1 is the block diagram of a typical digital audio device. Though not universal, and certainly not detailed, it will prove useful for describing the interfacing and measuring considerations with digital equipment. The real world signal at the input terminals is converted to a ground referenced, unbalanced signal. This is adjusted for the desired amplitude before being applied to the A/D conversion system. An anti-alias filter prevents out of band energy from entering the converter and producing incorrectly digitized signals. The continuously changing analog signal is held stationary by the sample and hold for later conversion by the A/D. The A/D converts each sample of the analog signal to a binary code. After appropriate storage and/or processing, the digital signal is output. This digital words are converted back to analog samples by the D/A converter. The de-glitcher prevents changes in output between samples, insuring an orderly transition from sample to sample. The sampled analog signal is then smoothed and band limited by the reconstruction filter. An output balancing amplifier interfaces the signal to the outside world for connection to other equipment.

Another path exists for signals which enter and exit the device in the digital domain. The digital signal is received by a format conversion circuit. This may be a standardized interface such as AES-EBU or MADI, or it may be a nonstandard interface such as SDIF2. This circuit must extract the audio data and relevant control bits. These control bits may include parity checking, sample validity, emphasis, sample rate, word size, time code, labeling, and other information necessary for later use by the processing and re-transmission. Errors may be corrected or concealed by a correction and interpolation circuit. After processing, the digital signal is reformatted for transmission in the desired digital format. The transmitted signal has the necessary control and labeling bits added at this point to the data bits.

As can be seen from the previous discussion, the signal assumes many formats during its journey through our hypothetical device. As examples, the input to and output from the system are balanced analog signals. The signal in the filters is unbalanced analog. The A/D converter typically outputs a serial data stream on clock, data and strobe lines in a non-standard format. Most digital signal processors (DSP's) operate on a parallel digital word access to memory and peripherals. The external digital inputs and outputs are one of the standards mentioned earlier. To properly test the design of each circuit and its operation requires the ability to interface to any of these formats, analog or digital.

Digital Formats

The most common external I/O formats in digital audio systems are the AES-EBU interface and the Sony SDIF2 interface. The AES-EBU interface is a two channel format where the words are channel sequential. The data line carries samples from alternate channels, first channel A then channel B. The data is sent LSB first, word width is selectable as 20 or 24 bit. The data is followed by 4 flag and status bits and preceded by a 4 bit preamble. This totals 32 bits for each channel of audio data. The entire 32 bit word is bi-phase encoded to guarantee a transition at the boundary of every bit location. The signal is output in a balanced form to improve noise rejection. Since the signal has been bi-phase encoded polarity is not important, eliminating problems from accidental interchange of the conductors. This interface has been widely accepted within the audio community and there are a growing number of professional audio devices incorporating it. The consumer SPDIF interface found on CD players is essentially the same except for some minor changes in the channel status bits and the balanced versus unbalanced characteristic of the interface.

The other interface that is common in professional applications is called SDIF2. It is a format developed by Sony for use on their PCM 1610's and 1630's. It is a two channel, word parallel format where left and right are transmitted simultaneously on separate cables instead of sequentially on the same cable. The interface requires a third cable for word clock information which is used by the receiver to lock its sample rate to the transmitted word clock. The individual channels carry a word sync post-amble to allow synchronization of bit clocks and data words by the receiver. This has wide following in CD mastering applications because of the large number of PCM1610's which have been sold. It has never been standardized and has no published specification from which to design. This has caused many designers to make invalid assumptions about timing and created many incompatible versions of the interface. Fortunately it is fading from prominence, unfortunately not fast enough.

MADI is a multi-channel version of the AES-EBU interface which runs at an extremely high data rate. It carries up to 56 channels of audio at a fixed 100 MHz data rate. Data is sent over an unbalanced coaxial cable with sync information conveyed by a dedicated sync signal, house sync, or an AES-EBU interface. It is a proposed standard for professional use within the studio environment or other areas in which many channels must be sent over short distances. Several manufacturers have announced products which follow the proposed standard and it can be expected to grow over the next few years.

Digital signal processor (DSP) integrated circuits are inherently parallel data devices. Some A/D and D/A converters also fall into this category. The audio samples are output or input as a individual data lines for each bit of the word. The word width is device dependent, but all are 2's complement binary. The currently available devices are either 16 bit, 24 bit, 16/6 floating point or 24/8 floating point. For the floating point widths the first number is mantissa and the second number is the exponent. When testing DSP algorithms for equalizers, compressors, limiters, etc. a parallel format is necessary to connect directly to the processor.

Most A/D and D/A converters manufactured today use serial data formats for interfacing. When testing converters or devices which communicate with converters a three line serial format is typically needed. The exact format is device dependent but all incorporate clock, data and strobe lines. The main differences are in frame size and strobe timing relative to the data word. All commercial parts used for audio have frame sizes of 16 or 32 bits. If a smaller number of bits are needed than one of these sizes the remaining bits are typically zero filled. The strobe is usually asserted at either the beginning of data (same as beginning of frame), end of data, or end of frame.

System Block Diagram

The Audio Precision DSP-1, diagrammed in Figure 2, provides a DSP core with both analog and digital input and output capabilities. The system is built around three digital signal processors, one for the DSP core functions, one for the analog interface and one for the digital interface. The DSP core and the analog input and output circuitry are on one circuit board. The digital input and output interfaces are on a optional second circuit board.

All DSP chips in the system are Motorola DSP56001 24 bit digital signal processors operating at 24.576 MHz. The manufacturers block diagram is reproduced in Figure 3. It uses a 6 bus architecture for internal processing, providing separate address and data busses for program memory, and for two data memories. On chip resources include two 256 word data memories, a 256 point sine table, 512 words of zero wait state program memory, 24 bit serial inputs and outputs, an 8 bit host interface, interrupt controller with two external interrupts, a 24x24 multiplier and an ALU which supports double precision operations. A single external address and data bus are provided for expanding program or data memory size off chip.

The external busses of all three processors are tied together on the same expansion memory. A bus arbitrator controls access to the shared resources and prevents conflicts. Program memory is expanded to 8k words of 0 wait state static ram and is used by the main DSP only. The other two DSP's rely completely on their internal 512 word program memory. The main board provides 32k words of additional static data ram, split equally between X and Y data memory spaces. The option board adds another 64k words of static data ram, again split equally between X and Y memory. All of the external data ram operates with one wait state, providing a good trade-off between cost and processing speed. The main board also contains a hardware random number generator and simple real time clock. The data ram and peripherals are available for use by all processors. Only the main and decimator DSP's can communicate directly with the host computer.

A single rom contains the program code which fills the internal program memory of all three processors. Each reads its respective portion of the rom when released from reset and fills internal program memory. The main DSP can then communicate with the host computer to download larger programs into external program ram as well as modifying its internal ram. This rom also contains serial number information and calibration constants.

The decimator DSP receives signals from two on board A/D converters and decimates the data to lower sampling rates. The A/D's are always run at a fixed 192 kilosample rate and the decimator reduces that rate to 48 kHz or 32 kHz for compatibility with professional audio sampling rates. If the option board is fitted, additional clocks are available. This allows the converters to operate at a 176.4 kilosample rate with decimation down to 44.1 kHz. This allows compatibility with all professional and consumer sampling rates. Analog signals from the various portions of the System One analog measurement modules or from dedicated front panel inputs are routed to the anti-alias filters by CMOS switch selectors. The filters are designed to remove energy above 80 kHz before feeding the A/D converters. The converters are dithered by an analog output from the same random number generator which supplies the DSP's with digital dither. After filtering, the decimator DSP passes its data serially to the main DSP for further processing.

The third processor is used for AES-EBU communications. Data encoding and decoding, synchronization, parity checking and extracting the serial status bit stream are accomplished by dedicated logic circuits. However, complete implementation of the AES-EBU interface requires the ability to write and read quite a few status bits in real time. These bits provide sample counts, time of day clocks, pre-emphasis flags, channel allocation information, source and destination codes, etc. Additionally, it is necessary to send and receive pre-emphasized data. In a conventional digital device which will output or input the signal in the analog domain, emphasis is provided in the analog domain. Since this instrument must measure these devices, without resorting to analog techniques, emphasis is produced digitally in this DSP. This architecture always passes linear data to the main DSP. Emphasis is provided via recursive (IIR) filters to duplicate both the amplitude and phase response of an analog filter. A linear phase non-recursive (FIR) filter could be used but would not produce a correct simulation of the analog circuit.

The digital option board also contains parallel input and output ports for connection to external test circuits and DSP devices. Each port multiplexes two channels of data onto one set of 24 data lines. Each output channel is buffered by a two stage FIFO to prevent jitter in the transmitted data. Each input channel consists of latches which are written by the external device and read by the main DSP. A sample clock generator is provided to time data input and output or an external clock may be used for either or both functions. Appropriate hardware generates interrupts for the processor when data is read from the output or received at the input.

The digital interface circuits, either AES-EBU or parallel, serve the same function as the balancing amplifiers in an analog system. They provide an interface between the format of the device under test and the internal hardware of the test equipment. They also double buffer data to allow for timing differences between the two pieces of equipment.

Measurement Architecture

The discussion thus far has centered on the need to make measurements of conventional audio parameters in the digital domain. To allow comparisons between signals at arbitrary points in a digital audio device requires that the measurements be made digitally in a manner similar to that in the analog domain. For example, suppose an unacceptably high value of THD+N is measured at the analog output of a digital tape recorder. A measurement of distortion on the digital signal out of the recorder will indicate whether the problem is in the record or reproduce sections of the machine. However, measuring distortion with an FFT of the digital data will provide incomplete results since it cannot be correlated with the reading on the analog signal. To avoid this problem the measurements in the two domains must be performed with comparable methods. The DSP measurement architecture will therefore be compared to a conventional analog audio measurement system.

A conventional analog signal generator and analyzer is diagrammed in Figure 4. One or more signal sources are summed to create the test signal. For harmonic or intermodulation distortion tests these are very low distortion sinewave oscillators. For other tests one of these may be a random noise generator or squarewave generator. The amplitude ratio between these signals is typically set by resistors used to sum the signals. After summation these signals are attenuated to the desired level and sent through a balancing amplifier or transformer to the device under test (DUT) input.

The signal received from the DUT is buffered and converted to a ground referenced signal for processing by the measurement hardware. The signal at this point is used by many different measurement circuits; an rms level meter, frequency counter, noise weighting filter and notch filter. The rms level meter provides the fundamental level reference for distortion measurements or simply to monitor the incoming signal. This meter may be used for general purpose level measurements or measuring frequency response when the stimulus is sweeping frequency. If THD+N is being measured the notch filter is used to remove the fundamental. The remaining signal is amplified if necessary and drives highpass and lowpass band-limiting filters. This filtered signal goes to a second level meter for display of the distortion component amplitudes. If distortion is needed as a percentage or dB value, the most common case, the amplitude of the distortion products are divided by the amplitude of the input signal before display. A frequency counter will typically be used to determine the incoming frequency. The measured frequency value tunes the notch filter frequency and informs the operator as to the signal being measured. This allows for measurements off test tapes or at the receiving end of a broadcast link when the test signal frequency is not known a-priori.

When noise measurements are needed the weighting filter is used to simulate the response of the human ear as a function of frequency and correlate the measured value with the audible nature of the noise. Several standards exist for noise measurements which recommend different weighting filters and different types of level meter. American and Japanese standards are written around A weighting networks and rms or average responding meters. European standards specify a CCIR weighting filter with a quasi-peak level meter. When changing between noise, distortion and response measurements the filters and detector must be switched.

A digital signal generator and analyzer is diagrammed in Figure 5. There is a strong similarity to the analog system in the previous figure. Again, one or more signal sources are summed to create the test signal. For harmonic or intermodulation distortion tests these are very low distortion sinewave generators. As with the analog system one of these may be replaced by a random noise generator or squarewave generator for other test needs. The amplitude of these components is set by multiplying each by the desired output level before summing.

Many digital signal generators have been developed in the past which simply read a sinewave or other test signal out of rom. This limits the resolution of the test signal to submultiples of the sampling frequency. The resulting resolution is simply the sampling frequency divided by the rom address size. A 64 k word rom would be needed to obtain resolution of better than 1 Hz. When used at 20 Hz this would be marginal at best. This would exercise less than 32k codes in a D/A converter, not adequate to test high resolution professional converters. Finger (1986) discusses the problems resulting from inadequate test signal frequency resolution.

The present design computes the values of the output signal from equations and coefficients stored in a table.

This allows any sinewave to be generated with 24 bit resolution and accuracy in both frequency and amplitude. The resulting waveforms are free from rom size limitations and suffer no degradation in accuracy when set to non-integer frequency values.

Finger (1987) describes some of the problems in generating accurate digital test signals. To eliminate distortion in the test signal and enable measurements at low levels the output must be correctly dithered. A random number source is scaled appropriately for the desired word size and added into the signal before being output. It can then be rounded to the correct word size in the interface without introducing quantization distortion. The most common word size is 16 bit, due in part to the popularity of the CD. The professional audio field has realized the inadequacy of this for recording and is slowly moving to larger word sizes. It is also becoming common to have larger word sizes inside products than is provided in their I/O to insure no loss of performance during processing. Therefore the word size must be adjustable to match the application.

The analysis portion of the DSP measurement system is less complicated than the corresponding part of the analog measurement system. The signal applied to the appropriate input is converted to the 24 bit parallel, simultaneous word, format used internally. The signal at this point corresponds to the buffered and ground referenced signal in the analog system. It is sent to an rms level measurement routine, a frequency counter routine and a filter routine. Instead of having a bank of filters that are switched in and out there is one multistage filter. It is a 10 pole design composed of 5 second order stages in cascade. The filter coefficients are programmed appropriately for any of the measurements to be made. When measuring THD+N this is programmed as an 8 pole notch filter. Two additional poles are used as a high pass to block incoming DC. The notch filter is then tuned by the frequency counter that measures the input frequency.

There is an additional routine which computes the necessary coefficients of the digital filter for the measurement being made. When performing THD+N measurements the frequency value is used to derive the 20 coefficients required for the 4 sections of filter. The 5 coefficients to program the two poles of highpass are set by the desired highpass frequency. When noise measurements are performed the filter is converted to a weighting filter by re-programming these coefficients with the appropriate values. All that is required for any desired filter response is to re-program the coefficients.

Software Architecture

Figure 6 illustrates the program flow for the digital distortion measurement program. The signal generation, scaling, and dither must all be done in real time. On the measurement side, the rms computation of the input signal, filtering, amplitude detection of the filtered signal and zero crossing detection for the frequency counter also must be done in real time. There is additional real time overhead for each input sample and output sample from each channel. Frequency computation, coefficient computation, coefficient loading, housekeeping tasks and communication with the PC happen at a lower rate.

The processor has 250 instructions available between samples at a 48 kHz sampling rate. The real time tasks are allocated 200 of these instructions and operate in a loop. The remaining 50 instructions are allocated to background tasks which run in a separate loop.

There are other programs for the DSP-1 hardware which run under a batch mode. These are slight misnomers since the real time programs have some non-real time operations and the batch mode programs have some real time tasks. However, these terms accurately describe the operation of the programs in the measurement tasks. Real time programs determine measurements from every input sample before the arrival of the next sample. Batch mode programs acquire data in real time but compute readings after all data has been acquired. The digital distortion analyzer described above is an example of a real time program. Time domain display and FFT analysis is performed with a batch mode program.

Filter Architecture

There are two broad classes of digital filters, recursive and non recursive. Although a slight misnomer, these are often referred to as IIR and FIR filters. Figure 7 illustrates the topology of a non-recursive filter. These have the advantages of linear phase response, freedom from limit cycles and guaranteed overload recovery. Recursive filters offer much sharper filtering action for an equivalent number of processor operations than non-recursive types. Their required number of operations is not nearly as large at low frequencies as are non-recursive filters.

As can be seen from the diagram, a recursive filter consists of a series of one sample period delay elements. Each delayed version of the input signal is multiplied by a constant and summed with all of the others. The multiplier coefficients are the sample values of the filter impulse response. Non-recursive filters implement their filtering by continuously convolving the signal with this impulse response. Clearly then, the desired impulse response must be shorter in time than the total delay time in the filter. For any given sample rate high frequency filters will require fewer delays, and therefore fewer multiply operations, than low frequency filters.

The approximate minimum number of taps for a non-recursive filter may be obtained by doubling the sample frequency and dividing by the filter transition bandwidth (the frequency where filtering stops - the frequency where its starts). For example, a 20 Hz low pass which must eliminate frequencies above 30 Hz would have approximately 9,600 taps at a 48kHz sample rate. The passband ripple and stopband attenuation also affect the number of taps but this is a simple estimation technique. This is well beyond the current state of the art in DSP hardware and therefore impractical for general purpose filtering. It is however quite well suited to filtering high frequency signals and is the approach used in the decimator DSP. These filters have an additional advantage when performing sampling rate reduction in that the multiply and sum operation is done at the output sampling rate. However the delay line must be serviced at the input sampling rate.

The direct form I topology recursive filter is diagrammed in Figure 8. An excellent introduction to recursive filters for audio applications and the fundamental problems in their design was given by D'Attoro (1988). The filter shown operates by summing delayed versions of the input and output signals after appropriate scaling by the filter coefficients. The multiplication process creates data words which are equal in width to the sum of the signal data and the coefficient widths. In the typical design, these words are added in their wide state and the output of the summer is truncated or rounded before being sent to the delay elements. This represents a quantization operation on the data and introduces quantization noise.

One major limitation of recursive filters is their high noise gain at low frequencies. The quantization noise is fed back into the filter when the delayed output values are multiplied and summed. The gain from this point to the filter's output can be quite large at low frequencies. This is similar to the large gains sometimes encountered from intermediate points of an analog active filter to its output. An estimate of the noise gain of a direct form bandpass filter is the ratio of Nyquist frequency to center frequency times the ratio of Nyquist frequency to the 20 dB bandwidth. For a 20Hz bandpass with a 10 Hz bandwidth this gives (24,000/20)*(24,000/10) = 129 dB. A 24 bit DSP will yield a S/N ratio of about 15 dB for this filter, a 16 bit DSP would not work at all.

This gain cannot be reduced without redesigning the topology of the filter. This has been done by many designers and some topologies with lower noise have been found. However all require more computations than the direct form filter. Many also have serious drawbacks when used as a continuously tuned filter over the entire audio band since they are optimized for low frequency operation only. Even the best of these have noise gains at low frequencies of several bits. The optimal form filter eliminates the noise gain dependence on filter bandwidth but maintains the dependence on center frequency. This gives our previous example of a 20 Hz filter a noise gain of 62 dB.

The programmable filter bank is required to accept 24 bit digital audio data and filter this data without introducing any distortion or noise. When used for distortion measurements the filter bank must produce a notch response at any center frequency from 20 Hz to 20 kHz. To provide acceptable accuracy the response must be less than 0.5 dB down at second harmonic. This translates to a bandwidth of approximately 20 Hz at a 20 Hz center frequency. Since 24 bit dynamic range was desired no filter using 24 bit mathematics can be used. This necessitates a much wider word width in the delay elements and the multipliers to reduce the amplitude of the noise source internally to avoid error. A 48 bit internal word width was chosen to allow use of the double precision support inherent in the DSP.

Applications

Figure 9 shows the results of an alias rejection test, something unique to digital devices. This device is an oversampling A/D converter. The converter is driven with a sine wave and the output is measured digitally with the DSP-1 operating as a digital RMS meter. The sinewave amplitude is set at 6 dB below full scale to prevent overdriving the input filters of the converter. The sinewave frequency is swept from 10 kilohertz to 200 kilohertz and the RMS level is graphed. This provides a plot of the response of the converter to in-band and out-of-band signals. All response above the Nyquist frequency of 48 kHz represents aliased components. The alias products that the converter lets through show quite clearly in the graph at approximately 50 dB below full scale or 44 dB below the input. They are mirror imaged around 96 kHz (2x the sampling rate) and 192 kHz (4x the sampling rate). There is also a strong alias product between 24 kHz and 28 kHz due to half band topology of the last stage of decimation filtering in the converter.

Figure 10 shows the same device under the same conditions except that the dashed line is a measurement of the frequency of the digital signal out of the converter. The DSP is now computing the frequency of the signal as well as the RMS amplitude. At frequencies above 73 kilohertz there is a significant alias product. The frequency counter indicates that the output from the A/D is 23 kilohertz (96-73). As the input frequency is increased the frequency of the alias is dropping. Finally, as the alias product reaches approximately 7 kHz it is 90 dB down and is obscured by the noise floor of the system and the frequency counter reading gets very noisy.

Level linearity on D/A converters may be measured by generating a digital sinewave using the DSP. The output of the converter is measured with an analog bandpass filter followed by an rms meter. Sweeping the amplitude of the digital signal and graphing the measured amplitude results in the plot of Figure 11. The sinewave frequency is 997 Hz to exercise the maximum number of codes in the converter. Subtracting the best fit straight line from the data and re-graphing gives the deviation from linearity plot shown in Figure 12. The deviation is within 1 dB down to 95 dB. Below one LSB (-108 dB) the linearity curve flattens out again as the system becomes a correctly dithered single bit converter. If the dither was set too small the linearity error would climb radically below 1 LSB.

Inverting the functions of the analog and digital portions of the measurement equipment allows linearity measurements on A/D converters. The low distortion analog generator is used to stimulate the A/D with a 997 Hz sinewave. The DSP is programmed to filter the digital signal from the converter with a bandpass centered at 997 Hz and to measure the rms amplitude of the filter output. The amplitude of the analog sinewave is swept and the digital rms level recorded. After subtracting the best fit straight line, the deviation from linearity graph is obtained (Figure 13). This converter is an oversampling 18 bit design which has exceptional low level linearity performance as evidenced by the results.

Using the DSP to generate a properly dithered low distortion digital sinewave, THD+N measurements may be performed on the D/A converter measured earlier. The analog distortion analyzer measures and tracks the frequency out of the D/A and tunes its notch filter appropriately. By sweeping the frequency of the digital sinewave the plot of THD+N vs frequency in Figure 14 is obtained.

The digital and analog functions of the system may again be interchanged to obtain a THD+N vs frequency plot of an A/D converter. The DSP measures the frequency and rms amplitude of the received data. It computes the necessary coefficients to form a notch filter centered on the incomming sinewave. The notch filter output is measured with a second rms computation. The resulting THD+N plot is shown in Figure 15. The rise in distortion at high frequencies was later traced to non-linear capacitance effects in the buffer amplifier ahead of the converter itself.

A sweeping spectrum analyzer view of the A/D converter output is graphed in Figure 16. The converter was driven with a 997 Hz signal from the generator at four different amplitudes. The DSP was used as a narrow-band bandpass filter and rms meter which swept in frequency to display the spectrum at each amplitude. The increase in distortion at the higher levels is clearly visible as the signal exceeds 24 dB below full scale.

Acknowledgments

Bruce Hofer designed the analog front-end circuitry, Bob Wright and Carl Hovey wrote the PC software, Tony DalMolin designed the analog output filtering and developed valuable filter response design software, Jerry Liebler wrote the original decimator DSP code. Studer A. G. provided substantial assistance on the AES-EBU interface, with special thanks to Mark Erne and Peter Joss.

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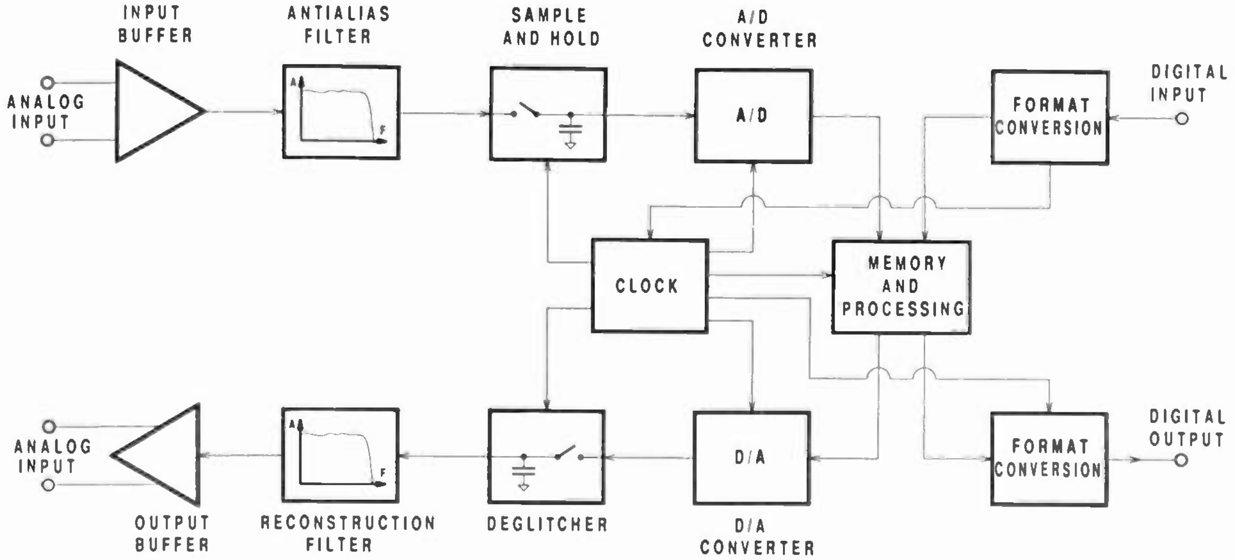


Figure 1. BLOCK DIAGRAM OF A TYPICAL DIGITAL AUDIO DEVICE

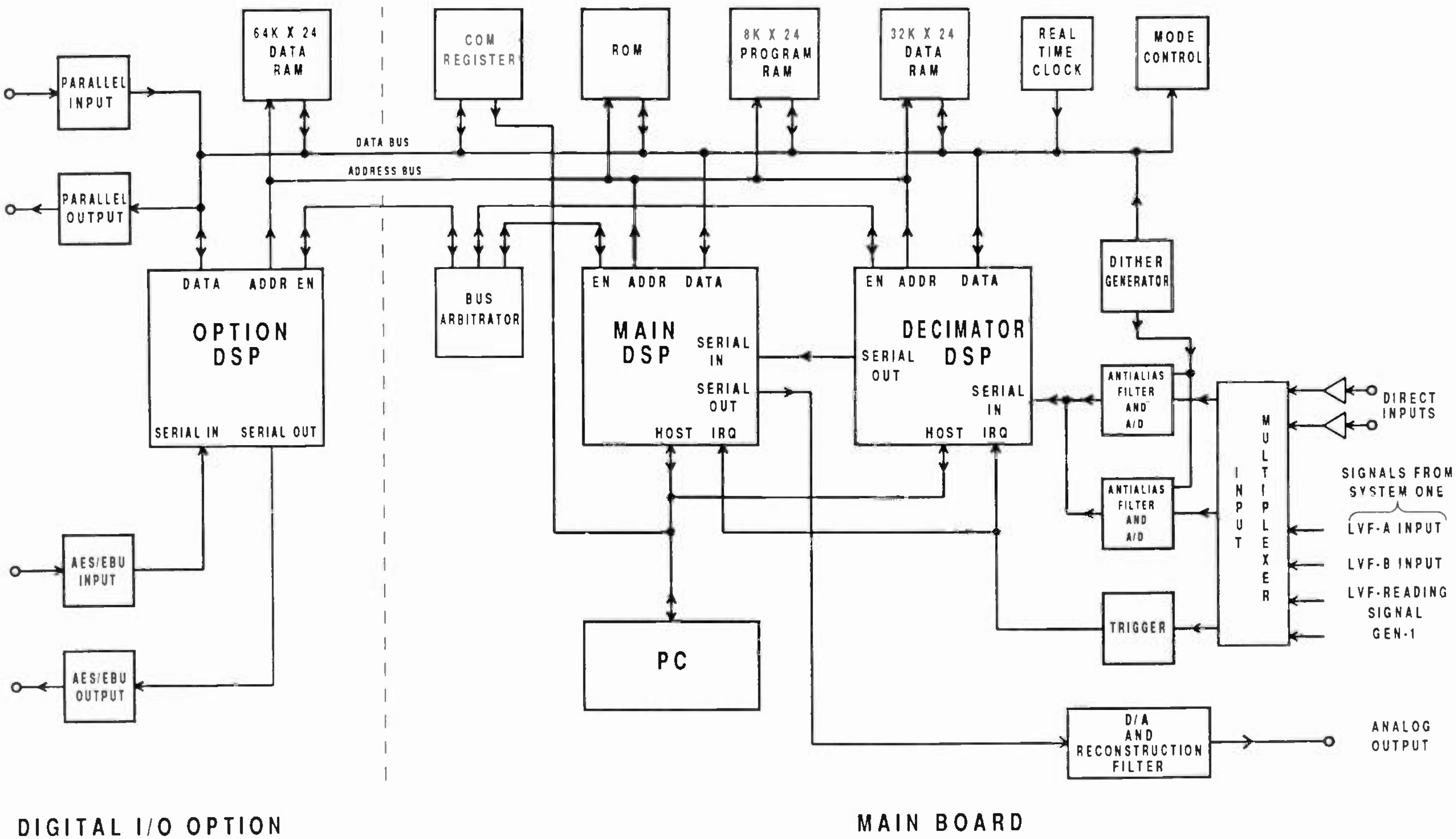


Figure 2. DSP-1 BLOCK DIAGRAM

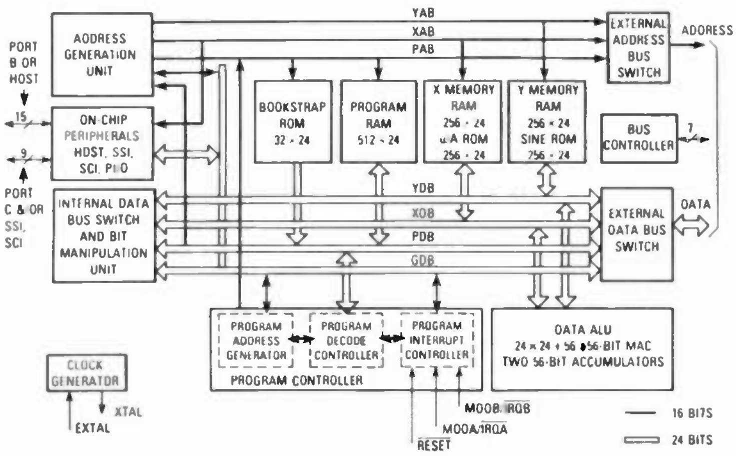


Figure 3. DSP56001 Block Diagram

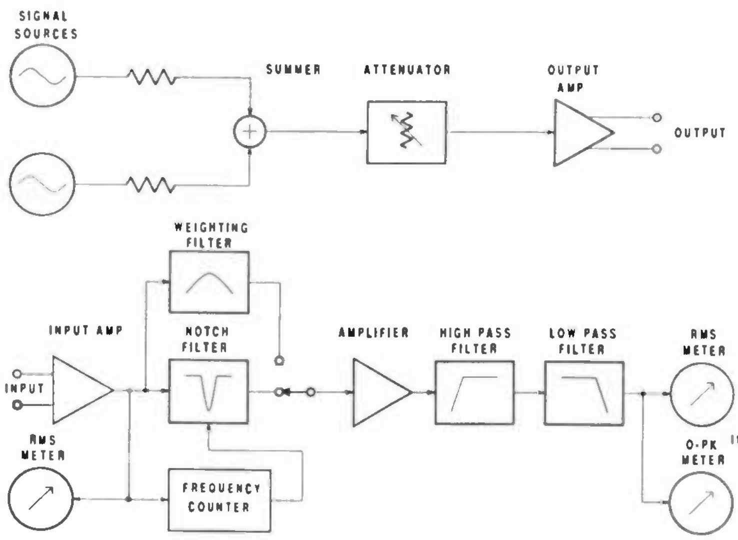


Figure 4. ANALOG DISTORTION ANALYZER

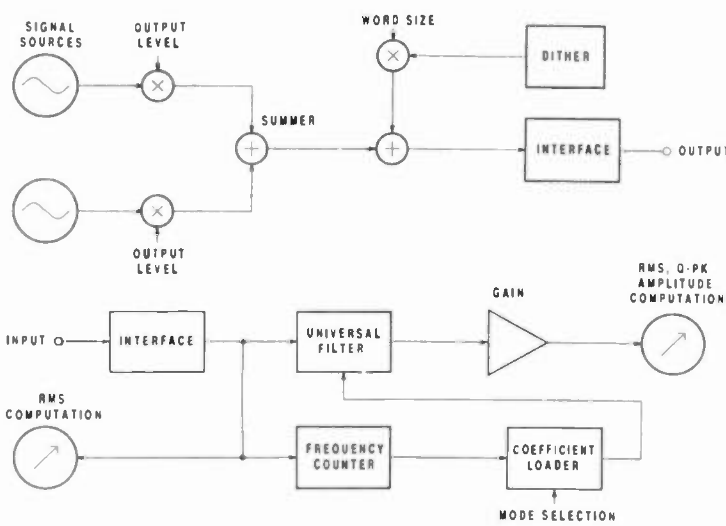


Figure 5. DIGITAL DISTORTION ANALYZER

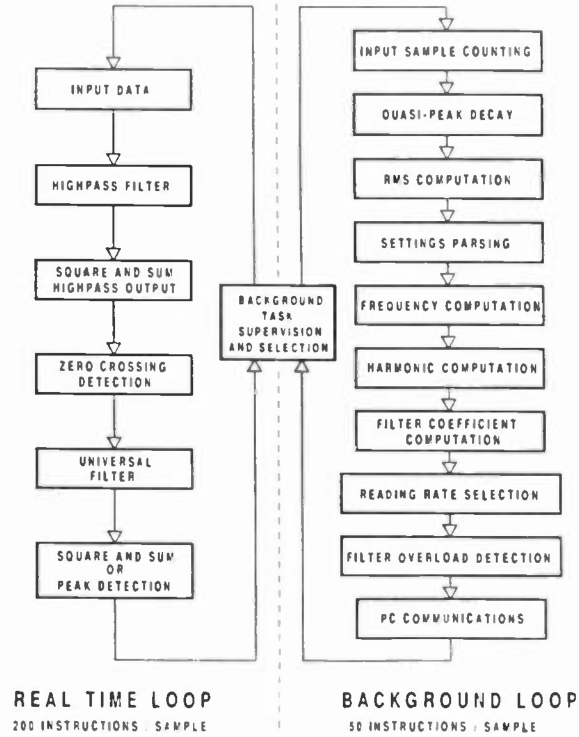


Figure 6. DIGITAL DISTORTION ANALYZER PROGRAM FLOW

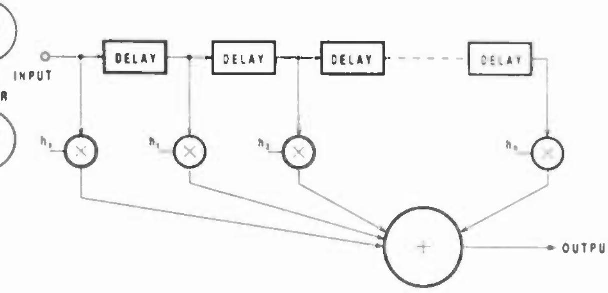


Figure 7. FIR DIGITAL FILTER

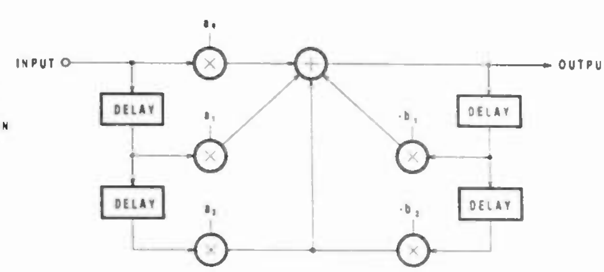


Figure 8. IIR DIGITAL FILTER

Figure 9   18-BIT A/D CONVERTER ALIAS REJECTION, -10 dB INPUT

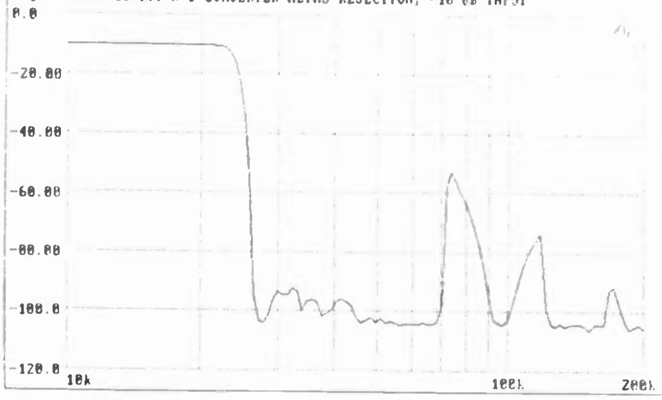


Figure 10   A/D ALIAS REJECTION   SOLID LINE AMPLITUDE, DASHED LINE FREQUENCY

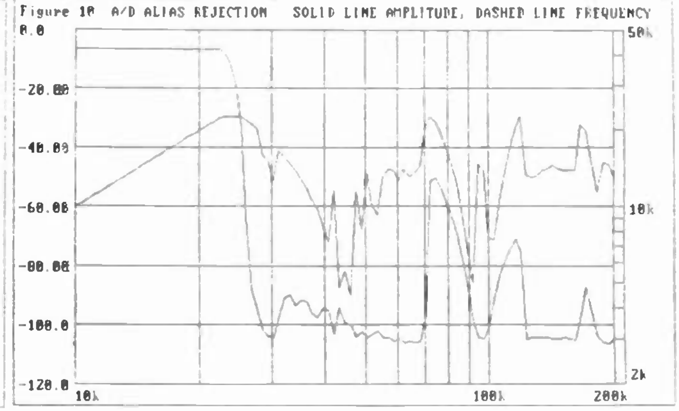


FIGURE 11   D/A ANALOG OUTPUT LEVEL (dBV) vs DIGITAL INPUT LEVEL (dBFS)

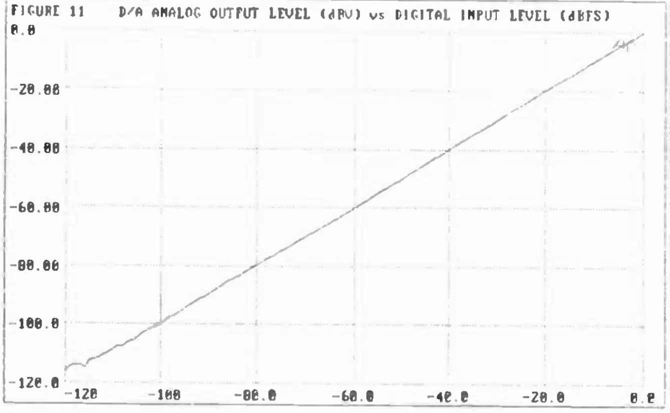


FIGURE 12   D/A ANALOG OUTPUT LEVEL ERROR (dB) vs DIGITAL INPUT LEVEL (dBFS)

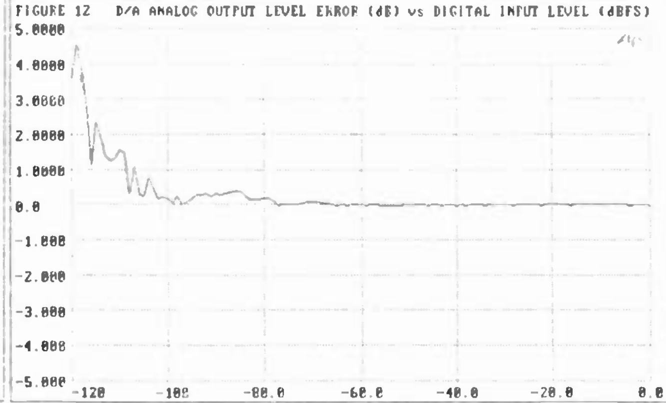


FIGURE 13   18 BIT A/D CONVERTER DEVIATION FROM LINEARITY vs INPUT LEVEL (dBFS)

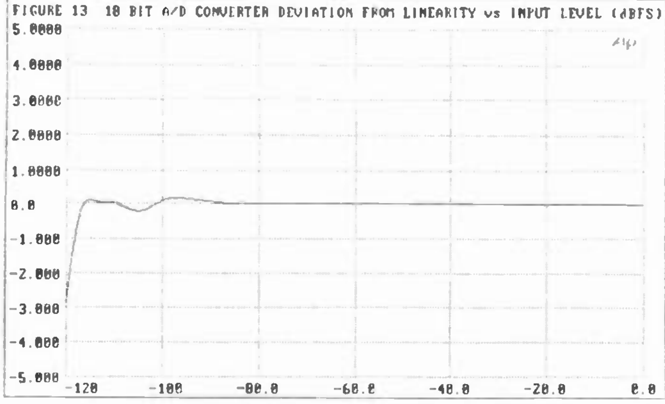


FIGURE 14   18-BIT D/A CONVERTER THD+N vs FREQ AT 0 dBFS OUTPUT

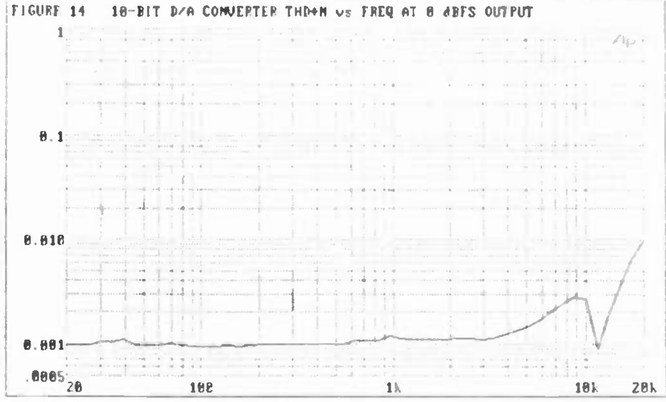


FIGURE 15   18-BIT A/D CONVERTER THD+N (dB) AT -3 dBFS OUTPUT

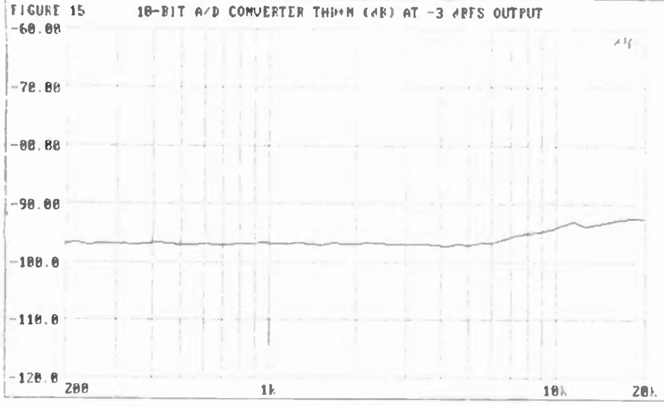
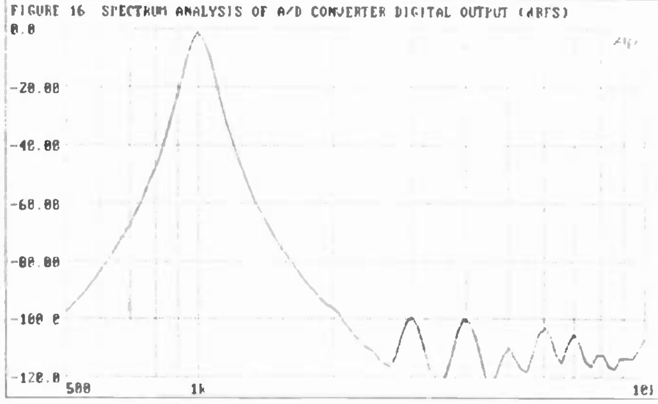


FIGURE 16   SPECTRUM ANALYSIS OF A/D CONVERTER DIGITAL OUTPUT (dBFS)



THE CD PLAYER IN THE BROADCAST ENVIRONMENT: A PROPRIETARY CARTRIDGE SYSTEM

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Laura J. Tyson
Denon America Inc.
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Abstract

A compact disc player has been developed specifically for broadcast use. This CD player utilizes a cartridge system to facilitate ease of operation and disc longevity. The cartridge system is proven to allow faster and more effecient performance by the broadcaster while enhancing the sound quality of the broadcast station. The cartridge system will prevent degradation of the compact disc due to mishandling. This CD player is designed to both psychologically and operationally resemble an anolog tape machine.

Introduction

In the Spring of 1983, the compact disc player made its first appearance in the United States. Since then, the CD player has achieved world wide acceptance as a high quality play-back medium in the consumer marketplace. The consumer CD player, however, does not sufficiently satisfy the demands of professional use. Without options such as Cue to Music, Track Preview, and wired Remote Control, the consumer CD player requires more operational steps for on air playback. In a situation where frequently used by many different people, the Cd player must also incorporate a system for protecting the disc from damage resulting from constant handling.

Plans for a Broadcast CD Player

The idea for a dedicated broadcast CD player was first initiated in February of 1987. We determined a CD player for broadcasters must fulfill specific requirements to render it successful. It must be durable, reliable, easy to service, and have universal controls. It must withstand daily use. Ideally, the broadcast CD player would minimize the steps required for the operator to perform, and provide a logical and familiar control panel.

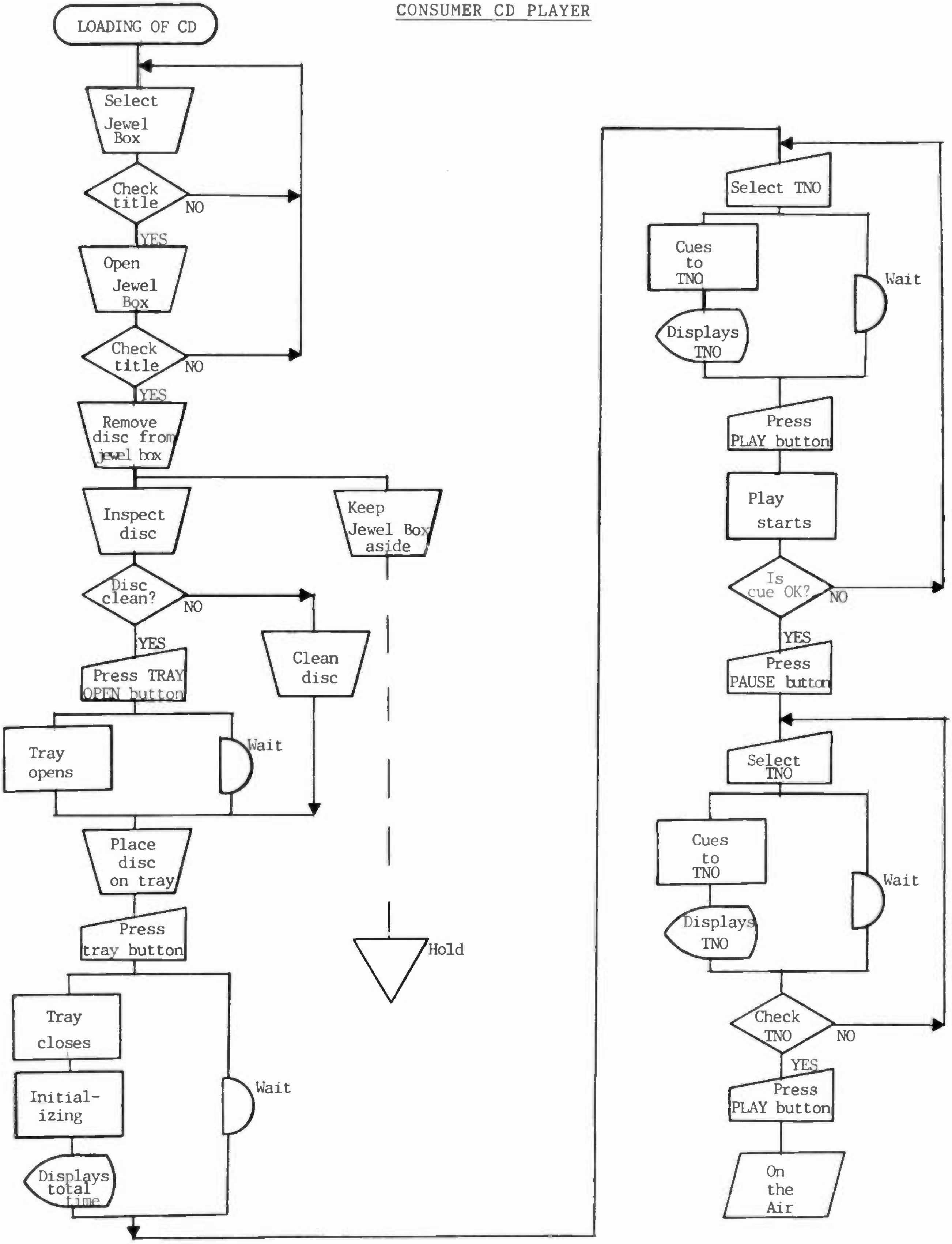
Operational Flow Chart

Figure A illustrates a flow chart analysis of the steps involved to use a typical consumer CD player in the broadcast environment. By modeling the broadcast CD player after a conventional tape cart machine, we were able to minimize the steps required for playback of a selection. Figure B illustrates similar use with a CD Cart Player. Upon loading the cartridge, the player automatically cues to the desired selection, and stops when the cartridge is unloaded, thus eliminating the requirement for a dedicated STOP button. Figure C compares the process of removing and storing the disc from a consumer CD player to that of a CD Cart Player.

The Cartridge System

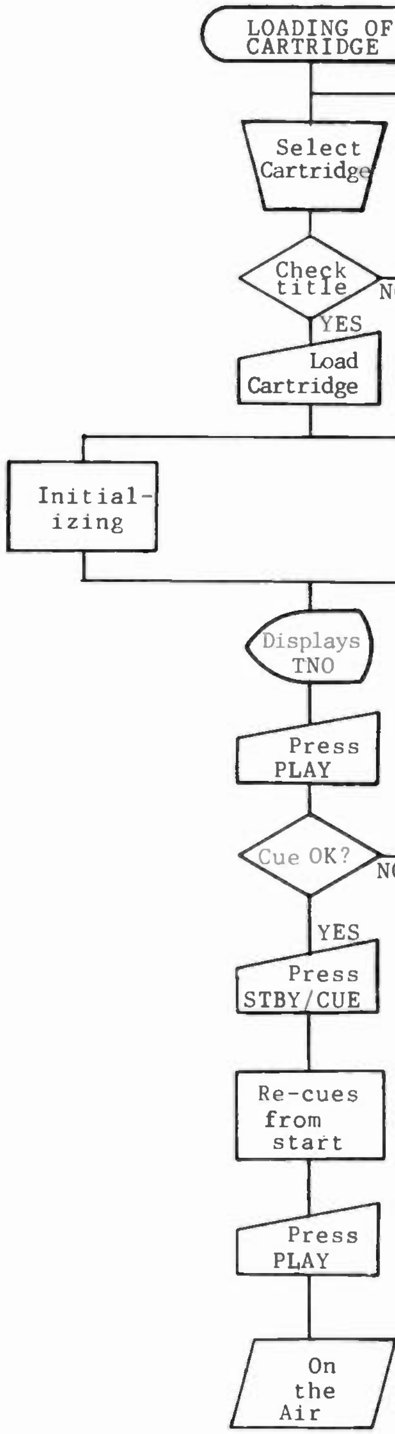
The basic purpose of the cartridge is to protect the disc from damages incurred from handling. The cartridge system also eliminates many of the steps involved in storing discs in conventional jewel boxes. The disc is permanently housed in the protective cartridge to eliminate the potential damage resulting from everyday handling. When placed in the cartridge, only the outer edges of the disc, where no data is recorded, comes in contact with the cartridge. Two mounting screws fix the cover of the cartridge to prevent the disc from being removed. The cartridge can be labeled for easy identification when loaded in the player and also when stored. The cartridge is manufactured from a clear, semi-soft plastic. Unlike the jewel box, the cartridge will not break or shatter when dropped. The contents of the cartridge can easily be confirmed from a quick visual inspection. During cueing and playback, the operator can confirm the disc is properly rotating in the player.

CONSUMER CD PLAYER

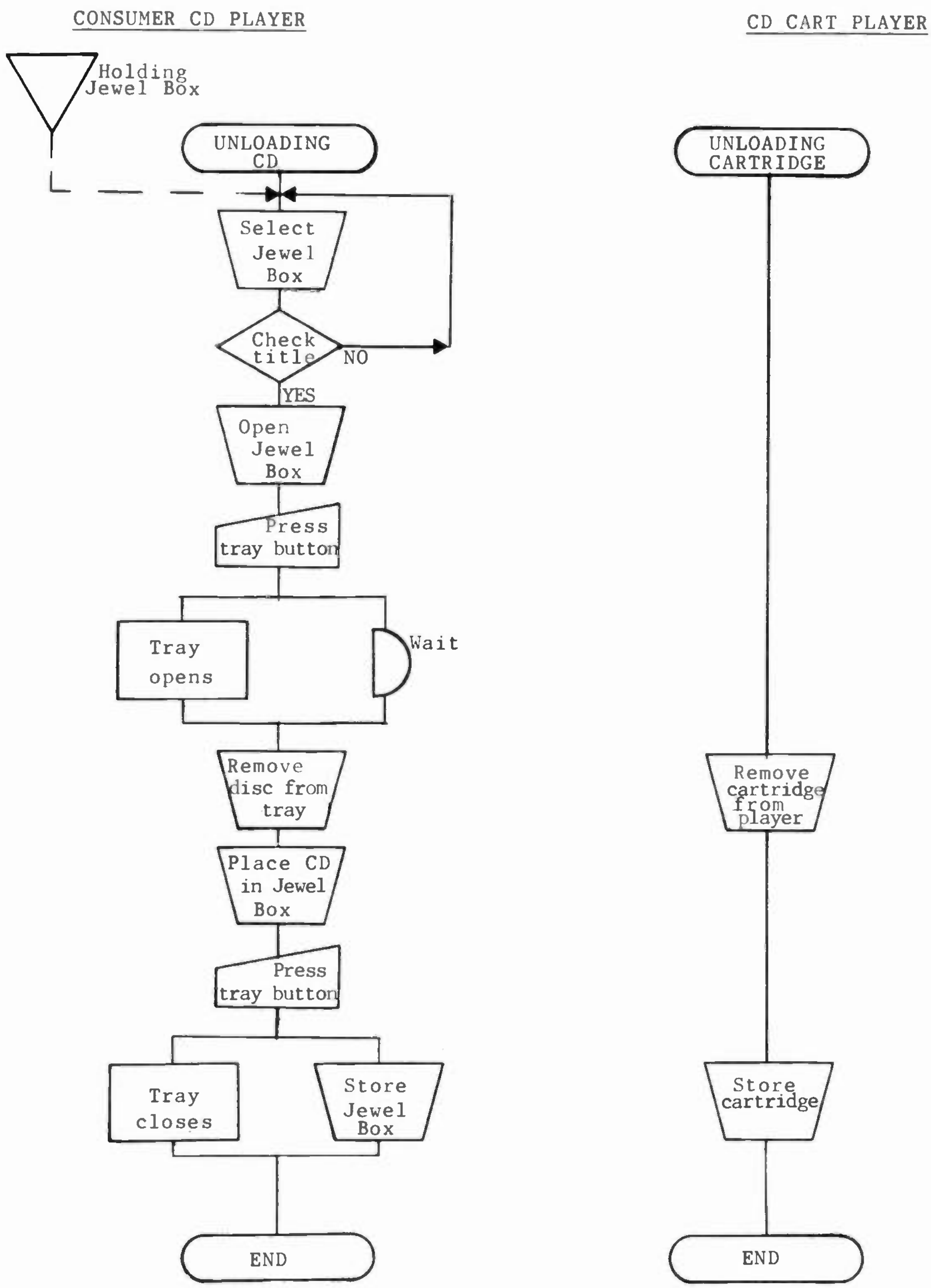


(Figure A)

CD CART PLAYER



(Figure B)


CONSUMER CD PLAYER
CD CART PLAYER
Holding Jewel Box
UNLOADING CD
Select Jewel Box
Check title
NO
YES
Open Jewel Box
Press tray button
Tray opens
Wait
Remove disc from tray
Place CD in Jewel Box
Press tray button
Tray closes
Store Jewel Box
END
UNLOADING CARTRIDGE
Remove cartridge from player
Store cartridge
END

(Figure C)

Broadcast CD Player Ergonomics

1. Pattern Recognition
The CD Cart Player was designed to both physically and psychologically resemble a tape cart machine. The fron panel layout is similar to the layout of a typical tape cart machine. By applying the principle of pattern recognition, the operator will immediately identify the functions and controls.

2. Control Sets
The desired track number for playback is selected either before or after the cartridge is loaded into the player. The track is selected with the rotary Select Knob, located on the left side of the front panel. The click function of the Select Knob allows the operator to select a specific track number without having to watch the display. A switch positioned around the Select Knob permits either Single or Continuous playback of the disc. The start and cue controls are placed on the right side of the front panel, with the PLAY/PAUSE button placed directly above the STDBY/CUE button. Four Search buttons placed under the cartridge slot allow the user to search into a selection, or to do a manual cue. The Slow Search button scans through the selection in increments of one Frame (13.3 milliseconds), which allows the operator to execute precise manual cues. (See Figure D)

3. Visual Displays
The front panel shows a seven-segment LED display of the selected track number, and also remaining time in Minutes, Seconds, and Frames during playback of the selected cut. If desired, the Frame indicator can be removed from the display via a dip-switch located on the rear panel. When a disc is loaded, the STDBY/CUE button flashes yellow to indicate the search mode. Upon locating the track, the flashing changes to steady illumination. During playback, the PLAY button remains illuminated red until the user-selected End Of Message warning occurs, at which point it begins flashing. Once the track is completed, the PLAY button flashes yellow to indicate the selection has already been played.

Functions of the CD Cart Player

The Cd Cart Player addresses some of the problems of playing CDs live on the air. The player automatically cues to the audio on the disc, utilizing a user-selected cue level, depending on the style of music played and the format of the station. Once the music is cued, the operator may preview an unlimited amount of the select-ion, and one press of the STDBY/CUE button returning him to the exact time address originally cued. This will increase the operators sense of preparation, and hopefully eliminate any unnecessary apprehension. The audio will start up with in 300 milliseconds after the PLAY button is pressed. The CD Cart Player can be controlled from the studio console in order to minimize operator errors and traffic in the air studio. Two different modes of wired remote control allow different levels of control from the console. A feature added to the latest version of the CD Cart Player gives a dry contact on the remote control connector upon recognition of Index NO 3 on specially formatted subscription service discs. When used, two players can be set up to do automatic segues.

Summary

The CD Cart Player made its on-air debut in the United States on July 21, 1987. Since then, many more features have been added to the current model, concurring with market demand. The cartridge system has proven its usefulness by protecting the discs during all stages of handling, and by providing broadcasters with a simple and efficient method of playing compact discs on the air.

Figure D

DIGITAL AUDIO FOR LINKS AND SUBCARRIERS

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ABSTRACT:

The recent development of "Slip Code " digital modulation makes it possible to transmit digitized information at rates approximately 10 times as fast as the previously used methods. Data at T1 (1.54 Megabits) and CEPT (2.05 MBs) can be transmitted in a 200 KHz bandwidth. These data rates make it possible to send compact disk audio in digital form over a 950 MHz Studio to Transmitter Link. When the technique is applied to SUPPLEMENTARY CARRIER service (SCA), data can be sent at rates up to about 400 KBs. Data obtained from tests utilizing experimental station KA2XVZ are reported in this paper.

BACKGROUND:

Data transmission efficiency is generally expressed in terms of the Nyquist Factor or Bits per Second per Hertz of Bandwidth. Conventional FM or AM links have a Nyquist Factor of 1. Some newer methods which are practical for use by broadcasters can improve this to 4. The recent development of slip code modems used with Single Sideband FM have improved this by values up to 10. The highest rate previously achieved was with Quadrature AM which can go as high as 8. In practice, the availability of filters to match F.C.C. requirements limit the results obtained to about 2/3 the theoretical value. The high cost, complexity and poor S/N performance of QAM make the method impractical in most cases for broadcaster use. Also, it is not possible to use QAM in a mobile environment, whereas slip code is SSB-FM and is not as sensitive to level variation.

Slip code is relatively free from multipath and fading problems when compared to the other methods requiring a much wider bandwidth. Since the signals are SSB-FM with no AM component and can be noise limited, some additional advantage is obtained.

The implimentation of the slip code method is relatively simple. An encoder chip is used at the transmitter and a decoder chip at the receiver. The remaining circuitry consists of very narrow bandpass filters, mixers and modified FM receivers. Other high Nyquist Factor methods require much more complex circuitry.

As a general rule, when the Nyquist Factor rises, the required Signal to Noise Ratio for a given error rate also rises. Slip code suffers a similar degradation, but not to the same degree as MPSK or QAM.

THE SLIP CODE METHOD

Slip coding derives its name from an algorithm that adds a fraction of the bit width time to each bit as transmitted. In this regard it is similar to MFM used for double density disk recording where the transmitted bit width varies from 1 to 2 units. At some stage of the transmit/receive process, both Slip Code and MFM are a form of Pulse Position Modulation. The amount of change from pulse position to pulse position is related to the resulting Nyquist Factor, hence bandwidth used

Assuming 4,5,6 slip encoding, the pulse widths stay 4/4 wide for no change, 5/4 wide for a change and 6/4 wide for a missed bit and resync change. At the end of 4 changes a bit will have been missed in the timing frame. The decoder senses this bit, reinserts the proper bit and resets the 0,1 level relationship on the receipt of the 6/4 time difference. 4,5,6 slip coding has a Nyquist Factor of 4. 10,11,12 slip coding would have a Nyquist Factor of 10.

The bandwidth required can be illustrated by the following example. Using 6,7,8 slip coding (NF=6), data at 192 KBs can be sent in a theoretical bandwidth of 32 KHz. Because of filter limitations an actual bandwidth of more than 40 KHz is required to meet F.C.C. specifications. At very high bit rates (1.5-2 MBs), saw filters can be used with some improvement in bandwidth efficiency. Only Gaussian and Saw filters can be used.

This same 32 KHz bandwidth can be used for a 192 KBs digital mobile phone system at 400-900 MHz in a 50 KHz channel. With an ideal bandpass filter (which does not exist) the maximum data rate could be around 200 KBs in a 25 KHz bandwidth with a Nyquist Factor of 8 (8,9,10 slip).

Figure 1 shows the block diagrams for a slip code transmitter and receiver. The critical element is the SSB filter which passes only a portion of one sideband centered around 1/2 the bit rate.

SLIP CODE IN FM-SCA

Figure 2 shows the spectrum of an FM transmitter with a 92 KHz subcarrier deviated the maximum amount allowed which is 82.5 Khz on the main and 8 KHz on the subcarrier. Note that the signal extends from 0-182 KHz on either side of the carrier and that the part between 160-180 KHz is 25 Db below the unmodulated main as allowed by F.C.C. Regulations, Part 73.317. The regulations state that all radiation from 120-240 KHz shall be 25 Db below the main and all radiation from 240-600 shall be 35 Db below the main.

When the spectrum of a slip code encoded signal before filtering is examined on the spectrum analyzer, it will be noted there are islands of information centered around 1/2, 3/2, 5/2, 7/2 etc. times the bit rate. Using a shaping filter, the important information can be extracted from the 1/2 BR island alone. It is not practical at baseband to use only the 1/2 BR island as is done at R.F. It is also necessary to preserve the phase relationship of the frequency components around the island which can be very difficult at baseband while obtaining any skirt selectivity.

The shaped slip code signal could be added at the normal SCA input. The parts of this island above 1/2 BR are 20 Db below the 1/2 BR island so the spectrum would appear as shown in Figure 3. Since the part extending from 180- 230 KHz is more than -25 Db below the main carrier, 73.317 is not violated. For noise improvement purposes, it might be desirable to raise the level of the higher frequency components to -25 (Preemphasis) and reduce it at the receiver (Deemphasis).

As anyone who has worked with digital SCA is aware, the linearity of the detector and IF filters is very critical and the effects of bleed through from the main channel cause a very high error rate on certain modulation peaks. The phase linearity problem associated with restoring slip code requires an unusually complex filter to remove the main channel, which is 20 DB stronger than the subcarrier.

The sum total of the problems make baseband operation with slip code at very high bit rates impractical.

There is a simpler way to accomplish the same end result and at the same time reduce the spectrum used. The F.C.C. has not passed judgement upon or approved the method to be discussed, which involves a supplementary carrier and not a subcarrier. All tests to date have been performed on laboratory equipment and over an EXPERIMENTAL station at 893 MHz. (KA2XVZ).

SCA stands for SUPPLEMENTARY CARRIER AUTHORIZATION, not SUBCARRIER authorization.

Instead of creating a subcarrier and mixing it with the audio and any other subcarriers in the exciter, the alternate method is to mix two transmitters at the exciter output. To do this, a small transmitter of about .1-.2 watts output is required. The mixing method is shown in Figure 4. The main channel can retain one subcarrier at 67 KHz if desired, but the supplementary transmitter replaces the 92 KHz subcarrier. Figure 5. shows the spectrum at the exciter output. Note that the main and subcarrier at 67 KHz now extend +- 150 KHz, whereas the spectrum with a 92 KHz subcarrier would have extended to 182 KHz. The space between 150 and 180 KHz is no longer used by the main, but a second transmitter has been added to occupy the same spectral space. The spectrum at the transmitter has not been increased in width or amplitude beyond the spectrum of a 92 KHz subcarrier alone as shown Fig. 2, but the data rate is 80 KBs which is 4 times that presently obtainable with standard SCA methods. Data at rates over 200 KBs has been transmitted without exceeding 73.317. Removing the 67 KHz sub-carrier can further reduce the spectrum used.

Figure 6 shows the spectrum inside an FM receiver measured after limiting. The limiter nonlinearity has made a double sideband signal out of the single sideband signal. Similar effects can be expected of a final amplifier with AM compression. This would lead one to expect the apparent subcarrier result could be detected in the normal manner. This is not the preferred manner however as will be noted later, due to the phase and interference problems mentioned above.

Regulation 73.319 requires that the subcarrier shall not cause interference greater than -60 Db to the main channel. The interference caused by the supplementary channel is easy to calculate. Figure 4(b) shows two vectors. The main channel vector M is 25 Db stronger than the supplementary channel vector S. If the vectors are added, a phase deviation of 2.9 degrees and an amplitude variation of about 5% results.

If a single audio tone of 1500 Hz is used to modulate the main channel to 75 KHz deviation, the angular modulation produced is 50 radians or 3,000 degrees. Dividing 3,000 by 2.9 yields a cross modulation of approximately 1/1000 or -60 db. This has been verified by measurements made on a low cost FM receiver utilizing a 3089 Quadrature Detector and ceramic filters. For modulation frequencies below 1500 Hz the interference becomes less and for frequencies above 1500 Hz the preemphasis network maintains it at -60 Db as required.

When slip code is used, there is no audible tone since the modulating frequency is 1/2 the bit rate , which we assume to be very high. Any cross talk occurs at a frequency above 50 KHz which is reduced by receiver roll off (Deemphasis) by an additional 40-60 Db. The result is that utilizing the supplementary carrier, bleed through or interference to the main is actually less than that obtained by the conventional subcarrier method.

Since the supplementary carrier signal leaving the FM station is an independent signal not dependant upon the main FM carrier, it can be detected by a narrow band SSB-FM receiver, which is the same type of receiver used for a studio to transmitter link.

Interference from the main channel to the supplementary channel depends on the filters used, but in general it is more than 60 db below the supplementary carrier unless the main channel is greatly overmodulated. With the conventional SCA method, this interference is seldom below -40 to -45 Db. Because of this low interference level, even QAM could be used as a supplementary carrier utilizing this method.

The advantages of this method are:

1) Lower main channel interference.
2) Better S/N ratio utilizing a narrow band SCA receiver.
3) Main channel to SCA interference is reduced.
4) Data rates far above those obtainable by normal SCA.
5) Reg. 73.317 is not violated.
6) Multipath interference is reduced due to the reduced bandwidth.

The immediate disadvantage is that some accomodation with the wording of 73.319 is required that will require action or modification by the F.C.C.

As broadcasters more and more enter the information era, the need for higher data rates becomes apparent. The search for better sound will inevitably require the use of digital transmission methods. To accomodate these requirements, new and better techniques are required such as those discussed above.

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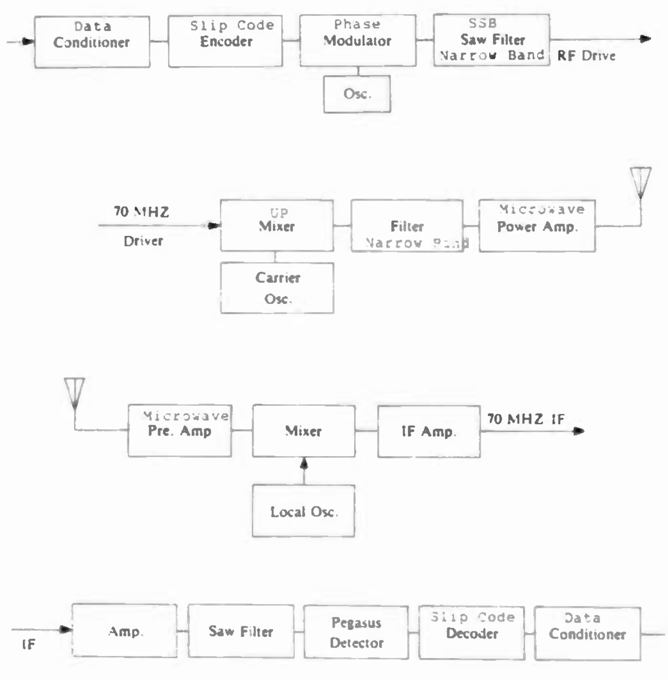

Figure 1. Slip Code transmitter and receiver for microwave or satellite use at T1 or CEPT rates. The bandwidth is less than 300 KHz.

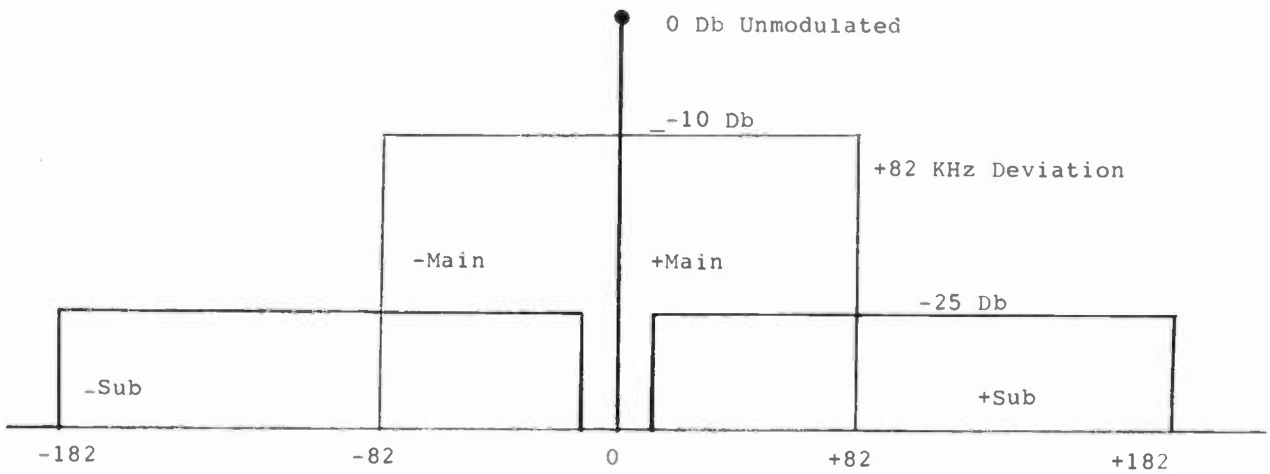

Figure 2. 82 KHz Main deviation plus 92 KHz Subcarrier deviated 8KHz.

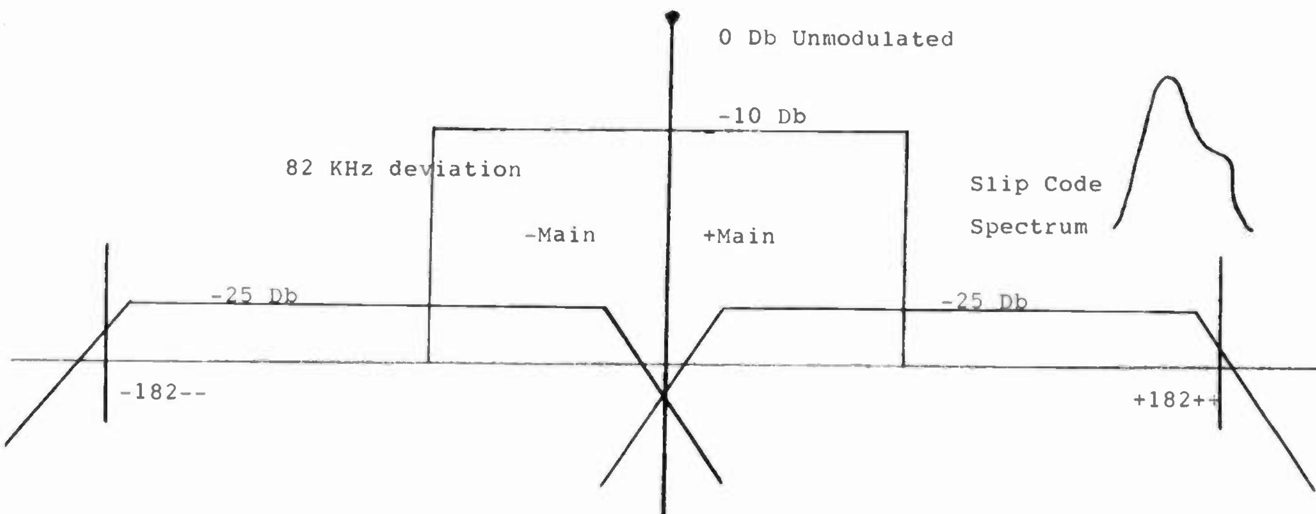

Figure 3 80 KBs Slip Code on 89 KHz centered Sub. Note overrun at edge

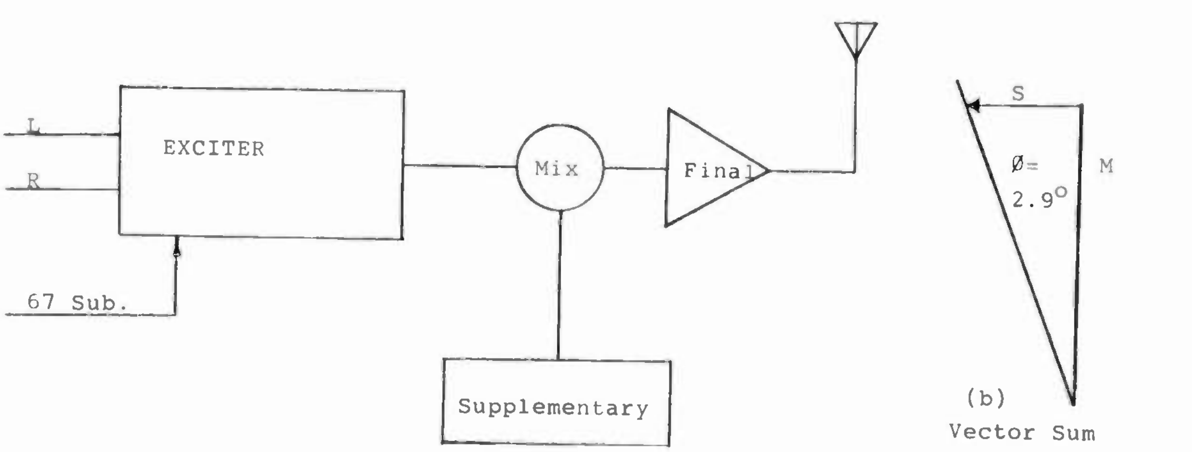

Figure 4. Conventional exciter at left and mixed supplementary transmitter.

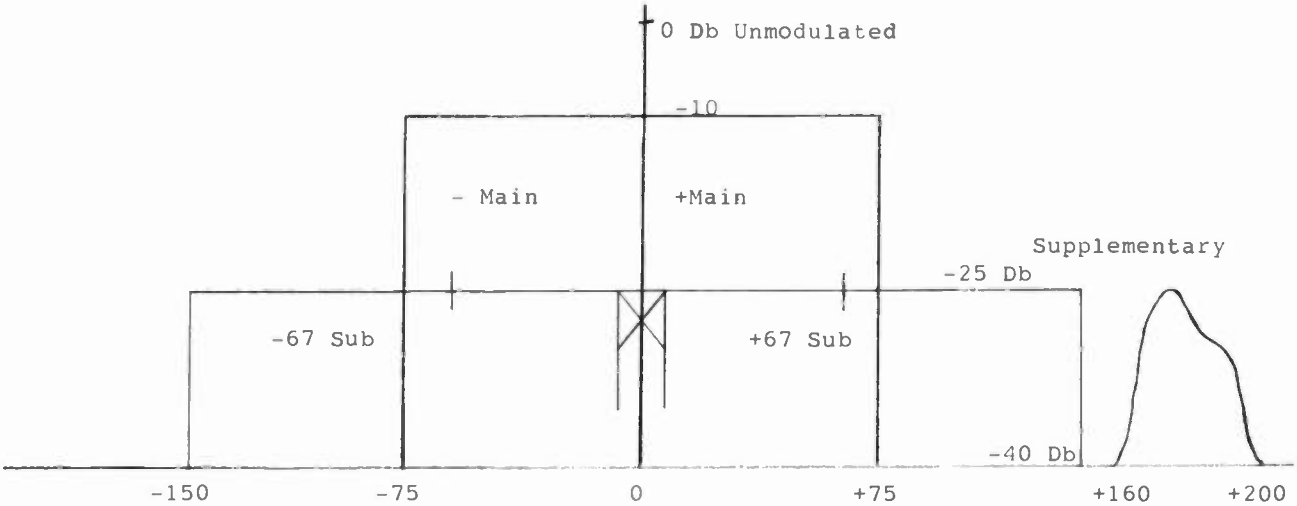

Figure 5. Main plus Sub deviated 75 KHz. Supplementary Carrier added at right after mixing as in Fig. 4.

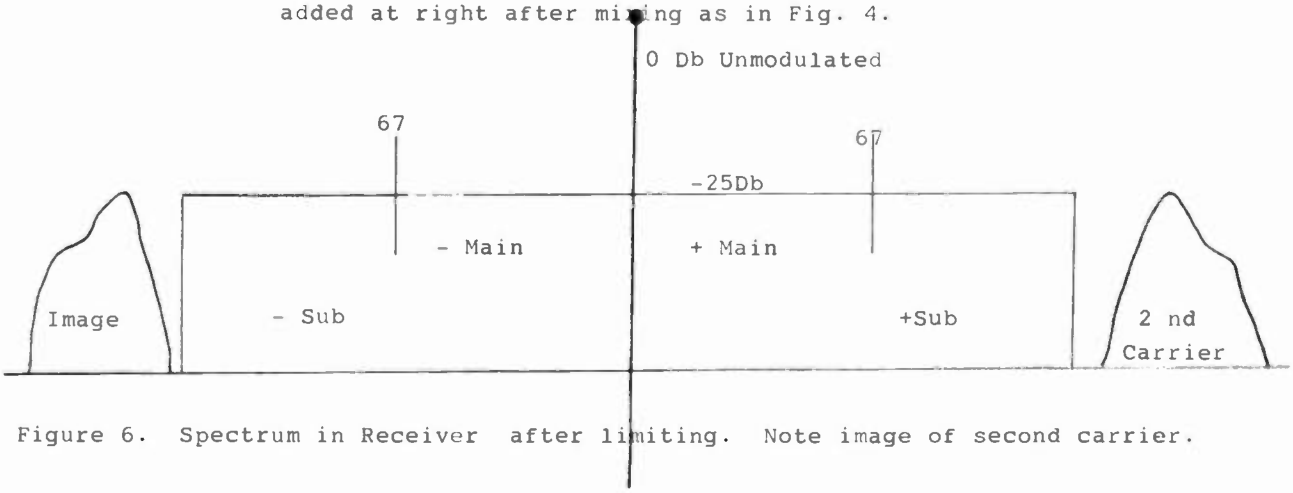

Figure 6. Spectrum in Receiver after limiting. Note image of second carrier.

AUDIO TO RF: A COMPLETELY DIGITAL FM BROADCAST STEREO SYSTEM

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Introduction

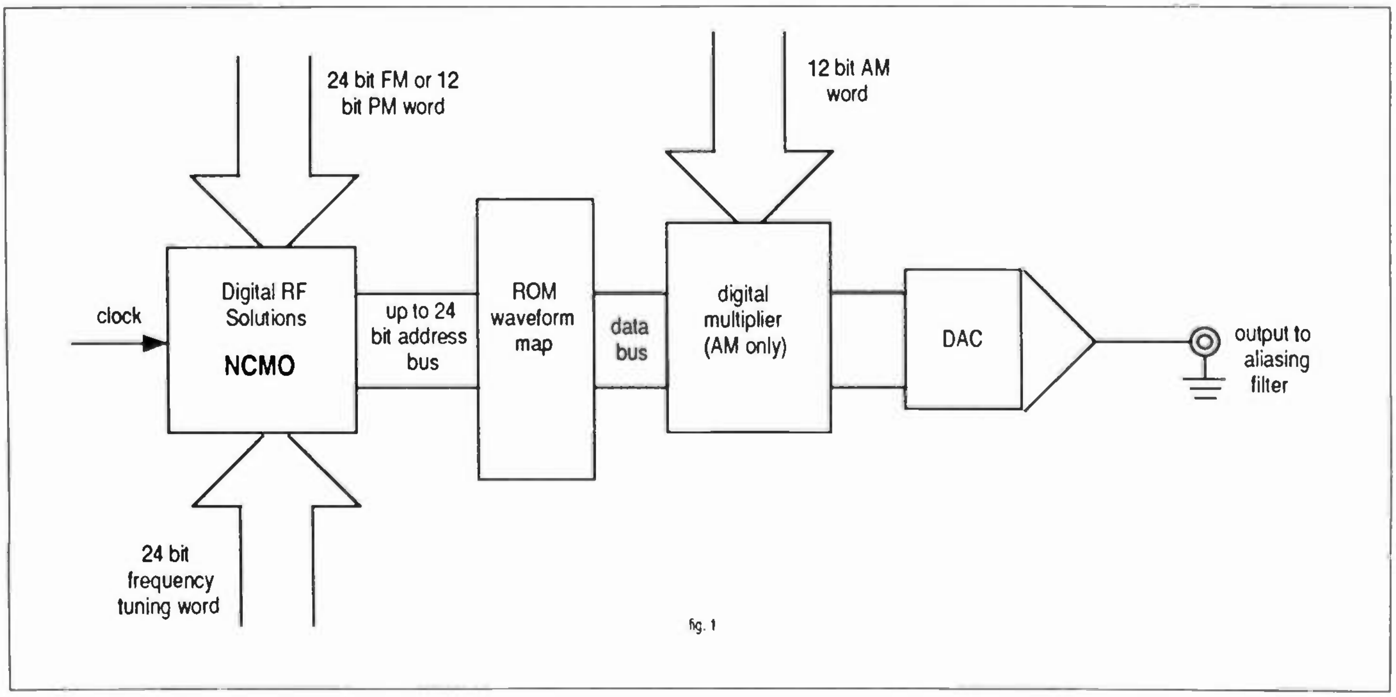
The NCMO™ approach to direct digital synthesis (DDS) has opened a multitude of application possibilities. One of these possibilities exists in the radio broadcast industry. This paper discusses the power of the DDS technique and its potential for the broadcast industry. The system outlined in the following pages is possible to build today with standard products. Refinements are required to construct a working system, but the high level of integration afforded by the NCMO will simplify the design of a completely digital stereo radio broadcast system.

Recent advances in the direct digital synthesizer (DDS) technology now permit digital technology to be used up to RF. The same advantages digital techniques offer to audio solutions can now apply to the two remaining blocks in an FM broadcast chain, the stereo generator and the RF exciter. The basics of DDS are covered in several articles and application notes from Digital RF Solutions Corporation [references 1 and 2].

Direct Digital Synthesis with the NCMO

Unlike analog LCs or crystal oscillators or phase-locked loop (PLL) synthesizers, all parameters of the signal (amplitude, phase, and frequency) are defined by digital numbers. Figure 1 displays a four IC DDS systems implementation using the NCMO as the system core.

4-Chip DDS System Using the NCMO



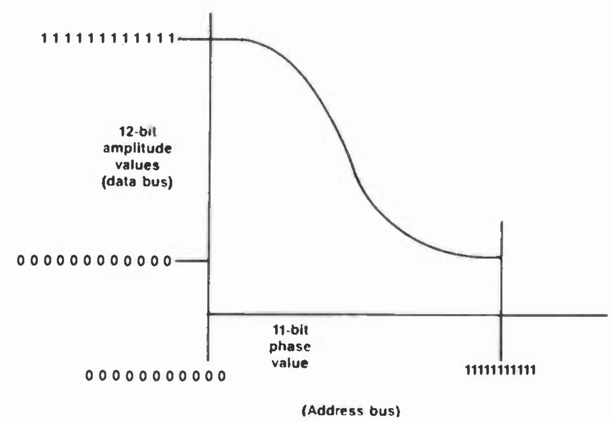


fig. 2

At the heart of the NCMO is the phase accumulator. For fixed clock frequency (assume 20MHz) there is a corresponding fixed phase change for any desired output frequency for every clock cycle. (36 degrees for 2KHz, 1.8 degrees for 100KHz, etc.) The phase accumulator accepts a 24-bit word which results in 24-bit frequency resolution. The NCMO then cyclically accesses ROM locations. The ROM acts as a "waveform map" that simply coverts the the phase position (calculated in the NCMO) to the appropriate instantaneous sinusoidal amplitude value. (figure 2) Stated another way, the ROM address corresponds to time and phase, while ROM data corresponds to the appropriate proper value. The ROM is actually a digital phase-to-amplitude converter.

The ROM typically contains 180 degrees of phase information. Cosine data is preferred to sine data because it is better to have amplitude errors at "peaks and valleys" rather than at zero crossing points. The NCMO clocks the ROM address locations up and then back down to complete a whole cycle. This data is applied to the digital-to-analog-convert (DAC). A 12-bit DAC system requires a ROM with 11 address bits and 12 data bits.

Tuning

The 24-bit frequency word can be presented to the NCMO by three methods: 24-bit parallel, three 8-bit words for microprocessor interface, or by two-bit quadrature serial sequencing allowing interface to optical encoders. These rotary encoders are mounted on a shaft and can be tuned similar to a radio dial. The tuning rate of the serial mode is selected through a three-bit word. The tuning rate of the serial mode is selected through a three-bit word. The 24-bit parallel mode is used when maximum speed is required for center frequency changes, as in frequency hopping radios. The eight-bit mode is used when easy interface to a microprocessor is desired. Finally the serial mode is used when the synthesizer needs to be tuned by a dial.

Twenty four-bit frequency resolution represents approximately 16.8 million possible frequency values. However, if the MSB on the tuning word is set high, the NCMO attempts to synthesize a frequency above the Nyquist Frequency (one-half the clock frequency). This will produce a "negative" frequency response which will appear under the Nyquist value. Consequently, there are actually about 8.4 million possible frequency values under the Nyquist Frequency. The frequency resolution of the DDS synthesizer equals the clock frequency/16.777216 MHz. If the clock frequency is 16.777216 MHz, the NCMO synthesizer will output a signal selected in exactly 1Hz steps. Finally, if this is not sufficient resolution, two

Functional Block Diagram of an FM Stereo Broadcast Facility

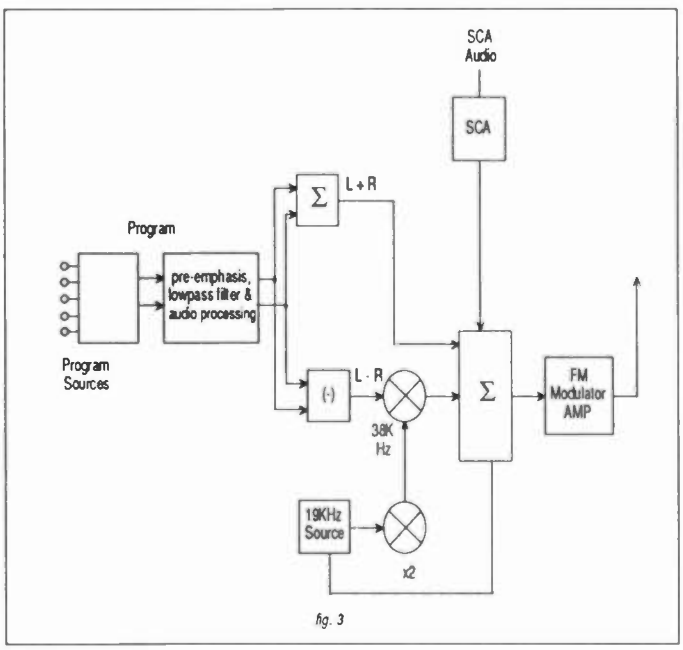


fig. 3

NCMOs can be cascaded for nanohertz resolution with no consequence to phase noise performance.

Modulation

The NCMO has on-board FM and PM capabilities with 24-bit resolution. When the FM mode is selected, the 24-bit modulation port number is simply added to the tuning word before the phase accumulator, thus "pulling" the frequency "up". In the FM mode the modulation and tuning port are interchangeable. Simultaneous FM and PM is possible by frequency modulating the tuning word and phase modulating the modulation port. (if anyone would want to do such a thing). For phase modulation 0 to 360 degree deviation is possible with 4096 phase steps. In the PM mode the modulation word is added to the output of the phase accumulator thus advancing the phase to the appropriate number of degrees. The FM and PM modes are selected by a single pin level.

Digital control of modulation parameters has some very desirable advantages. Since the numbers add to frequency values, frequency deviation can be held constant over the entire tuning range of the synthesizer. Bandwidth limitations are absolute, therefore preventing excessive deviations and over modulation. Furthermore, performance is identical from one unit to another eliminating many production line calibrations and real world operating digressions.

Finally, amplitude modulation can be affected by inserting a digital multiplier between the waveform map ROM and the DAC. The multiplier then adjusts the digital amplitude word applied to the DAC according to the modulating waveform. A four-quadrant multiplier produces a DSB suppressed carrier signal, while a single-quadrant multiplier produces a full carrier AM signal. Of course, AM can be accomplished in a parallel with PM and/or FM for a wide range of vector modulation schemes.

FM BROADCAST APPLICATIONS

Figure 3. illustrates a block diagram of a typical FM stereo broadcast system. The remaining three figures show the same basic system broken into the three basic parts using digital techniques only. The first analog signal is an FM stereo signal with SCA at about 4MHz. This RF signal is then upconverted to the FM channel frequency, filtered and amplified and then transmitted in the conventional manner.

Digitized Audio Signal

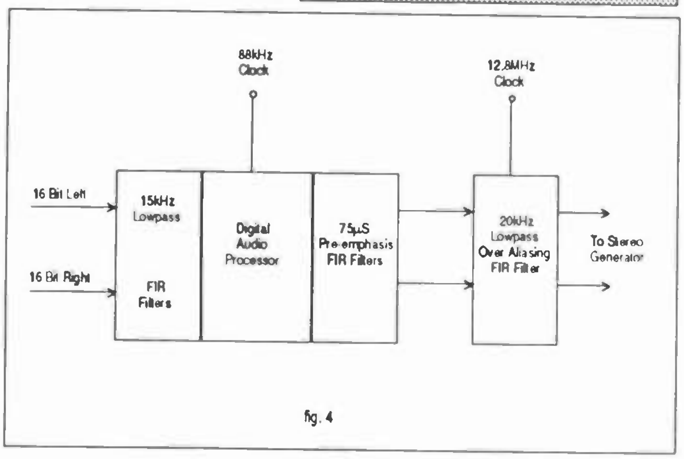


fig. 4

Digital Audio

The introduction of compact disks (CDs) has dramatized the superiority of digital audio over conventional analog techniques. Digital audio is here to stay, from the highest quality recording and broadcast studios to portable consumer CDs. The advantages of CDs and other digital technologies are many, including:

- **Very high dynamic range**
- **No degradation from age or playback wear (as with vinyl or analog magnetic recording)**
- **Outstanding phase linearity and channel balance in equalizers, pre-emphasis networks, compres sors, and other filters**
- **Noiseless mixing consoles and studio-transmitter links**

The filtering and other digital audio functions are commonly performed by digital signal processing techniques (DSP). Finite impulse response filters (FIRs) are capable of performance levels that are impossible with analog techniques. These techniques have been widely publicized.

The studio and audio DSP system can use a relatively low frequency sample rate (perhaps 88KHz). This allows for less expensive A/D converters and standard DSP filters. Also, data systems are easier to interface when timing errors are minimized. The timing errors can be minimized by using lower clock speeds. A high clock speed is required for the stereo generator and exciter to move the alias signals well outside the FM broadcast channel. A very convenient broadcast system clock is 12.8 MHz.

Figure 4. displays the additional low pass FIR filter operating with a 12.8 MHz clock. After this filter the alias signals are shifted up to over 12MHz, well outside the exciter's SAW filter bandpass.

Figure 4. also shows the three necessary audio processing steps for use with this system. FIR filters are used for the 15KHz low-pass and 75µS pre-emphasis circuits. Compressor design can become quite esoteric as anyone involved with radio broadcasting can attest. Digital processing offers many advantages over comparable analog techniques. However, the details of digital compressor design are not within the scope of this paper. Peak limiting is not required with this system! The final maximum frequency deviation is controlled by the nature of the numeric modulation technique.

Digitzed Stereo Generator

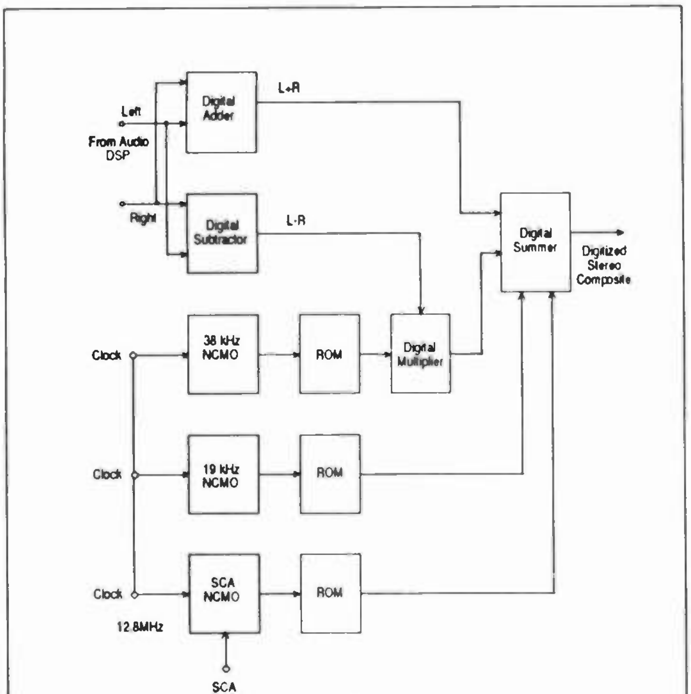


fig. 5

Digitized FM Stereo Exciter

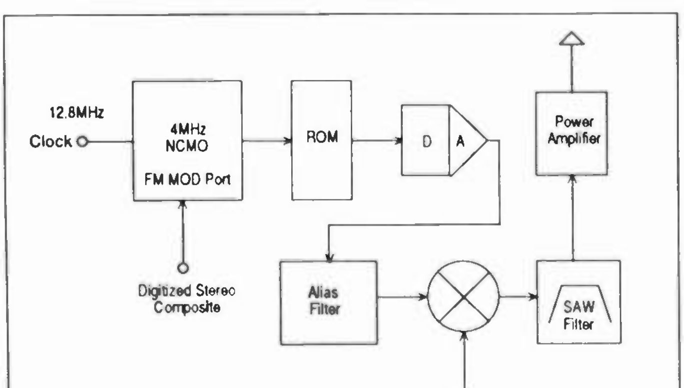


fig. 6

A Direct Digital Stereo Generator

Figure 5. shows a block diagram of a DDS stereo generator. The L+R and L-R digital signals are created with a digital adder and a subtracter respectively. The three digitized subcarriers are synthesized by three separate DDS circuits employing the NCMO™ system. The 38KHz subcarrier is digitally modulated by the L-R signal. If a 16x16 bit four-quadrant multiplier is used, a digitized double sideband suppressed carrier signal is generated. The carrier rejection exceeds 80dB. L-R subcarrier linearity is comparable to a 16 bit CD player!

A second NCMO-ROM generates the 19KHz CW pilot. If the 38 and 19KHz DDS synthesizers are run off the same clock they will be phase-locked. (the phase of either signal is adjustable to 12 bit resolution or <.1°) A third(and possibly more) NCMO can be added for an SCA generator. All subcarrier frequencies can be programmed with .76 Hz resolution using the 12.8MHz clock. The SCA program material is, of course, also digitized. The four digitized signals, L+R, L-R subcarrier, pilot, and SCA are then fed to a digital summer. The output of this summer is a digitized FM stereo composite signal.

A Direct Digital FM Stereo Exciter

Figure 6. illustrates the final steps in the DDS FM Broadcast system. The digitized FM stereo signal is fed to a fourth NCMO synthesizer. This DDS system works at about 4MHz. A two pole low-pass filter with an 5MHz cut-off frequency preceeds an up-conversion mixer. The LO is chosen for the IF to fall on the desired FM channel frequency. The desired FM frequency is defined by a SAW filter with about 500KHz bandpass. The SAW filter also provides proper image rejection and additional alias filtering. The wideband filter response keeps phase distortions to an absolute minimum. Power amplification and transmission follows using conventional techniques. The digital input to the FM modulation port on the NCMO adds Hertz in an exactly linear manner, rather than reactance. Consequently, the frequency deviation is exactly defined by the digital modulating word. This precludes the need for peak limiters.

The filter is narrow enough, however, to act as a frequency alias filter. This effect serves to "smooth" the discrete frequency "jumps" inherent in a sampled FM system. The effect is comparable to "smoothing" discrete amplitude steps in a sampled amplitude waveform. Aliasing works in both the frequency and time domains.

The deviation is absolutely limited to 200KHz if FM modulation bits 3 through 18 are used. The two least significant modulation bits are not used for a 16 bit modulation word. 200KHz maximum deviation will be maintained when a 12.8 MHz exciter clock is used. The worst-case spurious signals will appear close to the desired signal when the signal frequency is tuned to 1/3 or 1/4 the clock frequency or 4.267 and 3.2 MHz respectively. The exact DDS frequency should avoid these worst-case spurs. A 4MHz fundamental signal avoids these spur frequencies and will provide adequate image rejection in the exciter's up conversion process.

STL Considerations

A digital STL will not contribute noise to the broadcast system. It could be inserted before Figure 4, between Figures 4 and 5 or between Figures 5 and 6.

Conclusion

A completely digitized FM broadcast transmission system can now be configured using DDS systems built around the NCMO. All the parameters of the final RF waveform; amplitude, phase and frequency are strictly defined. With multi-bit resolution, the end product will be as good as the defining digitized bits. Spurious signals within the bandpass are typically -75dBc. The system displays outstanding linearity, typically .0035% THD. Linearity is independent of deviation, since the modulating waveform is adding or subtracting HERTZ deviatior and not a reactance. The various phase relationships among the signals are controlled within the constraints of a 16 bit resolution system. The phase error is very difficult to measure with existing test equipment since it would approach perfection. The phase performance would be limited by the analog circuitry.

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OPTIMUM BANDWIDTH FOR FM TRANSMISSION

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ABSTRACT

This paper researches the optimum RF bandwidth for high quality FM transmission. The penalty of increased RF intermodulation due to less "turn around loss" in wider than optimum transmitters is explained. This is especially important to closely spaced transmission facilities and community tower projects using hybrid combining techniques.

Complete audio performance data has been taken for various bandwidth transmission systems to determine the optimum bandwidth, and a complete system performance evaluation of a modern high power transmitter with optimum transmission bandwidth is presented.

INTRODUCTION

For theoretically perfect reception of any frequency modulated (FM) signal, an infinite transmission and reception bandwidth is required. This is due to the nature of FM, which creates an infinite number of sidebands whose structure is determined by the modulation index. In a perfect FM transmitter, the output power remains constant, but as the modulation index changes, the power distribution between the carrier and the sidebands changes.

Practical applications require finite bandwidth restrictions on the FM signal. For the broadcaster, several elements reduce the transmitted bandwidth of the FM signal, including tuned stages in the transmitter grid and output, and the transmitting antenna itself.

For the receiver, the desired signal must be selected, while all others are rejected. This is done primarily by the intermediate frequency (IF) filter. This IF filter is by far the largest contributor to the total RF bandwidth limitation, typically being less than 300 kHz wide (3 dB). Some receivers are available with selectable IF bandwidths of 1 MHz or more. As receiver technology advances, this typical IF bandwidth of less than 300 kHz may very well increase.

In any case, broadcasters should not allow receiver shortcomings to limit their efforts to transmit the best possible RF signal.

There is a wide diversity of opinion among both broadcasters and broadcast equipment manufacturers as to the required RF bandwidth for quality FM transmission. At first glance, the "more is better" assumption is likely to prevail. But a closer look reveals some practical considerations which show a need to limit the transmission bandwidth to reduce other problems, especially the ever increasing potential for RF intermodulation in broadcast transmitters.

Therefore, the purpose of this paper is to determine how much bandwidth is required for low distortion FM transmission, and at what bandwidth the point of diminished returns regarding distortion improvement is reached.

Bandwidth Limitations

Several factors contribute to limit the transmitted RF bandwidth of an FM transmission facility. Often, the limiting factor is the antenna system itself. For community tower applications, wideband panel antennas are available. In this case, the hybrid combiners and cavity tuned filters are predominantly the narrowest elements in the transmission path.

The transmitter also plays a role in the total RF bandwidth of a station. Several key areas determine the bandwidth limitations of a transmitter.

A solid state broadcast transmitter is rarely the limiting factor for RF bandwidth. It should be much wider than the antenna, combiners or cavity tuned filters. For tube transmitters the story is much more complex. The output of a high power tube transmitter consists of a frequency selective network in the form of a tuned cavity. The bandwidth of the cavity depends on its construction, the amount of tube output capacitance, and how heavily it is loaded.

The output (cavity) bandwidth is often considered the limiting factor for the whole transmitter. Oddly enough, this is not the case for the grid driven amplifier. The large grid input capacitance of a power vacuum tube causes the loaded Q of the grid circuit to be even higher than the output (1), (2). This fact is often ignored because the grid is driven into saturation which partially masks the amplitude variations of the grid matching network. The popular method of measuring transmitter bandwidth with a network analyzer is somewhat misleading, since the measured 3 dB amplitude bandwidth does not completely account for the grid circuit effects due to saturation. The non-linear response of the power tube further effects the response, especially close to the carrier frequency. This is why accurate predictions of the transmitter 3 dB bandwidth cannot be made by looking at synchronous AM performance, or visa-versa. The amplitude response of the transmitter can be made flatter over a ±75 kHz deviation from carrier due to heavy saturation, heavy loading, and tube impedance non-linearity (3), (4). Measuring this "0.1 dB" bandwidth (-45 dB synchronous AM) proves to be inaccurate when attempting to predict the 3dB bandwidth from this information. For a properly adjusted transmitter, the synchronous AM performance tends to predict a wider than actual 3 dB bandwidth.

Audio performance is also not completely predictable from a measured transmitter amplitude response. The problem arises from the group delay variations (phase response) of the grid circuit and the non-linear nature of the final tube, which can have serious effects on the distortion performance of the entire transmitter. Group delay variations degrade the composite amplitude response, which in turn limits stereo separation. A properly designed, broadband grid matching network is essential for proper operation of the entire transmitter. Even if the output bandwidth were not limited, the grid circuit could seriously affect the transmitter's performance.

This degradation due to phase response is true for any tuned circuit, even if that stage is run into saturation. Therefore it makes sense to eliminate as many tuned stages as possible. This is why a wideband, solid state exciter and intermediate power amplifier (IPA) are advantageous in high power FM transmitters, even though the output stage uses a tube (1), (2).

WHY LIMIT BANDWIDTH?

If there were only one radio signal being transmitted at any given time, there would be no need to limit the bandwidth. However, any time two signals are present, there exists the possibility of RF intermodulation between them. All that is required is a non-linear device acting as a mixer, which creates two more intermodulation products.

The transmitter final amplifier is that non-linear active device. If any other frequency finds its way back into the output stage, RF intermodulation will occur. This mixing will have some conversion loss, referred to as "turn-around-loss". There are three main contributors to the total turn-around-loss (5). They are:

1. The in-band conversion loss of the non-linear device.

2. The attenuation of the interfering signal due to the selectivity of the output stage.

3. The attenuation of the resulting IM products due to the selectivity of the output stage.

Notice that 2. and 3. relate to the transmitter output bandwidth. This clearly shows the desirability to have as much selectivity as possible in the output stage. This will be a design trade-off between system modulation performance and immunity from RF intermodulation. It is important to note that the broadband nature of a solid state broadcast transmitter makes its susceptibility to RF intermodulation greater than a tube/cavity output stage.

Broadcast engineers are faced with the following questions:

1. What is the optimum bandwidth for FM transmission?

2. At what point does the performance become acceptable?

3. What is the limit of diminishing returns where you pick up basically no more modulation performance, but continue to "open up the door" to increased RF intermodulation?

TESTING RF BANDWIDTH PERFORMANCE

How is the optimum bandwidth determined? There are models available (6),(7) to predict distortion performance, but these require that the transfer function of the network is known and assumed to be passive. It is practically impossible to model an FM broadcast transmitter operating class C, due to its nonlinear transfer function.

A straightforward empirical alternative is to measure the performance degradation of a "perfect" modulator when it is passed through a passive band limiting network. A real broadcast transmitter is not practical for this test, as there is only a very limited range of bandwidth variation available, and determining its true bandwidth is difficult due to grid saturation effects.

A test cavity was constructed to simulate the effects of band limiting. The tuning and loading range was sufficient to allow bandwidth testing from 400 kHz to 3 MHz (-3 dB). While the effects of the grid circuit were not seen, the output bandwidth effects were very accurately modeled. This was useful for several reasons. First, it showed the performance degradation caused by various bandwidth limitations. Second, it shows at what bandwidth performance ceased to improve. Third, it provides a good basis to compare to a real broadcast transmitter. Figure 1 shows the physical construction of this test cavity.

The resulting data gives a clearer insight into the effects of the grid circuit and the nonlinear effects of the output tube, based on actual performance vs. measured bandwidth of a real transmitter. It also shows that 3 dB bandwidth is not necessarily a good measure of synchronous AM performance due to the more complex response of the entire transmitter design.

The Test Equipment

Before a determination of performance degradation can be made, a benchmark must exist to define the desired goal, or "perfect" FM modulation. In a wideband RF environment, the system performance is limited only by the FM exciter used (the modulator), and the receiver (demodulator). The accuracy of the test is limited by the distortion, noise, and composite amplitude response of this test equipment.

Figure 2 shows the performance of the Broadcast Electronics, Inc. model FX-50, 50 watt FM Exciter, measured with the Belar Electronics model FMM-2 FM Modulation Monitor and model FMS-2 FM Stereo Modulation Monitor. Audio generation and measurement was done with the Audio Precision System One audio test set. Stereo encoding was accomplished with the Broadcast Electronics model FS-30 FM Stereo Generator.

This combination provided a guaranteed signal to noise ratio of -90 dB minimum, THD+N, SMPTE and CCIF IMD performance better than 0.005%, composite amplitude response of better than ±0.025 dB, composite phase response of ±0.1 degree, and stereo separation of 60 dB, 30 Hz to 5 kHz, greater than 52 dB, 5 kHz to 15 kHz.

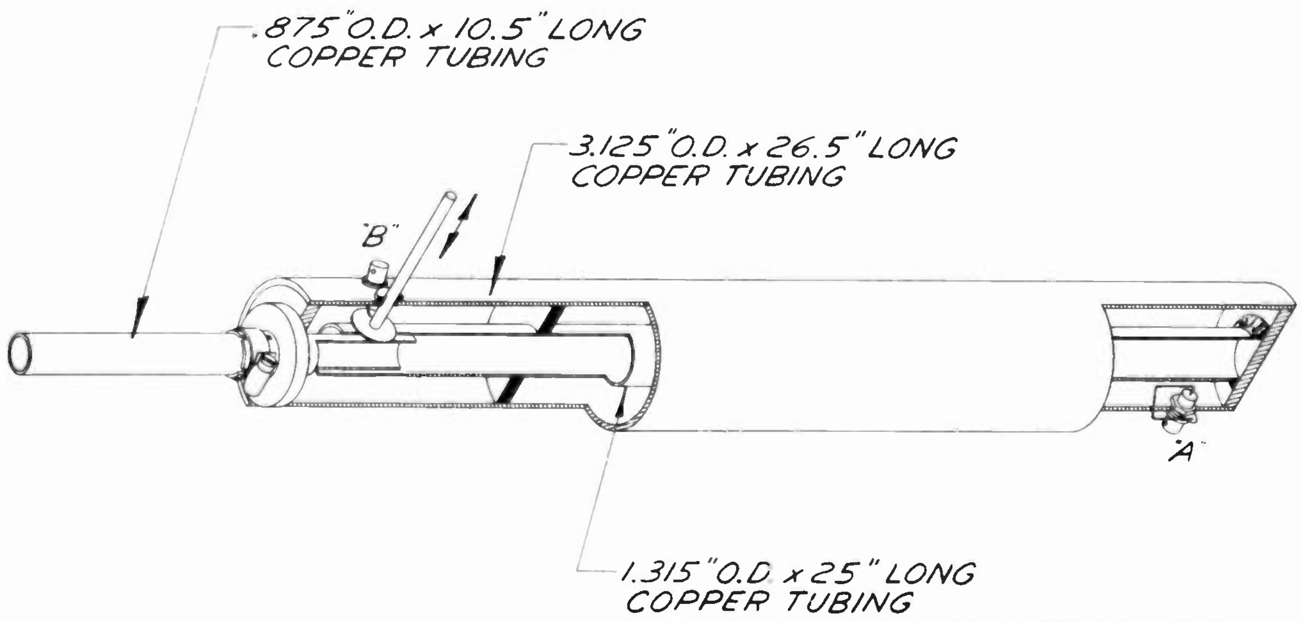
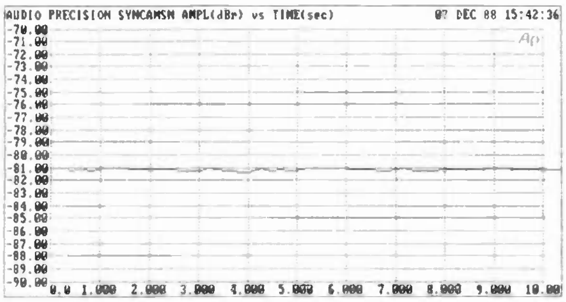
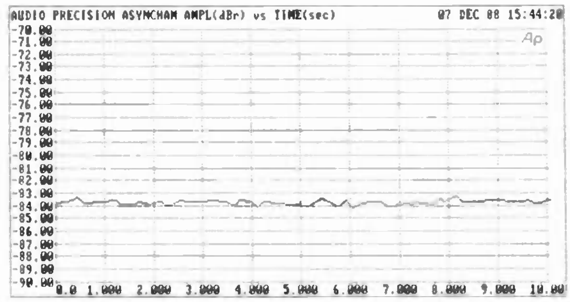


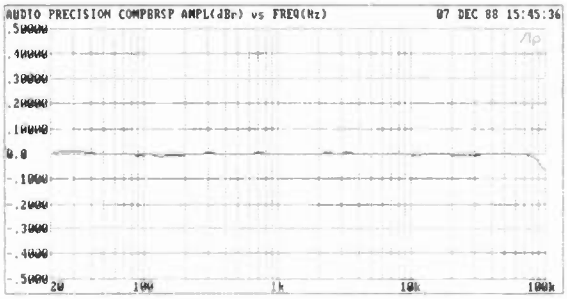
FIGURE 1. TEST CAVITY CONSTRUCTION



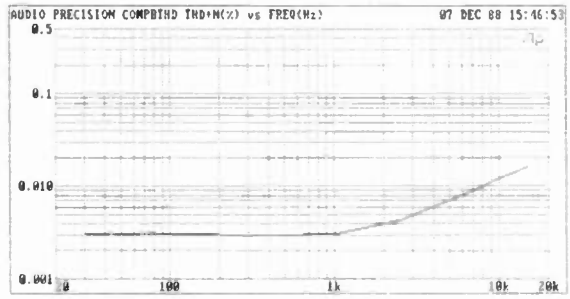
FX50 SYNCHRONOUS AM SIGNAL TO NOISE vs. TIME
10 SECOND SWEEP WITH 75uS DEEMPHASIS BELOW EQUIVALENT
100% AM MODULATION WITH +/-75KHz FM MODULATION AT 400Hz



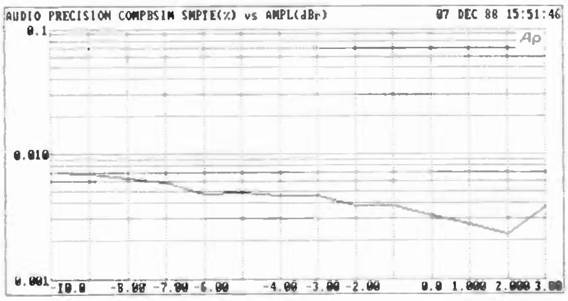
FX50 ASYNCHRONOUS AM SIGNAL TO NOISE vs. TIME
10 SECOND SWEEP WITH 75uS DEEMPHASIS BELOW EQUIVALENT
100% AM MODULATION



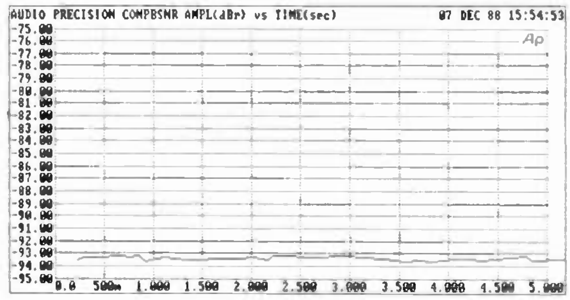
FX50 COMPOSITE FREQUENCY RESPONSE
20Hz TO 100KHz, GENERATOR/ANALYZER NORMALIZED
BELAR FMM-2 DEMODULATOR, STEREO OUTPUT



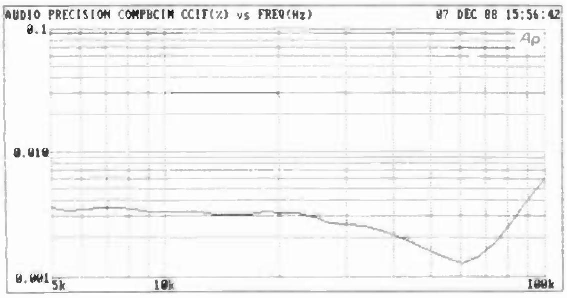
FX50 COMPOSITE THD+N vs. FREQUENCY
30 Hz TO 15 KHz, 75uS DEEMPHASIS
BELAR FMM-2 DEMODULATOR, BALANCED OUTPUT



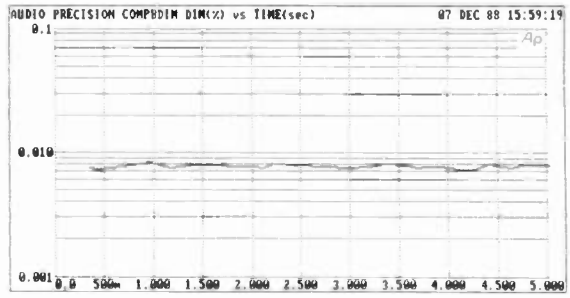
FX50 COMPOSITE SMPTE IMD vs. LEVEL
60Hz/7KHz, 1:1 RATIO FROM 3dB OVERMODULATION TO
-10dB UNDERMODULATION (0dBr = +/-75KHz)



FX50 COMPOSITE SIGNAL TO NOISE RATIO vs. TIME
5 SECOND SWEEP, BELAR FMM-2 DEMODULATOR,
75uS DEEMPHASIS, BALANCED OUTPUT

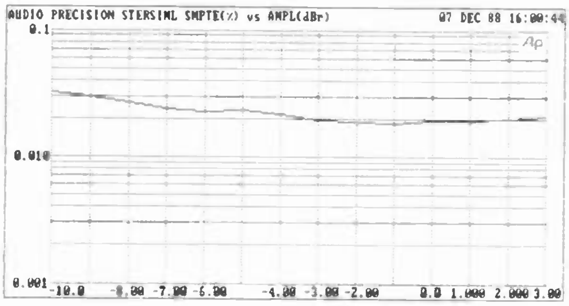


FX50 COMPOSITE CCIF IMD vs. FREQUENCY
TWIN TONE 1:1 RATIO FROM 99/100kHz TO 5/4kHz
AT +/- 75KHz DEVIATION, NO DEEMPHASIS

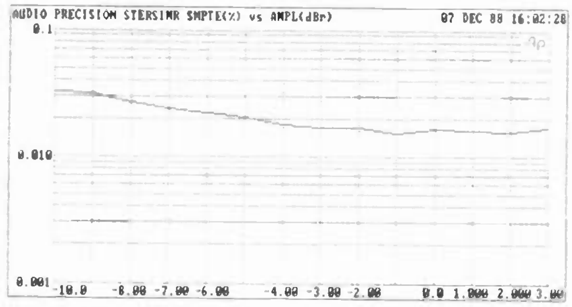


FX50 COMPOSITE DIM/TIM vs. TIME
5 SECOND SWEEP, 3.15KHz SQARE WAVE/ 15KHz SINE WAVE
NO PREEMPHASIS, NO DEEMPHASIS

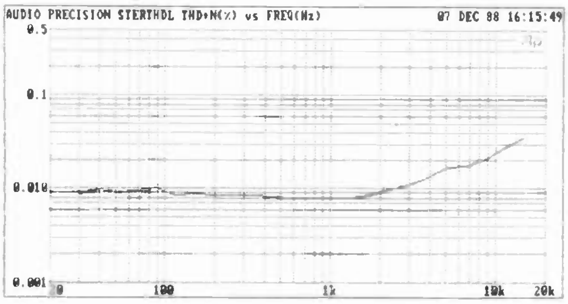
FIGURE 2. FX-50 FM EXCITER PERFORMANCE DATA
(Sheet 1 of 3)



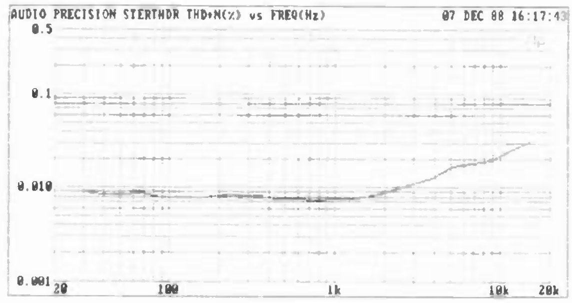
FX50/FS30 LEFT CHANNEL SMPTE IMD vs. LEVEL
60Hz/7KHz, 4:1 RATIO 75uS PREEMPHASIS, DEEMPHASIS
FROM 3dB OVERMODULATION TO -10dB BELOW 100% (0dB=100%)



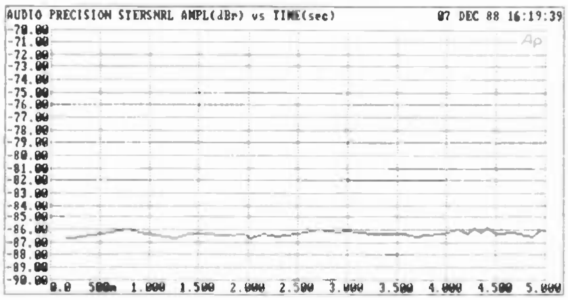
FX50/FS30 RIGHT CHANNEL SMPTE IMD vs. LEVEL
60Hz/7KHz, 4:1 RATIO 75uS PREEMPHASIS, DEEMPHASIS
FROM 3dB OVERMODULATION TO -10dB BELOW 100% (0dB=100%)



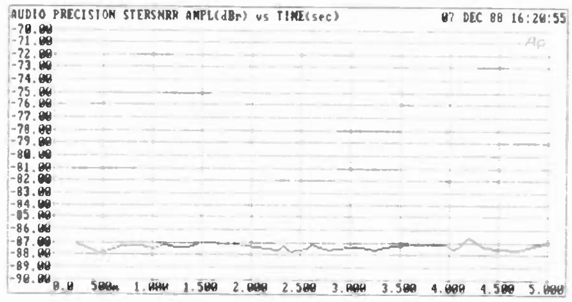
FX50/FS30 LEFT CHANNEL THD+N vs. FREQUENCY
30Hz TO 15KHz, 75uS PREEMPHASIS/DEEMPHASIS
BELAR FMM-2 DEMODULATOR, FMS-2 STEREO DECODER



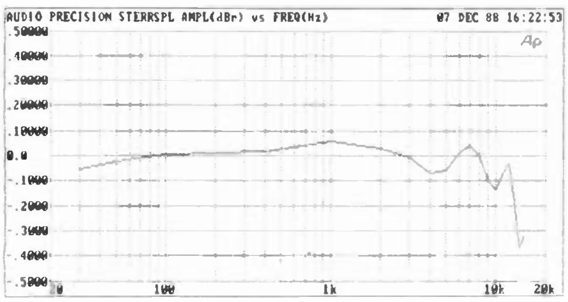
FX50/FS30 RIGHT CHANNEL THD+N vs. FREQUENCY
30Hz TO 15KHz, 75uS PREEMPHASIS/DEEMPHASIS
BELAR FMM-2 DEMODULATOR, FMS-2 STEREO DECODER



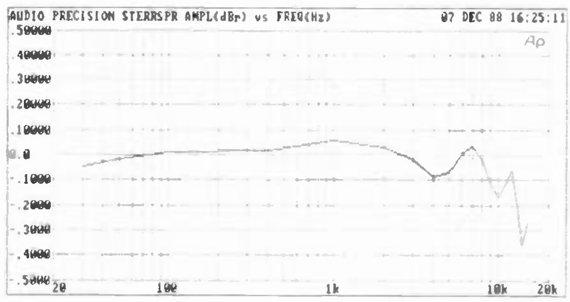
FX50/FS30 LEFT CHANNEL FM SIGNAL TO NOISE RATIO
5 SECOND SWEEP, 75uS DEEMPHASIS
BELAR FMM-2 DEMODULATOR, FMS-2 STEREO DECODER



FX50/FS30 RIGHT CHANNEL FM SIGNAL TO NOISE RATIO
5 SECOND SWEEP, 75uS DEEMPHASIS
BELAR FMM-2 DEMODULATOR, FMS-2 STEREO DECODER

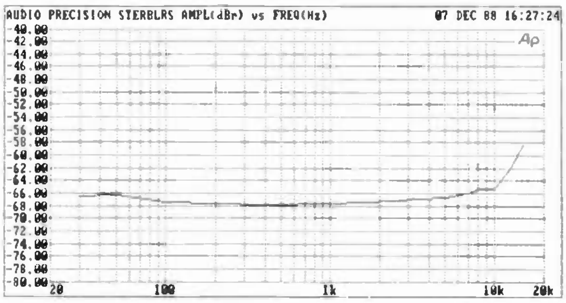


FX50/FS30 LEFT CHANNEL FREQUENCY RESPONSE
30Hz TO 15KHz, GENERATOR EQUALIZED FOR 75uS DEEMPHASIS

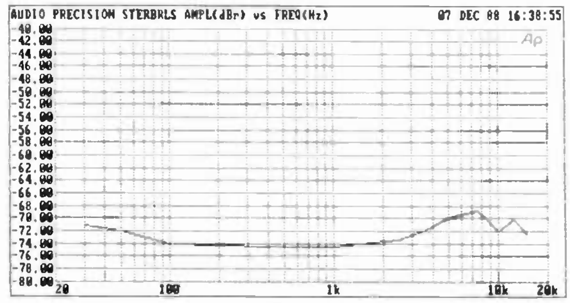


FX50/FS30 RIGHT CHANNEL FREQUENCY RESPONSE
30Hz TO 15KHz, GENERATOR EQUALIZED FOR 75uS DEEMPHASIS

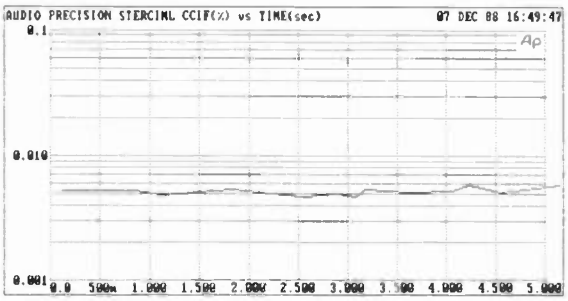
FIGURE 2. FX-50 FM EXCITER PERFORMANCE DATA
(Sheet 2 of 3)



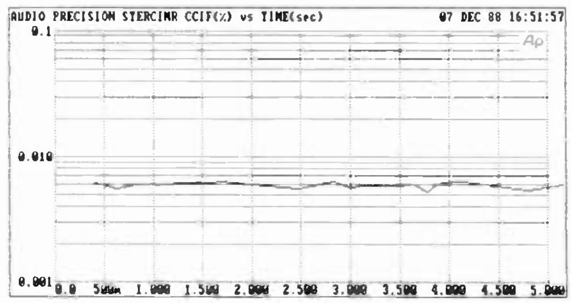
FX50/FS30 STEREO SEPARATATION, LEFT TO RIGHT
BELAR FMM-2 DEMODULATOR, FMS-2 STEREO DECODER



FX50/FS30 TRANSMITTER STEREO SEPARATATION, RIGHT TO LEFT
BELAR FMM-2 DEMODULATOR, FMS-2 STEREO DECODER



FX50/FS30 LEFT CHANNEL CCIF IMD vs. TIME
5 SECOND SWEEP, TWIN TONE 15KHz/14KHz 1:1
75uS PREEMPHASIS, NO DEEMPHASIS



FX50/FS30 RIGHT CHANNEL CCIF IMD vs. TIME
5 SECOND SWEEP, TWIN TONE 15KHz/14KHz 1:1
75uS PREEMPHASIS, NO DEEMPHASIS

FIGURE 2. FX-50 FM EXCITER PERFORMANCE DATA
(Sheet 3 of 3)

The Test Setup

Figure 3 shows the setup used to test the performance degradation of the FX-50 Exciter caused by various bandwidth restrictions. The Audio Precision System One test set was used to measure the audio performance. This allowed a very complete and consistent set of data to be compiled for each bandwidth test.

The System One audio oscillator fed either the FS-30 stereo generator for the stereo performance tests, or the FX-50 composite input directly for the baseband composite performance tests.

The output of the FX-50 was connected to the variable bandwidth test cavity. The cavity was adjusted for the desired -3 dB bandwidths of 400 kHz, 600 kHz, 800 kHz, 1 MHz, 1.5 MHz, 2 MHz, and 3 MHz. Figure 4 shows the amplitude and group delay responses of the test cavity at each setting.

The cavity was loaded by a 50 ohm, 20 dB attenuator, and a sample was connected to the FMM-2. The de-emphasized audio and wideband composite outputs were used for the composite tests. The composite baseband was also used to drive the FMS-2, and the Tektronix model 7L5 spectrum analyzer.

The decoded left and right outputs of the FMS-2 were used for the stereo performance tests.

PERFORMANCE DATA

Figure 5 - Synchronous AM

Figure 5 shows the synchronous AM performance of the test cavity, adjusted for the various bandwidths. Very close correlation between test results and computer modelling of predicted synchronous AM performance was obtained (3). This is because the band limiting network is completely passive, and can be accurately modeled.

Based on this data, better than 40 dB of synchronous AM should be achieved with only 800 kHz of RF bandwidth. This passive representation is only an approximate method of predicting synchronous AM performance.

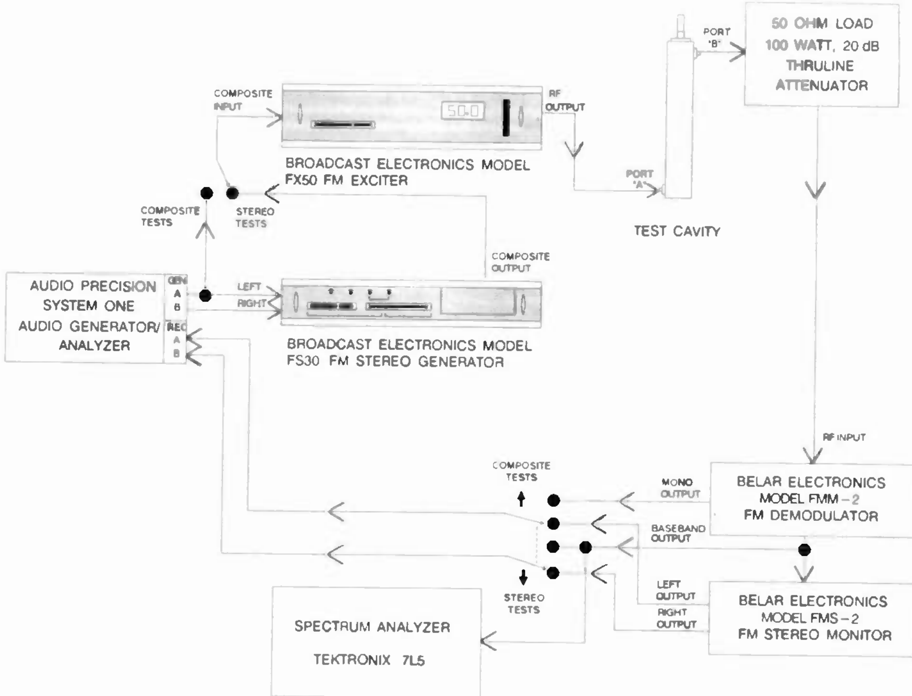


FIGURE 3. OPTIMUM BANDWIDTH TEST SETUP

Figure 6 - Asynchronous AM

Figure 6 confirms that there is no change in Asynchronous AM signal to noise ratio with bandwidth.

Figure 7 - Composite Frequency Response

A dramatic degradation in composite frequency response occurs below 600 kHz. Above 800 kHz to 1 MHz, very little improvement in response is seen. The effects of this parameter are more clearly illustrated by the stereo separation tests.

Figure 8 - Composite THD+N

Figure 8 shows the rise in THD+N with frequency as the bandwidth is varied. With more than 600 KHz bandwidth, the THD+N is better than 0.1%, 30 Hz to 15 kHz.

There is virtually no improvement in performance above 1.5 MHz bandwidth, as shown in the second graph.

Figure 9 - Composite SMPTE IMD

Even at 400 kHz bandwidth, SMPTE IMD is better than 0.1% (measured at 0.05%), and crosses the 0.01% mark at 1 MHz RF bandwidth.

This test is actually SMPTE IMD vs. Level, which shows the IMD performance from 10 dB below 100% modulation to 3 dB above 100% modulation. SMPTE IMD is specified at 100% modulation (Ø dBr the horizontal axis of figure 9A and 9B).

Figure 10 - Composite FM Signal to Noise Ratio

As expected, no change in composite FM signal to noise ratio was observed as bandwidth was varied.

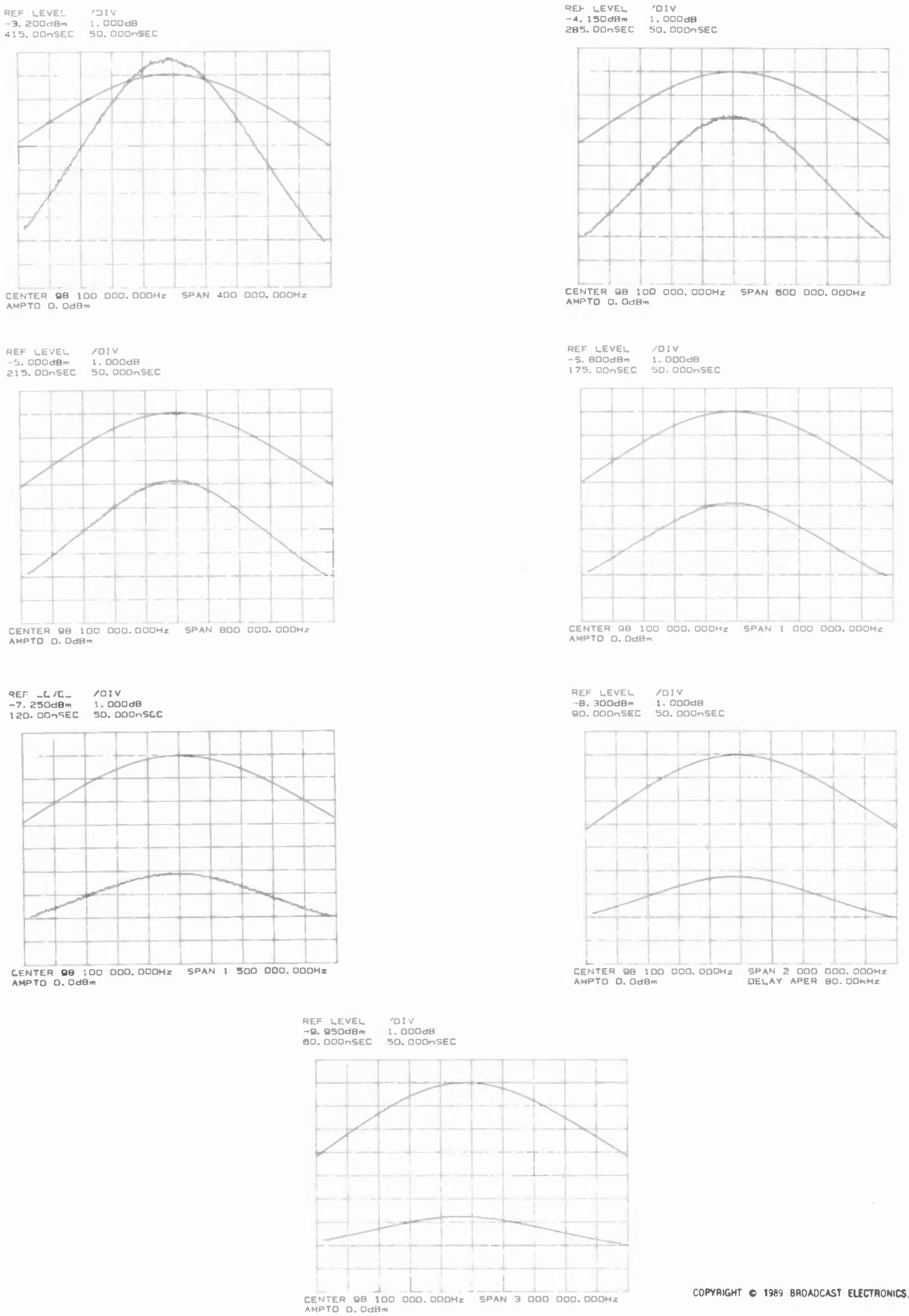
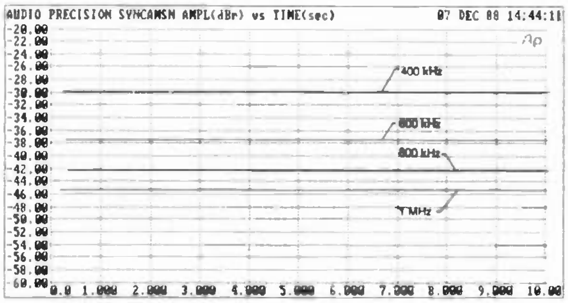
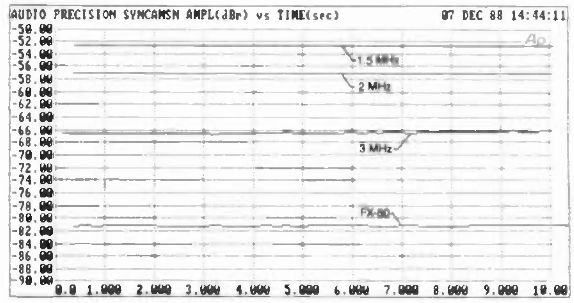


FIGURE 4. TEST CAVITY AMPLITUDE/GROUP DELAY SWEEPS vs. BANDWIDTH SETTING.

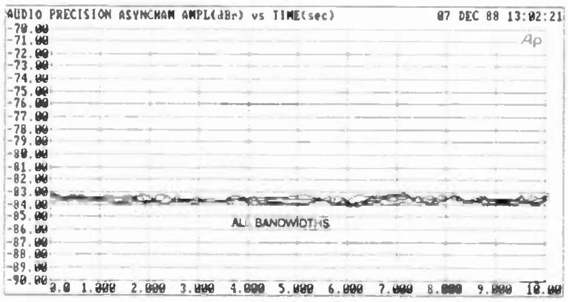


TRANSMITTER SYNCHRONOUS AM SIGNAL TO NOISE vs. TIME
10 SECOND SWEEP WITH 75uS DEEMPHASIS BELOW EQUIVALENT
100% AM MODULATION WITH +/-75KHz FM MODULATION AT 400Hz



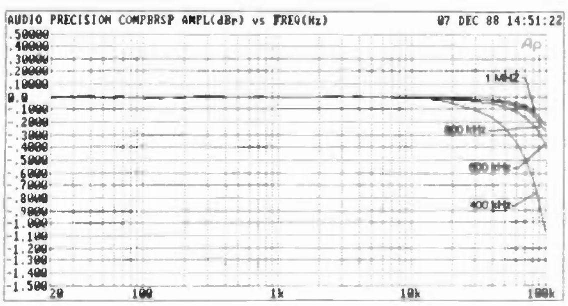
TRANSMITTER SYNCHRONOUS AM SIGNAL TO NOISE vs. TIME
10 SECOND SWEEP WITH 75uS DEEMPHASIS BELOW EQUIVALENT
100% AM MODULATION WITH +/-75KHz FM MODULATION AT 400Hz

FIGURE 5

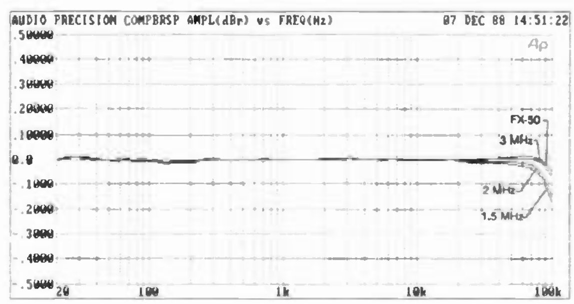


TRANSMITTER ASYNCHRONOUS AM SIGNAL TO NOISE vs. TIME
10 SECOND SWEEP WITH 75uS DEEMPHASIS BELOW EQUIVALENT
100% AM MODULATION

FIGURE 6

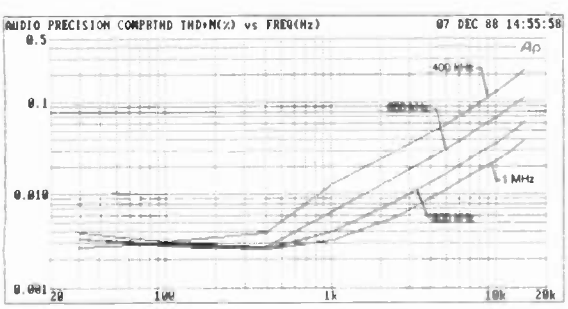


TRANSMITTER COMPOSITE FREQUENCY RESPONSE
20Hz TO 100KHz, GENERATOR/ANALYZER NORMALIZED
BELAR FMM-2 DEMODULATOR, STEREO OUTPUT

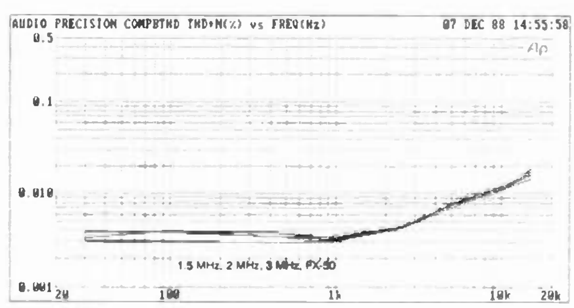


TRANSMITTER COMPOSITE FREQUENCY RESPONSE
20Hz TO 100KHz, GENERATOR/ANALYZER NORMALIZED
BELAR FMM-2 DEMODULATOR, STEREO OUTPUT

FIGURE 7

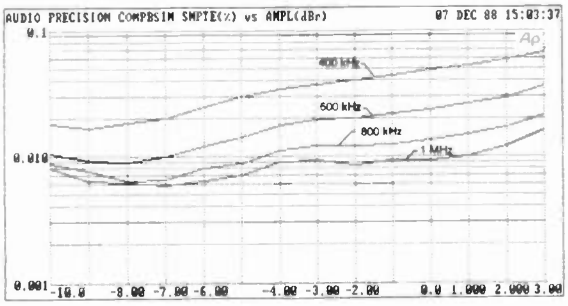


TRANSMITTER COMPOSITE THD+N vs. FREQUENCY
30 Hz TO 15 kHz, 75uS DEEMPHASIS
80 kHz AUDIO BANDWIDTH

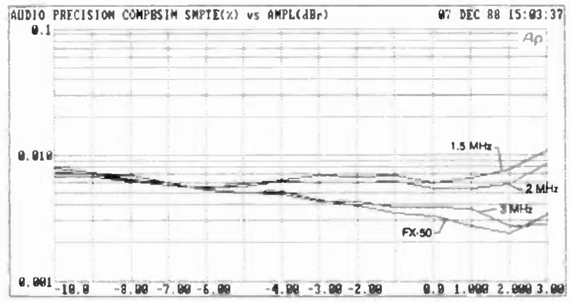


TRANSMITTER COMPOSITE THD+N vs. FREQUENCY
30 Hz TO 15 kHz, 75uS DEEMPHASIS
80 kHz AUDIO BANDWIDTH

FIGURE 8

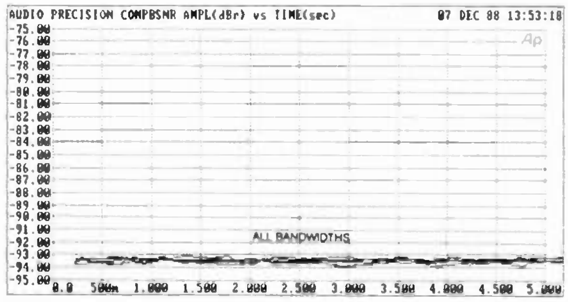


TRANSMITTER COMPOSITE SMPTE IMD vs. LEVEL
60Hz/7KHz. 1:1 RATIO FROM 3dB OVERMODULATION TO
-10dB UNDERMODULATION (0dBr = +/-75KHz)



TRANSMITTER COMPOSITE SMPTE IMD vs. LEVEL
60Hz/7KHz. 1:1 RATIO FROM 3dB OVERMODULATION TO
-10dB UNDERMODULATION (0dBr = +/-75KHz)

FIGURE 9



TRANSMITTER COMPOSITE SIGNAL TO NOISE RATIO vs. TIME
5 SECOND SWEEP. BELAR FMM-2 DEMODULATOR
75uS DEEMPHASIS. BALANCED OUTPUT

FIGURE 10

Figure 11 - Composite CCIF IMD

Surprising results were obtained during this test. Figure 11 shows little change in CCIF IMD performance as bandwidth is varied. Upon closer examination, it was found to be because the test tone is comprised of equal amplitude components, which keeps the individual modulation indexes low, thereby reducing the bandwidth required for low distortion.

Figure 12 - Composite DIM/TIM

There is virtually no change in DIM/TIM performance vs. bandwidth. In fact, the FX50 measurement of 0.008% turns out to be noise limited. No IM products could be found by spectrum analysis.

Figure 13 - Stereo SMPTE IMD

The stereo SMPTE IMD performance is better than 0.05% with 600 kHz or more bandwidth. Very little improvement is noticed above 1 MHz bandwidth.

Figure 14 - Stereo THD+N

There is practically no improvement above 1 MHz bandwidth, with better than 0.1% performance, even at the 600 kHz mark.

Figure 15 - Stereo FM Signal to Noise Ratio

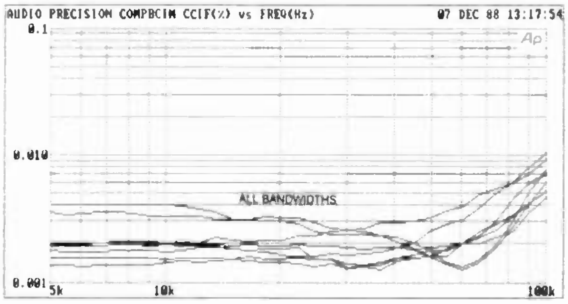
As expected, there was no change in stereo signal to noise ratio with bandwidth.

Figure 16 - Stereo Frequency Response

Stereo amplitude response is not effected with at least 400 kHz of RF bandwidth.

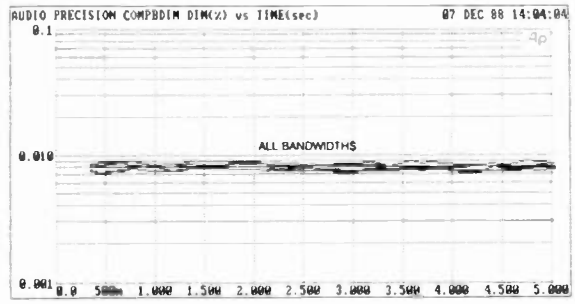
Figure 17 - Stereo Separation

This is an excellent example of the bandwidth effect on composite frequency response. At 400 kHz, separation is limited to slightly better than 40 dB, crossing the 50 dB performance mark at about 700 kHz bandwidth.



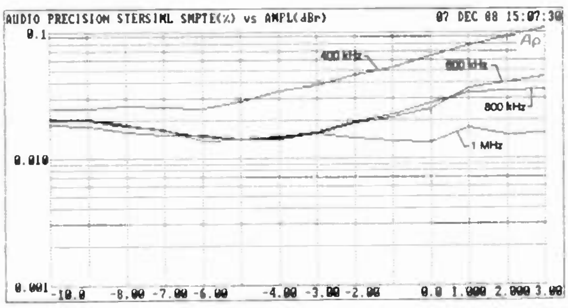
TRANSMITTER COMPOSITE CCIF IMD vs. FREQUENCY
TWIN TONE 1:1 RATIO FROM 99/100 KHz TO 5/4 KHz
AT +/- 75 KHz DEVIATION, NO DEEMPHASIS

FIGURE 11

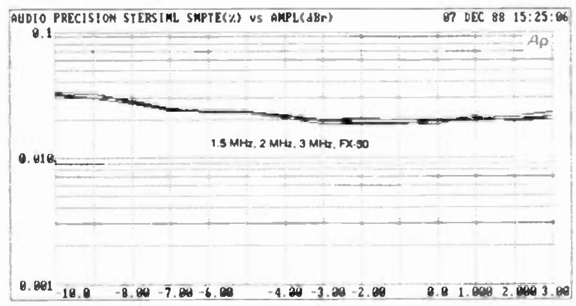


TRANSMITTER COMPOSITE DIM/TIM vs. TIME
5 SECOND SWEEP, 3.15 KHz SQUARE WAVE/ 15 KHz SINE WAVE
NO PREEMPHASIS, NO DEEMPHASIS

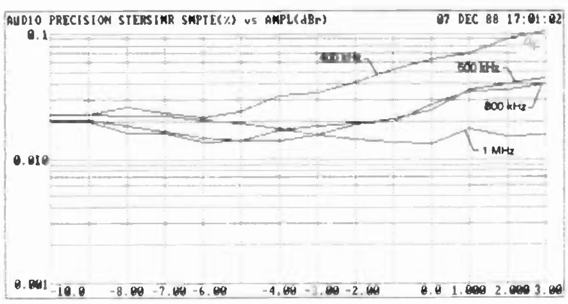
FIGURE 12



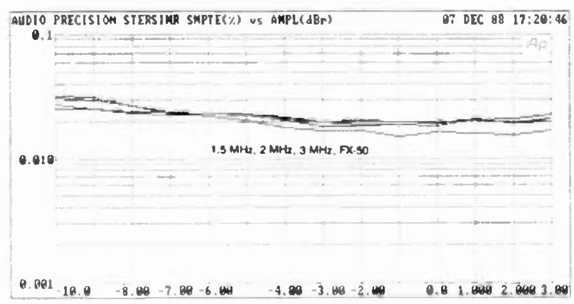
TRANSMITTER LEFT CHANNEL SMPTE IMD vs. LEVEL
60Hz/7KHz, 4:1 RATIO 75uS PREEMPHASIS, DEEMPHASIS
FROM 3dB OVERMODULATION TO -10dB BELOW 100% (0dB=100%)



TRANSMITTER LEFT CHANNEL SMPTE IMD vs. LEVEL
60Hz/7KHz, 4:1 RATIO 75uS PREEMPHASIS, DEEMPHASIS
FROM 3dB OVERMODULATION TO -10dB BELOW 100% (0dB=100%)

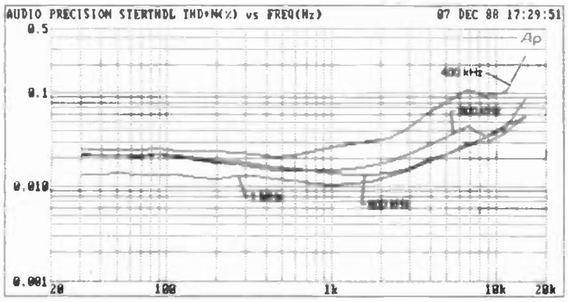


TRANSMITTER RIGHT CHANNEL SMPTE IMD vs. LEVEL
60Hz/7KHz, 4:1 RATIO 75uS PREEMPHASIS, DEEMPHASIS
FROM 3dB OVERMODULATION TO -10dB BELOW 100% (0dB=100%)

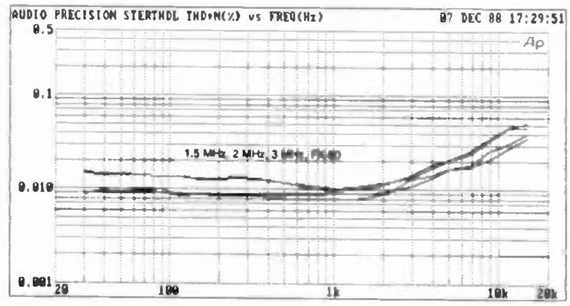


TRANSMITTER RIGHT CHANNEL SMPTE IMD vs. LEVEL
60Hz/7KHz, 4:1 RATIO 75uS PREEMPHASIS, DEEMPHASIS
FROM 3dB OVERMODULATION TO -10dB BELOW 100% (0dB=100%)

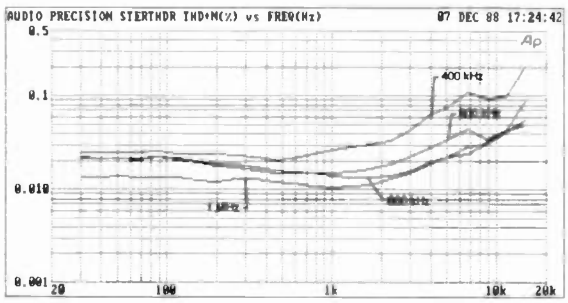
FIGURE 13



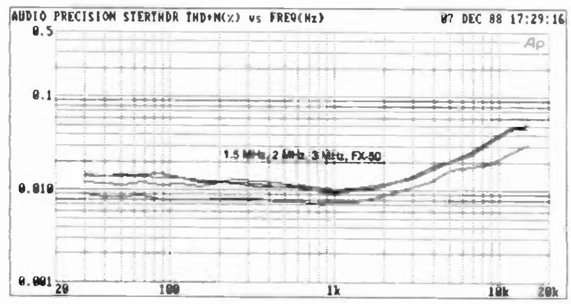
TRANSMITTER LEFT CHANNEL THD+N vs. FREQUENCY
30Hz TO 15KHz, 75uS PREEMPHASIS/DEEMPHASIS
BELAR FMM-2 DEMODULATOR, FMS-2 STEREO DECODER



TRANSMITTER LEFT CHANNEL THD+N vs. FREQUENCY
30Hz TO 15KHz, 75uS PREEMPHASIS/DEEMPHASIS
BELAR FMM-2 DEMODULATOR, FMS-2 STEREO DECODER

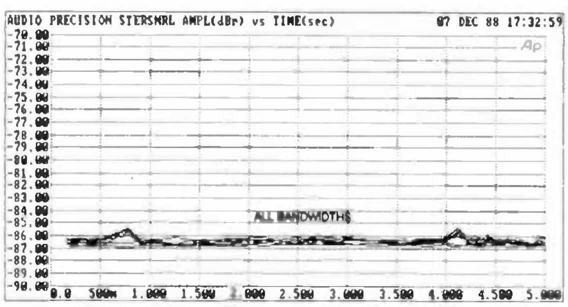


TRANSMITTER RIGHT CHANNEL THD+N vs. FREQUENCY
30Hz TO 15KHz, 75uS PREEMPHASIS/DEEMPHASIS
BELAR FMM-2 DEMODULATOR, FMS-2 STEREO DECODER

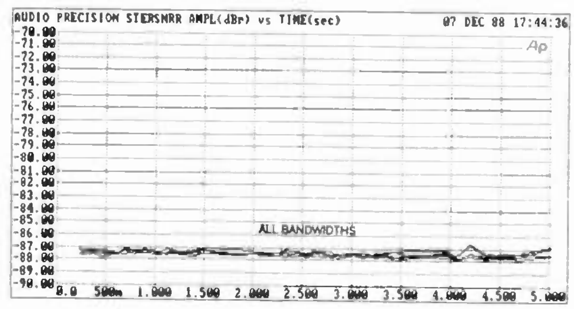


TRANSMITTER RIGHT CHANNEL THD+N vs. FREQUENCY
30Hz TO 15KHz, 75uS PREEMPHASIS/DEEMPHASIS
BELAR FMM-2 DEMODULATOR, FMS-2 STEREO DECODER

FIGURE 14

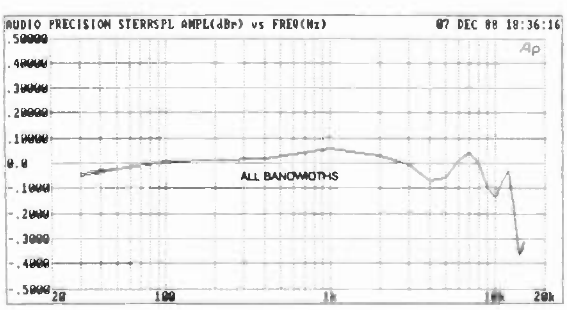


TRANSMITTER LEFT CHANNEL FM SIGNAL TO NOISE RATIO
5 SECOND SWEEP, 75uS DEEMPHASIS
BELAR FMM-2 DEMODULATOR, FMS-2 STEREO DECODER

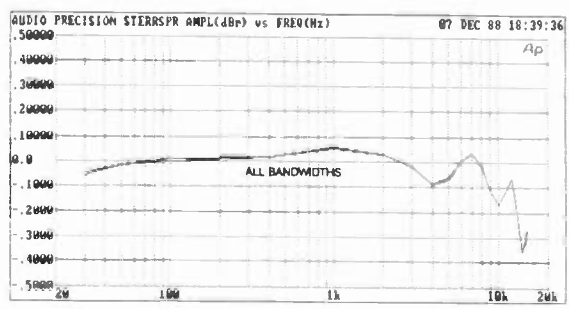


TRANSMITTER RIGHT CHANNEL FM SIGNAL TO NOISE RATIO
5 SECOND SWEEP, 75uS DEEMPHASIS
BELAR FMM-2 DEMODULATOR, FMS-2 STEREO DECODER

FIGURE 15

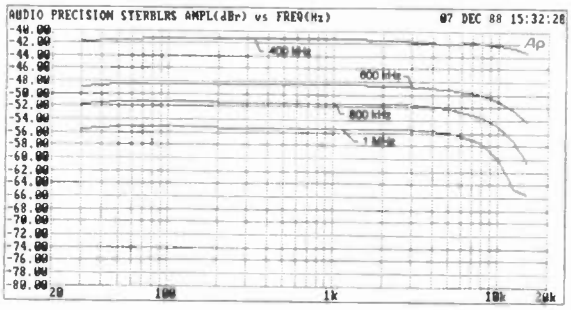


TRANSMITTER LEFT CHANNEL FREQUENCY RESPONSE
30Hz TO 15KHz, GENERATOR EQUALIZED FOR 75uS DEEMPHASIS

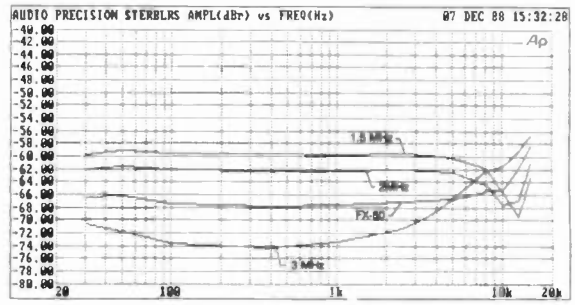


TRANSMITTER RIGHT CHANNEL FREQUENCY RESPONSE
30Hz TO 15KHz, GENERATOR EQUALIZED FOR 75uS DEEMPHASIS

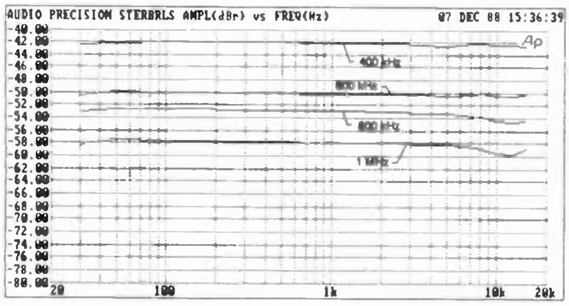
FIGURE 16



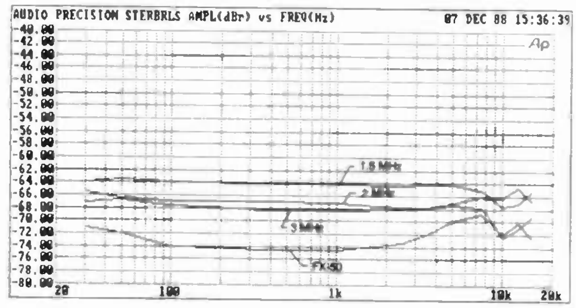
TRANSMITTER STEREO SEPARATATION, LEFT TO RIGHT
BELAR FMM-2 DEMODULATOR, FMS-2 STEREO DECODER
NO DE-EMPHASIS



TRANSMITTER STEREO SEPARATATION, LEFT TO RIGHT
BELAR FMM-2 DEMODULATOR, FMS-2 STEREO DECODER
NO DE-EMPHASIS



TRANSMITTER STEREO SEPARATATION, RIGHT TO LEFT
BELAR FMM-2 DEMODULATOR, FMS-2 STEREO DECODER
NO DE-EMPHASIS



TRANSMITTER STEREO SEPARATATION, RIGHT TO LEFT
BELAR FMM-2 DEMODULATOR, FMS-2 STEREO DECODER
NO DE-EMPHASIS

FIGURE 17

Stereo separation reaches the 60 dB mark at about 1.5 MHz, and actually measures better at 3 MHz, left into right, than it does with the full bandwidth. This is due to the small errors adding in one channel while subtracting in the other.

Figure 18 - Stereo CCIF IMD

Excellent Stereo CCIF IMD performance was achieved with at least 600 kHz RF bandwidth.

Figure 19 - Composite Baseband Spectrum Analysis

Spectrum analysis shows the distortion products generated with each bandwidth tested. The test was with 4.5 kHz single channel modulation. This produces several distortion products throughout the SCA frequency range. At 800 kHz bandwidth and above, all distortion products are more than 80 dB below 100% modulation.

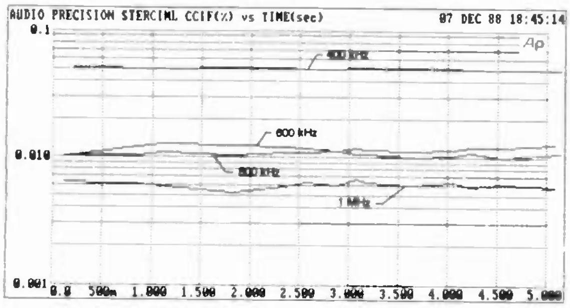
A REAL TRANSMITTER

Figure 20 shows the performance curves for the Broadcast Electronics model FM-20B. The data shown is representative of the entire "B" series of transmitters from Broadcast Electronics.

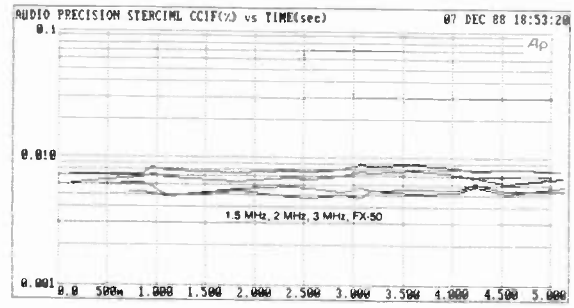
The FM-20B is a 20 KW broadcast transmitter using an Eimac 8989/4CX12,000A final tube in the patented folded half-wave cavity found in all Broadcast Electronics single tube transmitters.

The FM-20B also uses a patented broadband grid matching network to minimize the signal degradation caused by the grid circuit.

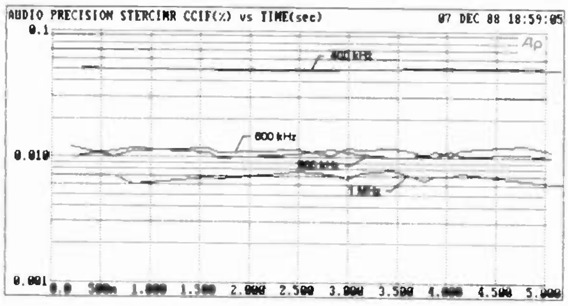
Figure 21 shows the actual measured bandwidth of the FM-20B. Notice that with the transmitter properly tuned for minimum synchronous AM, there is about a 1 dB difference between the upper and lower sidebands at ±700 KHz, respectively. This is perfectly normal and is due to the nature of a bandpass filter, which is symmetrical at the geometric upper and lower frequencies. In other words, the attenuation below center frequency is the mirror of the attenuation above center frequency when plotted on a logarithmic frequency axis, not a linear axis (11).



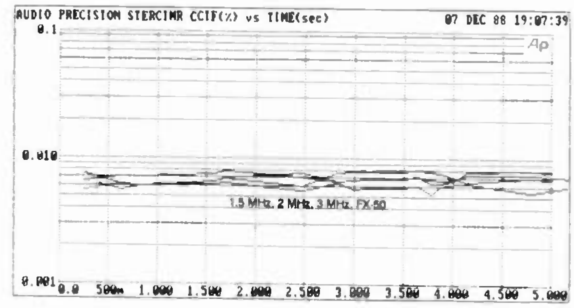
TRANSMITTER LEFT CHANNEL CCIF IMD vs. TIME
5 SECOND SWEEP, TWIN TONE 15KHz/14KHz 1:1
75uS PREEMPHASIS, NO DEEMPHASIS



TRANSMITTER LEFT CHANNEL CCIF IMD vs. TIME
5 SECOND SWEEP, TWIN TONE 15KHz/14KHz 1:1
75uS PREEMPHASIS, NO DEEMPHASIS



TRANSMITTER RIGHT CHANNEL CCIF IMD vs. TIME
5 SECOND SWEEP, TWIN TONE 15KHz/14KHz 1:1
75uS PREEMPHASIS, NO DEEMPHASIS



TRANSMITTER RIGHT CHANNEL CCIF IMD vs. TIME
5 SECOND SWEEP, TWIN TONE 15KHz/14KHz 1:1
75uS PREEMPHASIS, NO DEEMPHASIS

FIGURE 18

The second plot shows where the ±700 KHz points are symmetrical (-3 dB). The arithmetic mean (linear axis) center frequency is 78 KHz higher than the actual tuned frequency of the transmitter, but does show the 3 dB bandwidth to be 1.4 MHz.

The measured bandwidth and audio performance of the FM-20B do not exactly match the predictions based entirely on the passive band limiting tests, as expected. Slight variation is due to the non-linear input and output characteristics (transfer function) inherent to a class C tube power amplifier and matching networks.

The audio performance is excellent, with THD+N better than 0.01% at 400 Hz, with less than 0.1% at 15 kHz, SMPTE IMD better than 0.01%, greater than 54 dB stereo separation, and 92 dB signal-to-noise ratio, all with a -3 dB bandwidth of less than 1.5 MHz.

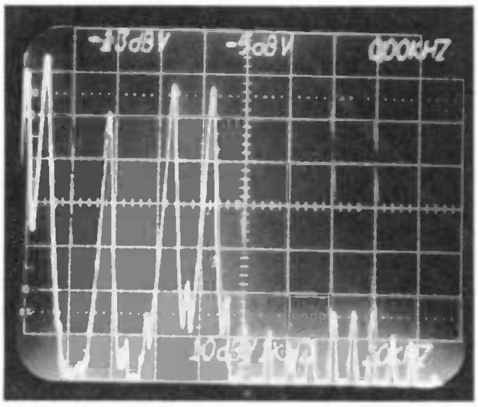
Also, as stated earlier, the synchronous AM performance is better than would be predicted from the measured 3 dB bandwidth.

CONCLUSIONS

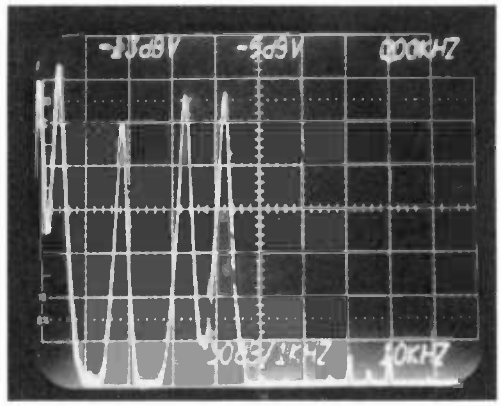
The accurate prediction of actual audio performance from measured RF bandwidth is a difficult task due to the masking effects of the grid circuit and non-linear nature of the output stage in a single tube transmitter. Carefully controlled testing of RF bandwidth limitation by a passive network tends to show acceptable performance with as little as 800 kHz bandwidth, and little, if any, improvement with more than 1.5 MHz bandwidth.

This premise is verified by actual tests on a typical, real world FM broadcast transmitter of less than 1.5 MHz bandwidth.

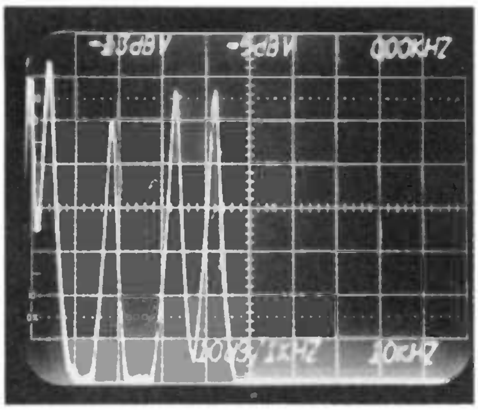
Therefore, it is concluded that good audio performance can be achieved with as little as 800 kHz bandwidth, and that with 1.0 to 1.5 MHz bandwidth, excellent audio performance results are obtained, gaining only slight improvement above 1.5 MHz. This optimum bandwidth will produce outstanding audio fidelity with maximum protection from RF intermodulation potential.



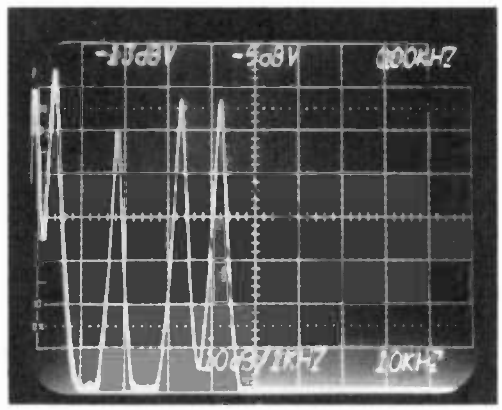
400 kHz



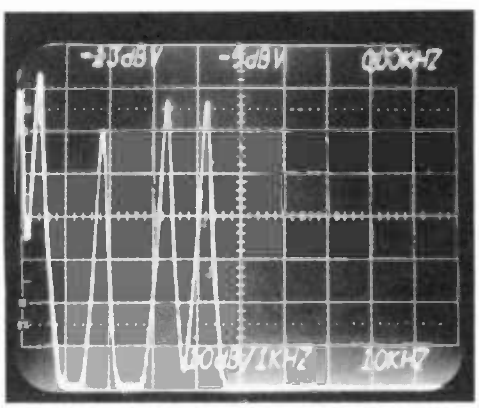
600 kHz



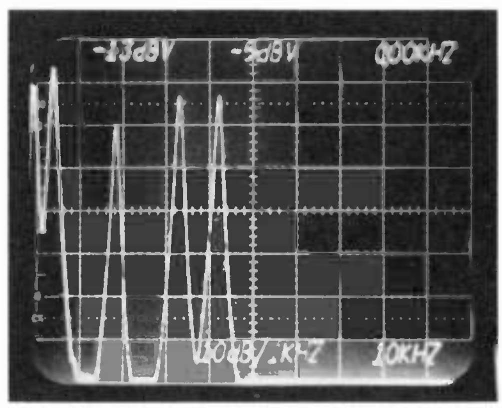
800 kHz



1 MHz



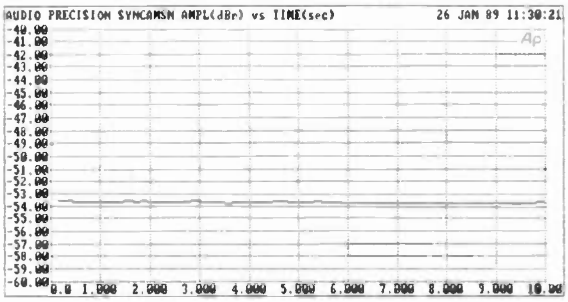
2 MHz



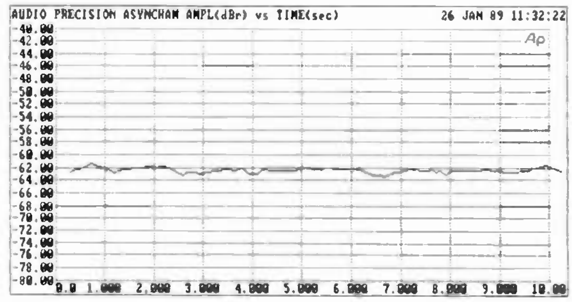
3 MHz

FIGURE 19. BASEBAND DISTORTION PRODUCTS
vs. BANDWIDTH

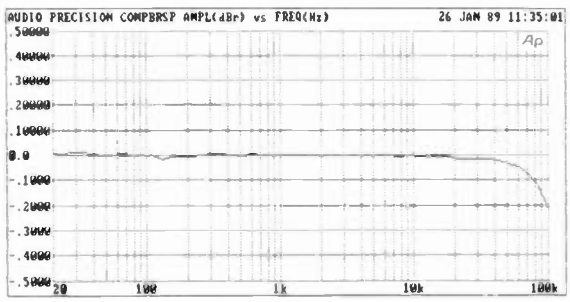
```
     FM-20B OPTIMUM BANDWIDTH TEST, 101.1 MHz     THU 01/26/89     11:25:02AM
     BROADCAST ELECTRONICS INCORPORATED    MODEL FM-20B     S/N 0000    VR2.2
-------------------------------------------------------------------------------
CONDITION: NORMAL OPERATION
-------------------------------------------------------------------------------
POWER AMPLIFIER (PA)    EFFICIENCY= 82%  | TRANSMITTER POWER OUTPUT
                                         |
                   PLATE    SCREEN   GRID | AUTHORIZED 20.00KW=100%      0.0KW ERP
VOLTAGE           9.66KV     667V   -302V | ACTUAL     20.00KW=100%      0.0KW ERP
CURRENT           2.51A      86MA    38MA |
POWER OUTPUT     20.00KW                  | REFLECTED   0.00KW=  0%
DISSIPATION       4.24KW      57W         | VSWR        1.0:1
-------------------------------------------------------------------------------
INTERMEDIATE POWER AMPLIFIER (IPA)        |
                       -1-        -2-     |
VOLTAGE               27.9V      27.7V    |
CURRENT               10.9A      11.2A    |
FORWARD POWER          203W       202W    |
REFLECTED POWER          2W         1W    |
DISSIPATION            101W       108W    |
TOTAL POWER        FWD= 364W  RFL=    3W  |                   *TOWER LITES*
----------------------------------------- | EXHAUST AIR TEMP=  47 C
EXCITER FORWARD POWER      38W            |                  REMOTE CONTROL ON
EXCITER REFLECTED POWER     1W            | MPU CONTROL
```



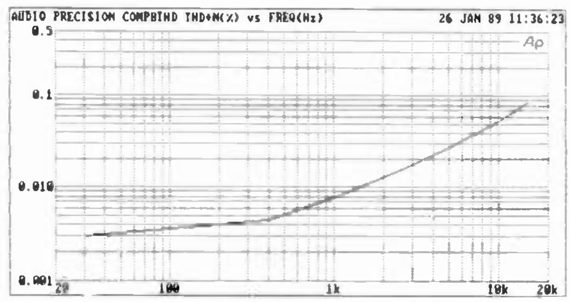
FM-20B SYNCHRONOUS AM SIGNAL TO NOISE vs. TIME
10 SECOND SWEEP WITH 75uS DEEMPHASIS BELOW EQUIVALENT
100% AM MODULATION WITH +/-75KHz FM MODULATION AT 400Hz



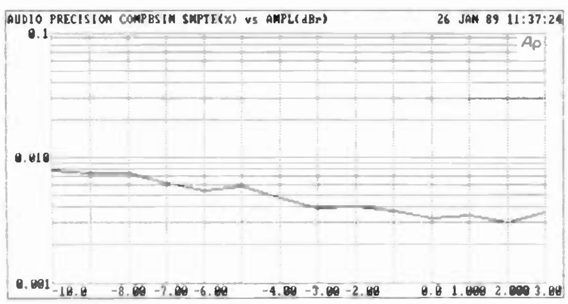
FM-20B ASYNCHRONOUS AM SIGNAL TO NOISE vs. TIME
10 SECOND SWEEP WITH 75uS DEEMPHASIS BELOW EQUIVALENT
100% AM MODULATION



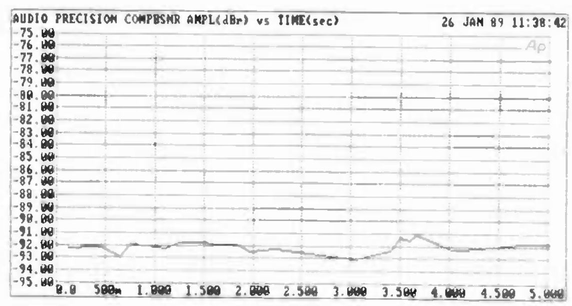
FM-20B COMPOSITE FREQUENCY RESPONSE
20Hz TO 100KHz, GENERATOR/ANALYZER NORMALIZED
BELAR FMM-2 DEMODULATOR, STEREO OUTPUT



FM-20B COMPOSITE THD+N vs. FREQUENCY
30 [illegible] 15 KHz, 75uS DEEMPHASIS
BELAR FMM-2 DEMODULATOR, BALANCED OUTPUT

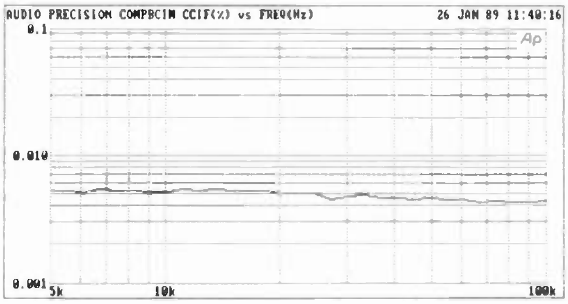


FM-20B COMPOSITE SMPTE IMD vs. LEVEL
60Hz/7KHz, 1:1 RATIO FROM 3dB OVERMODULATION TO
-10dB UNDERMODULATION (0dBr = +/-75KHz)

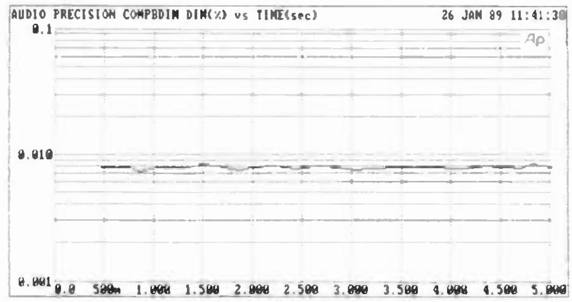


FM-20B COMPOSITE SIGNAL TO NOISE RATIO vs. TIME
5 SECOND SWEEP, BELAR FMM-2 DEMODULATOR,
75uS DEEMPHASIS, BALANCED OUTPUT

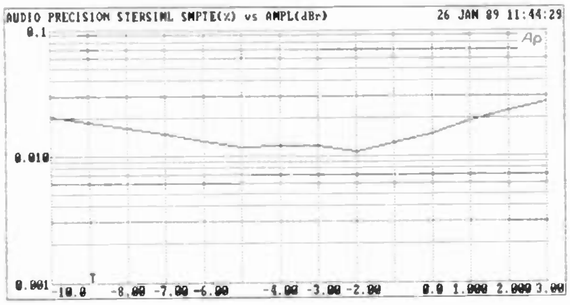
FIGURE 20. FM-20B 20kW TRANSMITTER
PERFORMANCE DATA
(Sheet 1 of 3)



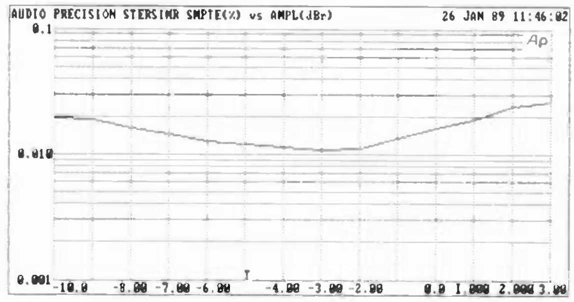
FM-20B COMPOSITE CCIF IMD vs. FREQUENCY
TWIN TONE 1:1 RATIO FROM 99/100KHz TO 5/4KHz
AT +/- 75KHz DEVIATION, NO DEEMPHASIS



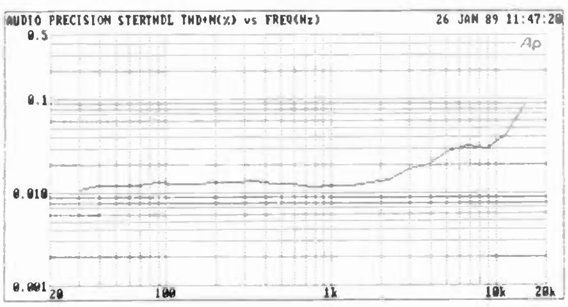
FM-20B COMPOSITE DIM/TIM vs. TIME
5 SECOND SWEEP, 3.15KHz SQARE WAVE/ 15KHz SINE WAVE
NO PREEMPHASIS, NO DEEMPHASIS



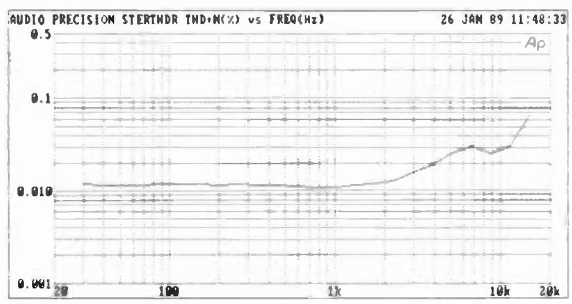
FM-20B LEFT CHANNEL SMPTE IMD vs. LEVEL
60Hz/7KHz, 4:1 RATIO 75uS PREEMPHASIS, DEEMPHASIS
FROM 3dB OVERMODULATION TO -10dB BELOW 100% (0dB=100%)



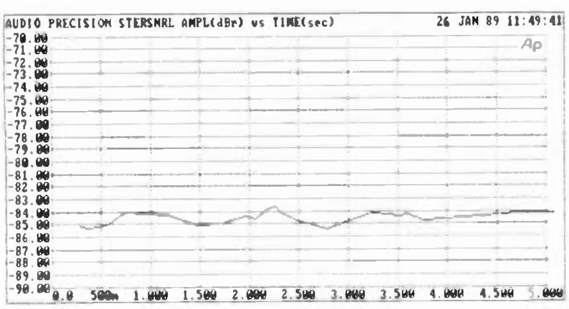
FM-20B RIGHT CHANNEL SMPTE IMD vs. LEVEL
60Hz/7KHz, 4:1 RATIO 75uS PREEMPHASIS, DEEMPHASIS
FROM 3dB OVERMODULATION TO -10dB BELOW 100% (0dB=100%)



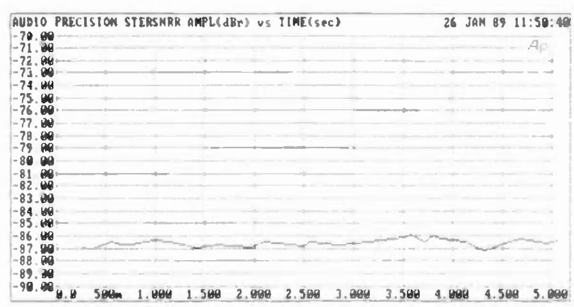
FM-20B LEFT CHANNEL THD+N vs. FREQUENCY
30Hz TO 15KHz, 75uS PREEMPHASIS/DEEMPHASIS
BELAR FMM-2 DEMODULATOR, FMS-2 STEREO DECODER



FM-20B RIGHT CHANNEL THD+N vs. FREQUENCY
30Hz TO 15KHz, 75uS PREEMPHASIS/DEEMPHASIS
BELAR FMM-2 DEMODULATOR, FMS-2 STEREO DECODER

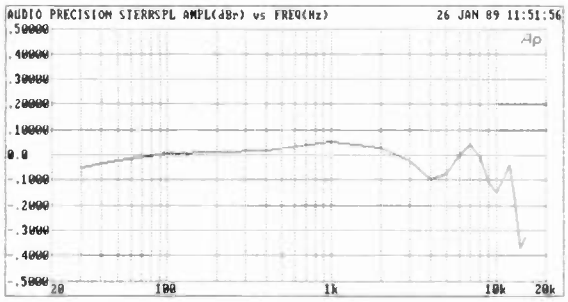


FM-20B LEFT CHANNEL FM SIGNAL TO NOISE RATIO
5 SECOND SWEEP, 75uS DEEMPHASIS
BELAR FMM-2 DEMODULATOR, FMS-2 STEREO DECODER

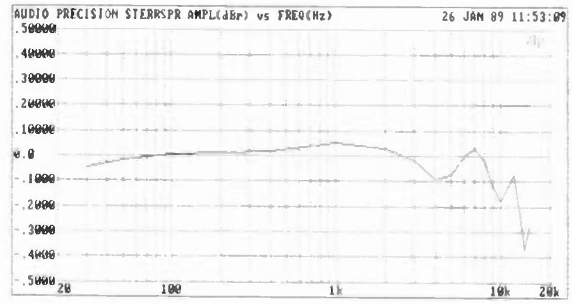


FM-20B RIGHT CHANNEL FM SIGNAL TO NOISE RATIO
5 SECOND SWEEP, 75uS DEEMPHASIS
BELAR FMM-2 DEMODULATOR, FMS-2 STEREO DECODER

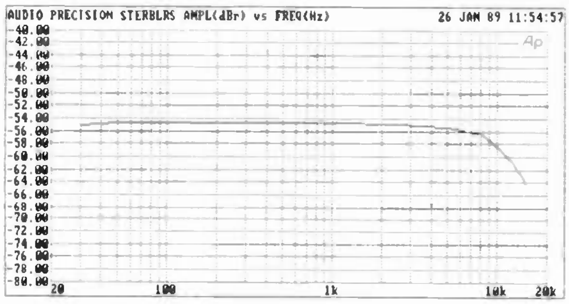
FIGURE 20. FM-20B 20kW TRANSMITTER
PERFORMANCE DATA
(Sheet 2 of 3)



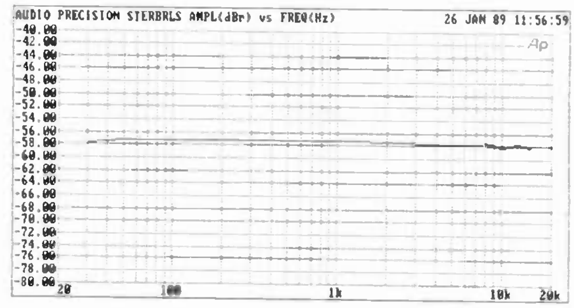
FM-20B LEFT CHANNEL FREQUENCY RESPONSE
30Hz TO 15KHz, GENERATOR EQUALIZED FOR 75uS DEEMPHASIS



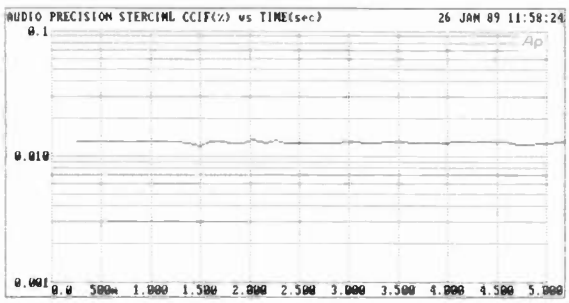
FM-20B RIGHT CHANNEL FREQUENCY RESPONSE
30Hz TO 15KHz, GENERATOR EQUALIZED FOR 75uS DEEMPHASIS



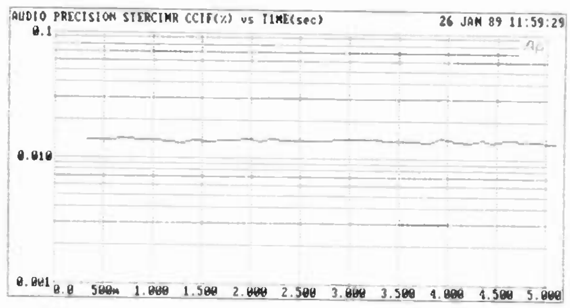
FM-20B STEREO SEPARATATION, LEFT TO RIGHT
BELAR FMM-2 DEMODULATOR, FMS-2 STEREO DECODER
NO DE-EMPHASIS



FM-20B STEREO SEPARATATION, RIGHT TO LEFT
BELAR FMM-2 DEMODULATOR, FMS-2 STEREO DECODER
NO DE-EMPHASIS

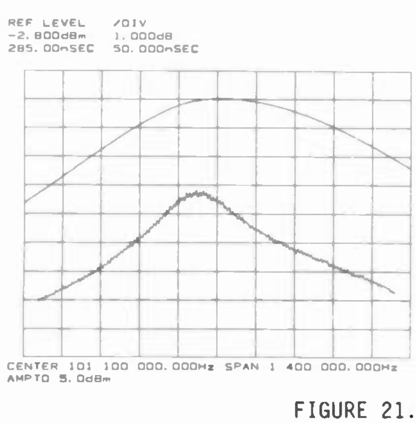


FM-20B LEFT CHANNEL CCIF IMD vs. TIME
5 SECOND SWEEP, TWIN TONE 15KHz/14KHz 1:1
75uS PREEMPHASIS, NO DEEMPHASIS



FM-20B RIGHT CHANNEL CCIF IMD vs. TIME
5 SECOND SWEEP, TWIN TONE 15KHz/14KHz 1:1
75uS PREEMPHASIS, NO DEEMPHASIS

FIGURE 20. FM-20B 20kW TRANSMITTER
PERFORMANCE DATA
(Sheet 3 of 3)



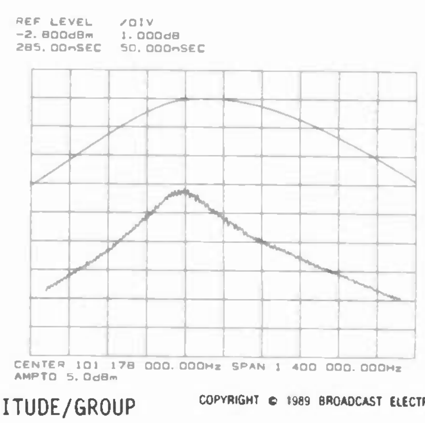


FIGURE 21. FM-20B TRANSMITTER AMPLITUDE/GROUP
DELAY RESPONSE

ACKNOWLEDGEMENTS

The author wishes to thank Geoff Mendenhall, Mukunda Shrestha, and Rick Carpenter for their invaluable assistance in understanding the complex operation of the modern, high power broadcast transmitter. I would also like to thank Charlotte Steffen for word processing, Larry Foster for editing, and William Glore for test cavity illustration.

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He is a member of the Phi Kappa Phi honor society and the National Radio Systems Committee (NRSC) FM Subgroup.

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A REPORT ON THE FORMATION OF THE NRSC FM SUBCOMMITTEE

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Group W Radio
Houston, Texas

HISTORY

Developed in the 1930's, inherently wideband and noise free, FM at first appeared to be the answer to many problems already facing AM broadcasting. But Major Armstrong's invention almost didn't succeed as a commercial broadcast service.

Off to a rocky start because of World War II and assignment in the skywave prone 42 to 50 mHz band, FM languished through the decades of the forties and fifties even after it was moved to its present 88 to 108 mHz position. Pre-war receivers were almost impossible to convert and the appearance of television coupled with simulcasting of AM programming by FM stations did little to foster growth. In fact, in the late forties and early fifties, over 250 FM stations ceased operation due to lack of interest.

With public interest lagging and growth stymied, FM operators turned to what was then controversial programming- storecasting on the main channel complete with paid subscribers and beep tones to control receivers. But development of multiplexing techniques and FCC authorization of Subsidiary Communications Authority in the mid 50's relegated this type programming to subcarriers.

FM's first real boost came in 1958 when the FCC received petitions to allow stereo broadcasting. Subsequent studies and testing of proposed systems were undertaken by the National Stereophonic Radio Committee, a precursor of today's National Radio Systems Committee. The many proposals were eventually reduced to six systems and field tested on Westinghouse Broadcasting station, KDKA-FM. Adoption of the US system and what we hear today was the result.

Stereo realism soon aroused the public's interest in the FM band. Listener growth increased enormously during the 1970's and today has outdistanced AM listening by at least three to one. Armstrong's invention had not died an early death after all.

AM DILEMMA

Soon after FM took off, a painful awareness began to develop in the listening public and the broadcasting community-the AM band had some major problems. Increased interference, poor quality, and rising noise levels made AM a place to tune away from. To counteract interference, manufacturers began to decrease the bandwidth of their receivers. Station engineers retaliated with audio pre-emphasis and increased compression. Interference grew worse. Receiver bandwidth narrowed. Engineers turned up the pre-emphasis and compression. Broadcasters and receiver manufacturers were hung in a loop without a break command.

NRSC

By the early 1980's AM problems were almost out of hand. At the 1985 NAB convention Michael Rau announced formation of an NAB AM Improvement Committee. Faced with perplexing, compound problems the committee had already worked for over a year identifying ways to technically improve AM broadcasting. It soon became obvious that to make major strides, receiver manufacturers should be a part of the solution. To accommodate this action, the National Radio Systems Committee formed it's own AM improvement committee. The work of these two committees during the last four years has been stellar. Thanks largely to their actions the AM band is slowly improving. Implementation of NRSC-1 bandwidth restrictions alone will go a long way toward cleaning up interference on the AM band.

NRSC FM COMMITTEE

Some say that the FM band isn't broken so don't fix it. There is truth in that statement. It isn't broken yet. But it could be and many believe it's well on it's way.

Late in 1988 responding to concerns about FM band deterioration, the full NRSC committee decided to form an FM subgroup. This group would be charged with investigating and identifying problems plaguing today's FM operations. Composed of broadcasters and equipment and receiver manufacturers, the committee went immediately to work. The first action was to poll members with a survey. Survey results were tallied and problem areas identified for committee consideration.

THE ISSUES

Although almost twenty areas, some as wide ranging as standardized audio processing for all stations were identified, ultimately four issues were placed before the committee for action. This first set with the committee's initial actions are identified below but others will surely follow.

Multipath Measurement

Multipath is an aggravating and fatiguing sound to a listener. Probably the most talked and written about FM phenomenon, a satisfactory way to eliminate it has never been discovered. Exacerbated by improper transmitter tuning as well as reflecting surfaces, it must be quantified by new measurement techniques and new types of equipment. Station engineers must also learn that continued vigilance is paramount in minimizing transmitted synchronous AM noise that can contribute to "multipath-like" response in a receiver.

Committee Action

Write a bibliography. Many articles and papers have been presented-this bibliography would serve as a reference compendium on the subject.

Investigate the use of a multipath simulator for receiver evaluation. Although not readily available, a simulator that provides multipath, amplitude and phase, median field strength variations, and vehicle speed simulation is manufactured by JRC.

Write a paper on basic receiver design that would allow engineers to gain understanding of receiver multipath mechanisms and effects of narrower IF responses in modern FM receivers.

Subcarriers

For years engineers have complained that subcarriers degrade the transmitted signal of FM stations. Under certain conditions, these are valid concerns. It is well known that additional channel loading by subcarriers can increase multipath conditions. And even though small, there is still the 0.5 to 1.0 db trade off in main channel modulation. A station engineer saddled with a couple of subcarriers leased to an aggressive operator wanting maximum everything and reporting to a GM that wants to be the loudest station in town probably has only two options-overmodulate or resign.

Committee Action

At the initial meeting, members felt that even though subcarrier effects had received extensive attention in the past, further investigation was warranted. Subsequently we have learned that tests on subcarrier effects will be performed in Canada during 1989. No action will be taken by the FM subcommittee until test results are published.

Receiver Performance

Recent trends toward co-located transmitter sites have generated yet another anomaly in the increasing set of FM band problems. These sites tend to present receivers with many signals of more or less equal field strength and some of the results are less than desirable. Of all the products generated at co-located sites, third order intermodulation is the most noxious. Not only can IMD be generated in a transmission system, but some receivers can produce their own internal set of products.

Committee Action

Identify sites that produce IM product interference.

Recommend that ANSI/IEEE 185-1975, Standard Methods of Testing Frequency Modulation Receivers, be updated to include measurement of the "IF taboo", third order IM and front end input levels, stereo separation, and SCA rejection.

Adjacent Channel Interference

A growing problem for some areas of the country, especially the east coast. Paying careful attention to modulation levels and judicious (or no) use of composite clipping can eliminate a large part of these problems for broadcasters. Better receiver selectivity could also help.

Committee Action

A working group of broadcasters and receiver manufacturers has been formed. This group will evaluate possible new methods for measuring frequency modulation levels and could even propose something as radical as an RF or composite audio mask for FM.

SUMMARY

The public expects clean, clear, high quality sound on an interference free channel from the FM service. Unfortunately what the public expects and what we deliver today falls short in some cases. Increased allotments, multipath, intermodulation product interference, adjacent and co-channel interference are quickly becoming a part of modern listening life. Listeners are fickle-cassettes, CDs, and DAT offer quality solutions to their problems. The gauntlet has been thrown and we are challenged to prevent AM-ization of the FM band. The NRSC FM subcommittee accepts this challenge and will be working diligently to technically improve FM broadcasting.

FM DIRECTIONAL ANTENNAS AND NEW FM SHORT-SPACING RULES

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Moffet, Larson and Johnson, Inc.
Falls Church, Virginia

INTRODUCTION

Directional FM transmitting antennas have been used successfully in the United States for many years, although fewer than one in ten commercial FM broadcast stations employ them.[1] Their numbers are limited because the rules of the Federal Communications Commission have previously given broadcasters little opportunity to employ FM directional antennas (FM DAs) in allocation matters. Recently, the FCC adopted new rules permitting limited short spacing of FM stations by using FM DAs.[2] This paper discusses some allocations issues, application of the Commission's rules, and the implementation of FM DAs.

PRESENT ALLOTMENT RULES AND POLICY

The Commission chose distance separations as its basis for commercial FM allocation over 25 years ago.[3] Allocation of FM stations on channels 221 to 300 is controlled by requiring certain minimum separation distances between adjacent and co-channel stations. This policy is graphically depicted in Figure 1, in which the proponent of a station must maintain a certain geographic distance ("Required Separation Distance") from an existing protected station. These distances are listed in the tables of Section 73.211 of the FCC Rules.

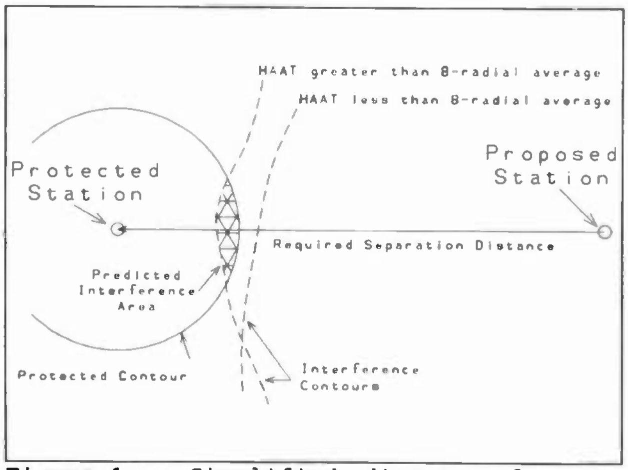

Figure 1 - Simplified diagram of FCC's nominal separation distance rules with resulting contours.

The minimum distance between the proponent and protected station are based on the frequency (channel) relationship and the class of the stations, while other factors, such as effective antenna heights and actual power radiated by both stations (within the class brackets) are not considered.

In the commercial FM allocation system, all stations are assumed to be operating at the maximum facilities (maximum power and antenna height) for their class, and the antennas are assumed to be non-directional (radiating equally in all horizontal directions). Since the antenna height above average terrain (HAAT) in any direction may vary from the average value of the eight standard radials, contour distances may differ from the assumed separations, resulting in contour overlap or underlap as shown in Figure 1.

It should be noted that the FCC derived its separation distances from nominal RF protection ratios and calculations of contour distances. By eliminating signal contour protection from consideration, however, the FCC simplified its system for allotment of channels to communities, as well as the processing of new applications. Thus, when the Commission required strict separation distances between critical FM stations, based on assumed maximum omni-directional facilities, there was little advantage to the broadcaster in employing FM DAs.

Separation between non-commercial FM stations on channels 200 through 220, known as the "reserved" portion of the FM band, are not based on strict distance separations. Rather, a new station applicant must not produce an interfering signal contour that overlaps existing stations' coverage contours, as shown in Figure 2.

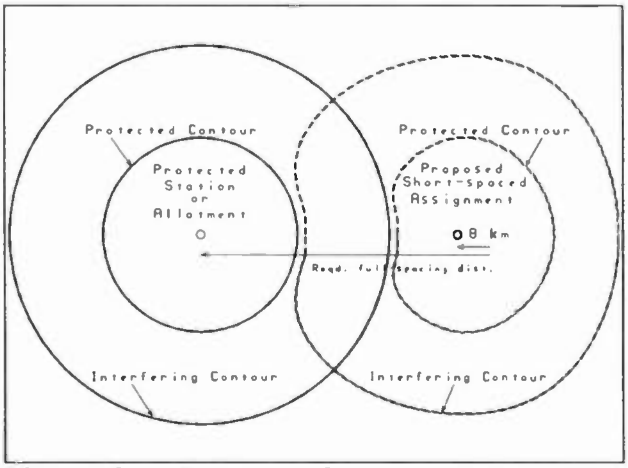

Figure 2 - Diagram of contour protection requirements for a short-spaced station.

Contour protection provides great flexibility in the location of NCE-FM stations, although the contour protection method may restrict coverage expansion or relocation of either critical station. Furthermore, contour protection merely avoids "objectionable interference", as defined by Commission rules, however, the interference may still be significant.[4]

Commercial FM DAs are currently permitted in one specific case, involving "grandfathered" short-spaced stations that were in existence at the inception of the 1964 Table of Allotments. Within certain separation distances, these stations may use directional antennas such that radiation in the direction of a short-spaced station is not increased. This principle is not related to contour protection, however, and does not necessarily maintain constant potential interference.

USE OF DIRECTIONAL ANTENNAS

In its Report & Order, the Commission notes that contour protection has been used for many years "with excellent results" by non-commercial educational FM broadcast services (in the *reserved* portion of the band).[5] In its earlier Notice of Proposed Rule Making (NPRM)[6] and again in the R&O, the Commission noted that the use of directional FM antennas could provide "greater flexibility..." to deal with "...site restrictions encountered by the applicant because of FAA clearance difficulties, government ownership and restrictions on use of desireable sites, as well as environmental, economic and coverage concerns...".[7]

The Commission repeatedly pointed out that while directional antennas were used for station allotment purposes in the reserved portion of the FM band, directional antennas would be allowed in the commercial (non-reserved) FM band for short-spacing only during the assignment process (not during the allotment process).

Nevertheless, several of the commenters responding to the NPRM expressed concern with the introduction of contour protection in the commercial FM band. Noting that the present rules are based on separation distance assuming maximum facilities, some commenters emphasized the importance of protecting existing stations operating at less than maximum facilities. They explained that contour protection of only the present operation would preclude future upgrade to the maximum facilities for its class.

The Commission stated that most of the arguments against contour protection are based on claims that it is fundamentally inferior to the distance separation requirements. Some argued that the imprecision of current signal-prediction methodology could increase interference levels in the FM service, while some expressed concern that contour protection would result in the "AM-ization of the FM band" similar to the interference conditions that exist in the AM service.

After consideration of issues surrounding contour protection, the Commission concluded "We have no reason to believe that the further application of contour protection in the FM service will have any adverse effects".[8] The Report & Order further explained that existing separation distance rules ignore the variations in terrain height in specific instances, tending to sometimes overprotect, and at other times underprotect, FM service.

Other issues raised in comments concerned accuracy of FM antenna directivity, antenna installation and maintenance procedures, and the relationship between the antenna horizontal and vertical radiation patterns. Some of these matters were incorporated into the rules. A number of restrictions were also placed on the implementation of the new FM short spacing rules, to be discussed later herein.

It is interesting to note that in its discussion of the FM DA issue the FCC revealed it considers FM service to be a "mature", "heavily populated" medium.[9] Despite the existence of some 5800 licensed and authorized operations, the Commission still expects a potential demand for relocation involving directional antennas so heavy it could significantly exceed its ability to process the applications.

IMPLEMENTING A SHORT-SPACING POLICY
Signal Contour Requirements

The Commission adopted limited short-spacing and FM DA rules to afford applicants some flexibility in antenna site selection. A number of restrictions were placed on these rules to protect existing stations to the maximum degree possible, to protect Class A stations pending the outcome of a proposal to increase power, to handle the anticipated influx of applications involving FM DAs, and to allow the FCC to respond to unforseen problems in the processing of these applications.

Applicants proposing to short space under the new rules will be required to provide a map showing the protected and interfering contours of all stations located at less than the standard minimum separation distance. Figure 2 is a simplified representation of this type of map. It is evident that the proponent must neither cause objectionable interference to, nor receive objectionable interference from other stations or allotments. The field strength values for the contours are listed in a table in Section 73.215 of the new rules.

Present rules require all commercial stations to provide a 3.16 mV/m (70 dBu) signal over the station's allotted community of license. The comments generally agreed with this policy, and the Commission maintained this requirement.

The FCC decided to limit for an indefinite period the amount by which applicants may short-space to 8 kilometers (5 miles). This was done primarily, the R&O notes, to restrict the number of applications to a number that the Commission staff can manage. The rule states that this temporary restriction "**...will be removed when the Commission determines that available resources are sufficient to allow the timely processing of additional applications...**".[emphasis added][10] Stations located within 320 kilometers (200 miles) of the Mexican-United States border must continue standard distance separations from Mexican stations.

Noncommercial stations on channels 218, 219 and 220 in the reserved portion of the band are also subject to the new rules, in so far as they protect commercial FM stations.

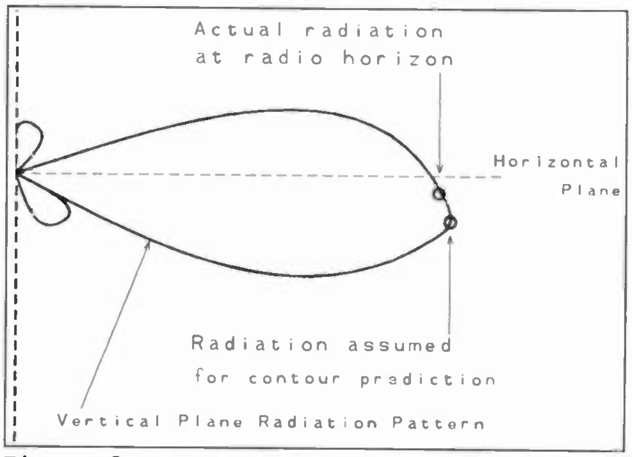

Figure 3 - Derivation of antenna radiation for antennas with beam tilt.

Contour distances will be predicted for the maximum radiation, as shown in Figure 3, even if the main beam is deflected "beam tilted" electrically or mechanically below the radio horizon.

The new FM DA rules will require that the applicant use as many radials as necessary to establish the lack of prohibited overlap. As Figure 4 illustrates, eight radials may not be enough if the antenna HAAT increases sharply on an intermediate azimuth, as for example, a valley running radially from the proposed site. The increase in HAAT could result in extensions of the contour that would no be evident on adjacent radials.

Applicants who comply with the minimum distance separation table of Section 73.207 of the rules need not be concerned with their interfering contour, whether or not it overlaps the protected contour of another station. This is shown in Figure 1 as in indication of the present assignment results, which may or may not cause contour overlap, depending on the antenna heights of the adjacently located stations.

Applicants will be required to protect the 1 mV/m (60 dBu contour of all classes of FM stations, except Class B and B1, which will require protection to the 0.5 mV/m (54 dBu) and 0.7 (57 dBu) contours. Although the protected contour is 1 mV/m for all stations on the reserved channels, the FCC's current commercial separation requirements for Class B and B1 stations were based on the 0.5 and 0.7 mV/m contours. Vacant allotments are to be protected to the applicable contour calculated for a hypothetical station operating at the allotment's reference coordinates.

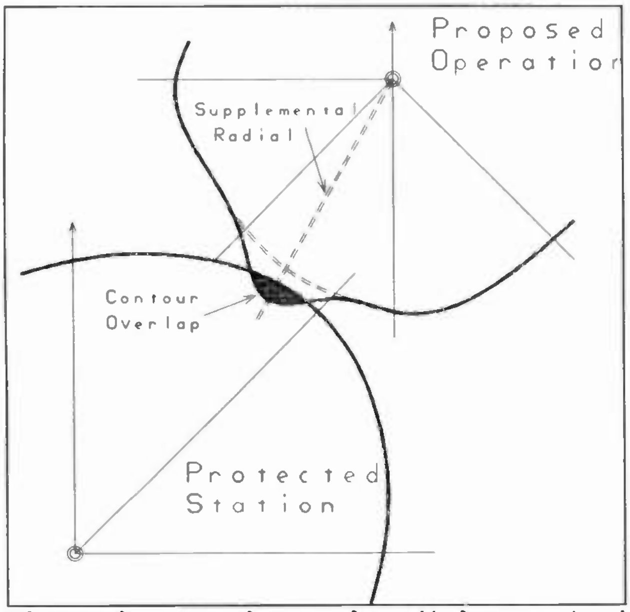

Figure 4 - Supplemental radials required to properly define contours.

All existing fully spaced stations will continue to be protected to contours presuming maximum effective radiated power (ERP) and reference height for their station class. This was provided, according to the Commission, to maintain the upgrade potential for stations that are operating at less than maximum facilities.

Stations that deliberately short space, however, will be protected only to the actual facilities for which they apply under the new rules. Since the antenna height above average terrain (HAAT) in any particular direction will not necessarily be the same as the standard eight-radial HAAT, the resulting variations in short-spaced station's protected contour will be the trade-off applicants must make in exchange for the choice to short space.

Figure 5 depicts the reduction in ERP across the critical arc, as determined by its interfering contour distances from the pertinent antenna HAATs in the direction of the protected station's contour. Across the remainder of the arc, the station proposing short spacing will determine its ERP based on the maximum permitted for its class based on the antenna HAAT of the eight standard radials.

Antenna Specifications

The new short-spacing rules add new technical requirements to applications involving FM DAs while maintaining the current performance standards, such as the maximum-to-minimum pattern ratio (still 15 dB) and pattern rate of change (still 2 dB per 10 degrees). For example, applicants will be required to furnish the following information:

- a single composite *plot* of horizontal plane relative field for both the horizontal and vertical polarizations;
- a *tabulation* of the relative field pattern values at least every 10 degrees plus all maximas and minimas;
- a *statement* that the DA will be mounted on the antenna tower as recommended by its manufacturer;
- a *statement* that the DA will not be mounted near a top-mounted platform which projects beyond the vertical face of the tower;
- a *statement* that no other antennas are mounted within a vertical clearance distance required for proper operation by the DA manufacturer.

After completion of construction, permittees must furnish a statement from a licensed surveyor that the DA has been installed in accordance with the manufacturer's instructions and is in the proper orientation. The FCC will continue to require proofs of performance to establish that the measured pattern complies with the authorized pattern. However, the contour distances will always be based on the authorized pattern, not the pattern provided in the subsequent license application.

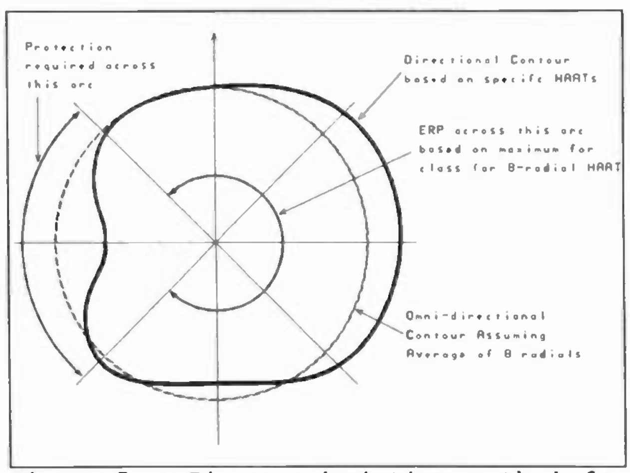

Figure 5 - Diagram depicting method for determining permissible radiation.

Design and Construction of Directional Antenna Systems

Designing The Tower Mounting

Antenna systems employed by FM stations have always required attention to the effects of tower mounting. This is because FM-band antennas usually are both circularly polarized and side mounted on a supporting structure of cross-sectional dimension which is an appreciable portion of the wavelength involved. Therefore, energy from the antenna, especially the vertically polarized component, is intercepted and reradiated by the tower, resulting in appreciable distortion of the patterns.

Proper design of the FM DA depends on information about the supporting structure being provided to the manufacturer. Certainly, manufacturers need information about the dimensions of the structure, cross members, transmission lines, ladders, materials used, and precise orientation of the tower.

Antenna manufacturers must combine information about the tower with a permissible radiation pattern prepared in accordance with the applicable FM DA rules to produce a nominal design. Other information is helpful, such as the mounting arrangement (mounted on a face or corner, at or between cross members, etc.) and the direction of the optimum service area.

It is notable that the FCC's R & O did not consider pattern distortion in omni-directional FM antennas. The Commission still assumes that these antennas radiate uniformly in the horizontal plane, although the pattern of these antennas are also influenced by the supporting structure.

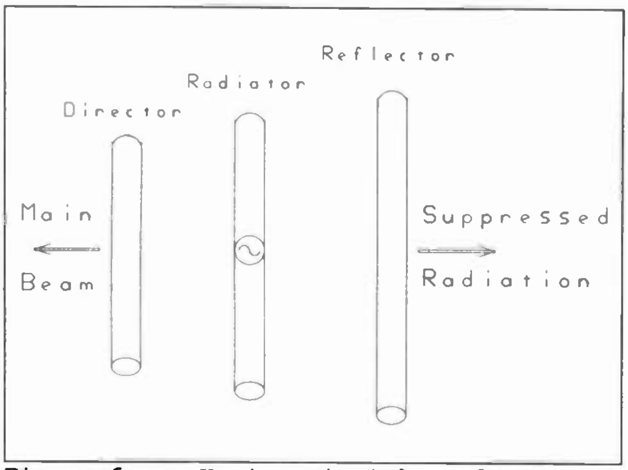

Figure 6 - Yagi principle of antenna directivity.

It is usually necessary to have a surveyor check the orientation of the tower legs prior to design; this can avoid an expensive antenna modification, if, after construction, the actual orientation is found to be different than was assumed.

Antenna Pattern Control

A technique which has gained acceptance is use of the principle of the Yagi antenna, shown in Figure 6, wherein parasitic elements are placed in the field of a dipole radiator to modify is directional characteristics. As is well known, a shortened dipole (director) placed in close proximity to a radiator reinforces radiation in the forward direction.

If the parasitic element (reflector) is longer than the radiator, the effect is reversed, that is the signal is suppressed on the side of the longer parasitic element and reinforced in the direction of the radiator.

Figure 7 - FM directional antenna system in final testing.

(courtesy Electronics Research, Inc.)

Figure 7 shows how a pair of parasitic elements have been positioned to modify the horizontally polarized signal. (The antenna is shown lying horizontal for final inspection before shipment.) The parasitics are visible on the support pole opposite the radiating horizontal elements. There are no vertical parasitics added for this antenna system; the support structure (the pole and interbay transmission line, in this case) can provide directional effects to the vertically polarized radiation.

FM antenna patterns are usually measured from a single bay of the antenna system, mounted as specified by the customer. Since the horizontal plane pattern is of primary interest in allocations matters, testing of a single bay of vertically-stacked multi-bay systems, rather than the entire array, is considered by the FCC to be sufficiently accurate. If the tower and its cross members, feedlines, ladders, and other conductive material is identical at each vertical level of the antenna, the horizontal plane pattern for the aggregate is expected to be the same as any single bay.

Antenna patterns are usually measured on a test "range" large enough to insure that an antenna pattern is measured in the "far field", that is, at a distance where the antenna the radiation moments from the antenna elements and any parasitic moments in the mounting structure effectively behave as a point source. This distance is conveniently found outdoors, at isolated locations relatively free of reflections from buildings, power lines, etc.

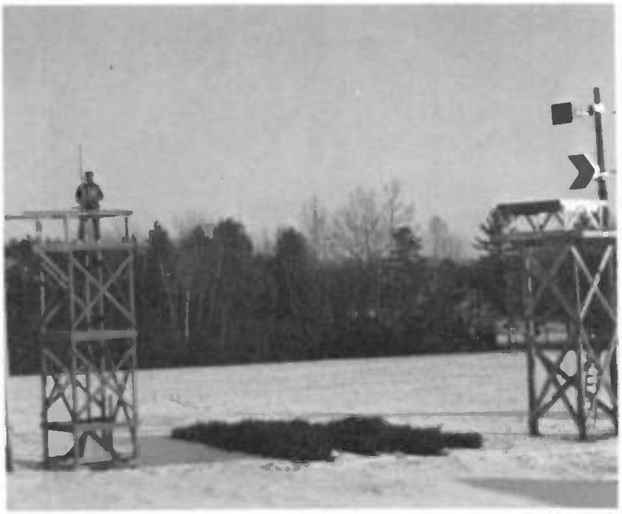

Figure 8 - Antenna manufacturer's test range. (courtesy Shiveley Labs)

Figure 8 shows the test range of one antenna manufacturer. The antenna under test is mounted on a rotatable structure in the tower at the left; the tower at the right supports the measurement antennas. Some manufacturers use the antenna under test to transmit the test signal for pattern measurements, while others use the antenna under test as a receive unit; since the process is essentially reversible the direction of the signal transfer is unimportant.

Figure 9 shows an antenna being set up on the test range; the reference measurement antennas are visible in the background. Note that the antenna under test is a reduced scale model. Some manufacturers work with full-scale models, while others prefer to test scale models of the antenna system. Both methods can be satisfactory, the accuracy of the overall pattern measurement depending on the faithfulness of the replica, whether full or reduced scale, to the actual antenna and supporting structure. This indicates that the customer must supply complete information about the tower, and must insure installation according to the manufacturer's directions. The new FM DA rules have promoted this procedure by requiring various certifications of compliance in the application process.

Figure 9 - Reduced-scale model antenna on manufacturer's test range. (courtesy Shiveley Labs)

The FM short-spacing and directional antenna rules are very new to the commercial FM industry. Many questions and clarifications will occur in the months ahead, as well as attempts by hopeful applicants to stretch the new limits for their own benefit. Some broadcast organizations reportedly are planning petitions to rescind the new rules in their present form. The remainder of 1989 will clearly be a busy time for parties on all sides of the short-spacing issue.

Endnotes

1. Currently there are approximately 5800 FM radio licensees and permittees, according to the FCC, of which only about 530 use directional antenna systems.

2. Report & Order, FCC MM Docket 87-121, *Amendment of Part 73 of the Commission's Rules to permit short-spaced FM station assignments by using Directional Antennas, adopted December 12, 1988.*

3. Third Report, Memorandum and Order, *Revision of FM Broadcast Rules, Particularly as to Allocation and Technical Standards*, Docket No. 14185, adopted July 25, 1963.

4. John C. Kean, *An Analysis of the FCC's FM Station Separation Methods in View of Docket 87-121*, NAB Engineering Conference Proceedings, pp. 177-185.

5. Report & Order, *op. cit.* para. 23.

6. Notice of Proposed Rule Making, FCC MM Docket 87-121, *Amendment of Part 73 of the Commission's Rules to permit short-spaced FM station assignments by using Directional Antennas, adopted February 25, 1988.*

7. Report & Order, *op. cit.*, para. 5.

8. *Ibid.*, para. 28.

9. *Ibid.*, para. 29.

10. *FCC Rules*, Section 73.215, part (b)(1).

NTIA IRREGULAR TERRAIN PROPAGATION STUDY

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INTRODUCTION

In the 1950s when FM Broadcast radio was in its infancy, methods of estimating FM signal coverage were needed that would be unambiguous and easy to implement. The FCC's FM field strength curves served just that purpose. In the same time period, the predecessors of the National Telecommunications and Information Administration (NTIA) were developing models that could predict signal coverage based on the environment[1,2]. The models are complex but with the availability of computers and environmental data bases, the models can be made simple for the user to calculate detailed signal-coverage maps of transmitters. This paper summarizes the conditions that affect FM Broadcast coverage, compares some of the models available to compute signal coverage, and provides some samples to show how signal coverage is dependent upon the surrounding terrain and how it can be accurately mapped.

FACTORS THAT AFFECT FM SIGNAL PROPAGATION

Antenna Heights

The heights of both the transmitting and receiving antennas affect the performance of FM radio. In general, FM transmitting antennas are placed at high locations relative to the receiving antennas. The spaces in front of the transmitting antennas are usually clear of obstructions, whereas the receiving antennas are normally surrounded by man-made and natural obstructions. As a rule of thumb, if the height of the transmitting antenna is doubled, the signal at many receiving antennas is increased by 6 dB. Doubling the antenna height at any one receiving site may or may not increase the received signal from a specific transmitter; too much depends upon the following factors.

Direct Ray and Indirect Signal Paths

If the transmitting and receiving antennas are well elevated above surrounding obstacles and the ground, then the two antennas are considered to be line-of-sight to one another. In this case signals from the transmitter reach the antenna by a direct ray path and are probably strong enough to overcome any indirect signals. Indirect signals are reflections from the atmosphere, ground, or terrain between the antennas, from buildings and other man-made obstacles, and from surrounding hills or mountains. If the two antennas are kept at constant heights but are moved further and further apart, the Earth's bulge will eventually pierce the direct path between transmitter and receiver. The signal then has to diffract over the bulge in order to reach the receiver. The signal that is diffracted over the horizon is greatly attenuated when compared to the signal that would be available in the line-of-sight situation. As the two antennas are further removed from each other, the attenuation due to the diffraction is so great that another indirect path provides a higher signal level at the receiver. That signal is due to atmospheric reflections or tropospheric scatter.

Terrain Profile

In the FM Broadcast band, the terrain between the transmitter and receiver locations affects the received signal level. With a relatively high transmitting antenna and low receiving antenna and with a smooth Earth between the antennas, one would expect that the terrain close to the receiving site would cause signal reflections at the receiving antenna. The reflected signal can cancel the direct signal depending upon the geometry of the signal paths. But if the terrain is irregular between the two antennas, intervening terrain obstacles can cause additional reflection points or can cause the receiver to be beyond line-of-sight and in the diffraction region. It is easy to imagine with irregular terrain that a receiver close to the transmitter, but shadowed by a terrain feature, receives a signal that is lower than a receiver that is further away but is high enough to be line-of-sight to the transmitter. Thus in irregular terrain and with a fixed receiving antenna height above ground, the received signal level will in general decrease as the receiver is moved further away from the transmitter; but the signal will vary considerably as terrain obstacles either reduce the signal by shadowing the receiving antenna or enhance the signal by raising the antenna above its surroundings and providing a better radio path.

Signal Variability

When an FM radio is moved over a short distance, say 1 to 2 wavelengths about 6 meters or 20 feet, the received signal can experience some very deep nulls. The signal fading is due to the direct and indirect signal addition as described previously. Because these multipath fades are geometry dependent and impossible to specify deterministically, they are usually defined by a statistical distribution, Rayleigh or Rician for example. The multipath fading statistics are usually independent of the terrain, the locations, or the time. Because of this independence, NTIA's radio propagation prediction models do not include deterministic multipath fading in their calculations; instead, long-term time-varying effects are of interest to prediction models. "Long term" means how the median of a signal varies in time from hour to hour, day to day, or season to season; it also means how the signal varies over many wavelengths, and over many paths that are similar in their terrain profiles. In the NTIA models, these long-term signal variabilities are included as statistics which will be defined and shown graphically later in the paper.

Ideal Conditions

The conditions which lead to the ideal situation for FM Broadcast radio are:

- high, unobstructed antennas
- line-of-sight paths
- no reflections from hills, buildings, etc.

Since these conditions are rarely met outside of the region closest to the transmitter, the following sections describe ways to predict FM signal coverage under various constraints of ease of use, complexity of model, and availability of environmental data.

APPROACHES TO DETERMINING SIGNAL COVERAGE AND INTERFERENCE

FCC Field Strength Curves

The FCC field strength curves were developed for their simplicity and were applied at a time when most calculations of signal coverage were done by hand using nomographs such as NBS Tech Note 101[1]. The advantages and disadvantages of using this method are listed below:

Advantages

- needs only HAAT (height of above average terrain) of the transmitter along each radial
- determines field strength vs. distance from table lookup or curve interpolation
- ensures all users should have same coverage calculations for same location of transmitter

Disadvantages

- assumes constant terrain irregularity for entire U.S.
- accounts for only 2-10 mile terrain heights in determining HAAT
- predicts monotonically decreasing values of field strength vs. distance

Irregular Terrain Models

By contrast, the irregular terrain models have a different set of advantages and disadvantages to offer, as follows:

Advantages

- calculates more realistic results
- allows actual coverage to be determined and "shaped" to population through transmitter antenna location and directional antenna patterns
- allows interference to be evaluated and controlled through directional antenna patterns
- allows new models with improved prediction techniques to be developed

Disadvantages

- allows models with conflicting results to be developed
- requires terrain and other environmental data

The NTIA irregular terrain model is described in greater detail in the following sections.

NTIA IRREGULAR TERRAIN MODEL

Signal Loss

The loss of the signal as it propagates over irregular terrain can be estimated by our knowledge of physics. As a signal propagates in free space from an isotropic source, the signal is distributed uniformly in all directions; the signal distribution is related to the distance, D, from the source by the area of the sphere associated with that distance:

$$\text{Signal loss at } D \approx 4 \pi D^2 \qquad (1)$$

So if D is doubled, then the signal loss increases by 4 times from Eq. 1. The common description for this loss term is the free space propagation loss (FSL) and one expression for FSL is:

$$FSL = 20 \log F_{MHz} + 20 \log D_{km} + 32.45 \qquad (2)$$

To account for the losses due to the irregular terrain, a term called the reference attenuation, A_r, is added to FSL; depending upon the terrain profile from the transmitter to the receiver, A_r estimates the signal loss using theoretically derived expressions from line-of-sight, diffraction, and tropospheric scatter theories. In the line-of-sight region, A_r considers the direct ray and the interference caused by the indirect ground-reflected ray. In the diffraction region, A_r accounts for a weighted combination of knife-edge and smooth-Earth diffraction. In the scattering region, A_r evaluates the signal losses that are available due scattering of the signal from turbulent atmospheric conditions. The model estimates the signal losses in each region and makes a smooth transition from one region to the next. The expression for signal loss now becomes:

$$\text{Signal loss at } D = FSL + A_r(D) \qquad (3)$$

A good description of the physics involved with signal propagation through turbulent atmosphere over irregular terrain is found in NBS Tech Note 101[1].

Figure 1 shows a plot of signal loss vs. distance. In the case to be analyzed, the transmitter is assumed to have 100 W of effective isotropic radiated power (EIRP), to have an antenna height of 500 ft HAAT, and be operating at 100 MHz. The receiving antenna is assumed to be 30 ft above the ground and the signal level is displayed as field strength (FS) in decibels relative to 1 microvolt per meter (dBu). The top curve gives the FS values if only free space conditions were imposed. The dashed curve gives the FCC's F(50,50) curve for a transmitting antenna with 500 ft HAAT. The lower solid curve shows the estimated FS for a very smooth terrain path in south Florida. For the first 50 mi, the FCC's F(50,50) curve gives a lower field strength than that due to the irregular terrain model. That would be expected since the FM Broadcast curves assume a specific terrain irregularity, a mean value for the whole of the U.S.; for southern Florida, the terrain is flat with virtually no irregularity which is certainly not the case for most of the U.S. For the assumed antenna heights, the Earth's bulge would appear as the horizon to the transmitter and receiver when the two antennas are separated by about 39.4 mi. At that point the antenna would move from the line-of-sight region to the diffraction region. At about 80 mi the slope of the irregular terrain model FS curve becomes less steep; this is region that the scatter attenuation is less than the diffraction attenuation. From this distance and beyond, the signals due to scattering dominate.

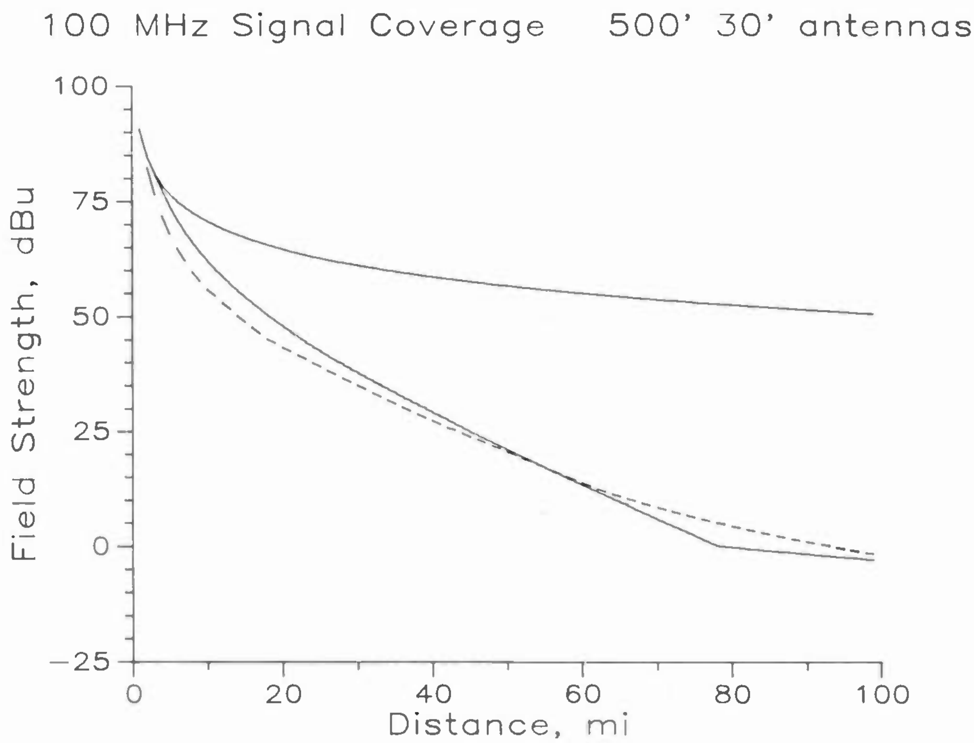

Figure 1. Comparison of predicted field strength for a smooth path in Florida using free space loss (top solid curve), FCC F(50,50) (dashed curve), and the NTIA irregular terrain models (bottom solid curve).

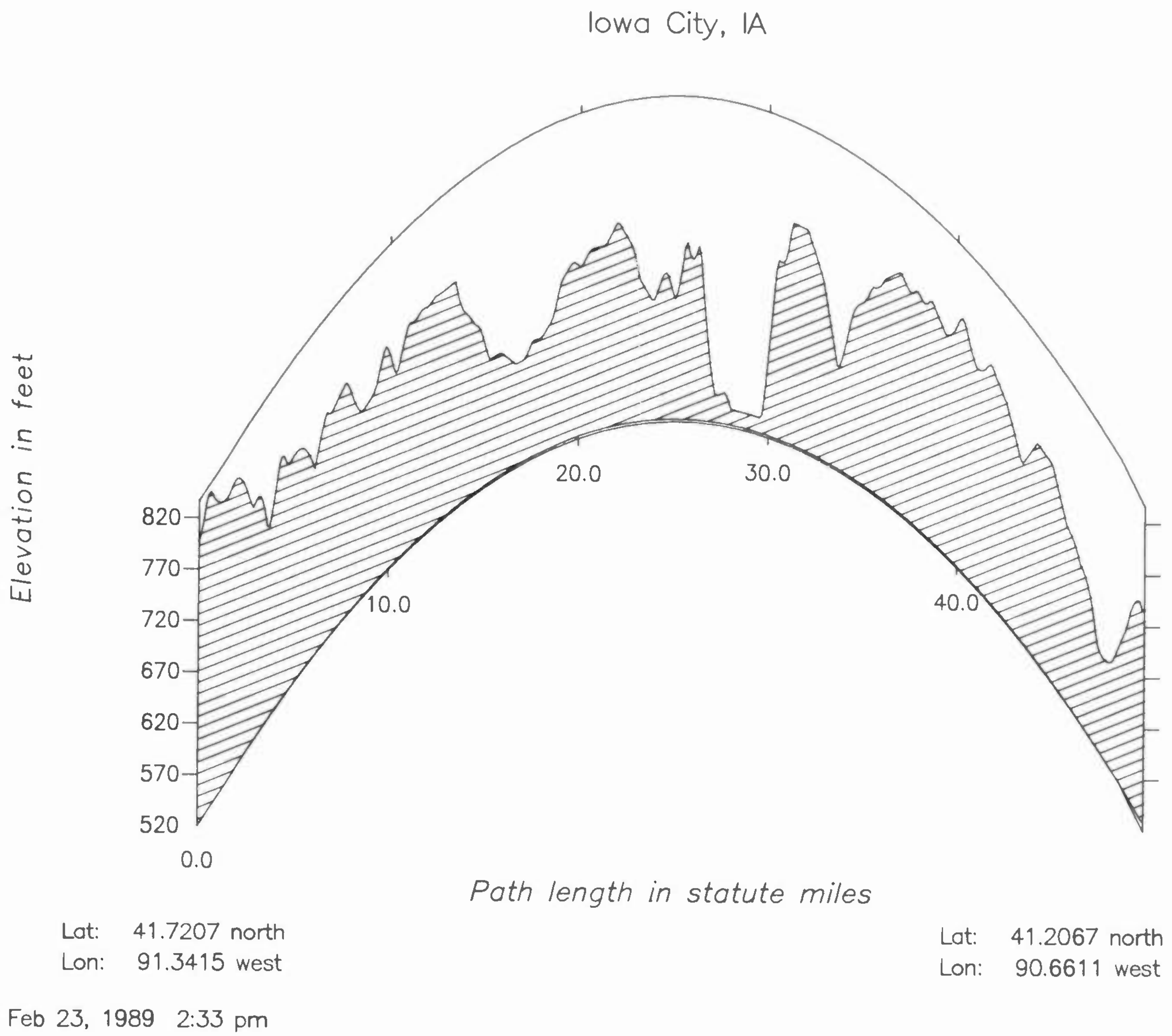

Figure 2. Terrain profile of a path near Iowa City, IA from a digitized terrain data base with 30 sec resolution.

The next example sets the same conditions for the transmitter but with the antenna now located near Iowa City, IA. At that location, the terrain would be described as rolling hills. Figure 2 shows the terrain profile out to 100 mi. Although the terrain appears to be rugged, there is a vertical exaggeration of 200:1 in the profile plot. Figure 3 shows the estimated field strength vs. distance for the path. Again the top curve is the FSL curve and the dashed curve is the FCC's F(50,50) curve for a 500 ft HAAT antenna. The lower solid curve is the estimated signal level at an antenna 30 ft above the ground as it is moved along the terrain on a radial away from the transmitter. Note that although the general trend is for the field strength to decrease as the receiver site is moved away from the transmitter, the signal does oscillate (as much as 25 dB in 5 mi at 25 mi, for example).

An example showing the comparison in more rugged terrain would reveal even larger oscillations of the irregular terrain model's estimated field strength and larger departures from the FCC's F(50,50) curve.

Signal Variations

Although the loss of the signal over irregular terrain can be estimated by our knowledge of physics and the terrain path profile from the transmitter to the receiver, there are too many possible indirect paths, too many man-made obstacles, and too many features created by nature to make a general purpose deterministic model of the signal loss. Based on many measurements and observations, the irregular terrain model can be

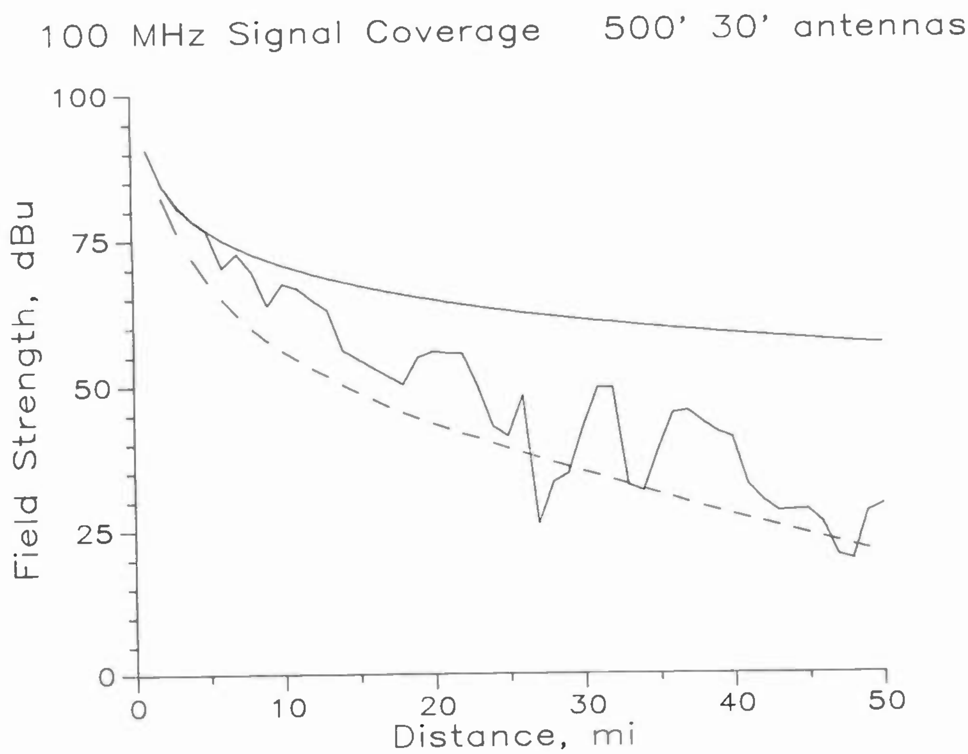

Figure 3. Comparison of predicted field strength for the Iowa City path using free space loss (top solid curve), FCC F(50,50) curves (dashed curve), and the NTIA irregular terrain models (bottom solid curve).

modified to account for these variations that happen over time and from location to location. The model considers these variations by specifying the quantiles of the observed variations; in other words, the signal loss value will not be exceeded for a given fraction of time and for a given fraction of the locations. The fractions of time and location are specified by the model's user and the model determines the loss associated with the path and the specified fractions of time and locations. A good description of the signal variability is found in Ref. 3. The signal loss expression at the distance D now becomes:

$$\text{Signal loss} = FSL + A_r(D) + V(D, q_T, q_L) \qquad (4)$$

To illustrate this variability in signal loss, consider the Iowa City path from Figure 2. In Figure 4, the FSL curve is again plotted as the top curve. The lower three curves show the variation with time; the top curve indicates the field strength at the receiver that would not be exceeded more than 10% of the time. The middle curve estimates the field strength for 50% of the time. The lower curve shows the field strength that would be exceeded for a minimum of 90% of the time. For these family of curves, the location variability was set to 50%. A similar set of curves would be plotted had the time variability been set to 50% (or any other value) and the location variability set to a range of values.

<u>Additional Factors</u>

Other factors can be added on to the signal loss expression that account for such losses as:

- urban attenuation
- building attenuation
- foliage attenuation

These require data bases or correction factors that would be independent of the losses associated with the irregular terrain. Note that in establishing the signal variability values in the irregular terrain model, the above factors were present during the signal measurements so that they are implicitly represented in the signal variation statistics.

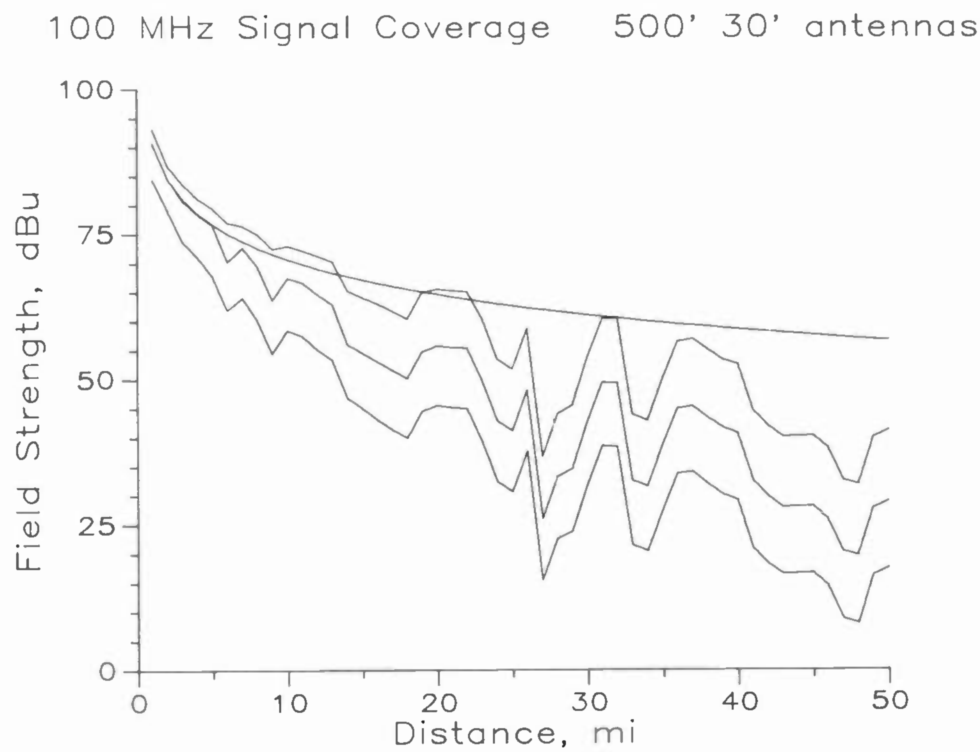

Figure 4. Comparison of predicted field strength for Iowa City path using free space loss (top curve), and three time variability values (10%, 50%, 90%) using the NTIA irregular terrain model.

APPLICATIONS

Although there are cases where the signal loss along a single radial is the desired result, the broadcaster is usually interested in the total area covered by a particular station or the locations of potential interference between two stations. The irregular terrain model evaluates the losses point to point along the radial from the transmitter to the receiver. By utilizing many radials, the losses in an area surrounding the transmitter can be estimated. A report by Jennings and Paulson[4] describes the general process of making a point-to-point irregular terrain model into one useful for analyzing a large area. In the samples given below, the NTIA Telecommunication Analysis Services' (TAS) program[5], the Communication System Performance Model (CSPM), was used to make the irregular terrain calculations and plots. Use of the TAS programs are available to consultants, private companies, public administrations, and government agencies by contacting the Institute for Telecommunication Sciences in Boulder, CO.

Single Station Coverage

KCRW, a public radio station in the Los Angeles area, was chosen to illustrate the use of the CSPM program for showing the coverage of a single station. The station operates at 89.9 MHz with 13.8 kW, an antenna height of 490 ft above ground, and a ground elevation of 1272 ft above msl. The CSPM model determined the regions that received the 60 dBu and 70 dBu signals for at least 50% of the locations and for at least 50% of the time within the region. Those areas were then plotted on the map as shown in Figure 5. The border of California was plotted in the background.

The station has an estimated 2-10 mi HAAT of 1100 ft, averaged over all radials, a maximum HAAT of 1600 ft along one radial and a minimum HAAT of 430 ft. The FCC F(50,50) curves indicate the coverage to 60 dBu to be approximately 32 mi and the coverage to 70 dBu to be about 22 mi for the given power and HAAT of 1100 ft.

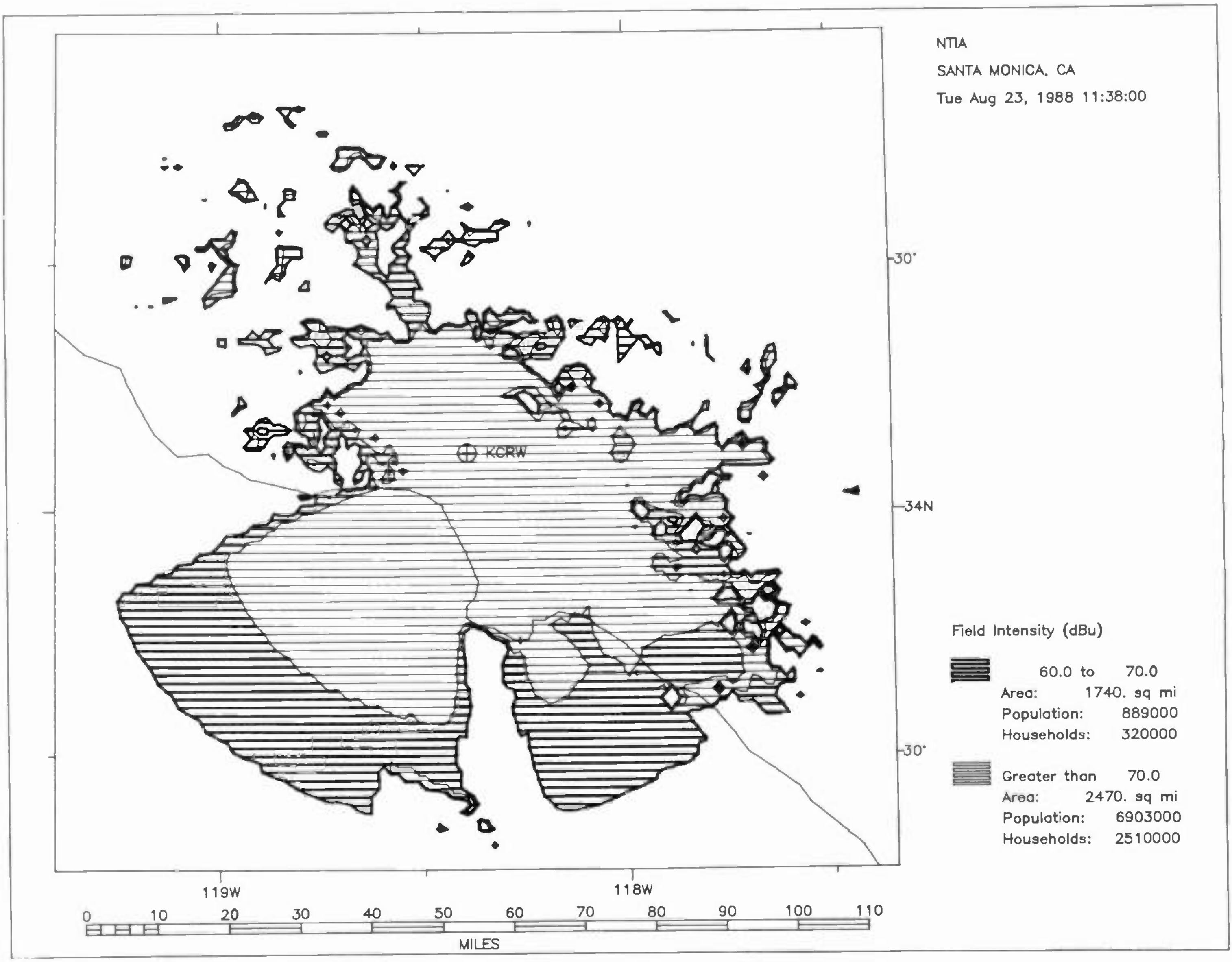

Figure 5. Signal coverage map for FM station KCRW.

The CSPM coverage map shown in Figure 5 illustrates how the mountains to the northeast limit KCRW's signal coverage and yet the signal is available 40 mi down the coast and up into the valleys to the northwest. Using the 1980 Census Data, the population and households that are covered by the 60 and 70 dBu signals were estimated. Those estimates along with the area within each contour are shown in the legend of Figure 5. Large transparent overlays of plots such as Figure 5 can be made to scale for placing over maps. The overlays assist in determining what towns or regions have adequate coverage and can be used to design antenna pattern requirements and other transmitter parameters.

Public Broadcasting Network

A second example using the CSPM program shows the combined coverage of many transmitters. Figure 6 shows the coverage of the public FM Broadcast radios that operate both within and outside the state of Ohio. The statistics on population, households, and area are for the defined coverage of all the stations serving the state of Ohio.

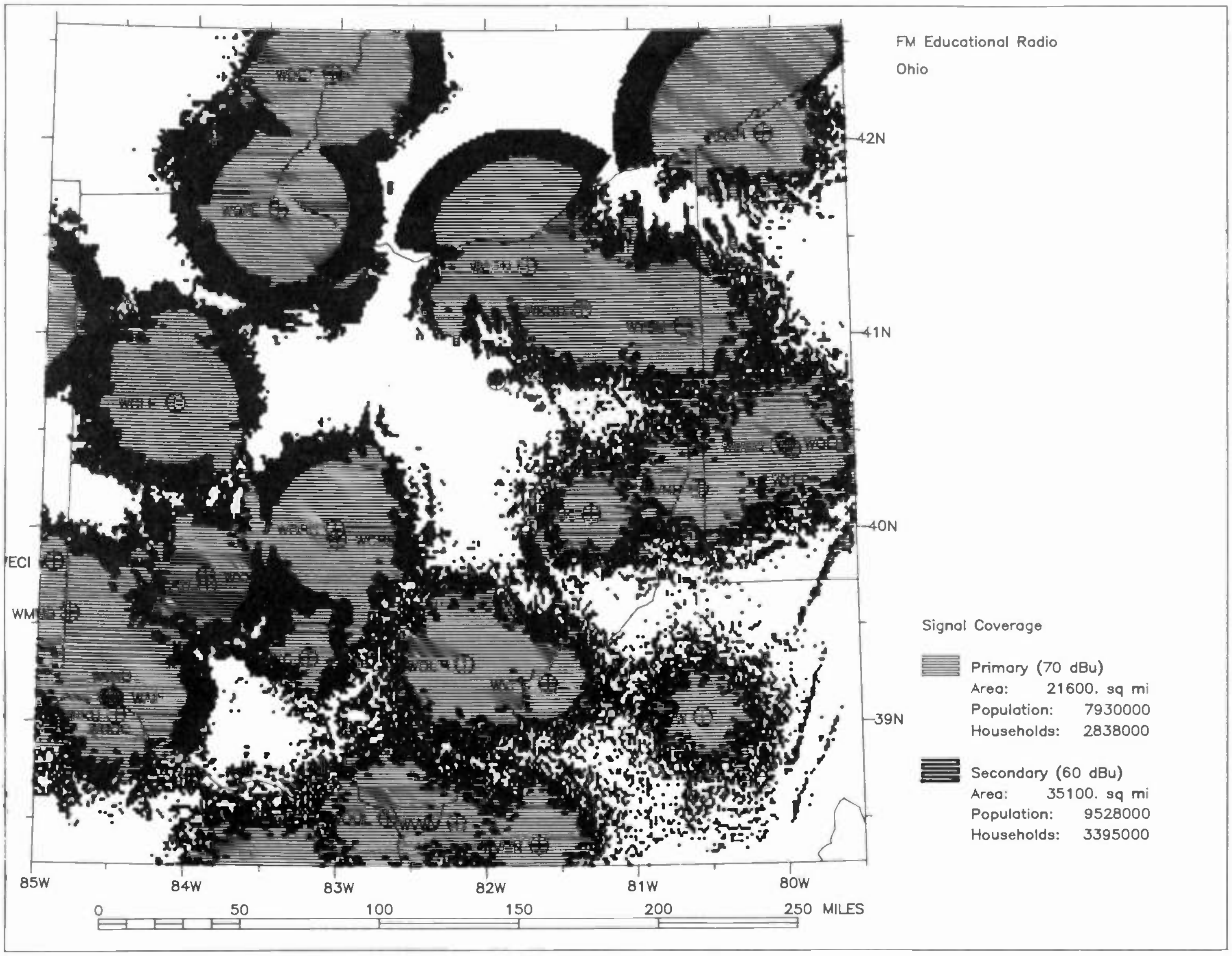

Figure 6. Signal coverage map for all public FM Broadcast stations in Ohio.

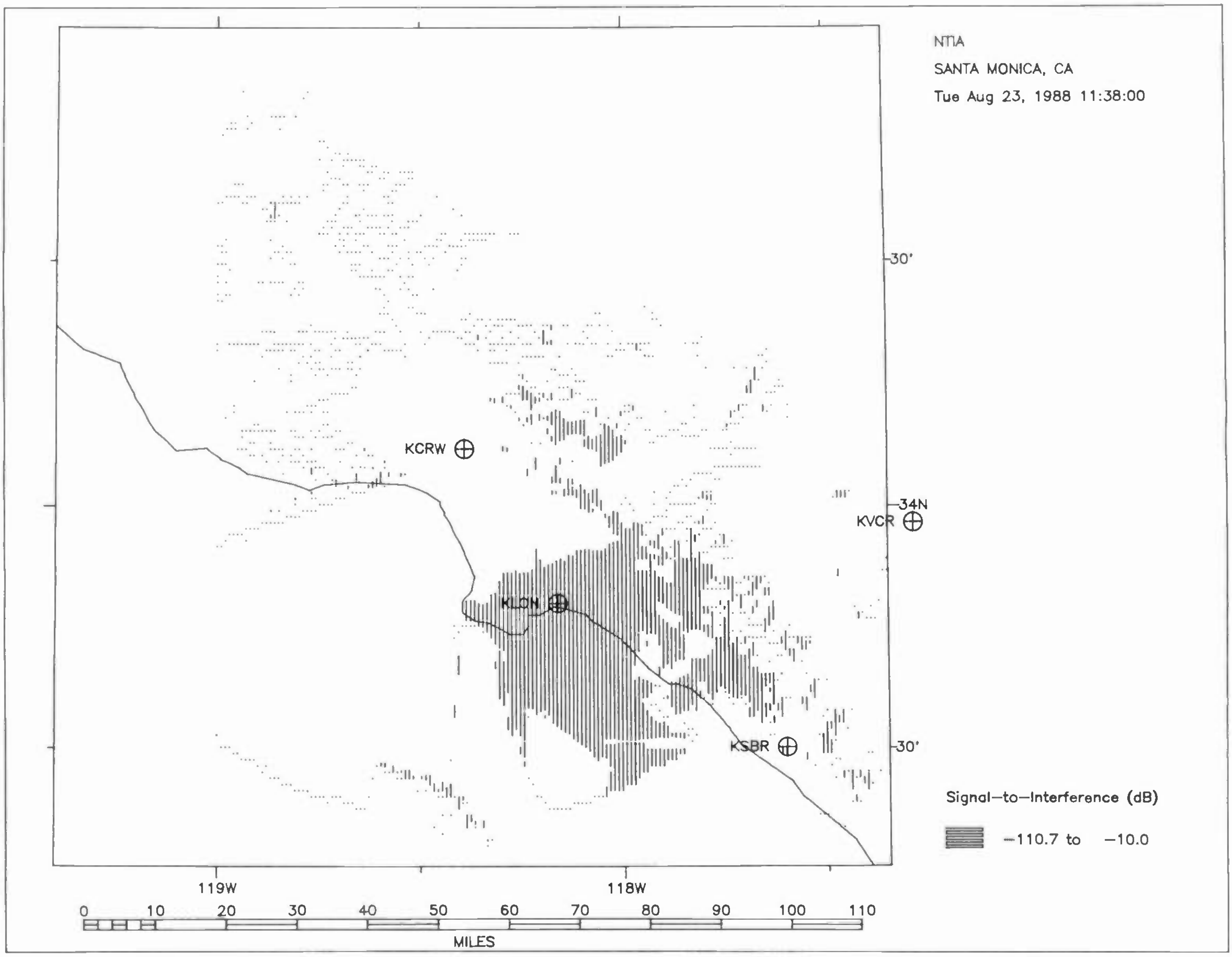

Figure 7. Signal-to-interference map for station KCRW with assumed nearby second-adjacent-channel stations.

Interference Regions

The CSPM program with its irregular terrain model can be used to analyze potential interference regions between two or more transmitters. Using the KCRW station as the desired signal station, Figure 7 shows regions of interference between KCRW and nearby FM stations. The signal statistics for the desired station KCRW were set for 50% of the locations and 50% of the time. Calculations of signal level of the other interfering FM stations in the area were made assuming statistics of 50% of the locations and 10% of the time. An assumed interference condition, such as second-adjacent-channel interference, is defined as that location where the desired signal is at least 60 dBu and where the interference signal is at least 10 dB greater than the desired signal. The interference regions shown in Figure 7 are fictitious since the transmitters are not co- or adjacent-channel stations; their locations and environmental conditions were chosen merely to illustrate an application of the irregular terrain model to a potentially real problem.

SUMMARY

The purpose of this paper has been to discuss and illustrate some of the available models for computing signal coverage from FM Broadcast stations. The models differ in their complexity and their ability to make realistic signal loss predictions. The models that use irregular terrain calculations along troposcatter paths give more realistic results than the FCC's F(50,50) FM Broadcast curves. The examples of this paper illustrate a few of the many possible applications using an irregular terrain model.

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4. Jennings, R. D., and S. J. Paulson, Communication system performance model for VHF and higher frequencies, OT Report 77-128. Available from NTIS, Access. No. PB 274458/AS.

5. Telecommunication Analysis Services, Institute for Telecommunication Sciences, Boulder CO 80303. Phone (303) 497-5301 for User's Guide to the Services or for more information.

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*National Technical Information Service, Springfield, VA 22161; phone (703) 487-4600.

CONSOLIDATING AM AND FM TRANSMITTER FACILITIES

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When economic belt tightening becomes a necessity in any industry, a common practice is to consolidate facilities. Often, real estate needs are reduced, maintenance expenses for facilities are decreased and management can more easily focus on problems, resulting in a more efficient operation.

This applies to many broadcast stations that may have two or more modes of transmission, such as AM and FM. Historically AM developed first, while FM was treated as a stepchild. AM radio transmitter sites were often set in low, swampy lands because of the radial ground system that is required and because the soil conductivity is usually better in such areas. The radio wave is launched, at least, over such terrain in hopes that the distant field intensity will be improved, even it it may be in dry, rough country. AM was king for many years, but over the last two decades, FM has overtaken AM in audience share for several reasons.

Because FM is about 100 times as high in frequency as AM, propagation of its signal is much different, needing as high an antenna as the combination of site and tower can provide. As a result, in many cases stations have two separate transmitter sites to maintain and on which to pay taxes, as well as a third site for the studios. This can represent a substantial investment in real estate, security and other factors, as well as adding a considerable amount of travel time to the engineer's usually full schedule.

What will be discussed here is a method by which one transmitter site and all its additional expenses may be eliminated by consolidating the two sites. The example given uses an FM transmitter site, but bear in mind that most any tall structure could perform the same function. Examples are communications towers, TV towers, water tanks, or most any tall structure that is located in an appropriate location to transmit the AM signal.

The key factor in the location of any transmitter site is antenna performance. Whether AM, FM, TV or any other form of transmission, maximum coverage area with no inteference to other stations is the main concern. The folded unipole has proven to be a very versatile antenna and a problem solver. With proper design and application, it can solve many problems which are either expensive, complex or impossible with conventional technology.

With conventional series fed antennas, height is very important, but the several advantages of the unipole preclude this importance, allowing the effective height to be tailored to suit the needs of the AM antenna. If the tower is too tall for the AM frequency, the upper portion may be made to effectively disappear, allowing the actual AM antenna height to be designed for the appropriate gain based on its height. (Fig. 1). If the tower is electrically short at the AM frequency, the impedance transformation capabilities of the unipole permit tuning the antenna to present a higher base resistance, thus improving the radiation efficiency and decreasing the transformation ratio required.

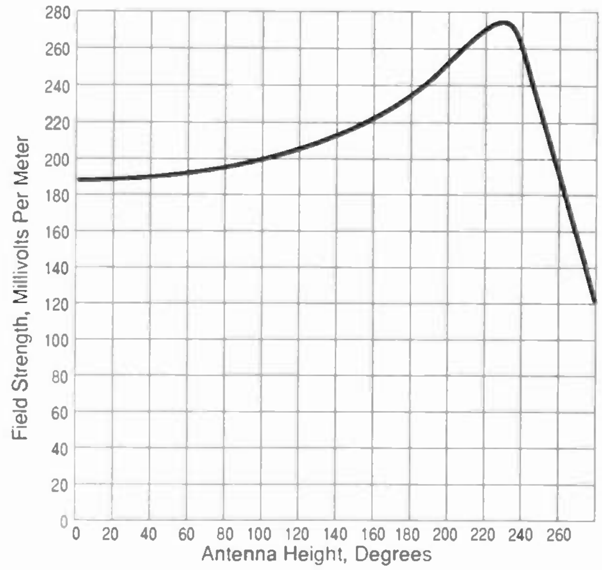

Figure 1. Inverse field strength at one mile for 1kw.

An important advantage of the folded unipole is that it is less ground dependent than the conventional series fed radiator. This characteristic has been noted as a result of empirical research, rather than scientific fact, but it has been very consistent. When conventional antennas with deteriorated or damaged ground systems have been converted to folded unipoles, the field strength has improved, often to a degree that is surprising. In one instance, a tower maintenance crew completely disconnected the ground system from an operating AM station's unipole antenna without any noticeable effect on the transmitter or the coverage area. The fact that the unipole combined with the tower forms a complete circuit in itself may possible contribute to this (Fig. 2)

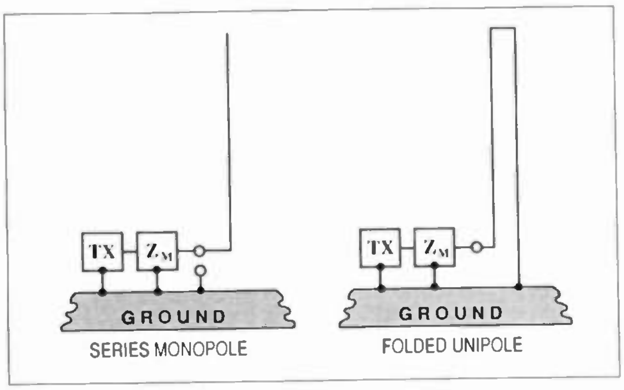

Figure 2. Antennas

The series fed antenna depends, of course, on the ground plane to complete its circuit, so a defective or deteriorated ground radial system would seem to have a greater effect on it. The preceding implies that the soil conductivity within the immediate area may be less important to a folded unipole antenna. In several cases this appears to be true.

OTHER CONSIDERATIONS

There are other aspects to consider when looking into the possibility of such a site consolidation. Probably the most expensive and difficult is that if a folded unipole is to be installed on an existing structure, all the tower guys must be broken up and insulators inserted at the proper intervals, or non-conductive guys used. If the guys are not broken up or non-conductive re-radiation from the guys will occur from them. Self-supporting towers, of course, have no problem in this regard.

The other expense would be the installation of the normal ground radial system as required by the FCC. It may be possible to utilize abbreviated ground systems if the velocity within the buried wires proves to be reduced by their proximity to the surrounding medium.

Experiments performed on buried antennas by the military indicate that the velocity may fall within the range of 25 to 60 percent of the speed of light when they are buried a few inches below the surface of the ground. Research needs to be done in this area, but it would be difficult to obtain funding for AM radio antenna research today.

Often, where an FM tower is located, there may be other towers nearby for communications and other services. Should these towers be of the proper physical dimensions, they could become parasitic radiators affecting the pattern of the AM antenna. They may be detuned with detuning skirts, which are simply a variation of the folded unipole. The flexibility of tuning the unipole allows it to perform as an excellent antenna or to be a "non antenna" to make any re-radiating structure effectively disappear from the near field of an AM antenna. This further supports the versatility of the folded unipole which allows the unipole to be able to utilize virtually any appropriate structure for a medium or short wave antenna.

PRACTICAL APPLICATIONS

So how may one approach the possibility of consolidating an AM antenna into another transmitting facility? Let's examine the following factors:

a) Describe the terrain surrounding the proposed structure under consideration. It need not be perfectly flat, as the ground radials can follow the surface. If its rocky, it may be more difficult but not impossible. Make a sketch of the area necessary for the ground system.

b) Do a survey of nearby structures, including their heights and distances from your tower. Don't overlook tower guys and power poles as potential re-radiators. Again, make a sketch showing their locations.

c) Make a map including the community of license and all territory that must be covered by the AM signal. Determine if this antenna will be in a good enough location to cover these areas.

d) Start the preliminary design work for the antenna. Determine the effective height that will be needed for the AM antenna. The broad latitude of the unipole allows you to select any height within reason. If the tower is too tall, determine what will be needed for the detuning section(s). Naturally, manufacturers of such products will be glad to assist in these areas.

e) Estimate the advantages of releasing the old transmitter site, i.e., its real estate value, elimination of security problems, utilities, tower lighting, etc. Don't forget that this can also simplify or eliminate one remote control system by controlling both transmitters with one system.

Compare and assess all the pros and cons from this evaluation to see if the advantages outweigh the trouble and expenses. Don't forget that this move will usually require the preparation and fee of an FCC Form 301, and upon completion, an FCC Form 302 must be submitted.

APPLICATION OF THE FOLDED UNIPOLE AS AN ANTENNA AND A DETUNING SKIRT

The simple construction and ease of tuning are what makes this method so attractive. A custom designed kit can usually be installed in a day by a two man crew and tuneup is simple and straight-forward. If there are VHF and UHF antennas on the tower, there is no problem if the skirt wires are kept way from them by a few inches. If necessary, skirt wires can be led around such antennas by additional brackets holding the wires. Since the structure on which the unipole is installed must be grounded, isocouplers to get these signals across a base insulator are not required.

For optimum performance and maximum service life, the folded unipole should be carefully designed to suit the needs and the details of the station. By doing so, the station will maximize the best coverage area and best audio quality, along with broad bandwidth for stability. Installing a unipole with whatever materials may be on hand, and without planning and installation knowledge can lead to disastrous results. This has already been done on several occasions. When an antenna is improperly tuned it can lead to a very high Q causing instability and erratic performance.

FEEDING THE FOLDED UNIPOLE ANTENNA

A commoning ring at the bottom ends of the skirt wires becomes the feed point for the antenna. The output of the antenna tuning unit is connected to this ring and impedance measurements are made at this point. The skirt appears to have a large effective diameter, decreasing the height to diameter ratio of the antenna and thereby broadening its bandwidth.

In the past, detuning skirts have been a problem in that a reactive element such as a coil or capacitor had to be mounted up on the tower at the detuning skirt. If such a reactor is used like this, it is desirable to be able to remotely tune it, so often a motor driven variable capacitor is installed for that purpose. However, this has been found to be unnecessary if conjugate detuning is employed. This simply means that rather than having the capacitor up on the tower in a weather-proof housing, it can be located in the antenna tuning unit housing or the transmitter building and connected to the detuning skirt with a length of coaxial cable (Fig. 3). Because the power levels and voltages are usually small, large coax is not necessary. Likewise, unless the station is high powered, the capacitor or coil may also be relatively small and inexpensive.

Naturally, the length of the coax is important in designing the conjugate system, but it can be easily calculated and the value and type of the detuning reactance determined. The simplest method would be to make the coax a half wavelength or a multiple of a half wavelength long, allowing for its velocity factor, because whatever impedance is seen looking into the detuning skirt will then be repeated at the input to the coax. Since the skirt input is alomst always inductive, a variable capacitor would then be connected across the input to the coax to detune the skirt.

A simple indicator of relative current in the coax would allow the system to be easily detuned. Thsis could consist of a toroidal pickup, a diode detector and a sensitive DC meter with a sensitivity control. Precise accuracy is unnecessary as you would simply tune for minimum current. All that is necessary is a relative indication. Such a detuning unit and indicator could be built on a standard rack panel and mounted t any convenient location. This would allow monitoring on a regular basis for optimum performance.

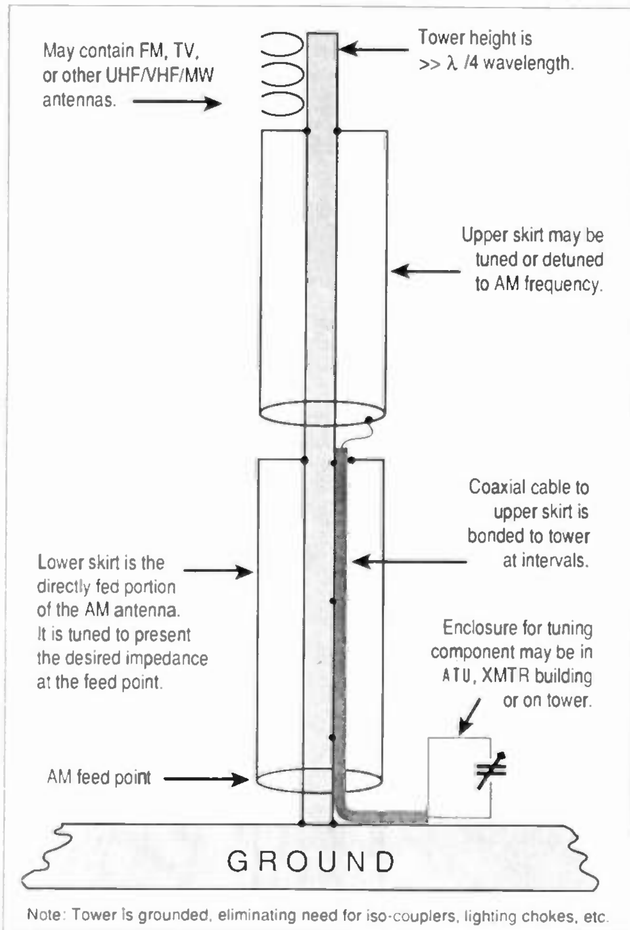

Figure 3. Conjugate tuning/detuning of upper antenna skirt

OTHER DESIGN CONSIDERATIONS

One factor that is often overlooked is the reduced propagation within any antenna with a decreased height to diameter (H/D) ratio. A tall, thin tower has a velocity > 95 to 98 percent of the speed of light. As an antenna becomes effectively "fatter", the H/D ratio decreases and the velocity also diminishes. This ratio may also be defined in terms of electrical diameter in degrees, the effects of which are illustrated in Figs. 4 and 5. Note that as the diameter of the antenna in degrees increases, the curves of the resistance and reactance deviate much less with the result that the slopes of these measurements in a practical antenna will also be gentler. What this means is improvement in bandwidth and greater stability with changes due to weather and the season.

In a conventional antenna, this diameter is a function of tower cross section, but when a unipole is installed, the skirt wires increase this diameter greatly. They form the "skeleton" of a circle which, for practical purposes, becomes the effective diameter of the antenna. Installation of a folded unipole may improve almost any antenna in several respects.

Tests indicate that the typical unipole has a velocity in the range of 84 to 90 percent of the speed of light. This must be taken into consideration when designing the antenna and its dimensions. Clearly, this becomes an advantage when an antenna must be electrically short. Whatever field gain that a folded unipole may have is due to this effect, but note that this is only a very small gain.

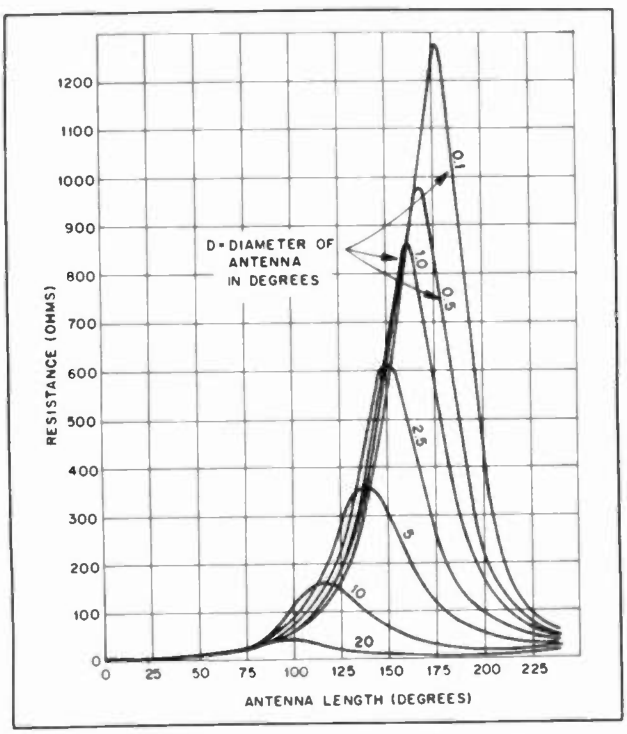

Figure 4. Feed-Point resistances (R_{rad} [2]) of monopoles with varying diameters. These values were computed for perfect ground. (360 degrees = 1 λ.)

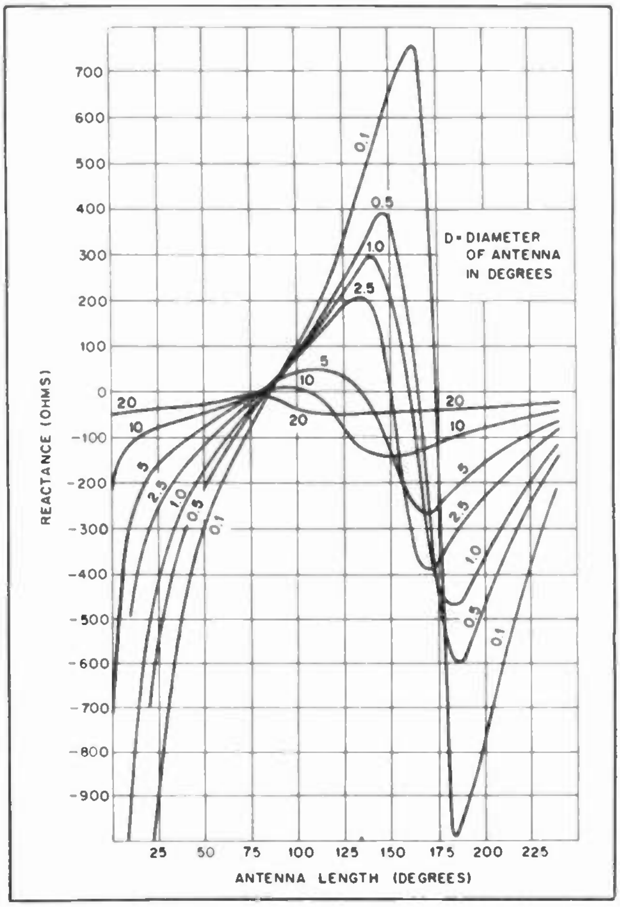

Figure 5. Feed-point reactances (over perfect ground) of monopoles with varying diameters.

The unipole can also be utilized as a impedance transformer, stepping up the small value of resistance characteristic of short antennas. When a conventional ATU is employed, this means that the transformation ratio can be minimized, reducing the Q, decreasing stored energy and broadening the bandwidth of the ATU. This advantage along with the inherent broad bandwidth of the unipole can greatly improve the performance of an existing short antenna.

BANDWIDTH

RF system bandwidth is very important and it includes the transmitter output network, the transmission line, the ATU and the antenna. We sometimes overlook the fact that if the output of the transmission line sees a poor bandpass, it can transform sideband energy into strange values seen at the transmitter output. Modulators may run hot, trying to pump energy into poor impedance matches. Audio processing cannot solve poor RF system bandwidth by trying to push more sideband energy into reactive loads. It may make the station sound louder at the expense of modulator power and sometimes a crunch in the audio. A broad, well matched RF system can solve a great many problems. Such an improvement may well be noted if a station elects to consolidate its AM onto another facility by this method.

SUMMARY

There exists a practical, proven method by which an AM broadcast station may share another structure for its antenna. While one advantage is cost reduction, it is possible that there may be an improvement in audio quality and even coverage area if designed, installed and tuned properly.

Expensive components such as iso-couplers, lighting chokes and base insulators are eliminated. The structure and antenna are directly connected to ground which can reduce lightning and static electricity problems.

AM broacasting still reaches a substantial audience, so few combined stations are ready to cast it aside. The folded unipole antenna permits a substantial reduction in operating costs, but it must be properly designed, installed, and tuned. It is hoped that the foregoing will aid managers and engineers in making decisions about both economizing for the survival and yet improving their facilities.

Assistance from Jim Burgess and Dick Ives of the staff of San Juan College, Farmington, New Mexico has been invaluable and is gratefully acknowledged.

GAIN FIGURE OF SIDE MOUNTED OMNI-DIRECTIONAL CP-FM ANTENNAS

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INTRODUCTION

In characterizing the radiation properties of any antenna, an accurate determination of its pattern and gain is of great importance. In broadcasting, accurate gain figures translate into maximum coverage efficiency, thus greater income for the station. A gain figure, which is less than the actual gain of the antenna, will increase ERP in areas where co-channel protection is critical and may cause legal problems. Similarly a gain figure that exceeds the actual gain of the antenna causes reduction in coverage and revenue.

In general, the gain of an antenna has a linear dependance on the directionality of its pattern. In fact, in some cases it is valid to ignore minimal losses in the antenna system and assume that the pattern directivity is indeed a good approximation to the actual gain of the antenna. In most practical situations, however, a more accurate gain figure is desired which accounts for such losses. It should be pointed out, that in most cases the nature of the losses are not as important as their effect on the antenna gain.

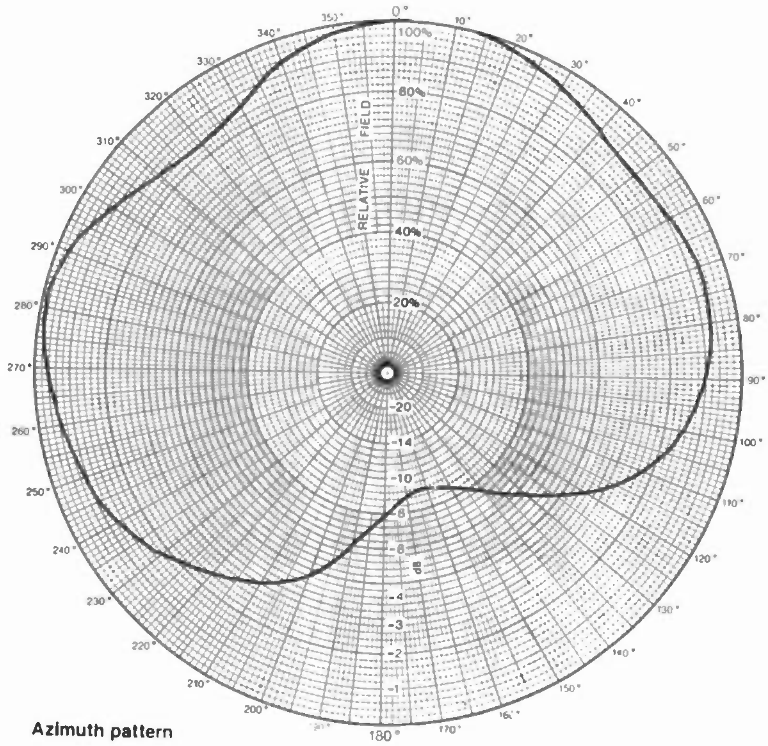
In situations where the elements of the antenna array do not interact with the mounting structures or if they do, such interactions do not significantly affect the performance characteristics of the elements. It is valid to assume a linear depenence of the antenna gain on the directivity of the pattern. The gain of panel antennas, which are side mounted on towers, for instance, are not affected by the tower structure to the extent that it be of concern when determining the antenna gain. A great majority of FM antennas, however, are side mounted omnidirectional CP antennas that interact strongly with their mounting structure. In such cases, there is no linear relation between the antenna gain and pattern directivity. Presence of the tower not only affects the vertical and horizontal polarization patterns, it also affects the power content in each polarization. This is a natural consequence of near field proximity of the tower to the antenna. An indication of such strong interaction is a drastic change in the internal impedance of the radiating CP antenna, caused directly by side mounting the antenna.

In this paper we establish the relation between pattern directivity and gain in both polarizations for such antennas. Before addressing gain, it is necessary to clarify certain aspects of pattern measurements of side mounted Omni-CP antennas.

PATTERN MEASUREMENTS IN OPEN RANGES

To measure the azimuth pattern of a side mount FM antenna, the support structure must be accounted for due to its complex interaction with the Omni-CP antenna. In almost all cases a CP element is mounted on a tower, which is of exact dimensions to the actual tower and the combined structure is positioned on a turntable. The antenna is then operated in the receive mode in the presence of a distant transmitting antenna. A scale version of the antenna structure at higher frequencies may also be used. However, such measurements are not dependable unless every facet of the structure is scaled. In most cases, such detail scaling is prohibitive, due to costs involved.

Figures 1, 2 and 3 show patterns of a typical side mounted CP-FM antenna. Figure 1 is a horizontal polarization pattern, normalized to its own maximum. Figure 2 is the same for vertical polarization. Figure 3 shows received voltages for both polarizations normalized to the maximum H-pol received voltage. It should be noted that the difference between the level of the vertical and horizontal maximum voltages is not necessarily due to the antenna alone. In fact, it is partly due to the difference in the ground reflection coefficients of vertically and horizontally polarized waves. This error that is introduced by ground reflections in an open range, may be corrected by using a cavity backed rotatable dipole, which is positioned at the location of the receiving antenna. By comparing the levels of the received vertical and horizontal polarization by this cavity, and subtracting it from that of the main antenna, we arrive at Figure 4, which indicates the true difference bewteen the levels of V-pol and H-pol radiations. (See Appendix)



Azimuth pattern

FIGURE 1

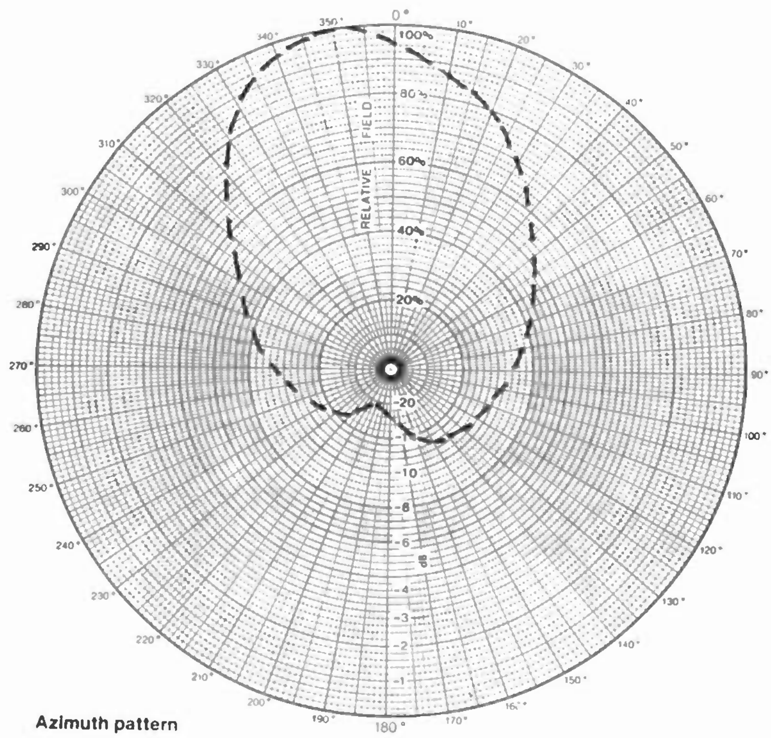
H-Polarized radiation pattern
of side mounted Omni-CP antenna

D = 1.5

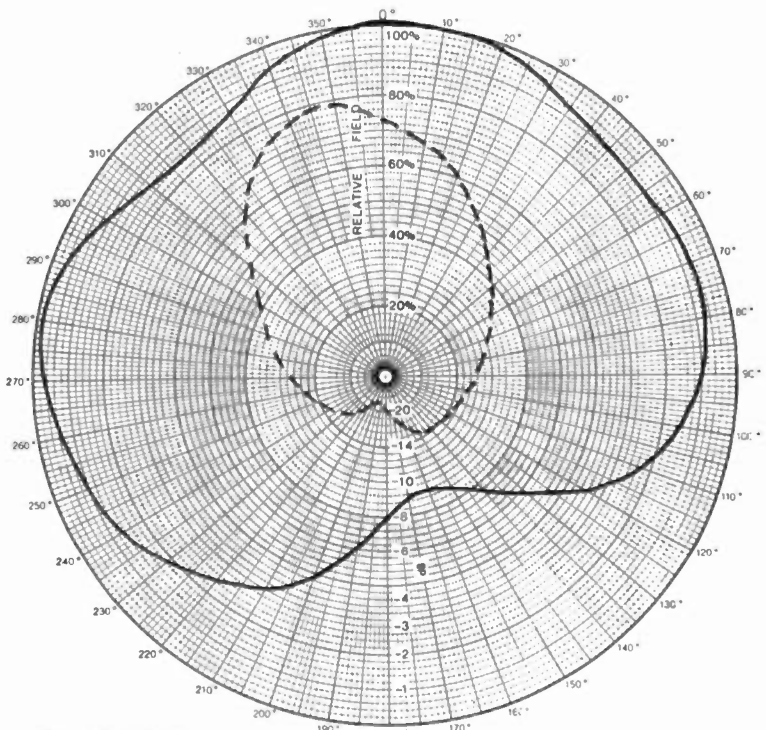
FIGURE 2

V-Polarized radiation of a side
mounted directionalized Omni-CP
antenna.

D = 3.9



Azimuth pattern



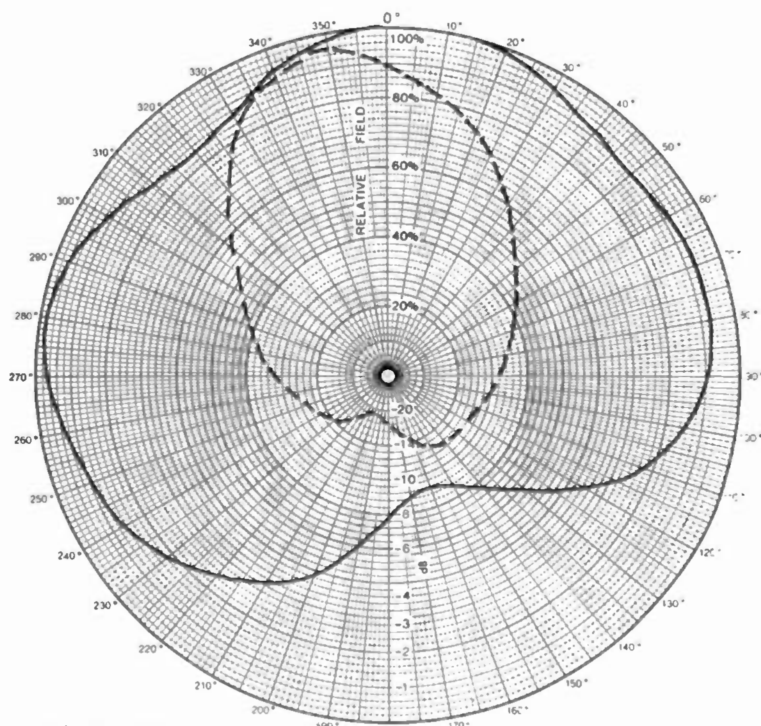
Azimuth pattern

FIGURE 3

Pattern of a side mounted Omni-CP antenna normalized to maximum level of H-Polarization (as measured)

FIGURE 4

Pattern of side mounted Omni-CP antenna normalized to maximum level of H-Polarization (corrected for range error)



Azimuth pattern

GAIN OF SIDE MOUNTED OMNI-CP ANTENNAS

Before proceeding any further, let us clarify the difference between the gain and directivity of an antenna. Directivity is a measure of the directionality of an antenna pattern. The more directional an antenna is, the higher the directivity. In broadcasting, when we talk about directivity, we usually mean directivity in the horizon plane. Directivities of Figures 1 and 2 are the ratios of the area of the circle containing the patterns to the area, contained by the patterns as is indicated by D in these Figures.

Gain on the other hand is a power ratio. More specifically, it is the ratio of the maximum radiated power of the antenna, to the maximum radiated power of a reference antenna (in our case an ideal 100% efficient dipole). In general, directivity and gain are related by:

$$G = \mu \times D$$

where:
G = Gain
D = Directivity
μ = Efficiency Factor

μ is factor that accounts for antenna losses such as mismatch losses, ohmic losses, aperture losses and polarization losses. For example; An ideal omnidirectional CP antenna has a directivity of 1 in both vertical and horizontal polarization, however, the gain of this ideal antenna is 0.5. For this antenna:

$$\mu = 0.5$$

which in this case is the polarization mismatch loss.

We may now go back to Figure 4. An interesting characteristic of this figure is that the horizontal polarization pattern has a higher gain even though the directivity of the V-pol is higher.

Clearly, the presence of the tower has disturbed the power split between the two polarization components. To be more exact, the presence of the tower has transformed the CP radiation to elliptically polarized radiation with varying axial ratio and polarization angles in the horizontal plane.

At this point, an assumption that μ = 0.5 would clearly run into contradiction when computing the gain of this antenna in H-pol or V-pol. The gain of V-pol under this assumption would exceed the gain of H-pol, while Figure 4 indicates the contrary.

The question is then, how, given Figure 4, one should determine the gain of the antenna in each polarization? A short treatment of the problem is given in Appendix A according to the results obtained in this appendix; the gain figures for each polarization is obtained using following relations:

$$G_H = \frac{\alpha D_V}{D_H + \alpha D_V} D_H$$

and

$$G_V = \frac{G_H}{\alpha}$$

where

α : Ratio of max H-pol to max V-pol

D_H : H-pol pattern directivity

D_V : V-pol pattern directivity

G_H : H-pol gain

G_V : V-pol gain

Certain aspects of these results are quite interesting:

a) Note that the polarization mismatch loss is not a constant (as is normally assumed), but is a function of both V-pol and H-pol directivity. This is a natural consequence of the strong interaction between the tower and the antenna.

b) μ = 0.5 only when:

$$\alpha = \frac{D_H}{D_V}$$

Under this condition, there is an even power split on both components and the gain of one over the other is primarily due to the difference in their directivity. This will strictly happen only when the tower is in the far field of the antenna and does not interact with the antenna. The effect of the tower in this scale is only a scattering effect, however, because of the near proximity of the tower to the omni element, the condition in (b) rarely occurs. It is recommended in these cases that the gain of the antenna be computed using the above equations, in order to achieve an accurate gain figure.

For instance, in the present example where:

$$D_H = 1.5$$

$$D_V = 3.9$$

$$\alpha = 0.92$$

We have

$$G_H = \frac{(0.92)(3.9)(1.5)}{(1.5) + (0.92)(3.9)} = 1.06$$

$$G_V = 1.15$$

which are consistant with the relative levels of the signals in the two polarizations, and yet quite different from the gain figures obtained by, erroneously, assuming μ = 0.5. In the case of H-polarization gain, the above result indicates a difference of 30% between the actual gain of the antenna and what would otherwise be given as gain by assuming $\mu = 0.5$. These results point out the significance of tower-antenna interaction in situations where an omnidirectional CP antenna is side mounted on a tower or pole. Because of this strong interaction, it is highly recommended that all measurements on such antennas and towers be carried out in full scale. This eliminates ambiguities associated with an accurate determination of gain at operating frequencies.

SUMMARY

In almost all situations where an omnidirectional antenna is side mounted on a pole or tower, there is a strong interaction between the tower and antenna. Such interaction disturbs the antennas CP-radiation and its gain in principal polarizations. Due to these interactions, linear relation between directivity and gain no longer applies. To arrive at reliable gain figures, we've proposed new expressions which take tower-antenna interaction into account. A detailed discussion of measurements and calculations related to the subject, is presented in the appendix.

APPENDIX A

In this appendix, we explain, in steps, the proposed technique in determining the gain figure, in principal polarizations, of a side mounted Omni-CP antenna. The first few steps explain the procedure to elminate the range error from measurements. Steps 4 through 6, give the details of the derivation of gain equations.

Steps

1. Set a cavity backed (with rotatable dipole) at the level of the antenna under the test. Measure the relative transmission loss for vertical and horizontal polarization normalize to vertical polarization. Call it:

$$\sigma_{RANGE}$$

2. Measure the relative maximum powers in vertical and horizontal polarization for the antenna under the test (normalize to vertical polarization). Call it:

$$G_m$$

3. Correct for the range error:

$$g_{(dB)} = G_{m(dB)} - \sigma_{(dB)}$$

and

$$\alpha = 10^{\left(\frac{g}{10}\right)}$$

4. Let D_H = horizontal polarization directivity

and

D_V = vertical polarization directivity

then

$$G_H = \eta_H D_H$$

$$G_V = \eta_V D_V$$

where η_H and η_V are the antenna efficiency factor for horizontal and vertical polarizations, respectively.

For an input power P_i to the antenna:

$$P_H = P_{max}\text{(H-pol)} = P_i \eta_H D_H$$

$$P_V = P_{max}\text{(V-pol)} = P_i \eta_V D_V$$

then

$$\frac{P_H}{P_V} = \left(\frac{\eta_H}{\eta_V}\right)\left(\frac{D_H}{D_V}\right) = \alpha$$

5. For an ideal CP antenna:

$$\eta_H + \eta_V = 1$$

Where we assume no ohmic losses. Further we assume that η_H and η_V are normalized to the power available to the antenna (do not include mismatch losses). In other words η_H and η_V are polarized mismatch losses.

Let us further define:

$$\frac{\eta_H}{\eta_V} = R$$

then

$$\alpha = R \frac{D_H}{D_V}$$

or

$$R = \alpha \frac{D_H}{D_V}$$

then

$$\eta_H = 1 - \eta_V$$

and

$$\eta_V = \frac{1}{1+R}$$

6. And consequently:

$$G_H = \frac{R}{1+R} D_H$$

$$G_V = \frac{1}{1+R} D_V$$

or

$$G = \frac{\alpha D_V}{D_H + \alpha D_V} D_H$$

and

$$G_V = \frac{G_H}{\alpha}$$

In the above equations

α is measured

D_H and D_V are computed by direct pattern integration of the antennas.

In most cases of interest, the elevation pattern of the element may be assumed to be close to that of the dipole. In these cases, the azimuth directivity is a good enough approximation to the actual element directivity W.R.T. a dipole.

THE NAB TEST CD—USE AND APPLICATIONS

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The NAB Broadcast and Audio System Test CD provides broadcast engineers with the means to quickly and conveniently generate many of the electronic test signals commonly used in the industry. Released in late-1988, the NAB Test CD is the first test disc designed to extend beyond player evaluation, with many of the provided signals intended to be useful in evaluating several other devices and systems in the broadcast transmission chain. Although useable with virtually any CD player, special tracks are provided to allow rapid investigation of important accuracy parameters of the player employed.

Introduction

The initial suggestion to produce the NAB Test CD came from the National Radio Systems Committee (NRSC), a joint NAB/EIA committee that was reformed in 1985 to find ways to improve the technical quality of AM broadcast transmission and reception. The NRSC suggested that a compact disc would be the ideal medium to distribute the specialized noise test signal required for compliance verification to the NRSC-1 and NRSC-2 AM technical standards. In addition to this signal, it was felt that other test waveforms could be included which would be useful to broadcast engineers. After several brainstorming sessions with a number of industry engineers, a track list was developed. The completed CD contains 99 tracks (the maximum allowed).

All signals on the CD are 100% digitally synthesized, except for initial announcement tracks 1 and 2. This means the test waveforms are mathematically perfect, within the 16 bit quantization limit and 20 kHz bandwidth allowed by compact disc specifications. Additionally, many of the signals were recorded at or near the maximum allowed CD peak level. This yields the greatest signal to noise ratio possible, nearly 100 dB on most players. Therefore, care should be exercised when using the CD, because several of the provided high frequency test signals can easily damage amplifiers and loudspeakers due to overload. (Most musical compact discs are recorded at a 15 or 20 dB lower average level to allow transient peaks to pass without causing clipping distortion.)

To provide maximum flexibility, 257 separate test signals are "indexed" within the 99 available disc tracks. Many players support index indication as an integral front panel display function. The use of this function allows greater ease in determining the location of test sequence breakpoints found within several of the tracks.

CD Player Performance Measurement

The first 13 tracks found on the NAB Test CD are designed to assist the user in confirming proper equipment setup and player performance. Tracks 1 and 2 contain voice announcements intended to assist in verifying the player left and right output channels have been correctly assigned and phased properly. Tracks 3 through 13 are provided to test specific elements of the player employed. Several signals on these tracks are recorded at seeming odd frequencies, such as 1001 Hz, because they are chosen to be at exact sub-multiples of the 44.1 kHz CD sampling frequency. Exact sub-multiple frequencies yield the best possible performance from any given player.

Tracks 3, 4 and 5 provide sinusoidal 1001 Hz reference, 40 Hz low limit and 19,999 Hz high limit signals respectively, recorded at maximum disc level (referred to as 0 dB hereafter), L=R. As viewed on a swept oscilloscope, each of these signals should be at the same peak-to-peak amplitude, without noticeable distortion. Additionally, maximum channel phase error can be evaluated when track 5 is viewed on an oscilloscope set up in X-Y mode.

Figure 1 shows the acceptable track 5 oscilloscope X-Y trace produced by one player. Note the minimal phase error (spreading) caused when this high frequency track is played. The trace of Figure 2 is produced by another player, playing the same track under identical test conditions. A severe phase error, approaching 90 degrees, is observed along with a widening of the trace outline.

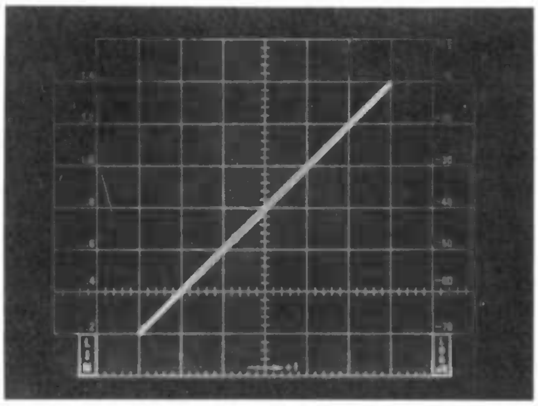

Figure 1. X-Y display of phase error, dual D/A CD player (19999 Hz, track 5).

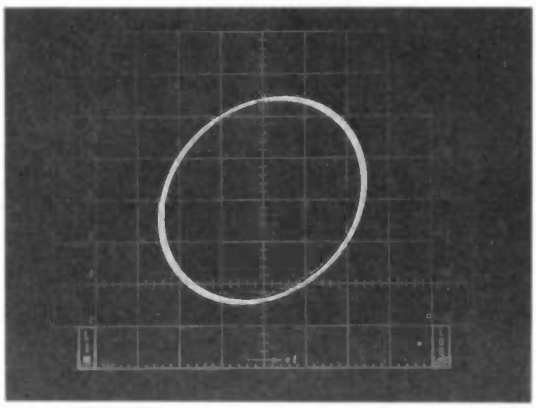

Figure 2. X-Y display of phase error, single D/A player (19999 Hz, track 5).

The reason for the major difference between the two selected players can be explained using the block diagrams of Figures 3 and 4. Figure 3 illustrates a simplified view of the digital-to-analog (D/A) converter section of the player exhibiting good phase response when track 5 is played. The "DISC AUDIO DATA" line, which consists of left channel/right channel information interleaved sequentially, is connected to the wiper of a switch. This switch is clocked back and forth between the inputs of two separate D/A converters by the "L/R CLOCK X 4" signal.

Generally, L/R CLOCK is defined as being twice the disc sampling frequency (88.2 kHz), and is provided by control circuits in the player. However, in this arrangement, the L/R CLOCK and DISC AUDIO DATA rates operate at four times their defined rates, or 352.8 kHz and 176.4 kHz respectively (for each 16 bit data word). This is known as an "oversampled" playback system. During the L/R clock period, each sequential left and right channel data word is loaded into a separate D/A converter, which converts the digital audio information into an analog level. Once each converter completes its task, another set of switches, clocked by the L/R CLOCK X 4 signal, divided by two, loads the analog signals into sample and hold amplifiers (sometimes called "deglitchers") which "freeze" the levels until the next audio samples are ready. Before exiting the player as recovered audio, however, identical low pass reconstruction filters remove high frequency images (or aliasing

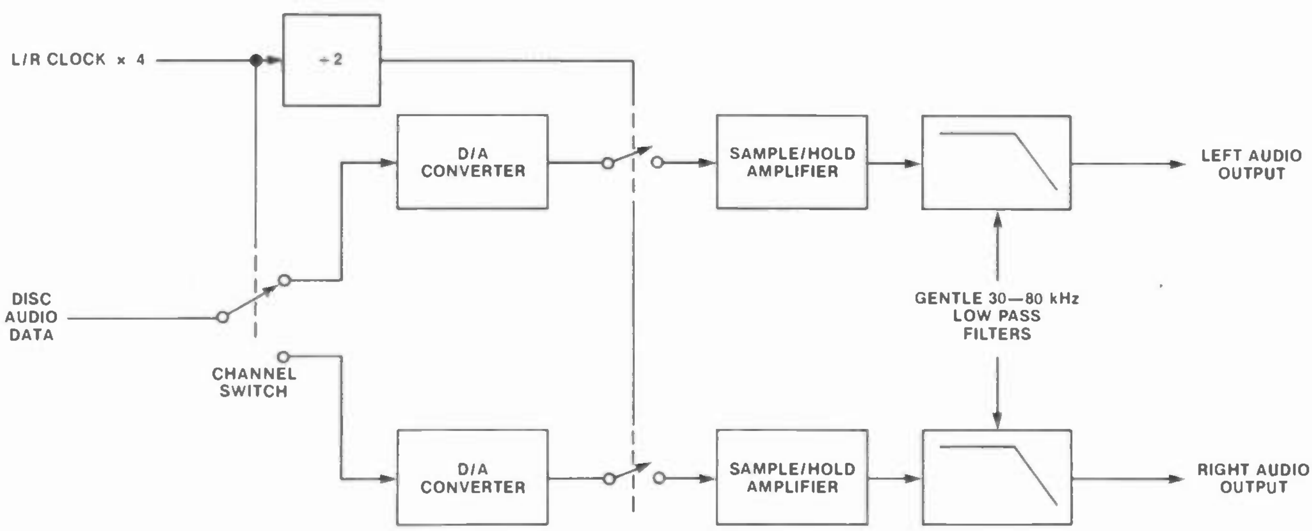

Figure 3. Compact Disc conversion system, 176.4 kHz conversion rate, dual D/A converters (simplified).

products) produced by the D/A conversion process. Since an oversampled system is used, these images fall above 88.2 kHz (half the 176.4 kHz audio data sampling rate). Therefore, relatively gentle and easily designed low pass filters are usually employed to remove these images.

The block diagram of Figure 4 illustrates the conversion system utilized in the player that produced the poor phase response to track 5, as seen in Figure 2. In a simpler and more conventional approach, the DISC AUDIO DATA and L/R CLOCK rates operate at 44.1 kHz and 88.2 kHz, respectively. The audio data is converted to an analog level sequentially, with a single D/A converter. As each conversion takes place, the recovered audio sample is loaded into a sample and hold amplifier, one for each channel. As in the previous system, low pass filters are used to remove high frequency images. However, these images are present above 22.05 kHz (half the 44.1 kHz sampling rate), and filters with a much steeper attenuation characteristic must be used. (The widening of the trace outline in the photo of Figure 2, as noted earlier, is caused by the inability of these filters to provide adequate suppression above 22.05 kHz.)

Other than the complexity of required low pass filters, the major difference between the two playback systems is that the audio information does not arrive at the sample and hold amplifiers concurrently in the system of Figure 4, causing the severe phase error to occur. When used with musical discs, this trade-off is thought to be acceptable in players, and a considerable cost savings can be realized by minimizing the amount of hardware required. When used with the NAB Test CD, however, this type of player is often likely to contribute a greater error to test signal accuracy. Fortunately, virtually all CD players intended for the broadcast environment employ the preferable oversampling and dual D/A conversion techniques.

Additional tracks in the player performance section of the NAB Test CD include a midband harmonic distortion test (3149 Hz, track 6), frequency and level sweeps (tracks 7 and 8), and high frequency separation tests (tracks 9 and 10).

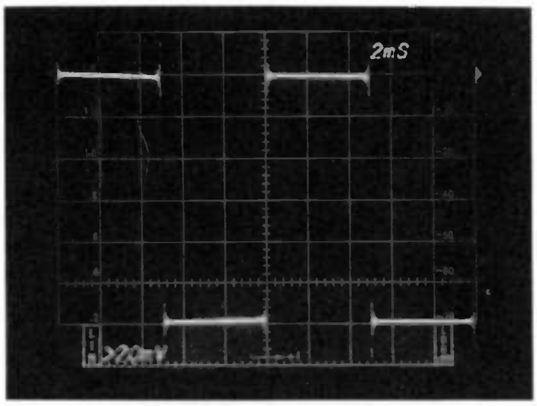

Figure 5. 100 Hz square wave, dual D/A conversion player.

The 100 Hz square wave signal found on track 11 can provide useful information with regard to the quality of the player low pass reconstruction filters and low frequency response. The photo of Figure 5 shows the swept oscilloscope trace (as

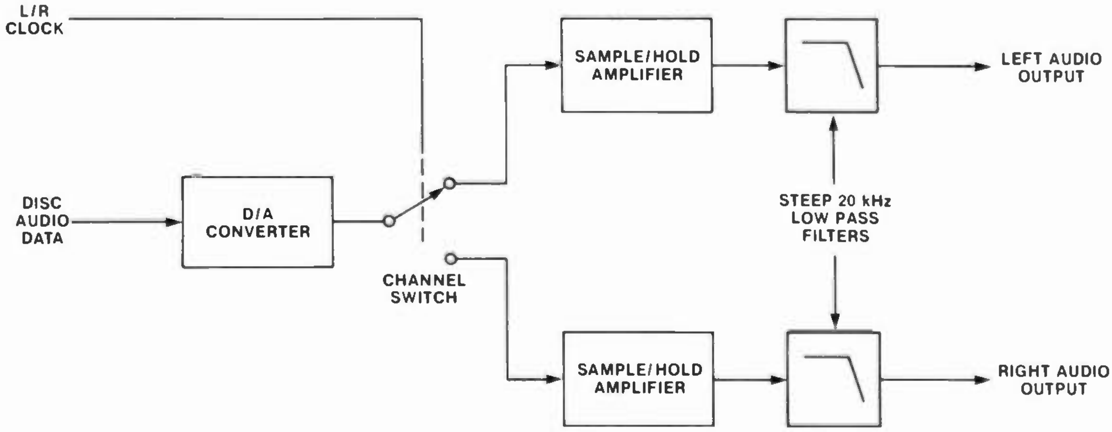

Figure 4. Compact Disc conversion system, 44.1 kHz conversion rate, single D/A converter (simplified).

monitored at one of the player output channels) of the track 11 square wave, as played on a high quality dual D/A converter machine. Note the minimum tilt of the horizontal waveform portion, and the equal minor ringing at all transition points. The lack of tilt indicates the player can reproduce very low frequency signals and complex waveforms accurately. The small amount of ringing is caused by the 20 kHz band limitation of the compact disc player, and its presence on all edges indicates the low pass reconstruction filters are designed for equal phase delay versus frequency (i.e., have constant group delay characteristics).

Figure 6 illustrates the result when the same track is played on a low quality CD player. The severe tilt, which distorts the waveform's proper amplitude, is an indication of poor low frequency response in the player analog output amplifier. The leading edge-only ringing is evidence that the player reconstruction filters introduce considerable phase delay at high frequencies, and will distort complex signal waveforms. Due to these errors, this player would be suitable for use only with sinusoidal single frequency test tracks on the disc.

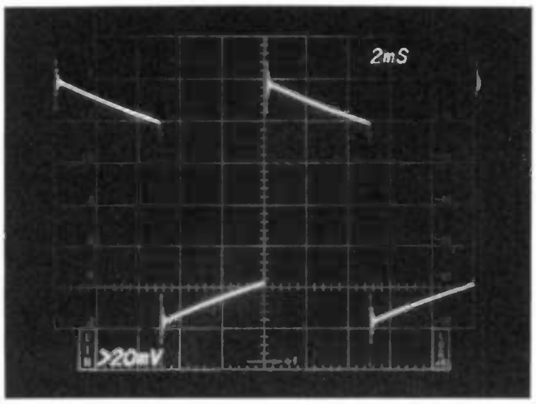

Figure 6. 100 Hz square wave, single D/A conversion player.

The final two tracks in the player performance section contain different forms of silence that are used to evaluate the ultimate signal-to-noise ratio of a chosen player. Track 12 is called "infinity zero" and consists of a silent track with all digital audio bits set to zero. On most players, the residual noise level on this track should be 80 to 90 dB below the 1001 Hz reference of track 3. Track 13, known as "silence+1 LSB" is the quietest signal the compact disc system can reproduce, and consists of the least-significant digital audio bit alternating at a 22.05 kHz rate. This generally allows for a more realistic reading of residual noise, since player analog muting circuits are not activated.

Broadcast System Test Signals

The remaining tracks on the NAB Test CD are intended to be used in testing various elements of broadcast or audio systems. Tracks 14 through 29 contain common discrete audio sine wave frequencies, all recorded at 0 dB, L+R. They are provided in ascending frequency order, and run from 20 Hz to 20 kHz, beginning with a 400 Hz level reference. With each frequency lasting for 30 seconds, they are useful for manually sweeping tape systems, audio amplifiers, or complete broadcast transmission chains. Frequency response and/or harmonic distortion checks can be made at each frequency.

Track 30 provides SMPTE (Society of Motion Picture and Television Engineers) intermodulation distortion (IMD) signals, consisting of two audio frequencies, 60 Hz and 7 kHz, that are linearly mixed. When this test signal is passed through a device or system exhibiting nonlinearities that cause IMD, the 60 Hz waveform mixes with the 7 kHz wave, causing 60 Hz sidebands to form around it. When these sidebands are demodulated, the recovered 60 Hz component can be related to a percentage of IMD. Virtually all distortion analyzers that include IMD functions will measure SMPTE IMD.

Two types of SMPTE IMD test signal are indexed onto track 30. The first consists of 60 Hz/7 kHz sine waves mixed at the typical 4:1 ratio for testing linear systems. The second index provides these same signals at a 1:1 ratio for characterizing composite FM systems. When directly connected to the composite signal path (without preemphasis, audio processing or stereo generator in line), distortion products can be measured directly at the deemphasized output of an FM modulation monitor. The insertion of deemphasis converts the 1:1 ratio back to 4:1 for measurement.

Tracks 31 through 36 contain tone pairs for CCIF (International Telephone Consultative Committee) IMD measurement. This method uses a combination of two sinusoidal signals of equal amplitude, separated in frequency by 1 kHz. When passed through an amplifier or other audio system under test, nonlinearities will cause frequencies at the sum and difference of the two input frequencies to form. The amplitudes of the residual sum and difference frequencies can be related to a percentage of distortion. Most analyzers, however, only measure the difference frequency since it is always found at 1 kHz.

CCIF IMD measurement has become popular in the measurement of AM broadcast transmission systems. It has been found that this type of distortion often increases rapidly at frequencies above 10 kHz, causing difference products to distort lower audio frequencies. Typically, the cause relates to transmitters, matching networks, and antenna systems that become greatly non-linear at 10 kHz or more removed from the carrier frequency of the station. Several stations that have employed the sharp filtering characteristic of the NRSC-1 technical standard have reported higher audio quality, even on narrow bandwidth receivers. It is believed that the removal of CCIF-type distortion products is responsible for this perceived improvement.

The NAB Test CD provides CCIF tone pairs ranging from 3/4 kHz to 13/14 kHz. The photos of Figures 7 and 8 show time domain representations of these lowest and highest CCIF test frequency pairs, respectively.

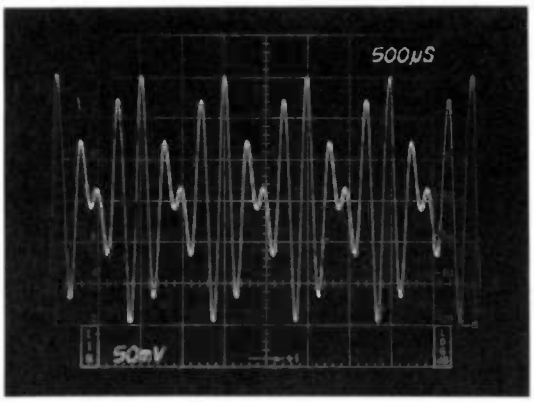

Figure 7. CCIF IMD test signal, 3/4 kHz (track 31).

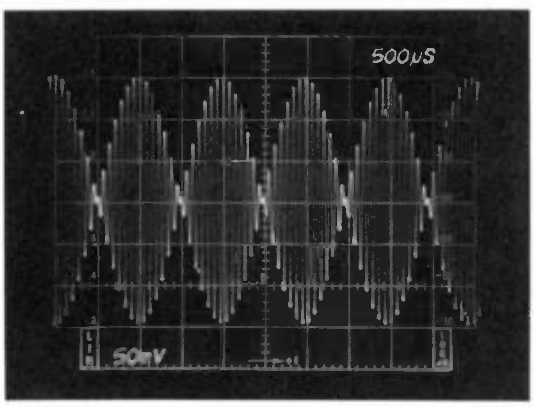

Figure 8. CCIF IMD test signal, 13/14 kHz (track 36).

Tracks 37 through 44 contain specific Bessel frequencies for absolute FM system modulation deviation calibration. Bessel functions are mathematical equations that can describe the amplitude and phase of the carrier as well as the sidebands of any frequency modulated signal, given the modulation index and modulating frequency. At certain modulation indices, the carrier amplitude goes to zero with all transmitted power distributed at frequencies other than the carrier frequency. This carrier null phemonenom is useful as an extremely accurate method for measuring frequency deviation and to check the calibration of modulation monitors.

The Test CD Bessel tone tracks provide the exact modulating frequencies required to assure the carrier will null at exactly 100% modulation for a given frequency deviation. These include 82.5 kHz (FM broadcast plus two subcarriers), 75 kHz (FM broadcast without subcarriers), 50 kHz (stereophonic television minus pilot and subcarriers), 25 kHz (monophonic television aural subcarrier), 10 kHz (stereophonic television SAP subcarrier), 6 and 4 kHz (popular FM subcarrier deviations), and 3 kHz (stereophonic television PRO subcarrier). The photo of Figure 9 shows a spectral view of the carrier null caused when an FM transmitter is modulated to 75 kHz using the 13,856.8 Hz Bessel frequency found on track 38.

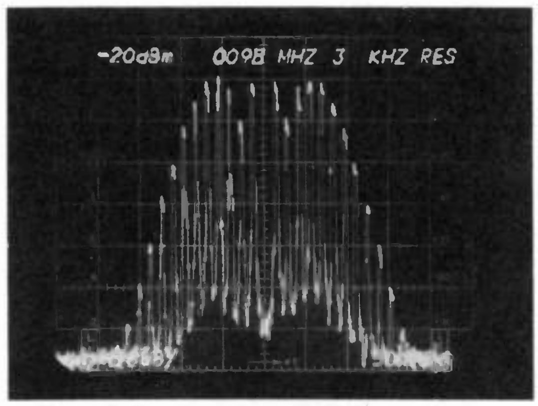

Figure 9. Bessel carrier null, FM transmitter modulated to +/-75 kHz (13856.8 Hz, track 38).

Track 45 provides a calibrated, indexed phase shift between left and right channels over a +/-360 degree range. This signal is useful for checking phase integrity of tape systems or broadcast audio chains. Phase meters can also be accurately calibrated with this test signal. All parts of the signal are recorded using a 1 kHz sine wave recorded at 0 dB. A 10 degree phase shift is indexed every 5 seconds.

Tracks 46 through 50 consist of five popular noise test signals, all recorded first for 30 seconds L+R, then 30 seconds L-R. Included is white, pink, USASI, CCIR, and partial synthetic program noise. The instruction booklet provided with the Test CD contains a detailed description and applications information for each of these types of noise signals.

Special NRSC test signals are provided on tracks 51 through 53. In addition to calibration tones, track 53 contains the ten minute pulsed-USASI noise test signal defined by the NRSC for testing AM broadcast transmission equipment and systems. Two NRSC standards, NRSC-1 and NRSC-2, make use of this signal. NRSC-1 uses the pulsed noise to verify proper operation of NRSC-compatible audio processing equipment, while NRSC-2 defines maximum occupied bandwidth for AM broadcast stations using NRSC processing. Figure 10 illustrates the block diagram of a typical NRSC-2 station measurement test configuration. The NRSC-1 and NRSC-2 standard documents should be consulted for detailed compliance measurement information (available from NAB).

Tracks 54 through 61 contain commonly used transmission preemphasis and deemphasis curve functions, each swept at 10 separate, indexed frequencies. These signals are useful for checking complementary networks. When a preemphasis curve is swept through a complementary deemphasis network, a flat response should occur. The same is true for a deemphasis curve swept through its corresponding preemphasis network.

The Test CD provides curves for NRSC AM transmission, 50 microseconds (FM broadcasting: Europe, Australia, etc.), 75 microseconds (FM, TV broadcasting: United States, Canada, etc.), and 150 microseconds (FM ancillary subcarrier transmission systems).

Tracks 62, 63 and 64 contain precision sinusoidal pilot and TV horizontal sweep frequencies. Included is 19.000 kHz (FM stereo pilot), 15.734 kHz (NTSC sweep/BTSC stereo pilot), and 15.625 kHz (PAL/SECAM horizontal sweep). The signals are useful for testing receiver/monitor decoder activation functions, as well as broadcast encoding systems that synchronize to these frequencies using phase-locking techniques.

Track 65 consists of the 25 Hz sinusoidal tone used as a radio broadcast automation transfer/stop tone in reel-to-reel tape playback systems. This test signal is intended to be used for troubleshooting the tone decoders used in these systems.

Track 66 provides a test signal intended to be used to check the phase linearity of a device or system at low frequencies compared to high frequencies. A 50 Hz sinusoidal tone is mixed with a 15 kHz sinusoidal tone, with the 15 kHz tone zero crossings exactly aligned with the 50 Hz zero crossings. When passed through a test device, any shift in the zero cross alignment indicates phase error in the system. It should be noted that CD player phase accuracy is critical for this test signal to be useful, and the player should be tested independently before this signal is used to test other devices.

Track 67 contains a precision 20 Hz to 20 kHz frequency sweep, first swept in the L+R domain, then L-R. The sweep rate is accurately controlled to 1 octave per 5 seconds, to permit the use of logarithmic scale paper in a chart recorder. This allows the frequency response of virtually

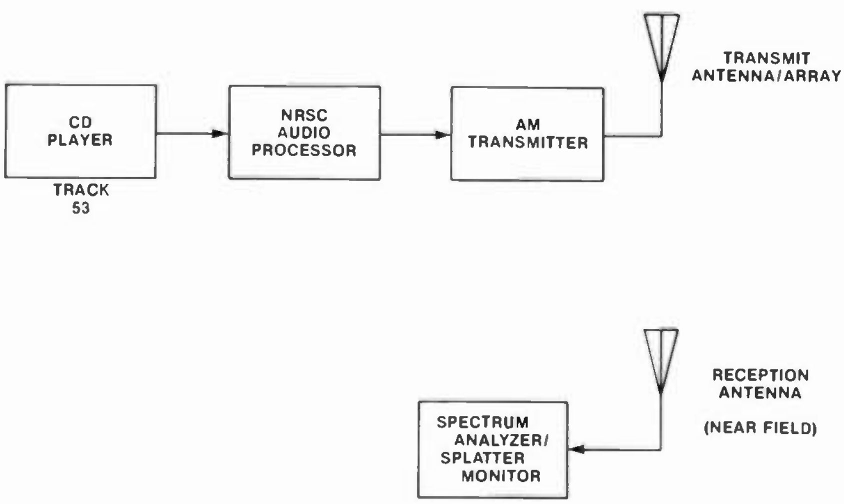

Figure 10. NRSC-2 compliance measurement (using pulsed-noise signal of track 53).

any amplifier or broadcast system to be rapidly characterized.

Track 68 consists of an indexed level sweep that provides a means of precisely calibrating level indicating devices, as well as checking the dynamic range of audio systems. A 400 Hz sine wave is recorded L+R from 0 dB to -60 dB in precise 5.0 dB steps, 5 seconds each.

Track 69 contains an indexed discrete sweep of increasing frequency square waves, which are all subharmonics of the 44.1 kHz CD sampling rate. Due to the 20 kHz bandwidth limitation of the compact disc system, however, square wave frequencies above 1 kHz can only approximate the "square" shape. At 6300 Hz (the highest square wave frequency provided on the disc), only the fundamental and third harmonic can be preserved. The photo of Figure 11 shows the time domain response of the 501 Hz square wave as reproduced on a high quality player. The square shape begins to disappear in the photo of Figure 12, when the 3675 Hz square wave is played on the same machine. However, its symmetrical appearance indicates good phase linearity in the playback system.

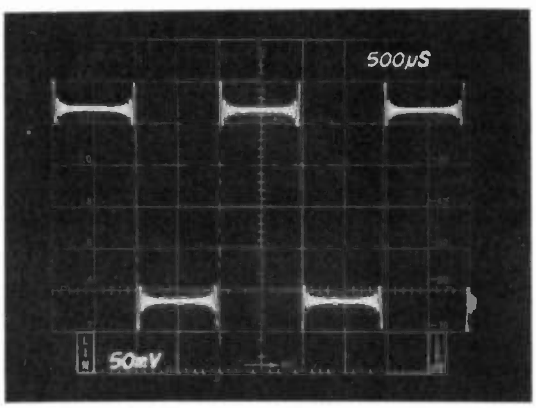

Figure 11. Square wave response, player only, 501 Hz (track 69/5).

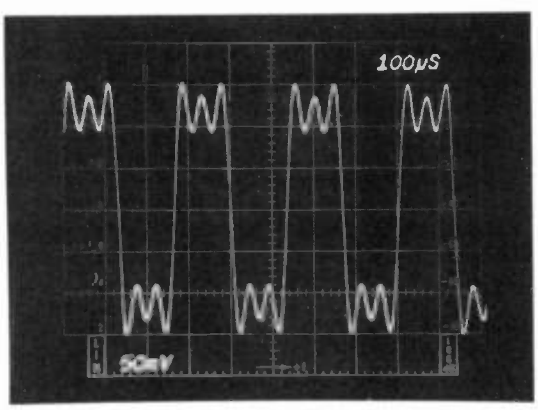

Figure 12. Square wave response, player only, 3675 Hz (track 69/8).

Sweeping these square wave frequencies through a similar band-limited system, such as a broadcast transmission chain, allows simultaneous relative measurement of both amplitude and phase errors versus frequency. The oscilloscope trace photo of Figure 13 shows the same 3675 Hz waveform of Figure 12, but passed through a system with considerable high frequency phase error and poor low frequency response.

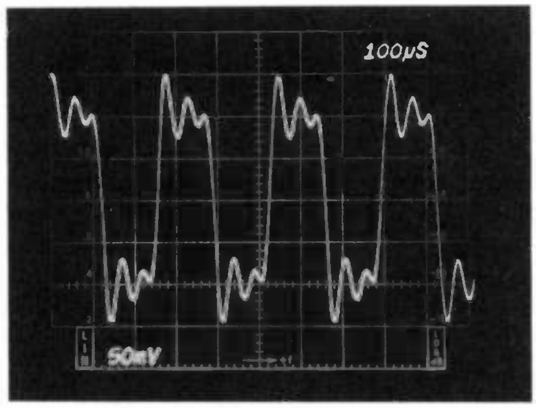

Figure 13. Square wave response, through test system, 3675 Hz (track 69/8).

Very linear triangle waves provide a means of testing AM transmitter modulator linearity at various modulation levels. Track 70 provides a 100 Hz linear triangle waveform for this purpose. As viewed on a swept oscilloscope trace (as shown in the photo of Figure 14), the formed straight edges and transitions of the diamond-shaped modulation envelope should remain well defined as the modulation level is increased. Any bending or discontinuity indicates a nonlinearity that could be caused by weak modulator or final amplifier tubes, a defective module in solid state transmitters, or external loading problems.

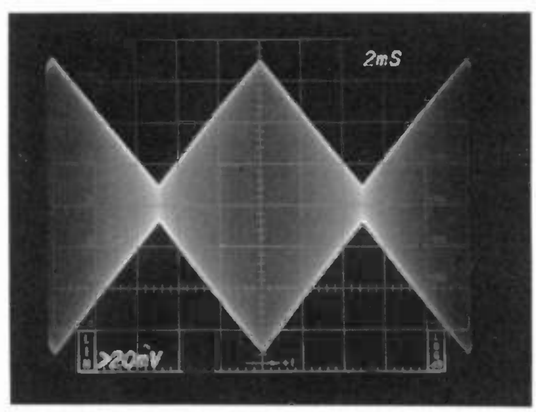

Figure 14. AM carrier, triangle wave modulated (by track 70).

Tracks 71 through 73 contain the most commonly employed sine wave frequencies used by wow and flutter analysis meters in conjunction with magnetic tape recording systems. The superior stability and accuracy of these frequencies (3 kHz, 3.15 kHz, and 12.5 kHz), make them suitable for use with in-house test tape generation projects.

Track 74 contains all the DTMF (dual-tone multiple frequency) tones commonly used in telephone and broadcast remote control systems. Supervisory tones, not commonly found on telephone-type keypads, are also included. Each tone pair is recorded for 1 second, followed by 2 seconds silence.

Also a dual-tone pair, the EBS (Emergency Broadcast System) attention tone is recorded on track 75. This signal is intended to serve as a troubleshooting aid for EBS decoding receivers and monitors.

Tracks 76 through 79 provide 400 Hz sine waves with a calculated amplitude of second harmonic frequency (800 Hz) added to provide precise amounts of harmonic distortion. The calibration of THD analyzers can be verified with these test signals which produce 0.1, 0.3, 1.0, and 3.0 percent THD respectively.

Tracks 80 through 95 contain specialized signals that follow established test procedures for PPM (peak program meter) and VU (volume unit) meter testing as described in IEEE, IEC and EBU standards documents. PPM tests include precision level reference and scale calibration tones, return time tone burst, delay time test tone, threshold test, and dynamic response tone bursts. A VU tone burst response signal is also provided. The reversibility-error signal (as shown in the oscilloscope trace photo of Figure 15), is provided to confirm that PPM or VU meters do not lose accuracy when asymmetrical waveforms are metered. This signal is also useful for investigating possible signal polarity inversions in broadcast audio equipment chains.

Two modes of tone bursts are provided on tracks 96 and 97 to facilitate testing of dynamic systems, such as AGC amplifiers and audio limiters. The action of attack and decay circuitry in these devices can be viewed on a swept oscilloscope by observing the effect they cause to the provided repetitious rectangular burst segments.

The final two tracks, 98 and 99, contain tone bursts for peak flasher calibration in FM modulation monitors. Until 1983, the peak flasher characteristics of FM broadcast modulation monitors in the United States were governed by FCC Rule 73.332. Since many monitors from that time period are still in use, and new monitors continue

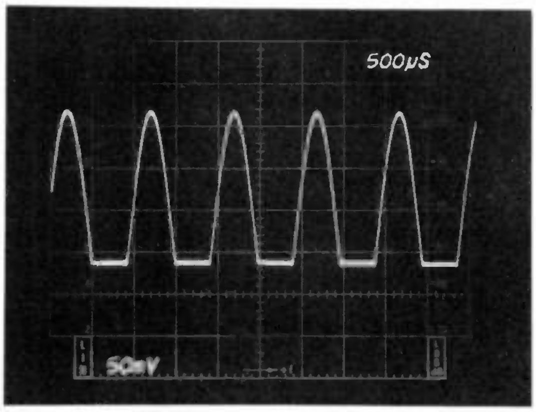

Figure 15. Reversibility-error test signal (track 93).

to be designed by these guidelines, the specified calibration bursts are provided.

Summary

The signals contained on the NAB Test CD are designed to provide the broadcast engineer with a convenient method of generating many complex as well as commonplace test signals. User comment is encouraged as to further applications for this disc, as are suggestions for signals to be included on future test disc volumes.

Acknowledgement

NAB thanks the National Radio Systems Committee and the Engineering Department of Broadcast Electronics, Inc. for their assistance in the formulation of the test signals contained on the NAB Test CD.

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ALTERNATE PRODUCTION OF GROUNDWAVE BY STRUCTURES OF INHERENTLY LOW SKYWAVE POTENTIAL

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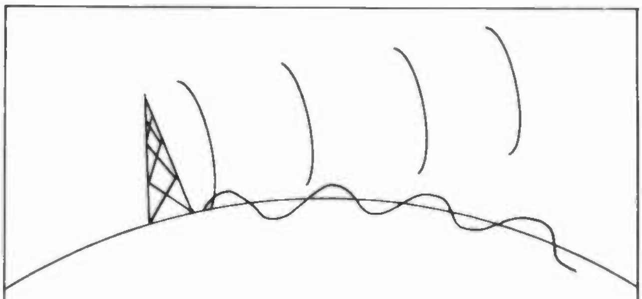
ABSTRACT
Since the earliest days of broadcasting nighttime groundwave service has been limited by skywave interference from distant cochannel facilties. This paper reviews the known characteristics of groundwave propagation and describes development and testing of a new technology for producing groundwave by structures having inherently low skywave potential.

GROUNDWAVE AND ANTENNAS

The most important thing I want you to understand from this paper is the fact that groundwave is already independent of skywave. It is only our traditional antennas that link the two together.

SEPARATE GROUNDWAVE IS LONG RECOGNIZED

This fact that groundwave and skywave are seperate phenomenon is best illustrated by the observation that groundwave continues along the earth's surface well past the horizon while the skywave signal departs the earth's surface at the horizon. It is the fact that groundwave continues after separating from the skywave component at the horizon that allows the AM broadcast band to have coverage that extends beyond the horizon in the first place. The original studies leading to the development of the FCC groundwave propagation curves recognized the existence of a separate groundwave. The familiar daytime propagation model has a close-in zone where both the groundwave and the free space wave of the traditional vertical antenna coexist and a zone beyond the horizon where only the groundwave exists. The resulting propagation graph was smoothed in by hand in the transition region not specifically fitting either part of the model. It has long been known that groundwave and skywave do exist separately at locations beyond the transmitting site. Although all of the common antennas used for medium wave broadcasting to date have



GROUNDWAVE AND SKYWAVE FIG. 1

created both the free space skywave and the ground wave simultaneously there is no overriding reason why groundwave must be the exclusive byproduct of the creation of a skywave by an efficient free space antenna conveniently located near the surface of the earth. Since the groundwave clearly exists separate from the free space wave at other locations it is reasonable that methods to create the groundwave directly can be discovered from basic priciples.

ESSENTIAL GROUNDWAVE CHARACTERISTICS

Groundwave travels as a current moving along the surface of the earth and yet the resulting field is characterized as a vertically polarized E field. With a tower it is easily visualized that the radial ground current is generated as a reflection of the vertical current in the tower. This is not the only way to generate a radial ground current.

It is usually assumed that a vertical structure is necessary to cause the vertically polarized groundwave E field component but think for a moment... is it necessary to have a vertical structure nearby for you to receive the vertically polarized groundwave on your field strength meter located beyond the horizon? Of course not! The traveling ground current creates a moving H field and because of the basic laws of field physics the moving H field is accompanied by a traveling E field and the combination is our familiar groundwave. The vertically polarized characteristic of groundwave persists as the wave propagates along the surface of the earth without a vertical structure for its propagation. Likewise there is no compelling reason that there must be a vertical structure present to initiate the vertically polarized E field that accompanies the traveling groundwave current.

LIMITATIONS OF TRADITIONAL ANTENNAS

Since the beginning of broadcasting all significant improvements in the groundwave coverage for nighttime have been based on reductions of skywave radiation. Reduced skywave radiation has been the result of either improved vertical radiation characteristic such as occurs with the 210 degree optimum antiskywave antenna or from selective phasing of multiple sources to cancel radiation in a selected direction as is commonly accomplished with the directional antenna array. The real problem is that all of these have started with an inherently efficient free space antenna. The existing technology has allowed us to reduce the free space signal in one or more selected arcs of interest. The potential for interference still exists in all other directions. With each additional station added the protection of the previous existing stations becomes more complex and coverage potential for each new station shrinks as the overall skywave background level increases.

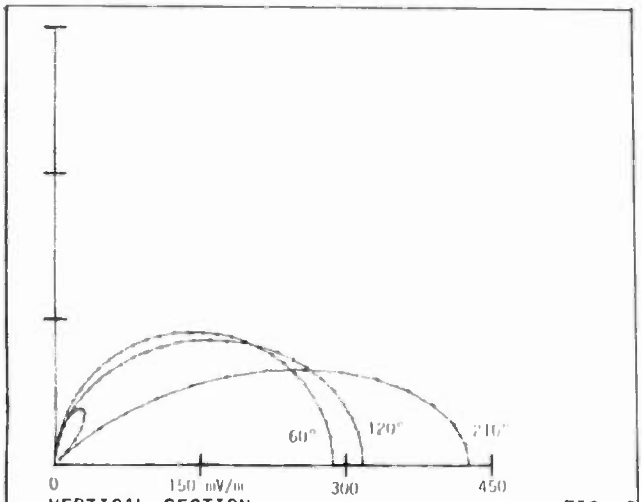
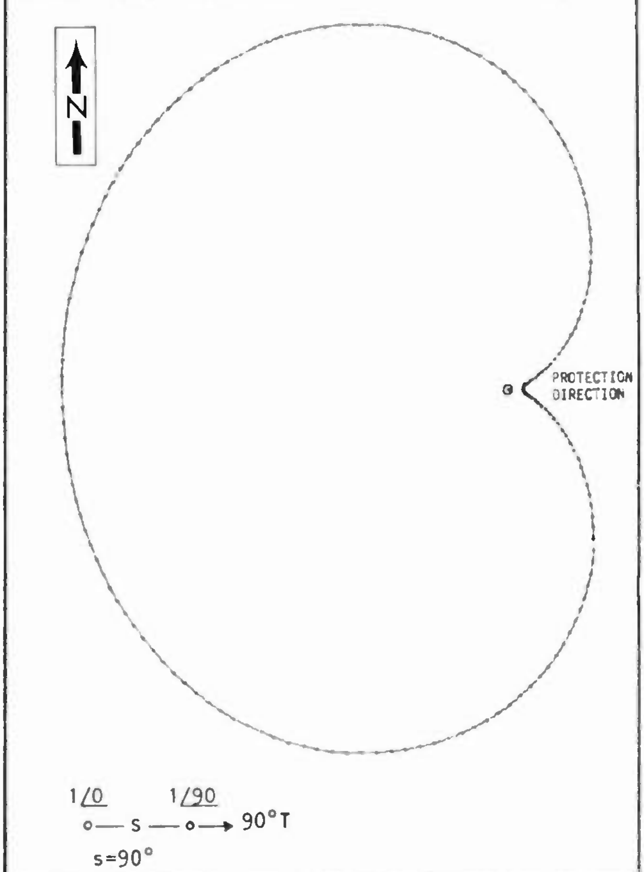


FIG. 2

DIRECTIONAL PROTECTION FIG. 3



GROUNDWAVE VERSUS ANTISKYWAVE

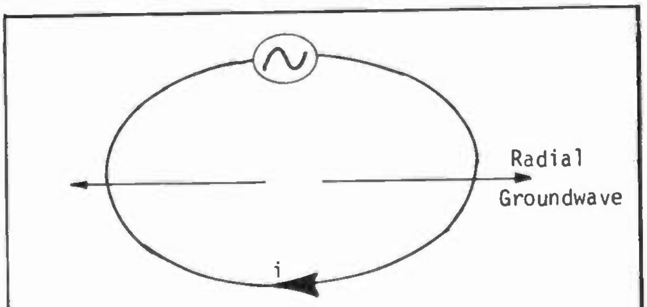
Traditionally we have started with a vertical antenna with excellent skywave potential and then tried to minimize the skywave. I have taken the opposite approach of exploring structures of inherently low free space radiation potential to discover how to create the traveling groundwave currents without simultaneously generating the pesky skywave.

BASIC CRITERIA FOR A GROUNDWAVE ANTENNA

The basic criteria for selecting a structure for analysis as a groundwave producer is that the structure have very little potential to be a skywave generator while still creating surface currents in the earth. This has limited my search to structures with minimal vertical currents, and for the most part to nonvertical structures. In order to minimize the possibility of horizontally polarized skywave being efficiently created any horizontal currents should be kept a small fraction of a wavelength above the ground. When horizontal current elements are a small fraction of a wavelength above the ground the horizontally polarized skywave fields are largely canceled by the ground reflection and reach their rather low maximum at relatively high angles above the horizon.

SPECIFIC GROUNDWAVE PRODUCER TESTED

The one type of groundwave producer specifically tested and analyzed to date is a horizontal loop laid directly on the surface of the earth driven with RF. The first test involved a 15 m diameter loop of insulated wire driven on the frequency of 1.82 mHz. Although method of moments analysis of this small loop over a reflecting surface indicated the potential fields produced should be on the order of 0.0000001 mV/m at 1 km for 1 kW input this antenna measured about 1 mV/m of groundwave produced. This is about 140 dB greater than the free space analysis would predict. Furthermore doubling the size of the loop to 30 m in diameter resulted in a measured 2 mV/m for an easy 6 dB improvement on the first effort.



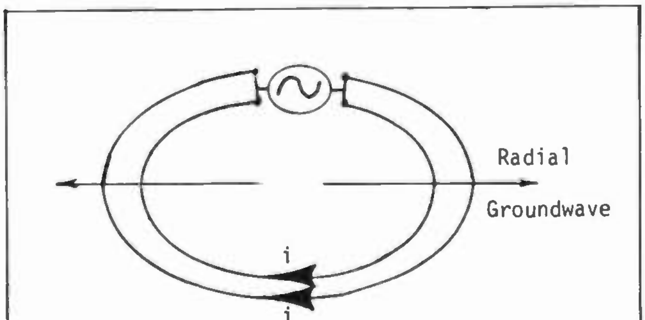
DRIVEN LOOP ON GROUND FIG. 4

HORIZONTAL WIRE AND VERTICAL POLARIZATION

Of special note is the fact that the measured field was definitely vertically polarized even when measured a small fraction of a wavelength from the loop. This would indicate that the mechanism for groundwave production is not the familiar free space radiation of a loop. A small horizontal loop should produce a horizontally polarized E field in the free space wave. The current in the loop creates an intense near field magnetic component which because of the close proximity to the conductive earth causes a current to flow in the earth at right angles to the loop current. This results in a current in the earth that is radial from the center of the loop. This radial current caused by the near field induction of the loop current then propagates as a groundwave.

FURTHER TESTING ON BROADCAST FREQUENCIES

FCC authority was obtained in May 1988 to make similar tests in the AM broadcast band. The sites selected for these tests were in a relatively isolated area of rural Eastern Colorado. These tests were conducted at two locations with widely differing soil conditions to explore what effect soil conductivity might have on the ultimate efficiency. The tests were conducted on frequencies of 540 kHz, 820 kHz, 1180 kHz, and 1570 kHz to make it possible to scale the physical properties measured without constructing a large number of antennas. Care was taken to select the sites so as to be clear of vertical structures. The tallest object around was a power line on wooden poles about 10 m tall along the road adjacent to the each of the test sites. To eliminate the possibility of contaminating the measurements by coupling the RF energy to power lines the tests were done with battery or portable generator power. The tallest part of the test apparatus was the power generator which was less than 1 metre tall. The test antenna itself was less than 1 cm tall.



Two Driven Loops on Ground FIG. 5

MEASURED RESULTS

The measured efficiencies peaked at about 20 mV/m per kW at 1 km at both sites for the several sizes and configurations tested but with different antenna sizes having the best performance at the different sites. The good soil site reached its efficiency peak with a loop size considerably smaller than the best performing loop for the poor soil conductivity site. The largest loop size tested was 100 m in diameter which is no larger than a typical radial ground system. Multiple concentric loops of slightly different diameter driven in parallel with equal currents were found to be somewhat more effective than a single loop of similar diameter. In all the tested configurations ranged from a single loop to twelve concentric loops. While 20 mV/m is still below the efficiency needed for most applications this represents a 26 dB improvement over the first proof of concept prototype tested in the back yard.

FEEDPOINT AND RADIATION CHARACTERISTICS

Of special interest to broadcasters is the fact that the driving point resistance of most of the loops tested have ranged from 30 to 250 ohms with rather low reactance making matching for the tests quite easy and noncritical. Also note that the maximum signal achieved was stable over about one octave indicating that the new antenna technology should lend itself to extremely broadband application.

RECEIVING TESTS

Several times during the testing period the setup time ran into night hours and the test antenna was connected to the external input of a Potomac Instruments FIM-41 field strength meter. Using the built in field strength meter antenna as a reference for comparison it was clear that the test antennas received the local stations available by groundwave much better than it received the distant stations coming in by skywave. The sensitivity to skywave signals was consistently 20 dB worse than the sensitivity to the more local groundwave signals. From this it would appear that the coupling to groundwave is at least 20 dB better than the coupling to skywave. Since an incoming skywave signal undoubtedly creates some measureable traveling groundwave when it reaches the earth it is likely that the ultimate groundwave to skywave superiority for transmitting may be far greater than 20 dB for this antenna configuration.

NONDIRECTIONAL NIGHT OPERATION

Any new antenna with 20 dB or more of skywave suppression would dramatically change night allocations. A 20 dB suppression of skywave would allow an omnidirectional groundwave equivalent to 1 kW nighttime for any station now allowed a 10 Watt nondirectional secondary nighttime or postsunset operation.

FUTURE IMPACT

Although predictions of the future are notoriously inacurate and the development of direct groundwave antennas is still an infant technology I will boldly attempt to look into the future. I plan to personally continue my research into groundwave antennas and fully expect to have a design that is within 10 dB of present tower efficiency within the next two to three years. At that time the vertical section will be fully documented. The antenna will allow improved nighttime omnidirectional operation for use by daytimers, secondary fulltime facilities, and fulltime facilities with deep nulls in populated areas often using the present daytime transmitter. As fulltime facilities convert to groundwave antennas the nighttime interference level will decrease. Nighttime service areas will increase nationwide. Ultimately as AM broadcasting converts to groundwave technology the AM allocations model will reduce to a simple single fulltime model. Fulltime coverage areas will be the same as presently possible daytime. Critical hours and nighttime skywave will no longer limit service. The FCC groundwave propagation curves may have to be revised again to account for antennas with no free space wave in the close in region. I expect long term research over the next ten years to result in a perfected groundwave antenna which is capable of coupling most of the power now being wasted in skywave into the groundwave for as much as 10 dB superior radiation efficiency over the present tower based technology. Even neighborhood zoning will be touched when towers disappear as unnecessary for AM broadcasting.

AM DIRECTIONAL ANTENNA TUNING, NEW METHODOLOGY, NEW TOOLS

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Abstract

AM directional tuning is a labor intensive operation relying on measurement techniques, and methodology developed in the 1920's. The need to tune AM arrays precisely is greater now than ever before.

Many antenna systems are in operation with pattern nulls adjusted for minimum field intensity, where the authorization, or the potential authorization permits a substantial signal. When the antenna system of a station was installed in the 40's, 50's, 60's or 70s, there may have been no consideration to antenna performance in the pattern null areas, now these areas may be densely populated and suffering with low signals, and distorted signals.

Phase shift networks must operate at or near the design shift, and transmission lines must be matched. Unfortunately, the real world rarely gives an antenna system which is as the computer predicted. It is all too easy, using conventional methods of adjusting an array to tune the system to meet FCC requirements, and be far from the antenna system design parameters.

The results of mistuning an array are pattern bandwidth limitations, poor transmitter load impedance, and high power losses due to excessive circulating current, and over stressed components subject to early failure.

An alternative methodology for initial tuning using only a medium power portable signal source, the station antenna monitor, direct reading VSWR and Power meters, and a portable plug in phase meter is described.

Final tuning may be accomplished by a real-time multi point per radial talkdown procedure using two way radios.

The methods described in this paper represent a direct method for tuning antenna systems more accurately in less time than presently achievable. This is particularly beneficial in minimizing the amount of off air time for station rehabilitation, and in controlling costs in tuning an antenna system.

Theoretical Pattern Parameters

Each directional antenna system consists of a number of vertical radiators. The FCC construction permit specifies a field ratio, phase, height, and location for each tower. The initial task is to establish the desired field vectors from each tower based upon the design, and computer model of the antenna system.

Traditionally, this has been accomplished by setting each branch of each network to the calculated values of impedance called for in the design of the phasing system. Unfortunately, stray inductance, capacitance, and variations in the antenna impedance from the predicted value conspire to make this route very circuitous. The values measured by a bridge for branch values are not able to closely predict actual operating values, particularly at the high end of the band where lead length adds inductance, and stray capacitance shunts each node[1]. The requirement of the tuning process at this stage is to produce the desired field vectors with each network adjusted to target phase shift, with all lines matched (provided that the design intends line match).

An alternative to the traditional

approach is to divide the system into subsystems. The first subsystem is the phasor itself. Assuming that the phasor is designed to operate into matched lines, the easiest way to set up this large section of the system is to connect the station antenna monitor in place of the antenna feed lines, and feed the common point with a low power transmitter or generator. When the antenna monitor reads the desired voltage output ratios, determined by the predicted power distributions, at the appropriate phase angles for the output terminals, this building block is set.

Several "helps" are available in getting the networks at the desired shifts. Zero degree series L-C networks may be shunted by a heavy clip lead, full T networks may be checked by connecting the antenna monitor at the power divider, then connecting the T network, and terminating the monitor at the output of the network. By adjusting the T network for the same voltage ratio, and the desired increase (or decrease) in phase, across the the network then the network is known to be adjusted to design. When all networks in the phasor have been adjusted in this manner, no changes should be made in any phasor adjustment except common point match until all lines are adjusted, and antenna parameters are within a few percent or degrees of design values.

Transmission lines are an important building block. Line lengths may be measured with a bridge by finding the frequency at which the reactance of an open line is zero, and the resistance is very low. The line is an electrical quarter wavelength at the lowest frequency this occurs. The frequency for three quarter wavelengths should be calculated, and the line should be checked at this frequency for a similar impedance. This simple check should be undertaken for both sample and transmission lines, as it will discover errors in length, or faults in the lines before conducting a wild goose chase for problems in the rest of the system.

The antenna tuning units require the most finesse in adjustment under this system. Each network may be adjusted by "eye" or by using the conjugate impedance method initially. The conjugate match method consists of connecting a 50 Ohm resistor at the line input, adjusting the shunt arm to the design value of impedance for the network, and measuring the impedance at the antenna terminal of the tuning unit. The series arms are adjusted to produce the conjugate (calculated resistance, and opposite sign but same magnitude of reactance) of the predicted antenna operating impedance.

The system is then interconnected, and the antenna coupler with the greatest operating power is adjusted for good VSWR (better than 1.5:1). The adjusted phase shift should be established by measurements either directly, or by calculation from the currents in the three branches of the network. Once the match is established at the high power (not necessarily high field) tower, then the same procedure is repeated at the next lower power tower, and so on for each tower in the system. When an array is close coupled (high RSS/RMS ratio), built with particularly tall (over 125 degree) or short (less than 75 degree) towers, then it will usually be necessary to "touch up" the match and phase shift of the higher power towers. "Touching up" will be less likely to be required if the lower power towers initially are adjusted with antenna phase in the target quadrant.

Once all towers are adjusted within a few percent of desired field and a few degrees of desired phase, and the match on all tower feed lines intended to be matched are adjusted for good VSWR, then the phasor controls may be adjusted for final trim. Each control should be checked to determine sensitivity of the control, and the common point network should be adjusted for the target impedance.

Field Adjustment

The array is now ready for field adjustment. Adequate non-directional data for null, minor lobe, and center of major lobe radials must be available to determine the Non-directional radiation at these bearings, and for several points between 3-6 miles from the station (further for low end stations). Target DA-ND field ratios should be calculated for each radial. For each radial the theoretical field intensity variations with

5% and 3 degree variations should be calculated and charted to show expected null variations with adjustment.

As many field technicians as are available with two way radios, field meters and maps should be dispatched to as many radials as possible. They should station themselves at previously measured non directional locations. An initial measurement should be taken as reference. Each phasor crank should be adjusted in order 1/2 turn, clockwise, the antenna parameters noted, and the field ratio calculated for each measurement location. The crank should be returned to its original position, and the next crank adjusted, parameters logged, and directional ratios noted until each control has been adjusted, and returned to its original position. The field technicians are then dispatched to a second measurement location on the same radial. The same crank adjustment, and logging procedure is repeated for these new locations. If the DA-ND ratios are the same for both points on a radial for each of the adjustment conditions, then either point may be initially considered a "good" point for the radial. If the ratios are different, or vary in a different way with adjustment, then additional measurement locations on the radial must be tested.

Once the antenna adjustment matrix is complete, the system is checked to make sure that the changes in measured field intensity correspond with the array adjustments predicted. The information predicted by the theoretical calculations, and the adjustment matrix are analyzed together, along with viewing a pattern plot of the target parameters (as corrected for monitoring offsets) to assure that no unintended nulls are generated in the array. A new set of crank positions/antenna parameters are chosen to closer approximate the desired parameters. If antenna parameters are to be used as a guide, then the cranks should be adjusted to bring in the parameters by halves, ie, each parameter should be adjusted only half way from its present value toward the target, then all other values should be restored to the original value, and then the parameter should again be adjusted 3/4ths of the way to the target and the other parameters restored. This procedure is repeated until the value is essentially at the target.

If a radial does not produce a consistent ratio from point to point, then there may be reradiation from external sources which may have to be detuned, or perhaps the radial may have to be adjusted using the "Silliman Talkdown" method [2]. The method of adjustment described above utilizes a simplified and stripped down version of the Silliman procedure for all radials to attempt to bring the inverse field intensity to a value as close as possible to the target, without exceeding the standard pattern limits.

It should be noted that the field intensity measured from a directional antenna at as much as 6 - 7 miles away may be somewhat higher or lower than that predicted by the conventional far field "theoretical" pattern of an antenna. For systems with tight nulls, and large antenna dimensions, there are errors due to parallax, where the phase distance to each radiator is not the same as that assumed in the far field case, causing a "phase error" in the summation of field intensity at each specific measurement location. Additionally, when off the end of an antenna array, the distance from the closest tower to the monitor location is less than the distance from the furthest tower in the array. The attenuation of the radiation from the closer tower is less than the attenuation from the further tower, appearing to upset the field ratios, while the antenna system may be properly adjusted. There are several ways of compensating for this effect utilizing correction factors for each measurement location.

Documentation

Once the antenna system is adjusted, the full set of measurements of field intensities must be collected, and put into presentation form for the license application to the FCC. The difficulty in making this happen is greater today than ever before. The number of technicians regularly employed in AM broadcasting is at an all time low. We must be able to employ relatively inexperienced people to conduct the measurements, collect the data and check it for errors so faulty data taking can be spotted before too much work must be thrown out. The biggest problems come from: 1. Failure to log date, and time correctly. 2. Miscalibration of the field intensity meter in areas with strong fields. 3. Taking readings at too great an interval, or not completing radials. 4. Misreading the multiplying factor of the meter. 5. Inadequate point descriptions. 6. Losing data sheets, or forgetting to do some radials.

These problems can be alleviated to some extent by training, but the use of printed operator instructions, clear field sheets with each required entry made only once and a "no blanks" policy decrease the number of initial errors. Once the data are taken, then the entry of the data into an computer database program which permits ongoing analysis of the data by sorting all data by operator, date, and time, or by date, and time to spot otherwise not apparent errors in logging. The program will also evaluate the radials for standard deviation of the data to spot obvious errors in the values, and make interim evaluation of the measured field of each radial before all data are in.

When the antenna system is being adjusted it is essential to keep track of the steps of adjustment, and interim readings, as well as crank numbers at each of the various stages of adjustment.

Tools

The tools of the trade have changed with this method of adjustment of AM antenna arrays. The old low power bridge is rarely used, supplanted by the operating impedance bridge, useful for measuring operating impedances, and line match.

The computing true VSWR meter is a particularly useful device. Line match may be continuously monitored, and the indications are independent of the operating power. This is particularly important in the early stages of interconnected setup, when the common point impedance varies, causing the operating power to change significantly. The unit in use at Radiotechniques is manufactured for Amateur Radio use, and a simple modification of the coupler permits it to work through the AM band. It also indicates power level, which shows how close the power distribution is to the theoretical values.

Frank Colligan uses a special antenna monitor which includes a small VHF transmitter [3] to be able to keep track of the antenna parameters while at the antenna couplers. The techniques used at Radiotechniques include the use of a portable battery powered antenna monitor which can be connected to the sampling system, clip on current probes, or a reflectometer. The unit is designed to operate with high sensitivity for low power testing, and is of the superhetrodyne type to reject interfering signals picked up by the antenna system under test. When the unit is connected to clip on current probes, it indicates current phase shift across a network, when connected to a reflectometer, it provides the magnitude *and phase* of the reflection coefficient providing a "no null" bridge replacement.

Another invaluable set of tools in use at Radiotechniques are a set of large vacuum variable capacitors in insulated cases with calibrated dials and clip leads, and rotary inductors with clip leads. These permit continuous adjustment of component values in antenna couplers under power.

A modified Yaesu FT-757 Amateur transceiver permits tuning of the antenna system at the modest power levels of 20-30 Watts, providing enough signal to drive the low power antenna monitor, the VSWR meter, and other instruments without posing the potential hazard of using the station transmitter whether adjusted for low power or not. When operating through a 3 db pad, the Yaesu will operate steadily, independent of common point impedance variations during adjustment.

Caveats

In spite of the new technologies in adjusting an antenna system, there are still many pitfalls in tuning an antenna system. The most important is that the station must be constructed to comply with the physical and electrical requirements, as well as to comply with the pattern performance requirements. When a station is newly constructed it is essential that the details of construction have been properly implemented.

If a station has been constructed with errors, or the antenna equipment is in poor condition, the engineering expense in rectifying the problems can exceed the engineering expense of tuning the antenna, and conducting the proof. Examples of problems encountered by myself and by my associates upon arrival at a station include:

Towers not properly oriented with respect to true North

Sampling system corroded causing intermittent readings

Guy insulators cracked causing drifting in wet conditions

Deteriorated ground system

Antenna couplers installed at wrong tower bases

Antenna feed lines cut to wrong length

Intermittent antenna monitor

Tower base insulator shorted

Tower base insulator full of water

Variable phasor coils shafts seized

Loose phasor counter dials and knobs

Control of pattern switching miswired

Sampling system not installed

Transmitter defective

Contractor left with antenna coupler keys

No power in transmitter building

Phasor installed at tower base with no provision for connecting antenna monitor, or powering test equipment.

Old shielded type sample loops full of water.

Different types of Toroidal Samplers at bases.

These problems should have been resolved during the construction phase. A consulting engineer should be involved at least by phone in the planning and layout stages to be sure that the facilities not only comply with the authorization, but provide for the operational needs of the proof.

Managements of stations require that the rebuild of the facilities interfere with station operations only minimally. One job required that all low power adjustments, and off air periods were between 2-5 AM on Saturday nights. If pushbutton non-directional operation from two different towers in the array was not available, the job still would not be complete.

Conclusion

The tuning of AM directional antenna systems is more straightforward than previously due to a new systems approach to the adjustment. The patterns may be more closely adjusted to the desired pattern shape, and the system can be more accurately adjusted to the design parameters. In order to achieve the best performance from any design of an antenna system, it is essential that the line match, power divider adjustment, and individual networks be adjusted to the design parameters, and not to some arbitrary value which will meet FCC muster.

The costs of tuning systems have decreased in some ways due to the improvement in methods, but the shortage of local technical personnel can make the human intensive portion of the process more difficult than in years past. It is more difficult to properly construct, and complete the field measurements today due to this shortage.

The performance of directional

antenna systems today can be much better due to improved design, but more importantly by being adjusted to the design parameters, instead of to values which simply meet the FCC requirements.

1. Colligan, Frank, A New and Systems Oriented Method of Directional Antenna Tuning, IEEE Transactions on Broadcasting, March 1988, p 63

2. Colligan, Frank S., A Talk-In Procedure for Critical A.M. Directional Antenna Adjustment, IEEE Transactions on Broadcasting, Vol BC-26, June 1980, p 17.

3. Colligan, Frank, Antenna Monitor Telemetered Data Memory as an Adjustment and Maintenance Tool for A.M. Directional Antenna Systems, IEEE Transactions on Broadcasting, Vol BC-30 Number 4, December 1984, p117

DIPLEXER DESIGN: Q-MATCHING TECHNIQUES

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Abstract

Many radio stations are considering diplexed systems because the costs of constructing a new site are higher than the cost of diplexing. Diplexing eliminates the need and cost of a separate tower, ground system, transmitter building, and site. To take advantage of the cost savings, a diplexer is required.

The Q-Matching technique is a method for choosing the value of each component to provide a diplex system with optimized bandwidth characteristics. It is the purpose of this paper to demonstrate the Q- Matching optimization technique that will minimize the effect the traps have on the bandwidth of both stations. At the same time, this technique will generally increase the isolation between the two stations over other designs.

Introduction

A diplexer is a set of traps that will allow the operation of two frequencies on the same antenna. Figure 1 shows a schematic representation of a diplexer. Although all parts are not necessary in every case, there are basically 5 parts to a diplexer.

The Antenna Coupling Unit (ACU) is used to match the impedance of the antenna to the Z_o of the transmission line. There are two ACUs; one for each frequency. Their design will not be discussed.

The Auxiliary Trap is used to further attenuate the reject frequency. It provides a high impedance to ground at the pass frequency (on the order of 10K ohms) and a low impedance to ground at the reject frequency (on the order of 1 ohm). There is one Auxiliary trap for each frequency.

The Antenna Resonator is a coil if the impedance at that point is capacitive, and a capacitor if the impedance is inductive. It is used to bring the impedance to the point where the Auxiliary Trap will be connected to the circuit close to resonance. This near resonance condition is necessary when employing the Q-Matching technique. There may be one antenna resonator for each frequency.

The Main Trap is used to attenuate the reject frequency. It provides a high impedance at the reject frequency (on the order of 10K ohms) and a low impedance at the pass frequency (on the order of 1 ohm). There are two Main Traps in a diplexer, one for each frequency.

The Diplex Point is where the two signals first come together. The impedances at this point are what is used in the calculations of

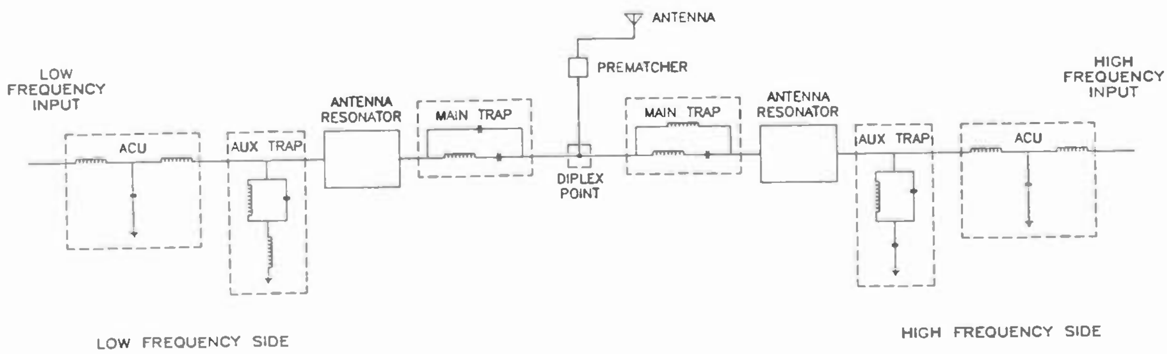


Figure 1 - Diplexer Schematic

stresses and Qs, for the trap circuitries. It is desirable to have the branch point impedance at each frequency to be close to resonance. Also, it is desirable for the parallel resistance to be somewhere between 50 and 200 ohms. Each Main Trap stores energy at the pass and the reject frequency. These criteria will provide enough series resistance so the Main Trap's Q will not be too high at the pass frequency. Also, a parallel resistance of 50 to 200 ohms is not too high so the Main Trap's Q will not be too high at the reject frequency. To aid in providing an improved impedance at the diplex point, a prematcher may be used. It consists of a combination of coils and capacitors which will make the diplex point impedances better at both frequencies. There is no set way for designing a prematcher, but an example will be given later in the paper to demonstrate its purpose.

Filter Classes

There are four different types of notch filters or traps that are generally used in the design of diplexers: two Main Trap types and two Auxiliary Trap types.

Figure 2 shows the two types of Main Traps. The first, which will be referred to as the Series Main Trap, consists of a coil and capacitor in series resonance at the pass frequency. Also, a third component is placed in parallel with the series combination of the first two. The third component is in parallel resonance with the series combination of the first two components C1 and L1 at the reject frequency. The third component is a capacitor if the reject frequency is higher than the pass frequency and is a coil if the reject frequency is lower than the pass frequency.

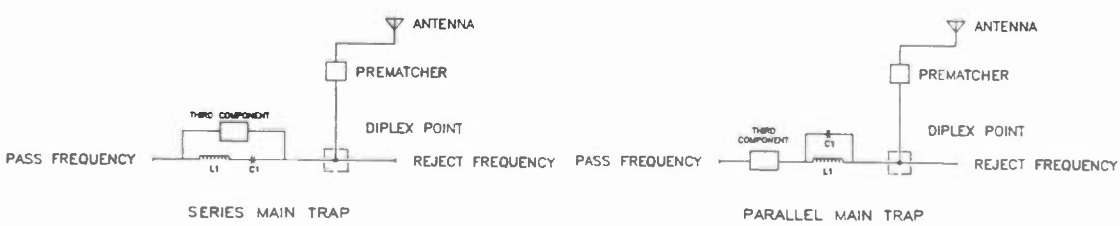


Figure 2 - Main Trap Types

The second Main Trap class which will be referred to as the Parallel Main Trap, consists of a coil and capacitor in parallel resonances at the reject frequency. A third component is placed in series with the parallel combination of the first two. The third component is in series resonance with the parallel combination of the first two components C1 and L1 at the pass frequency. The third component is a capacitor if the reject frequency is higher than the pass frequency, and is a coil if the reject frequency is lower than the pass frequency.

Figure 3 shows the two types of Auxiliary Traps. The first, which will be referred to as the Series Auxiliary Trap, consists of a coil and capacitor in series resonances to ground at the reject frequency. A third component is placed in parallel with the series combination of the first two. The third component is in parallel resonance with the series combination of the first two components C1 and L1 at the pass frequency. The third component is a capacitor if the pass frequency is higher than the reject frequency, and is a coil if the pass frequency is lower than the reject frequency.

The second Auxiliary Trap type which will be referred to as the Parallel Auxiliary Trap, consists of a coil and capacitor in parallel resonances at the pass frequency. A third component is placed in series to ground with the parallel combination of the first two. The third component is in series resonance with the parallel combination of the first two components C1 and L1 at the reject frequency. The third component is a capacitor if the pass frequency is higher than the reject frequency, and is a coil if the pass frequency is lower than the reject frequency.

Component Values

In this section the equations for the component values of each type of trap will be given. Each trap component is defined when the component C1 is chosen and the frequencies are known. In the equations below, W_L will be the radian frequency of the lower frequency. W_H will be the radian frequency of the higher frequency. F will be the ratio of the low frequency to the high frequency.

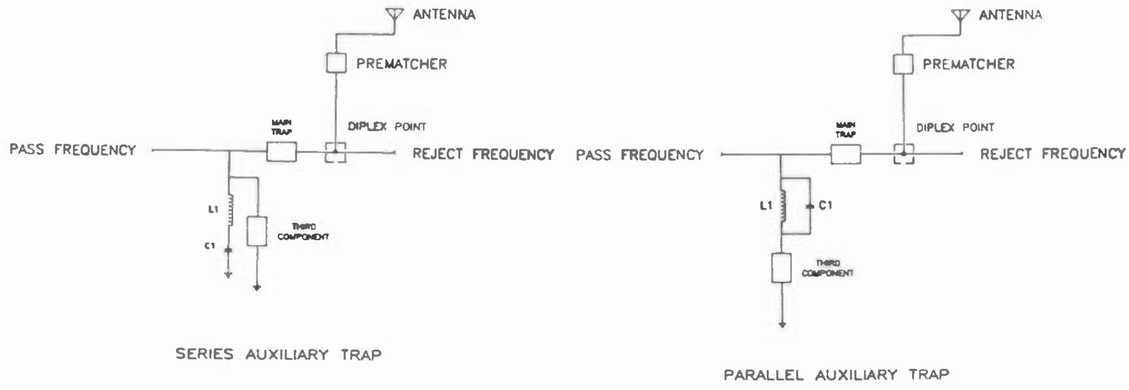


Figure 3 - Auxiliary Trap Types

$W_L = 2 \times PI \times F_L$ where F_L is the low frequency

$W_H = 2 \times PI \times F_H$ where F_H is the high frequency

$F = F_L / F_H$ F is the frequency ratio

Refer to Figures 4-7 on pages 4 and 5 for a schematic drawing. The equations for the components of each type of trap are given below.

Series Main Trap Low Frequency Side

$$L1 = 1 / (W_L^2 \times C1)$$

$$C2 = (C1 \times F^2) / 1 - F^2$$

Series Main Trap High Frequency Side

$$L1 = 1 / (W_H^2 \times C1)$$

$$L2 = (1 - F^2) / (W_L^2 \times C1)$$

Parallel Main Trap Low Frequency Side

$$L1 = 1 / (W_H^2 \times C1)$$

$$C2 = C1 \times (1 - F^2) / F^2$$

Parallel Main Trap High Frequency Side

$$L1 = 1 / (W_L^2 \times C1)$$

$$L2 = 1 / [(W_H^2 - W_L^2) \times C1]$$

Series Auxiliary Trap Low Frequency Side

$$L1 = 1 / (W_H^2 \times C1)$$

$$L2 = (1 - F^2) / (W_L^2 \times C1)$$

Series Auxiliary Trap High Frequency Side

$$L1 = 1 / (W_L^2 \times C1)$$

$$C2 = (C1 \times F^2) / (1 - F^2)$$

Parallel Auxiliary Trap Low Frequency Side

$$L1 = 1 / (W_L^2 \times C1)$$

$$L2 = 1 / [(W_H^2 - W_L^2) \times C1]$$

Parallel Auxiliary Trap High Frequency Side

$$L1 = 1 / (W_H^2 \times C1)$$

$$C2 = C1 \times (1 - F^2) / F^2$$

Bandwidth of Diplex System

It will be shown that different choices for the component C1 will affect the Loaded Q of the trap at both frequencies, thus the bandwidth of the system. Loaded Q is defined in the usual way. The equation for the Loaded Q is given below.

$$\text{Loaded } Q = (2 \times PI \times MSE) / DE$$

where MSE is the maximum stored energy and DE is the energy dissipated per cycle.

Consider the Loaded Qs for a Parallel Main Trap on the high frequency side. The Loaded Qs are as follows.

$$Q_L = RP_L \times W_L \times C1$$

$$Q_H = (W_H \times L2 \times Q_M) / R_H$$

where RP_L is the parallel resistance of the diplex point at the low frequency, R_H is the series resistance of the diplex point at the high frequency, and Q_M is the Q multiplier.

If a substitution for L2 is made, the second equation becomes as follows.

$$Q_H = (W_H \times Q_M) / [R_H \times C1 \times (W_L^2 - W_H^2)]$$

LOW FREQUENCY SIDE

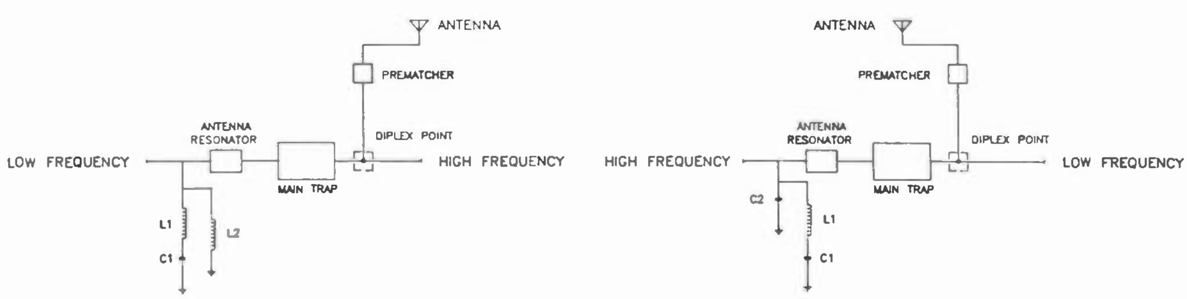
HIGH FREQUENCY SIDE

Figure 4 - Series Main Trap

LOW FREQUENCY SIDE

HIGH FREQUENCY SIDE

Figure 5 - Parallel Main Trap



LOW FREQUENCY SIDE

HIGH FREQUENCY SIDE

Figure 6 - Series Auxiliary Trap

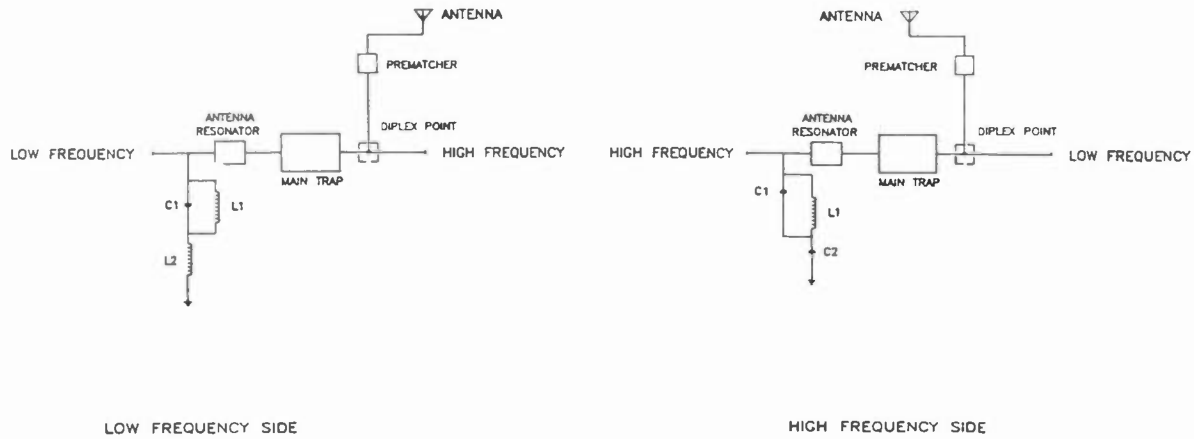


Figure 7 - Parallel Auxiliary Trap

Note that Cl is in the numerator for the equation for Q_L but in the denominator for the equation for Q_H. This means that as Cl gets smaller, Q_L gets smaller, and Q_H gets larger. Also as Cl gets larger, Q_L gets larger, and Q_H gets smaller. Therefore, in this case, if Cl is chosen to be relatively small, the stored energy in the high frequency system would increase, thus negatively affecting the bandwidth. Also if Cl is chosen to be relatively large, the bandwidth of the low frequency would be adversely affected.

One solution to the problem would be to choose a capacitor such that the Loaded Q for each frequency is the Same. This could be done by setting Q_L equal to Q_H and solving for Cl. For this case, Cl^2 would be as follows.

$$Cl^2 = (W_H \times Q_M) / [RP_L \times W_L \times R_H \times (W_H^2 - W_L^2)]$$

This design method for choosing Cl will be referred to later as the standard design. When the Q-Matching technique is discussed, it will be shown that this is not the best solution for Cl.

Unloaded Q and Rejection

To estimate the isolation that a Main Trap will produce, it is necessary to calculate its parallel resistance at the reject frequency. This parallel resistance will be called the reject resistance (using the symbol RR) to avoid the confusion with the parallel resistance of the diplex point. The magnitude of this reject resistance is generally on the order of 10 K ohms. The reject resistance is a function of the Unloaded Q (Q_U) of the coil. The Unloaded Q is the ratio of the reactance of the coil and its resistance. Unloaded Qs of coils used for diplexers are typically between 200 and 800.

To estimate the isolation that an Auxiliary Trap will produce, it is necessary to calculate its series resistance (RS) to ground at the reject frequency. The magnitude of the series resistance is on the order of 1 ohm. The series resistance is also a function of the Unloaded Q of the coils used in the trap.

Loaded Q and Rejection

The isolation that a trap will produce is also a function of the Loaded Q of the trap at the pass frequency. It was shown earlier that the Loaded Q of a Parallel Main Trap on the high frequency side could be expressed with the following equation.

$$Q_H = (W_H \times Q_M) / [R_H \times Cl \times (W_H^2 - W_L^2)]$$

The reject resistance of this trap is $W_L \times Ll \times Q_U$. If a substitution is made for Ll, the reject resistance for a parallel trap on the high frequency side would be as follows.

$$RR_H = Q_U / (W_L \times Cl)$$

It can be seen by inspection that if Cl is small, the Loaded Q of the trap and its reject resistance are high. This means that the higher the Loaded Q of the trap, the better the rejection of the trap.

Q_U is not constant for all coils. As the inductance of a coil increases, its Q_U will generally increase. For the above example, the reject resistance would increase faster than 1/Cl as Cl gets smaller.

Q-Matching Techniques

In the previous sections it was shown that the Loaded Q of a Main Trap affects the bandwidth of the system and its rejection resistance. In this section the Q-Matching technique will be discussed. To assist in the understanding of the Q-Matching technique, several circuits will be analyzed.

Consider the circuit in Figure 8. The circuit has a Loaded Q of 4 and produces a 1.083:1 VSWR at the 10 kHz sideband frequencies. Compare this to the circuit in Figure 9. This circuit has a Loaded Q of 8 but produces a VSWR at the 10 kHz sideband frequencies of only 1.006:1. The circuit in Figure 9 contains two parts. The first part is a series network with a Loaded Q of 4. The second part is a parallel circuit with a Loaded Q of 4. This circuit, which has a Loaded Q of 8, is producing VSWRs of the same magnitude as a series circuit with a Loaded Q of .32.

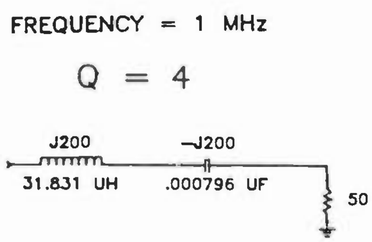


Figure 8
VSWR at the 10 kHz Sideband
Frequencies is 1.083:1

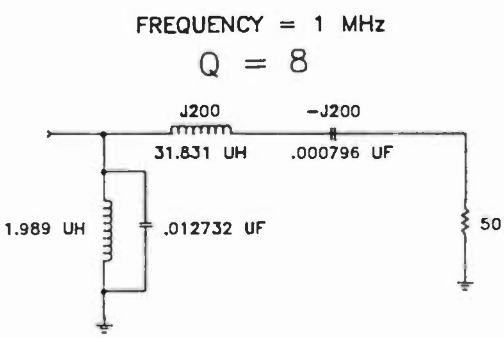


Figure 9
VSWR at the 10 kHz Sideband
Frequencies is 1.006:1

Equivalent Q (Q_E) of a circuit or system will be defined as the Loaded Q of a series resonant circuit which produces the same VSWRs at the 10 kHz sideband frequencies as the original circuit. The equation for equivalent Q is given below.

$$Q_E = (F_c \ / \ .02) \times [\mathrm{SQRT(VSWR)} - 1 \ / \ \mathrm{SQRT(VSWR)}]$$

Where F_C is the carrier frequency in MHz.

The circuit in Figure 9 would have a Q_E of .32 and a Loaded Q of 8. By making the Loaded Q of the series part of the circuit equal to the Loaded Q of the parallel part of the circuit, the Q_E of the circuit is much lower than the Loaded Q. This technique of reducing the Q_E of a circuit in this manner will be referred to as Q-Matching.

Table 1 below shows the degree to which the Q_E of a series resonant circuit could be reduced by adding an additional parallel circuit of the same Loaded Q. This table is for 1 MHz. To adjust this table for different frequencies, multiply each column by the new frequency in MHz. For example, a circuit with a Q of 10 could be reduced to a Q_E of 1.96 at 1 MHz. At .5 MHz a circuit with a Q of 10 (20 x .5) could only be reduced to 3.72 (7.43 x .5).

TABLE 1

Loaded Q of Series Circuit	Q_E of Series and Parallel Circuit
2	.08
4	.32
6	.71
8	1.26
10	1.93
15	4.31
20	7.43
25	11.18
30	15.43
35	20.07
40	24.99
50	35.36

It can be seen from Table 1 that the Q_Es of circuits that have a lower Loaded Q are reduced proportionally more than the circuits with higher Loaded Qs. Where the ratio of Q to Q_E for a network with a Loaded Q of 2 is 25:1, the Q to Q_E ratio of a network with a Loaded Q of 10 is only 5:1.

To obtain these reductions in Q_E, the point at which the parallel circuit attaches must be a purely resistive load. Using a network with a Loaded Q of the series part is 10 and a load impedance of 50 + j 50 ohms instead of 50 ohms, the Q_E of the circuit calculates to be 22. This is much higher than the Q_E of 1.96 which was calculated with the circuit looking into a purely resistive load.

This same principle can be used in the design of a diplexer. A Main Trap of a diplexer could be looked at as a resonant circuit in series with a load. The Auxiliary Trap is a parallel resonant circuit shunt to the load. Consider the following example:

Station Number	Frequency (MHz)	Diplex Point Resistance (ohms)	Impedance Reactance (ohms)
1	1.0	50	70
2	1.2	200	-100

Table 2 contains the loaded Qs of a Parallel Main Trap on the low frequency side of the diplexer for several values of C1.

TABLE 2

C1 (uf)	Q_L	Q_H
.001	23.7	1.9
.0015	15.8	2.8
.002	11.8	3.8
.003	7.9	5.7
.0035	6.8	6.6
.004	5.9	7.5
.005	4.7	9.4

As suggested earlier, one choice for C1 might be .0035 uf. This value gives about the same Loaded Qs for each frequency. For this case it is expected that the Q_E of the load as seen by each transmitter would increase by about 6.7 due to this trap alone. Add to this the effects of the Main Trap on the high frequency side, and the Auxiliary Traps on both sides. The total system bandwidth would be greatly reduced.

Very little can be done to reduce the Q_H of this network. But, the Q_L in Table 2 could be reduced to a much lower Q_E by using the Auxiliary Trap to Q-Match the Main Trap. To apply this technique, an Auxiliary Trap is chosen that has the same Loaded Q as the Main Trap. Also, it is necessary for the impedance at which the Auxiliary Trap is to be attached to be close to resonance. Table 3 would result if Table 2 is redone assuming the Main Trap is to be Q-Matched with the Auxiliary Trap.

TABLE 3

C1 (uf)	Q_E (Q-Matched)	Q_H
.001	10.1	1.9
.0015	4.8	2.8
.002	2.7	3.8
.003	1.2	5.7
.0035	0.9	6.6
.004	0.7	7.5
.005	0.5	9.4

Using Table 3 above, a better choice for C1 would be a .002 uf capacitor. The Q_Es of Table 3 contain the effects of both the Main and Auxiliary Trap on the low frequency side. The Qs in Table 2 show only the effect of the Main Trap.

Employing the Q-Matching technique, the trap would have a higher Q. The reject resistance for this trap would then be larger. Assuming an Unloaded Q of the coils in this example to be 500, the reject resistance for the trap choosing C1 as a .0035 uf capacitor would be 19 K ohms. If C1 is a .002 uf capacitor, the reject resistance would be 33 K ohms. By choosing the .002 uf capacitor over the .0035 uf capacitor, the isolation from the high to low frequency port would be improved.

By employing the Q-Matching technique, not only will the impedance bandwidth of the system be improved over other designs, but the isolation will also be improved.

Prematching Circuits

As mentioned earlier, there is no set way to design a prematching circuit. The purpose of a prematching circuit is to bring the tower resistance between 50 ohms and 200 ohms and keep the reactance near zero for both frequencies at the same time. This is rarely possible.

Consider an example where the frequencies of operation are 1 MHz and 1.2 MHz. Let the tower height be 300 feet tall. At 1 MHz the electrical height would be about 110 degrees. At 1.2 MHz the tower height would be about 130 degrees. These tower heights have impedances of about 135 + j 210 ohms and 420 + j 310 ohms. Let these impedances be used for the diplex point impedances. Assume that C1 is chosen for the Main Traps so that the Qs at the low and high frequencies are the same. Then the Qs for the Main Traps on the low and high frequency sides would be 6.6 and 3.1 respectively.

By inserting a capacitor of .00075 uf in series with the tower, the new diplex point impedances for the low and high frequencies would be 135 - j 2 ohms and 420 + j 133 ohms. Although the series resistance was not changed by this prematching network, the parallel resistance was lowered. For 1 MHz the parallel resistance went from 462 ohms to 135 ohms. For 1.2 MHz the parallel resistance went from 649 ohms to 462 ohms. Because the parallel resistances of the diplex point were reduced and the series resistance was unchanged, the bandwidth of the diplexer can be expected to be improved over a system which does not include a prematching network. Again assume that C1 is chosen so that the Qs at the low and high frequencies are the same. Then the Qs for the Main Traps on the low and high frequency sides for these diplex point impedances would be 5.5 and 1.7 respectively. Thus the bandwidth potential for the diplexed system would be increased with the capacitor placed in series with the tower.

Example

In this section the results of analyses will be shown for two diplex systems. One system will be designed using the standard design technique discussed at the end of the Bandwidth of Diplex System section. The second system will employ the Q-Matching technique for diplexer design. For simplicity, all unloaded Qs for the analyses are assumed to be 500.

The design paramaters for the sample problem are given below.

Frequency (kHz)	Tower Height (Degrees)	Impedance (ohms)	Q_Z
1000	90	55 + j 60	5.4
1100	99	82 + j 116	4.6

Figures 10 and 11 show the results of the bandwidth analysis done at the low and high frequencies respectively. For the low frequency, the standard design system had an equivalent Q of 8.2 compared to the equivalent Q of the Q-Matched system of 6.6. Both of these numbers can be compared to the equivalent Q of the antenna itself of 5.4. The equivalent Q of 5.4 is what could be expected if the system was not diplexed.

EQUIVALENT Q

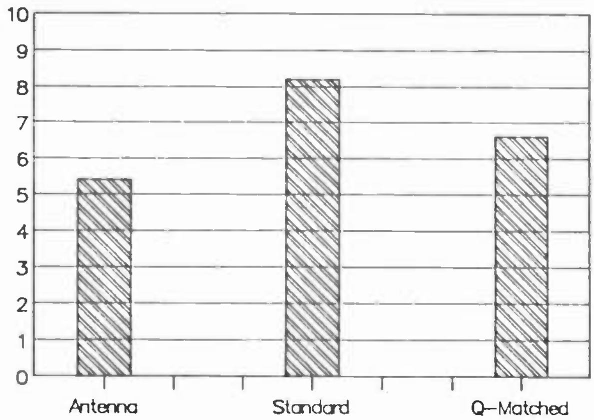


Figure 10 - Low Frequency

EQUIVALENT Q

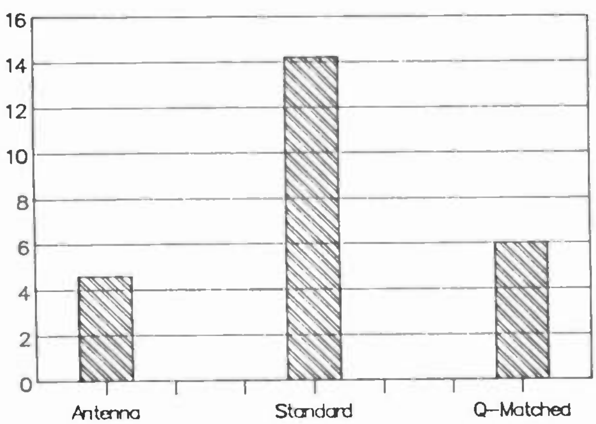


Figure 11 - High Frequency

For the high frequency, the standard design system had an equivalent Q of 14.2 compared to the equivalent Q of 6.0 for the Q-Matched system. Both of these numbers can be compared to the equivalent Q of the antenna itself of 4.6.

Figure 12 compares the port to port isolation of the two diplexer designs. The low to high frequency isolation for the standard design is 66 dB. For the Q-Matched system the port to port isolation is 88 dB. This is over a 20 dB increase in isolation.

PORT TO PORT ISOLATION

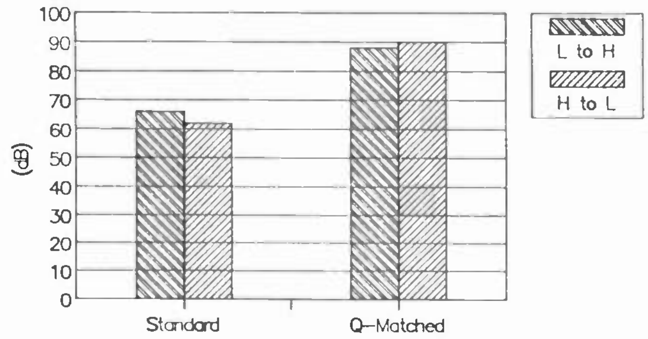


Figure 12

The high to low frequency isolation for the standard design is 62 dB. For the Q-Matched system the port to port isolation is 90 dB. This is nearly a 30 dB increase in isolation.

Conclusion

In this paper, equations are given that are relevant to the design of diplexers for AM radio stations. In the section titled Component Values, the equations for each of the four trap types are expressed as a function of C1. These equations can be used to design the most common traps used in diplexed systems.

It was shown that the bandwidth of a diplex system is affected by the choice of filter components used in the design. By the use of the Q-Matching technique the negative effect of the diplexer on the bandwidth of the system can be minimized. Figures 10 and 11 show the improvement in Q_E at the low and high frequencies by employing the Q-Matching technique.

Also, by using the Q-Matching technique, the port to port isolation will be improved. For the sample problem presented, the port to port isolation was increased by more than 20 dB.

THE SPLATTER MONITOR AND SPECTRUM ANALYZER—MEASUREMENT COMPARISONS

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Introduction

Can spectrum measurements made with a Splatter Monitor compare favorably with measurements made with a much more expensive spectrum analyzer? The short answer to this question is yes; comparison of spectrum measurements between the Splatter Monitor and the spectrum analyzer agree well.

This paper will present field measurement comparisons between the Delta Electronics Model SM-1 Splatter Monitor and the popular Tektronix Model 7L5 spectrum analyzer. The measurement techniques are described along with a theoretical discussion of spectrum measurements.

The Splatter Monitor

Since the Splatter Monitor employs a fundamentally different measurement scheme than that used in spectrum analyzers, a brief review of Splatter Monitor operation is necessary for the theoretical discussion.

Figure 1 is a simplified block diagram of the Splatter Monitor operating in the offset mode, the mode used in spectrum measurements. The local oscillator is phase locked to the input carrier frequency. Both in-phase (I) and quadrature (Q) mixers directly convert the RF input to baseband. Carrier and up converted products are removed by a 100 kHz low pass filter. This circuit is called a direct conversion or homodyne receiver.

Two important characteristics of this receiver should be noted. One, the in-phase (I) and quadrature (Q) mixers recover all of the available RF energy. Two, this homodyne receiver looks at both sides of the carrier simultaneously. The only practical disadvantage of this circuit occurs when an interfering signal corrupts the measurement. The spectrum analyzer shows each sideband independently so that the uncorrupted sideband can be used to estimate the level of the corrupted sideband. The Splatter Monitor cannot do this. Unfortunately, the ability to view both sidebands independently accounts for much of the cost of a spectrum analyzer.

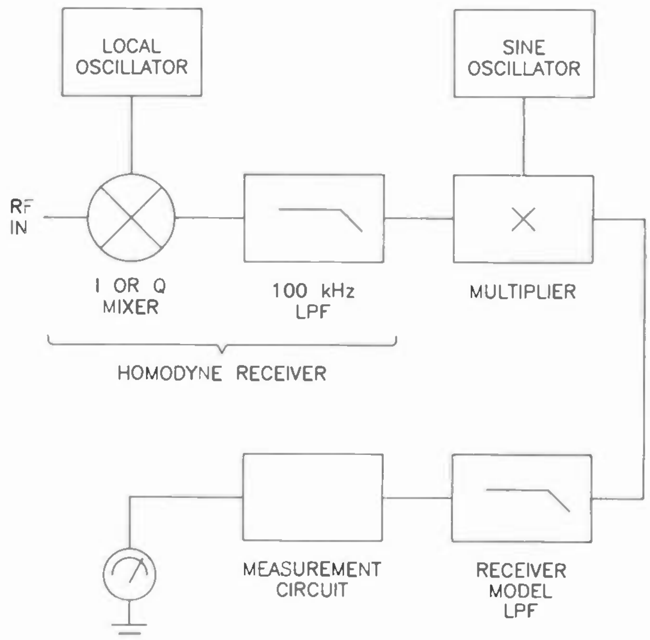

Figure 1

One might think that since both sidebands are viewed simultaneously by the Splatter Monitor, an imbalance in the sidebands would result in an erroneous Splatter Monitor reading. This is not the case as a simple example illustrates. Starting with a 100% amplitude modulated signal, the in-phase (I) detector will have a one volt peak demodulated signal and the quadrature (Q) detector will have zero volts output. If all of the energy from the lower sideband were added to the upper sideband, a single sideband signal would result. An examination of the appropriate vector modulation diagram reveals that the output of the in-phase detector would decrease by 3 dB and that the output of the quadrature (Q) detector would increase to this same level. Thus the recovered energy is simply redistributed between the two detectors.

The instantaneous magnitude of the baseband signal is the vector sum of the in-phase (I) and quadrature (Q) detector outputs. If one detector's output is 10 dB or more above the other detector's output, the lower level detector output can be ignored. A 10 dB level difference results in a maximum error of only 0.4 dB. For Splatter Monitor measurements, the in-phase detector usually dominates by at least 10 dB.

The output of the homodyne receiver is multiplied by a pure sine wave. The sine wave frequency is the frequency offset from the carrier. Thus, if we wish to examine spectrum components 20 kHz removed from the carrier, the sine wave frequency would be set to 20 kHz. A receiver model low pass filter isolates the spectrum components of interest. This is equivalent to tuning the modelled receiver to the two spectrum regions above and below the carrier.

A measurement circuit consisting of an absolute value circuit, a peak detector, and a logarithmic amplifier drives the meter. The splatter indications are displayed in decibels referenced to the carrier level (dBc).

Bandwidth Effects

The obvious, underlying reason for spectrum emission limitations is reduction of interference to other radio stations. This interference is perceived by people listening to their radio receivers. Thus, a sensible approach might be modelling the real world interference condition by using measurement bandwidths analogous to real radio receivers and by using meter ballistics that approximate the psychoacoustic response of the human ear.

This is done in the Splatter Monitor in two of the four selectable receiver model low pass filters. A 3 kHz low pass filter models the bandwidth of a low cost, narrow bandwidth receiver and an 8 kHz low pass filter with NRSC deemphasis models a high performance receiver. When either of these filters is selected, the peak detector in the measurement circuit is switched to slower attack and faster decay times for the required meter ballistics.

The National Radio Systems Committee (NRSC) recommends 300 Hz resolution bandwidth for spectrum analyzer measurements of emission spectrum. This is, of course, much narrower than the bandwidth of a typical radio receiver. Consequently, the spectrum analyzer will intercept a much smaller fraction of the RF energy in a splatter burst than a typical radio receiver. One might surmise that such spectrum analyzer readings are, somehow, invalid since the readings would not reflect real world receiver interference levels.

The 300 Hz resolution bandwidth, however, permits measurement of the expected attenuation slope of the spectrum at 10 kHz from the carrier when using a NRSC cutoff filter. A wider resolution bandwidth would smear and shift this slope showing the spectrum analyzer IF filter skirts rather than the desired measurement. The maximum spectrum emission levels specified by the National Radio Systems Committee are based upon the use of 300 Hz resolution bandwidth. A wider resolution bandwidth would simply require higher maximum permissible emission levels. If, for example, a 10 kHz resolution bandwidth were employed in an effort to model a typical radio receiver, few radio stations would comply with today's FCC emission regulations.

One of the selectable receiver model low pass filters in the Splatter Monitor has 0.5 kHz audio bandwidth. This is equivalent to a 1 kHz RF bandwidth. Since the frequency offset from the carrier is selectable in 1 kHz steps, the use of the 0.5 kHz receiver model neatly divides the spectrum into 1 kHz segments and guarantees that no transmitter spurious output is overlooked. When this receiver model is selected, the peak detector is switched to a very fast attack time with a very slow decay time. Thus, the Splatter Monitor meter shows the highest peaks with little decay between peaks. This makes the meter easier to read and corresponds with the fast peak readings of the spectrum analyzer.

Field Measurement Technique

Delta conducted field tests and made observations of stations broadcasting in the Washington, D.C. vicinity. These measurements were made without the knowledge of the stations. A mobile van was equipped with an AC generator, a Tektronix oscilloscope with a 7L5 plug-in spectrum analyzer module, a Delta Splatter Monitor, a Delta Model AWA-1 Active Whip Antenna, a Potomac Instruments Model QA-100 Quantaural Audio Program Analyzer, and an oscilloscope camera. Figure 2 is a block diagram of the test setup.

Using the active antenna, the van was positioned near the transmitting site of the AM station. The RF spectrum was then simultaneously measured with the spectrum analyzer and the Splatter Monitor. The Model QA-100 permitted monitoring of the audio processing characteristics of each

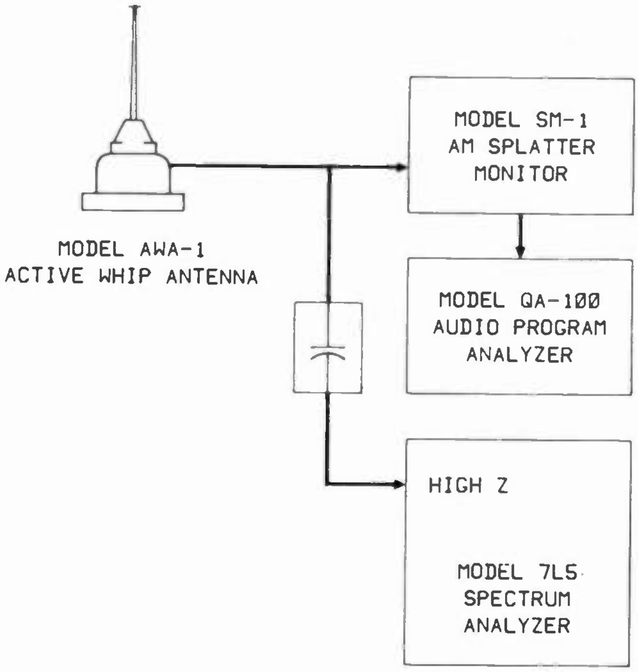

Figure 2

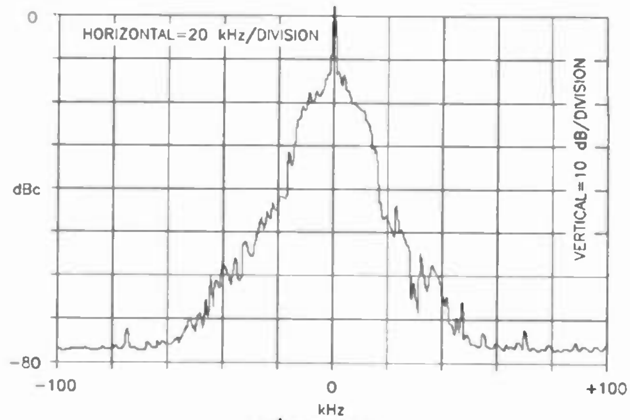

Figure 3

Table 1

Offset kHz	0.5 kHz I, dBc	0.5 kHz Q, dBc
15	-30	-49
20	-40	-58
30	-65	-78
40	-65	-72
50	-71	<-85
60	-80	<-85
70	-85	<-85
80	<-85	<-85
90	<-85	<-85
99	<-85	<-85

station measured. The spectrum analyzer was set for operation in accordance with the recommendations of NRSC-2. That is, the spectrum analyzer was set for a resolution bandwidth of 300 Hz, span of 20 kHz per horizontal division, peak hold with no video filtering, and sweep measurement for 10 minutes. During this 10 minute measurement period, Splatter Monitor readings were taken at 0.5 kHz, 3 kHz and NRSC bandwidths using both the in-phase (I) and quadrature (Q) detectors. For this paper, only the 0.5 kHz readings will be presented since these readings most closely approximate spectrum analyzer measurements.

The Splatter Monitor readings were taken by observing the "peaks of frequent occurrence" similar to the method used in modulation monitor measurements where an occasional overmodulation condition is tolerated. Since the spectrum analyzer is sweeping the band of interest and only observing any given frequency segment a fraction of the measurement time, the spectrum analyzer is likely to miss the rare maximum peak and observe the more frequent peaks in any given spectrum segment. In essence, the Splatter Monitor operator and the sweeping spectrum analyzer are exercising similar judgement.

Figures 3, 4 and 5 are digitized versions of the oscilloscope pictures taken at three of the stations visited. Tables 1, 2 and 3 are the corresponding Splatter Monitor readings, respectively, for the same stations. Observe that the carrier peak of each spectrum picture appears to lie above the reference level. This is due to low frequency modulation energy falling within the 300 Hz resolution bandwidth around the carrier. The carrier reference was obtained with the 10 Hz resolution bandwidth to overcome this problem.

Figure 3 is a typical spectrum picture for a non-NRSC, monophonic station. Note in Table 1 that all of the quadrature modulation readings by the Splatter Monitor are well below the in-phase readings at the same offset frequencies, so that we may ignore the quadrature readings. The first data point, 15 kHz from the carrier, shows -30 dBc. Judging from the spectrum picture, the upper sideband appears to cross +15 kHz at about -30 dBc and the lower sideband appears to cross -15 kHz at about -28 dBc. Although this is in good agreement with the corresponding Splatter Monitor measurement, the picture illustrates one of the difficulties with making spectrum analyzer measurements. The slope of the spectrum is falling so fast at 15 kHz from the carrier that it is difficult to judge the exact spectrum level.

The second measurement point at 20 kHz from the carrier reads -40 dBc on the Splatter Monitor. The spectrum analyzer picture shows about -47 dBc on the upper sideband and somewhere between -42 and -46 dBc on the lower sideband. Thus, the jagged line on the lower sideband produces uncertainty in the measurement. Do we interpret the spectrum analyzer as the highest peak or as the average slope of the trace? A good case can be made that the highest peak should be used since every point on the display is, presumably, a valid measurement. This highest peak at -42 dBc agrees well with the Splatter Monitor.

The third measurement point at 30 kHz from the carrier illustrates another spectrum analyzer measurement problem. We know from the Splatter Monitor that this transmitter is a good AM source because the quadrature modulation (IPM) level is low. The upper and lower sidebands should, therefore, be symmetrical. However, the spectrum analyzer readings of the upper and lower sidebands at 30 kHz from the carrier are dramatically different. The lower sideband is between -52 dBc and -56 dBc whereas the upper sideband reads between -62 dBc and -67 dBc. Clearly, this is not just a case of reading interpretation.

The explanation of this phenomenon lies with the sweep measurement technique. The upper sideband reads less than the lower sideband because the bursts of splatter energy were absent whenever the spectrum analyzer was sweeping the upper sideband. When sweeping the lower sideband, the spectrum analyzer did catch one or more stronger splatter bursts. Of course, the same phenomenon applies to the Splatter Monitor as shown by the data. Neither instrument will measure where they are not looking. The Splatter Monitor measurement was, evidently, taken when the spectrum analyzer was recording the splatter levels of the upper sideband since these measurements agree.

Figure 4 is the spectrum analyzer measurement of monophonic station using NRSC preemphasis and filtering with a mostly talk format. First, notice the presence of two carriers from other radio stations which invalidate our Splatter Monitor's field antenna measurements at 40 kHz and 99 kHz offset from the carrier. The presence of these other stations is revealed by the appearance of the other stations' programming in the Splatter Monitor's speaker. As discussed above, the spectrum analyzer easily handles this problem. When using the Splatter Monitor, this difficulty may be overcome by taking readings on both sides of the invalid offset frequency and interpolating to the correct reading.

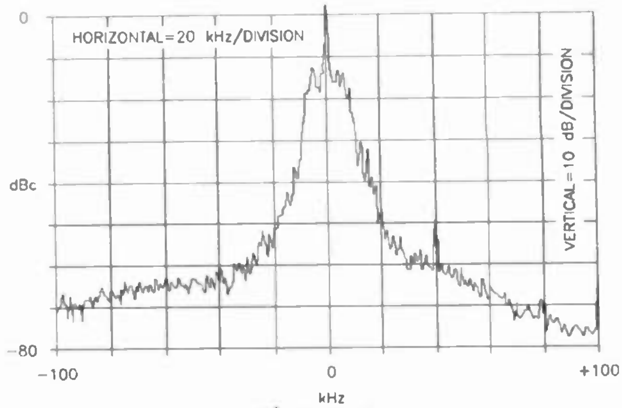

Figure 4

Table 2

Offset kHz	0.5 kHz I, dBc	0.5 kHz Q, dBc	Vector Sum, dBc
15	-40	-42	-38
20	-49	-52	-47
30	-62	-56	-55
40	-55	-53	-51
50	-70	-64	-63
60	-70	-66	-64
70	-70	-67	-65
80	-73	-69	-67
90	-76	-74	-72
99	-79	-78	-75

Second, observe from Table 2 that the output of the quadrature (Q) detector is almost as strong as the output of the in-phase (I) detector. This is due to a high level of incidental phase modulation which has compromised the effectiveness of the NRSC filtering. These high quadrature readings must be taken into account. Since peaks of incidental phase modulation are associated with peaks of envelope modulation and since in-phase splatter is associated with peak clipping on the same modulation peaks, one is justified in assuming that in-phase (I) and quadrature (Q) peaks occur simultaneously. Thus, the magnitude of the resulting peak is the vector sum of the in-phase (I) and quadrature (Q) peaks. This calculation appears in the sum column of Table 2.

The vector sum of the first data point at 15 kHz offset is -38 dBc. The upper and lower sideband peaks of the spectrum analyzer display are -35 dBc and -38 dBc, respectively. The second data point has a vector sum of -47 dBc and the upper and lower sideband peaks are -48 dBc and -49 dBc respectively. For the third data point at 30 kHz offset, the numbers are -55 dBc versus -57 dBc and -58 dBc. On the surface, this vector sum approach appears to yield

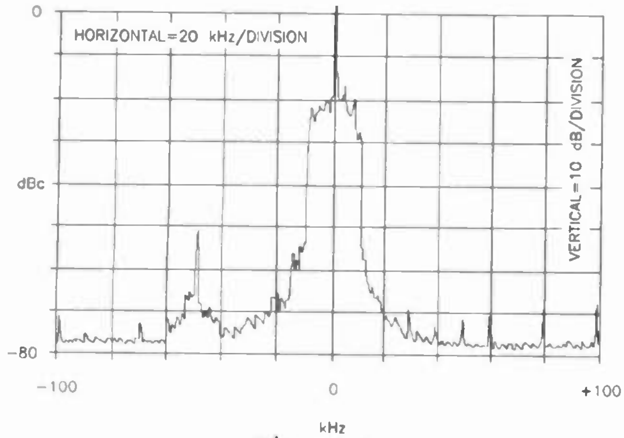

Figure 5

Table 3

Offset kHz	0.5 kHz I, dBc	0.5 kHz Q, dBc
15	-55	-70
20	-64	-80
30	-75	-80
40	-81	-82
50	-64	-65
60	-80	-80
70	-83	-82
80	-83	-79
90	-84	-82
99	-85	-83

close agreement between Splatter Monitor readings and the spectrum analyzer's highest peak readings. However, due to the fact that in-phase (I) and quadrature (Q) peaks do not necessarily occur at the same time as sideband peaks, this method is difficult to theoretically justify. Delta is pursuing this issue with special attention to AM stereo radio stations.

Figure 5 shows spectrum analyzer measurements of an exceptionally clean monophonic station employing NRSC preemphasis and filtering. This station's spectrum is so clean that the sidebands of another station only 50 kHz away are visible down to -75 dBc. This is attributable to the NRSC filtering, exceptionally low incidental phase modulation, and conservative modulation practices.

Again, the quadrature (Q) detector output is well below the level of the in-phase (I) detector and the quadrature contribution can be ignored. Examination of the first three data points shows that the Splatter Monitor's in-phase detector output agrees closely with the highest corresponding sideband peak observed on the spectrum analyzer.

The last observation about these field measurements is the level of the measurements far from the carrier. The spectrum analyzer readings at 90 kHz offset, for example, show a peak at about -75 dBc whereas the Splatter Monitor readings are in the -80 dBc range. This difference has not been fully explained except to note that the 7L5 spectrum analyzer is rated for -75 dB intermodulation products. Delta is still investigating this phenomenon.

Conclusion

Careful operation of the Splatter Monitor and the spectrum analyzer yield measurements in substantial agreement. In the presence of significant quadrature modulation or high levels of incidental phase modulation, the vector sum of the in-phase and quadrature peaks appears to yield good agreement with the highest corresponding spectrum analyzer sideband peak. In the presence of interfering signals from other stations, splatter levels at the interfering frequencies may be estimated by using an interpolation method.

Obviously the most careful means of determining peak splatter levels at any given frequency is to tune a measurement instrument to that frequency and observe the maximum peak. This avoids missing these maximum peaks in a sweeping frequency measurement and avoids judgement of "peaks of frequent occurrence." This is accomplished with a spectrum analyzer by setting the spectrum analyzer in zero span (i.e., zero Hz per horizontal division), tuning to the frequency of interest, and activating the peak hold function.

Delta has developed a digital peak hold circuit for long term retention of the maximum readings of the Splatter Monitor. Additionally, Delta has constructed a receiver model filter circuit for installation in the optional filter location of the Splatter Monitor. This filter closely matches the response of the synchronously tuned IF filter in the 7L5 spectrum analyzer with 300 Hz resolution bandwidth. With these new tools Delta hopes to develop even more accurate spectrum measurement techniques.

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2001 Eye Street, N.W.
Washington, DC 20006

BROADCAST APPLICATIONS FOR VOICE ACTIVATED MICROPHONES

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Abstract

The engineering of broadcast quality audio for radio and television talk shows becomes progressively more difficult as the number of on-air microphones increases. To handle these multiple microphone situations, a number of broadcast engineers have begun employing voice-activated microphone systems (also known as automatic mixing systems) to improve the audio quality of these productions. This paper describes the problems of multiple open microphones and how a variety of broadcast productions involving multiple microphones were handled with automatic mixing.

The Problems of Multiple Open Microphones

Broadcast engineers face the same problems as recording and PA engineers when employing multiple open microphones. Comb filtering, ambient noise and reverberation build-up, and acoustic feedback can plague remote broadcasts and studio productions. Since audio quality rapidly deteriorates as the number of open microphones increases, the remedy is to keep open the minimum number of microphones that will handle the program audio. Voice-activated systems help the engineer accomplish this by keeping unused microphones attenuated and activating microphones (when needed) within milliseconds.

Comb filtering occurs when open microphones at different distances from a sound source are mixed together. (See figure 1.) Since sound travels at a finite speed, the sound waves from the source arrive at the microphones at different times. As a result, the outputs of the microphones are not in phase with each other. When combined in a mixer, these out-of-phase microphone signals produce a combined frequency response very different from the frequency response of any single microphone. Figures 2a-2d clearly illustrate comb filtering.

The frequency response curve of a Shure model SM80 omnidirectional condenser microphone is shown in figure 2a. The SM80 was placed two feet away from the source loudspeaker. (All measurements were made in an anechoic chamber.) Note the flat, smooth response curve of the single SM80 microphone.

Figure 2b is the combined frequency response of two SM80 microphones. One microphone was two feet from the loudspeaker and the other was four feet. The gain settings on the mixer were set the same for both microphones. Comb filtering is easily seen in this response curve.

Figure 2c is the combined frequency response of three SM80 microphones. The microphones were placed two feet, four feet and six feet from the loudspeaker. As before, the gain settings on the mixer were set the same for all three microphones.

Figure 2d is the same setup as figure 2c, except the gain settings were adjusted to provide equal output levels from each microphone. Note that the comb filtering is the most severe in this situation.

The aural result of comb filtering is an audio signal that sounds hollow, diffuse, and "phasey." Voice-activated systems reduce comb filtering by keeping unused microphones attenuated. It is not necessary to keep unused microphones completely off to gain the aural advantages of automatic mixing. On most automatic mixers, the amount of attenuation applied to a microphone when it is not in-use is adjustable. Typical adjustments range from 8 dB of attenuation to completely off. An attenuation level of 15 dB has been found to be satisfactory for most situations. Keeping a microphone attenuated 10 to 15 dB instead of completely off makes the gating-on of that microphone sound smoother. It typically takes less than five milliseconds to raise a voice-activated microphone from the attenuated state ("gated off") to the in-use un-attenuated state ("gated on").

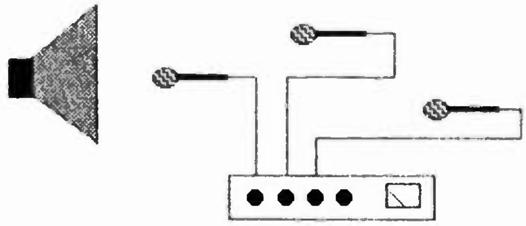

Figure 1.
Comb filtering occurs when open microphones at different distances from a sound source are mixed together.

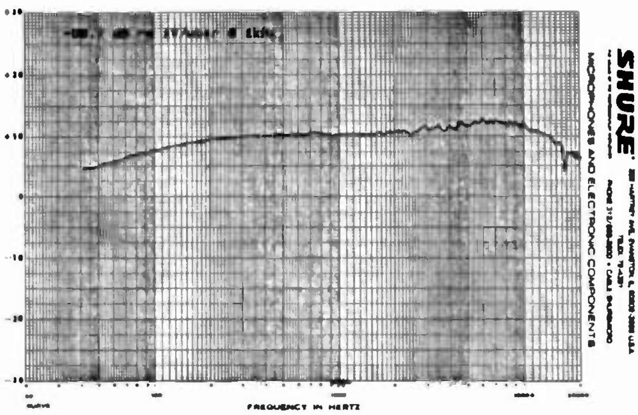

Figure 2a.
Frequency response of SM80 omnidirectional microphone at 2 feet from sound source

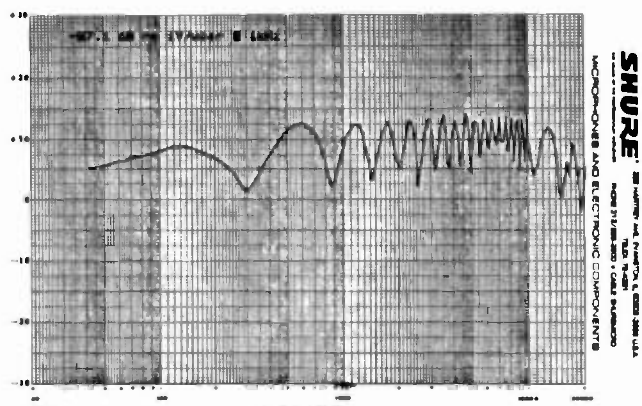

Figure 2b.
Combined frequency response curve for two SM80 microphones; one at 2 feet and one at 4 feet; equal gain settings on mixer

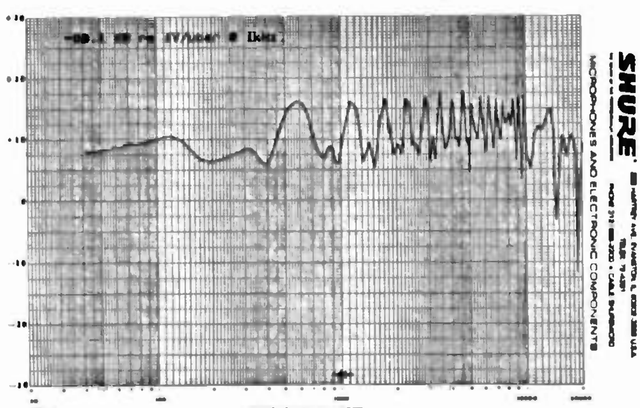

Figure 2c.
Combined frequency response curve for three SM80 microphones; one at 2 feet, one at 4 feet, and one at 6 feet; equal gain settings on mixer

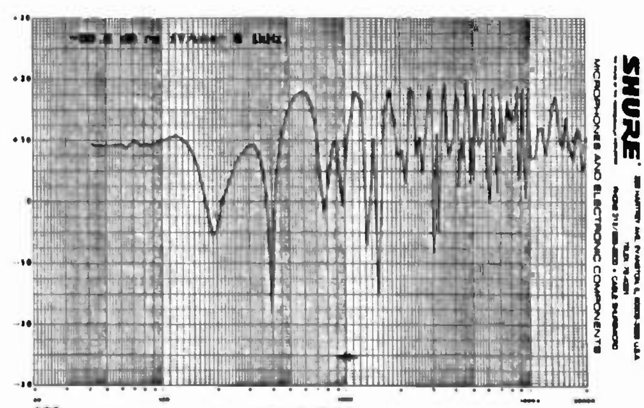

Figure 2d.
Combined frequency response curve for three SM80 microphones; one at 2 feet, one at 4 feet, and one at 6 feet; gain settings on mixer set for equal output from each microphone

Another effect of multiple open microphones is the build-up of ambient noise and reverberation. If eight microphones are open when only one microphone is needed, the mixed audio output will contain the ambient noise and reverberation of eight microphones, but only the desired signal from the one microphone located closest to the sound source. (It is assumed that the additional seven microphones would not add any useful signal as they are located at various distances from the sound source and would not be in phase with the source microphone. This explanation is overly simplified for the sake of clarity.) This increase of ambient noise and reverberation deteriorates the overall signal-to-noise ratio as the mixed audio output now contains a higher percentage of ambient noise and reverberation than if only the microphone closest to the sound source were open.

Each time the number of open microphones is doubled, e.g. one to two, two to four, four to eight, etc., the overall system gain is raised 3 dB. As a result of this gain change, ambient noise and reverberation also increase as the number of open microphones grows. To negate this 3 dB increase and control the build-up of unwanted noise, automatic mixing systems employ a circuit referred to as "NOMA" which stands for Number of Open Microphones Attenuation. NOMA lowers the overall gain by 3 dB every time the number of open microphones doubles. With a voice-activated mixing system, ambient noise and reverberation remain constant as microphones gate on and off because of the NOMA circuit action. Without NOMA, noise modulation ("pumping") would be objectionable as ambient noise and reverberation would increase and decrease as the number of open microphone varied.

Acoustic feedback or "howling" can also be a problem when a sound reinforcement (PA) system is used, for example, on a remote broadcast with an audience. The safety margin below the PA system's feedback point is reduced each time another microphone is opened. If a sound reinforcement system is being operated at 5 dB below the feedback point with one microphone open, the system would feed back if four microphones were opened as the overall gain would rise by 6 dB. "Howling" or "ringing" would result as the system would then be 1 dB above the feedback point.

The solution, once again, is NOMA. As more microphones gate on, the overall gain will remain constant as previously described. With a NOMA circuit, the audio engineer can be assured that if the sound reinforcement system does not feed back when any one microphone is open, the system should remain stable if all the microphones are open.

Since comb filtering, build-up of ambient noise, and feedback can be controlled by using voice-activated microphones, broadcast engineers are employing automatic mixers with greater frequency. The remainder of this paper documents unique broadcast applications of voice-activated microphones.

Engineer-less Public Affairs Programs

Public affairs programming can be a problem for radio stations. The taping of interviews or panel discussions often has to be done at inconvenient times when engineering personnel are in short supply. Ron Turner, chief engineer of WCLR (Chicago), decided to dedicate a small studio to public affairs and design it to be used at any time without the need for an engineer's presence.

In the center of the studio, Turner placed a round table that could comfortably accommodate a program host and three guests. To handle the automatic mixing for the studio, Turner chose a voice-activated system manufactured by Shure Brothers Inc. A unique feature of this automatic system is that microphone gating is direction-sensitive: a microphone can be gated on only when the sound source is located within a 120° gating zone, known as an acceptance window. This window is ±60° from 0° on axis. (See figure 3.) In addition to being within the acceptance window, the sound source must also be approximately 5 dB above the ambient noise (as measured at the microphone position) in order to activate the microphone gate.

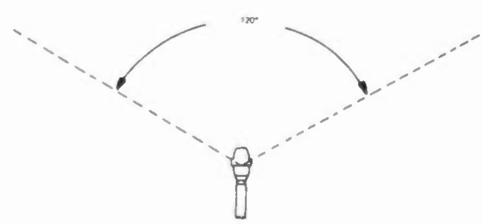

Figure 3.
120° acceptance window for
direction-sensitive gated microphone

For this studio, four low-profile boundary microphones were positioned every 90° in the center of the table. With this arrangement, if two people are in the studio, only two microphones will be activated, provided each person stays within his microphone's 120° acceptance window. Three people would activate three microphones, etc. This arrangement solved the problem of how many microphones should be on for each interview.

Another concern for Turner was losing studio ambience if all microphones gated off during a lull in the conversation. This was solved by using TTL compatible logic control terminals on the rear panel of the automatic mixer. By using an external transistor as a switch, the mixer was modified to activate the host's microphone if the three guests' microphones were off. With this arrangement, at least one microphone would be on at all times. This eliminated any possibility of ambience dropout.

The output of the automatic mixer was fed into a compressor to handle level changes and to provide a consistent audio signal to a reel-to-reel tape recorder located in an adjacent production studio. A remote record/stop switch was provided to allow the host to control the recorder from the studio. Since the studio is left powered up, all a program host needs to do is load a reel of tape, go into the studio with the guests, and start the recorder when ready. (See figure 4.)

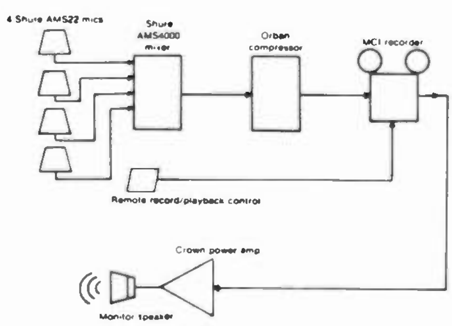

Figure 4.
Audio flowchart for WCLR public affairs studio

Recently, Turner added phone-in capability. The addition of a multi-line phone, digital hybrid, power amplifier, and tabletop loudspeaker allows a fifth guest to participate in the show without physically being in the studio.

WCLR's public affairs studio has been in operation for over four years and, according to Turner, "operates without a hitch." An additional bonus for Turner was a marked improvement in the audio quality of the public affairs programs due to the voice-activated system keeping the unused microphones attenuated.

Radio Talk Shows using Voice-Activated Microphones

Lou Ludovici of WICC Radio (Bridgeport, Connecticut) installed voice-activated microphones in the fall of 1988. The studio that received the automatic mixing is used for talk shows (studio and call-in) as well as for production work. Four microphone positions handle a host and up to three guests. Like Ron Turner at WCLR, Ludovici initially selected a low-profile boundary microphone to help reduce "mic fright" in guests not accustomed to being in front of a microphone. However, he later switched to more conventional looking microphones as guests were placing papers and other items on top of the low-profile microphones!

The four gated signals from the voice-activated microphones are mixed and fed into a single channel of a large studio console for the talk shows. In addition, a direct output (i.e., pre-gating and pre-fader) from each microphone is fed to an individual channel of the same console and employed if the studio is used for production.

TTL compatible logic terminals on the automatic mixer are used to provide cough buttons for each microphone location. The host has additional switches; one to mute all the microphones, and another to talkback to the producer using the on-air microphone. Of course, the talkback feature mutes the guests' microphones at the same time so no talkback goes over the air.

As mentioned, the studio is used for call-in shows. A tabletop loudspeaker is utilized to allow studio guests to hear the caller. This arrangement has eliminated the need for the guests to wear headphones. During commercial breaks, the loudspeaker automatically switches to program audio so that the people in the studio can tell when it is time to go back on air. The host does wear headphones so he may hear the producer while on the air.

Ludovici remarks, "The voice-activated system is working very well for our purposes and even the skeptics have been converted by it."

A similar system is in operation at WNWS/WLYF (Miami). Engineer Al Byers installed a voice-activated microphone system to help reduce the effect of serious acoustical deficiencies in the station's talk show studio - a 13 dB bump at 240 Hz and two glass walls were the worst problems!

Byers designed a four-channel system using black gooseneck microphones. He modified the automatic mixer so that the direct outputs were gated. These four individual gated microphone signals were fed into separate channels on the main studio console.

According to Byers, the voice-activated microphones have made a "dramatic difference" in the audio quality, and the problems of "hollowness" and "loss of articulation" have been solved.

Television Talk Show in the Round

When WTBS (Turner Broadcasting System, Atlanta) decided to produce a public affairs program dealing with current economic and cultural problems, Bill Tullis, chief audio engineer, faced an unusual problem. The set was to be a round table large enough to seat 14 participants. A camera was placed every 90° behind the participants and hidden by thin black scrim and selective lighting techniques. Mixing audio posed a problem as there was no one position where the audio engineer could see the faces of all the participants, and leaving 14 microphones open was aurally unacceptable. Also, the show was an unrehearsed and open forum, so quick verbal exchanges were a distinct possibility. Tullis decided to use voice-activated microphones to help solve this unusual audio situation.

A low-profile boundary microphone was placed between every two participants and located several feet from the edge of the table. Using a low-profile microphone made the set look "cleaner" and enhanced the output of each channel by using the 6 dB increase of sound pressure that occurs at any large acoustical boundary such as a table surface. (See figure 5.)

The channel LEDs located on the automatic mixer, which illuminated whenever a microphone would gate on, were useful to Tullis and the director. Tullis knew immediately what volume control to adjust if he needed to ride gain. And the director employed the channel-on LEDs to assist him in determining what camera to call up when someone new started to speak.

WTBS now routinely uses voice-activated microphones to handle the audio for panel discussions, conferences, and in-house meetings, as well as selected productions. Voice-activated microphones have relieved the audio engineer of trying to predict who will speak next. More attention can now be paid to proper levels and the overall quality of the mix.

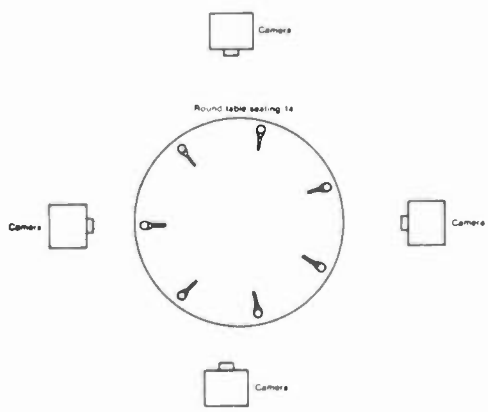

Figure 5.
Microphone and camera positions
for WTBS talk show

Radio Remotes with Voice-Activated Microphones

WCAU (CBS Radio, Philadelphia) regularly features call-in talk shows originating from remote sites. Jack Miller, the former technical operations manager for WCAU (now retired), decided to utilize voice-activated microphones to help control echo problems resulting from satellite transmission delay.

The cause of the echo problem was the acoustic link that existed at the remote site between the PA system and the talent microphones. If these microphones were left open, the listeners in Philadelphia would hear an echo. For example, a caller's voice would be sent to the remote site via telephone lines. The caller's voice would go out over the PA system and, of course, be picked up by any open talent microphone. Because of this acoustic link, the caller's voice would then be sent back to WCAU via satellite and re-broadcast 1/4 second later as a distinct echo because of satellite transmission time. The solution was to keep the talent microphones off when they were not needed. This would break the acoustic link between the PA system and the talent microphones. (See figure 6.)

Miller had considered using noise gates for the microphones but rejected the idea because of the time-consuming threshold adjustments that are necessary for proper gating. He settled on an automatic mixing system manufactured by Shure whose operating principle eliminates threshold adjustments. This system made the setup of remotes faster and easier.

According to Miller, the use of voice-activated microphones eliminated the echo problems and remotes with call-in questions became "duck soup".

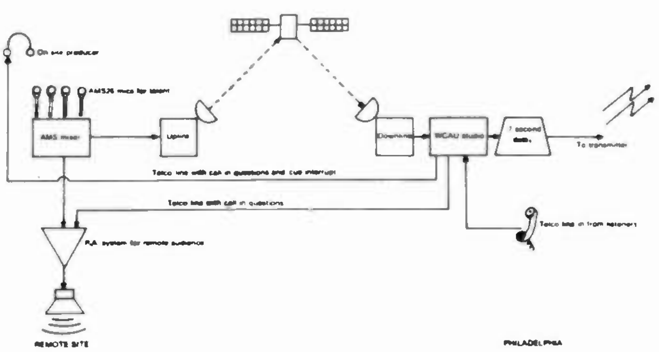

Figure 6.
Diagram for WCAU radio remote with call-in

Television Remote Using 47 Voice-Activated Microphones

In late 1983, ABC News (Washington D.C.) utilized a 47-channel voice-activated microphone system to facilitate the taping of a news special. "Voting for Democracy" documented an important symposium on voting practices in America. The program was co-sponsored by ABC News and Harvard University's Kennedy Institute of Government.

The symposium brought 47 politicians and academicians together in a caucus room of Washington's Russell Senate Office Building. Participants included former Presidents Ford and Carter. Panelists were seated around an imposing oval table, where they presented papers and discussed at length the various problems of participatory government. ABC taped 15 hours of discussion, which were edited down to a one-hour special.

Because of the program's unrehearsed format, the unusual oval set, and the large number of participants, the audio portion of the symposium presented considerable difficulties. Marc Drazin, ABC technical coordinator for this production, decided that voice-activated microphones would be the answer...but it wasn't quite that easy.

First, there would be no rehearsal, so it was not feasible to use an automatic mixer that required threshold settings determined before the symposium began. Second, the participants would be seated tightly together, and it was very likely that several microphones would be activated when someone spoke if sound pressure level was the only requirement for gating. Drazin solved these two major problems by using an automatic mixing system with a unique operating principle. He decided on a voice-activated system manufactured by Shure that provides direction-sensitive gating. Thus, any single microphone activation area would be confined to a 120° acceptance window. Direction-sensitive gating reduced the possibility of a participant activating other microphones than his own. In addition, the gating threshold automatically "floated" with the ambient noise around each microphone.

Each participant was seated in front of a microphone mounted ten inches above the table surface. As each person had a 100-page briefing binder, the microphone needed to be elevated so that paper noise would originate outside the gating window. With this arrangement, the participant could not gate on a microphone by shuffling paper or turning pages.

The 47 microphones were divided into three sub-groups, each of which were sent to one of three separate mixing locations. (See figure 7.) At each mixing location, two eight-channel automatic mixers were linked together to control the microphones, with the combined output feeding a 24-channel manual mixing console. For redundancy, the non-gated direct output of each automatic mixer channel fed an individual console channel. These redundant feeds were never used, as the voice-activated system operated without a problem. According to Drazin, "The (automatic) mixer's electronics switched silently, swiftly, and correctly every time. No one was up-cut or clipped at any time."

Two of the 24-channel consoles fed the third console which acted as the master. This master console also controlled a sub-group of 15 voice-activated microphones. The output of the master console supplied audio for the remote production truck, the microwave transmitter to the satellite uplink, the audience PA system, and the participant's monitor loudspeakers. (See figure 8.) These speakers were placed inside the opening of the oval set and hidden by plants. A mix-minus system was used to avoid feedback, e.g., sub-group 1 microphones were fed to sub-group 2 and 3 monitors, but not to sub-group 1 monitors. This worked well, as symposium participants sitting in each sub-group could hear each other acoustically and did not need to hear their neighbors through the monitor system.

The audio engineer located at each mix position was responsible only for his sub-group of microphones. The job of riding gain was made much easier by the channel-on LEDs located on the automatic mixers. Whenever someone spoke, the LED above that channel's gain control would illuminate and act as a beacon, guiding the engineer to the right knob if level adjustments were necessary. This feature was very important considering the audio engineers had poor sight lines and no video monitors to assist in visual identification of whom was speaking.

ABC News was so pleased with the operation of the voice operated microphones, they used a similar system to cover the Iran Contra hearings in 1987.

Summary

Voice-activated microphones are fast becoming a valuable tool to the broadcast engineer. As the complexity of studio and remote productions increases, automatic mixers will become indispensable in providing the best audio possible. Marc Drazin of ABC News sums it up nicely, "I shudder to think of the numerous problems we would have had if we had tried to cover 'Voting for Democracy' with conventional audio equipment and operators."

Acknowledgements

The following broadcast engineers have kindly consented to share their experience and knowledge of broadcast applications for voice-activated microphones with readers of this paper. My thanks to each of them for their invaluable assistance in providing details for this paper.

Al Byers	WNWS	Miami	305-653-8811
Bob Dove	WTTW	Chicago	312-583-5000
Marc Drazin	ABC News	Atlanta	404-431-7754
Lou Ludovici	WICC	Bridgeport	203-366-9383
Bruce Miller	ABC News	Washington	202-887-7785
Bill Tullis	WTBS	Atlanta	404-827-1226
Ron Turner	WCLR	Chicago	312-677-5900

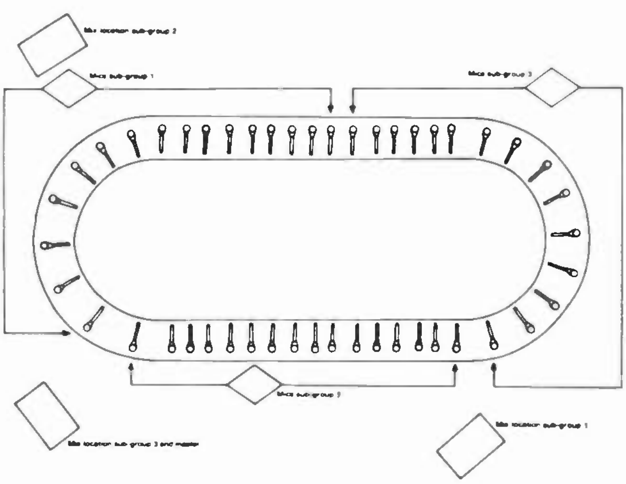

Figure 7.
Setup for ABC News symposium

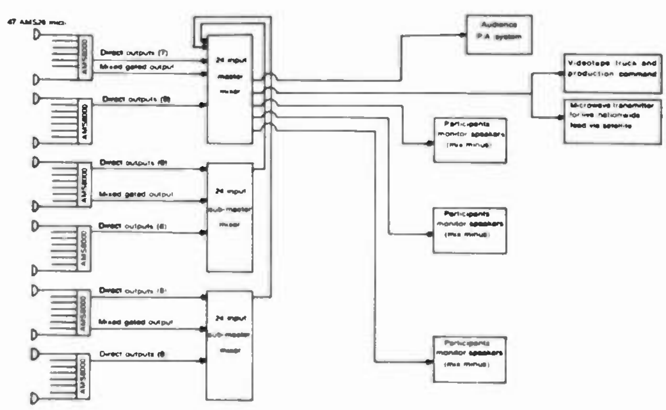

Figure 8.
Audio flowchart for ABC News symposium

ACOUSTIC NOISE LEVEL MEASUREMENT AND CONTROL TECHNIQUES FOR BROADCAST EQUIPMENT

Jeffrey H. Steinkamp
Broadcast Electronics Inc.
Quincy, Illinois

INTRODUCTION

At some point in time whether you're in management or on the technical staff of a broadcasting organization you will need to understand the basics of acoustic noise measurement and control. This may occur when working with the acoustical engineering consultant in the design and construction of your new control and/or production room. It may happen when you're evaluating new equipment such as transmitters or heat exchangers and understanding the manufacturers data concerning noise level will be essential to the selection process. Even that low noise emitted by the tape cartridge machine is critical when the machine is installed in that new high-tech production room. Plus, an occasional visit from the EPA (Environmental Protection Agency) or OSHA (Occupational Safety and Health Administration) concerning noise pollution coming from your facility will immediately heighten your interest about noise measurement and control. Nevertheless, no matter what your reason, a basic understanding of noise measurement and control will definitely be an asset to your job skills and overall technical knowledge.

This paper presents a general engineering review of sound properties and sound measurement with attention to techniques used to control noise levels of broadcasting equipment.

Points discussed will include the physical properties of sound, measurement of sound properties, the use of A, B and C weighting networks, speech interference levels, and noise criteria curves. Included in this paper will be a section on noise control methods with two examples of noise reduction used during the design of a tape cartridge machine and a FM transmitter.

PHYSICAL PROPERTIES OF SOUND

Sound is an alteration or oscillation in pressure, stress, particle displacement and particle velocity in an elastic or viscous medium. In air, sound waves take the form of alternating condensations and rarefactions. These density changes are the direct results of particle displacement. They can best be defined and measured in terms of pressure change.

The frequency of a sound is the number of periods (cycles) occurring in unit time, usually expressed as cycles-per-second (CPS) or hertz (Hz). Sound frequencies of interest in this paper are in the audible range of 20 to 20,000 cycles-per-second. Few sounds are of a single frequency. Those sounds with one frequency are called pure tones. Most music, for instance, are sounds of many tones and overtones. On the otherhand, most noise where noise is defined as unwanted or disturbing sound, consist of broadband sounds.

In noise measurement and control it is usually much more useful to know the individual pressure and corresponding frequency of the sound than to know just the overall pressure. Such a statement of individual pressures and frequencies is called a sound spectrum. Various bandwidths can be used to describe this sound spectrum. An octave band is a band ranging from one frequency to twice that frequency. Standard octave bands have been established for the audible range of hearing and are shown in Table 1.

TABLE 1. OCTAVE BAND STANDARDS

	FREQUENCY - Hz							
LOW	45	88	177	354	707	1414	2828	5657
MEAN	63	125	250	500	1000	2000	4000	8000
HIGH	88	177	354	707	1414	2828	5657	11314

MEASUREMENT OF SOUND PROPERTIES

The properties of sound, namely pressure level and frequency, are usually measured with a sound level meter and an octave band analyzer.

Sound pressure level is measured with a sound level meter. The sound pressure generates an electrical signal in the microphone, which is then amplified and transmitted through an adjustable attenuator to an indicating meter.

The frequency spectrum of a sound is determined with an octave band analyzer together with a sound level meter. A series of filters is used to greatly attenuate signal components above and below certain frequencies. When the output of a sound level meter is fed into an analyzer, the sum of the attenuator setting and the analyzer meter reading is the sound pressure level for the band of frequencies indicated by the band selector. There exist many sound pressure level meters with built-in octave band analyzers that have excellent performance at a reasonable price.

Keep in mind that the accuracy of any sound measurement depends on the acoustical response of the microphone and the electrical response of the meters. Meters should be calibrated often. Microphones are usually non-directional at low frequencies. When the wavelength is comparable to the size of the microphone, the response varies with wave length and angle of incidence. For highest accuracy, these effects should also be determined by calibration.

The unit of measure of a sound's frequency is cycles-per-second or hertz (Hz). When dealing with the audible range of hearing, 20 to 10,000 Hz will adequately cover the spectrum. Unfortunately, due to the remarkable sensitivity of the human ear, the sound pressure level cannot be so easily stated. The average range of sound pressure discernibility of the ear is from the threshold of hearing 0.0002 microbars to the threshold of pain 2000 microbars. This range of 10 million-to-one is quite cumbersome to work with from a mathematical stand point so sound pressure levels are now universally expressed in decibels (dB).

Decibels are dimensionless units for conveniently measuring power and or pressure whenever the range of values is very large. The decibel is a measure of ratio only and can be expressed as follows:

LEVEL IN DECIBELS (dB)=

$$20 \log_{10} \frac{(\text{Measured Quantity})}{(\text{Reference Value})}$$

In sound pressure level readings the reference value is always 0.0002 microbars or basically the threshold of human hearing, thus:

$$\text{SOUND PRESSURE LEVEL (dB)} = 20 \log_{10} \frac{(P)}{(0.0002)}$$

Where P = Sound Pressure in microbars.

The use of the decibel scale condenses the previous 10 million-to-one range to a more convenient and workable span of Ø to 140 dB. See Table 2 for typical sound pressure level situations.

TABLE 2. TYPICAL OVERALL SOUND PRESSURES AND SOUND PRESSURE LEVELS

PRESSURE (MICROBARS)	PRESSURE LEVEL (dB)	SOURCE
2000	140	Threshold of Pain
	130	
200	120	Threshold of Discomfort
	110	
20	100	Automobile Horn
	90	
2	80	Automobile at 40 mph
	70	
0.2	60	Conversational Speech
	50	
0.02	40	Quiet Residence
	30	
0.002	20	Whisper
	10	
0.0002	0	Threshold of Hearing

Although more convenient for numerical expression, using the decibel does make it more difficult to perceive the difference between two sound pressure levels.

It is frequently necessary to combine the effect of two sound sources or sound pressure levels. Since decibels are logarithmic units and cannot be added algebraically, it is necessary to convert each decibel reading to its equivalent power, add or subtract the powers and then convert to a combined decibel level. All these calculations may become quite involved and produce more accurate results than the situation may require. Thus, the graphical approach may be more appropriate and is shown in Figure 1.

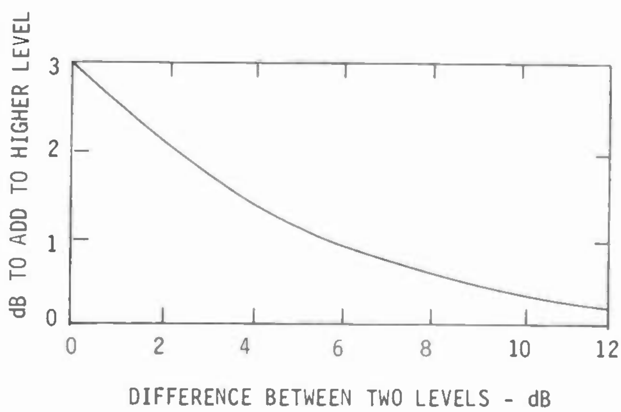

FIGURE 1. GRAPH FOR COMBINING DECIBELS

As stated before, there are many manufactures of sound level meters with or without built-in octave band analyzers. For help in selecting the proper performing, cost effective instrumentation that you may require here are two respected sources:

BRUEL & KJAER
185 Forest St.
Marlborough, MA 01752
(617) 481-7000

GenRad
300 Baker Ave.
Concord, MA 01742
(617) 369-4400

Keep in mind the physical properties of sound (pressure and frequency) and how they are measured (dB and Hz) because understanding these basics is essential before numerical values can be assigned to generated sound, acceptable sound, and the reduction required in a particular situation.

UNDERSTANDING A, B AND C NETWORK WEIGHTING, NOISE CRITERIA (NC) CURVES AND SPEECH INTERFERENCE LEVELS (SIL)

In the realm of sound pressure level measurements of which there are many, the following three types seem to be the most commonly used: A, B and C network weighting; noise criteria (NC) curves; and speech interference levels (SIL). A general review of each type of technique will be discussed.

A, B and C Network Weighting

Most sound level meters contain provisions to take measurements with either A, B or C network weighting. Why are these weightings used and how are they achieved?

During extensive testing of human hearing it was discovered that sounds or noises at frequencies under 1000 Hz do not sound as loud to human ears as equally intense (loud) noises at higher frequencies. In other words, the human ear is less sensitive at lower frequencies than at a frequency of 1000 Hz or greater. This effect is more pronounced for lower-level sounds than for louder sounds. Thus, it makes sense to reduce the sensitivity of the sound level meter, especially in the lower frequencies, so that its readings follow the characteristics of the ear more closely. The sound level meter has provisions to compensate for this human hearing "prejudice" with three weighted network circuits A, B and C which discriminate against the lower frequencies. To insure uniformity among the manufacturers of sound level meters the United States Standards Institute has established a standard to which all meters conform. The sound level meter achieves this response modification by the attenuation of certain low frequency components before display on the indicating meter.

Any sound or noise measurement should thus be identified as to which weighting was used; such as "50 dB(A) or "the A-weighted sound level is 50 dB". Note that the reading is said to be a sound level, not sound pressure level. "Sound Pressure Level" is read only when the sound meter frequency response is flat or uniform over the entire audible range and not weighted by A, B or C settings to fit the characteristics of the human ear. For A, B and C networks the reference level of 0.0002 microbars is implied.

A-weighted readings are used quite widely, and many noise level graphs include an A-weighted scale. Various damage-risk criteria have been proposed to prevent hearing impairment due to noise exposure. Perhaps the most well known criteria comes from (OSHA) standards and is defined by the A-weighted or dB(A) sound level.

TABLE 3. OSHA PERMISSIBLE NOISE EXPOSURES

DURATION (Hours)	SOUND LEVEL (dBA)
8	90
6	92
4	95
3	97
2	100
1.5	102
1	105
0.5	110
0.25 or less	115

Shown below are the actual A, B and C scale weightings in both graphical and tabular form.

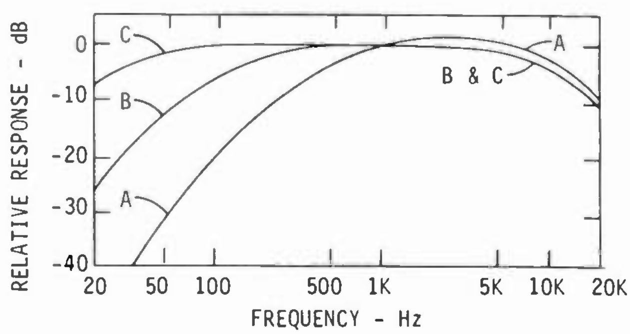

FIGURE 2. FREQUENCY RESPONSE OF A, B AND C NETWORKS

TABLE 4. FREQUENCY RESPONSE OF A, B AND C NETWORKS

WEIGHTING - dB

Hz	63	125	250	500	1000	2000	4000	8000
A-Scale	-26.2	-16.1	-8.6	-3.2	0	+1.2	+1.0	-1.1
B-Scale	-9.3	-4.2	-1.3	-0.3	0	-0.1	-0.7	-2.9
C-Scale	-0.8	-0.2	0	0	0	-0.2	-0.8	-3.0

Table 5 gives general suggestions as to which weighting to use for different sound level ranges.

TABLE 5. USE OF WEIGHTING NETWORKS

SOUND LEVEL RANGE, dB	RECOMMENDED WEIGHTING NETWORK
20 - 55	A
55 - 85	B
85 - 140	C

SPEECH INTERFERENCE LEVELS

At times a noise level environment may exist that is not loud enough to be considered a health threat but could still be a safety hazard. This situation may exist around equipment that produces noise that is loud enough to interfere with critical speech communication. Working around high voltage equipment (Broadcasting Transmitters) during installation, operation or maintenance requires a critical level of communications among co-workers.

Speech sounds are distributed over the frequency range from 100 to 10,000 Hz with most of the intelligence or information in the 200 to 6000 Hz band. If we measure the noise energy in that region only we will have some measure of its ability to interfere with speech.

A three band analysis of the octave bands centered on 500, 1000, and 2000 Hz will permit us to determine the arithmetic average of the sound pressure levels in these three bands. The resulting numerical average, in decibels, is defined at the speech interference level (SIL).

Example: What is the speech interference level of a blower that has the following characteristics at a distance of 3 feet?

Hz	SOUND PRESSURE LEVEL
500	80 dB
1000	72 dB
2000	70 dB

Answer: $\frac{80+72+70}{3} = 74$ dB SIL

Note that we have introduced a new factor to sound level readings, namely distance. In the above blower example it was given that the sound pressure levels were taken at a distance of 3 feet. Although it was not mentioned previously, it was implied that all sound readings are taken at some well defined distance from the point of origin.

During some situations the sound pressure level may be known at a certain distance but needs to be converted to a greater distance from the source. When assuming spherical divergence from a point sound source, pressure drops off as the first power of distance. Thus when dealing with sound pressure levels the difference between these pressure levels at two points can be expressed as:

$$\text{DIFFERENCE} = 20 \log_{10} \frac{(d_1)}{(d_2)} \text{ dB}$$

For example, if distance d_1 is twice d_2, then the difference in sound pressure level is:

$$20 \log_{10} \frac{(2)}{(1)} = 6 \text{ dB}$$

Because we know that the sound is weaker at twice the distance from the source we know the sound pressure level is 6 dB lower. This is the 6 dB-per-distance-doubled statement. Of course this is for pure spherical divergence with no reflections. In practice the falloff of sound pressure level with distance will almost always be less due to reflecting sound waves.

TABLE 6. SPEECH INTERFERENCE LEVELS (dB)

DISTANCE BETWEEN TALKER AND LISTENER (FT)	TALKER'S VOICE EFFORT			
	NORMAL	RAISED	VERY LOUD	SHOUTING
0.5	74	80	86	92
1	68	74	80	86
2	62	68	74	80
4	56	62	68	74
6	52	58	64	70
12	46	52	58	64

Once the speech interference level for a piece of equipment is known, how does it effect speech communications? The two determining components between talker and listener are the talkers voice effort and the distance between talker and listener. This talker's voice effort and talker/listener distance can best be explained in Table 6. This table is based on the 500, 1000, and 2000 Hz bands, and the levels are for average male voices with the speaker and listener facing each other and using unexpected word material.

From review of Table 6 and the blower example, which had a Speech Interference Level (SIL) of 74 dB, the following distance vs. effort matrix would result.

TALKER/LISTENER DISTANCE (FT)	VOICE EFFORT
0.5	Normal
1	Raised
2	Very Loud
4	Shouting
6	---
12	---

But loss of communication is only one of the problems created by noisy equipment. Noise well below the "hazard" level can cause fatigue and errors. Prolonged discussion in an excessively noisy environment can be very tiring; degrading judgement. Studies have shown that the effect of noise on work output depends greatly upon the nature of the work. The task requiring close attention for a long operation cycle is especially vulnerable to noise, with likely resultant higher rates of operator error and product rejects.

NOISE CRITERIA CURVES

Probably the most sophisticated way to measure sound and/or noise is by use of noise criteria curves. The noise criteria curve (shown in Figure 3) has an abscissa of the octave band center frequencies 63, 125, 250, 500, 1000, 2000, 4000, 8000 Hz and an ordinate of sound pressure level in decibels from 10 to 90 dB. In the field of the graph are a series of curved parallel lines labeled from NC20 to NC70. These lines represent "equal-loudness" curves relative to the human ear. The downward slope of these contours reflects both the lower sensitivity of the human ear at low frequencies and the fact that most noises having distributed energy drop off in a similar way with frequency. These curves may be used as a basis for rating the effective loudness of a noise. For example: The NC60 curve shows that a 67 dB level at 250 Hz sounds just as loud as a 58 dB 4000 Hz tone. The beauty of the noise criteria curves is that a spectrum analysis specification is inherent in a single NC number. Considering spectrum shapes of noises, the noise criteria analysis is far superior to using a single wideband noise level reading such as A, B or C weighting.

The noise criteria curves are implemented by plotting the sound pressure level of your noise source at each octave on the graph. The NC rating of that noise producing device would be equal to the point of highest penetration into the NC contours.

Noise criteria (NC) ratings are used quite frequently as design specifications for quietness required in studios, recording and sound reproduction rooms. Below is a list of recommended noise criteria for various types of space. See Table 7.

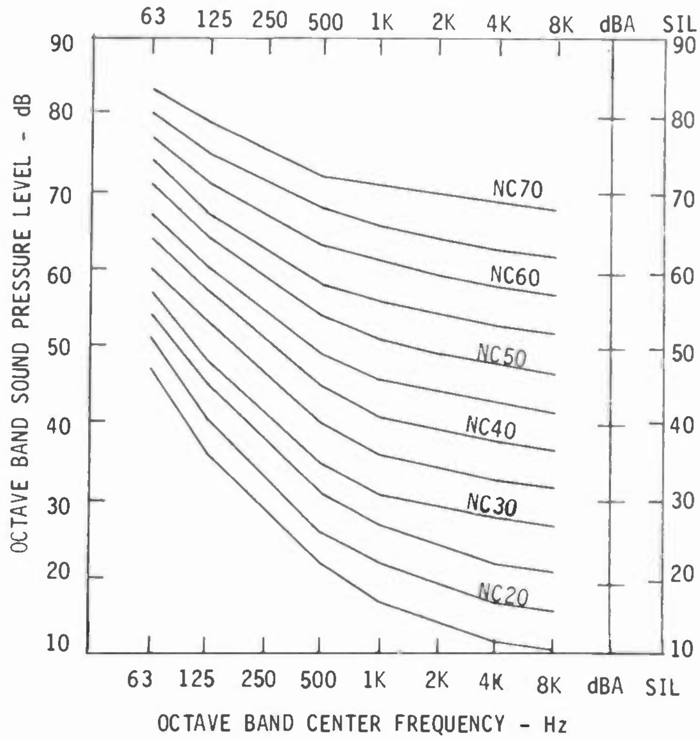

FIGURE 3. NOISE CRITERIA (NC) CURVE

TABLE 7. RECOMMENDED NOISE CRITERIA

MASS COMMUNICATION WITHOUT AMPLIFICATION	
Concert Halls	NC 15-25
Legitimate Theaters	NC 25-30
Conference Rooms	NC 25-35
School Rooms	NC 30-40
Churches and Courtrooms	NC 30-40
MASS COMMUNICATION WITH AMPLIFICATION	
Broadcast Studios	NC 15-20
Assembly Halls	NC 25-30
Motion Picture Theaters	NC 30-35
INDIVIDUAL COMMUNICATION	
Homes, Apartments, and Hotels	NC 25-35
Hospitals and Libraries	NC 30-40
Private Offices	NC 30-40
General Offices	NC 35-45
Restaurants and Department Stores	NC 40-50
Coliseums	NC 50-60
Factories	NC 50-70

NOISE CONTROL

In order to determine whether the noise at a particular listener location will be acceptable according to one of the criteria discussed above, it is necessary to study the transmission path over which the noise will travel from the source to the listener.

There may be several sources of different magnitudes contributing to the noise at any single location and several paths over which these noises may travel to reach the listener.

There are noise control problems that present themselves every day. Each one must be carefully reviewed, analyzed, and hopefully solved. This process may be very simple and straight forward or may take months and months of calculations with tedious trial and error testing.

In summary, the general procedure that should be followed in solving any noise control problem is:

1. Determine the sound pressure levels and directivity factors of all sources of noise by test or from the manufacturers data.

2. Determine the various listening areas that might be affected by the various sources and establish the allowable noise levels at these locations from applicable criteria.

3. Determine the paths by which the noise will travel to the listener and calculate the noise levels that can be expected, taking into account divergence, reflection and absorption.

4. If the expected noise levels exceed what can be tolerated, consider whether means of reducing noise at the source, means of altering the transmission path, or means of otherwise protecting the listener are available or desirable.

Lets look at two examples of noise control techniques used during the design phase of a tape cartridge machine and a 30 kW FM transmitter.

TAPE CARTRIDGE MACHINE

Very early in the engineering effort on the PHASE-TRAK 90 (PT-90) series cart machines the Broadcast Electronics design team knew that this top of the line product had to have the quietest running noise level possible. A design target was set at NC17 at 1 meter. To achieve this goal, every component that would produce noise and every component that could be used to attenuate this noise was thoroughly reviewed.

The major source of noise in a running cart machine is the motor. The two areas of concern are the motor bearings and motor vibration. To assure the quietest long life bearings, the design engineer must carefully select the proper bearing relative to load capacity, reliability, speed limitations, manufacturing tolerances, mechanical clearances, and lubrication. With careful analysis and testing, the perfect combination can be found. Also critical to the design is the control of noise causing vibration. The motor must be specified to maintain a high level of balance and dimensionally true rotation during operation.

Another source of unwanted sound is the solenoid which in this design was outfitted with an adjustable air dampening system that helps greatly to reduce the resultant impact noise when the solenoid is engaged. Attached to the solenoid is the drive cable which is different than the normal chain link system found in most cart machines. This flexible stainless steel cable is much quieter during activation than the rattling links of the earlier chain drive designs.

Another bearing that can be a source of noise is the pressure roller bearing. The use of a non-metallic self lubricating bearing is quieter than the normal sintered bronze bearing. By eliminating the metal-to-metal contact between the pressure roller bearing and pressure roller shaft the offensive clicking noise is reduced during cartridge start-up.

Listed above are a few of the active design specifications used to produce a quiet machine. There are also some passive design steps used to help attenuate the noise produced by the previously mentioned components. These steps can basically be summed up in two words, "solid construction". The use of a 0.5 inch thick aluminum deck plate insures a structural solid platform for the motor. The use of softer, thicker aluminum side panels versus thin resonating steel panels helps contain internal noise. The front die casting is manufactured from zinc which is also a dead soft material. Even the polycarbonate overlay front panel is reinforced with an 0.125 inch thick aluminum back panel to help retard vibration and absorb sound. When steel material is used on the rear modules attention was given to proper folding and mechanical fastening to maintain adequate stiffness and good noise containment. Even the quantity, quality and location of the hardware used for assembly was well thought out to guarantee solid mating of the various sheet metal components.

With all of the design ideas from above and many more subtle additions implemented into the product, the cart machine was ready for noise level testing.

The cart machine was tested in an anechoic chamber located at the laboratories of the Central Institute for the Deaf in St. Louis, Mo. Sound emissions were measured from three model PT90PS machines and two model PT90RPS machines. Sound pressure levels, A-weighted and in octave bands were measured on the axis of symmetry of each machine at a distance of 1 meter from the front panel. The machines were located in the center of the 18' X 18' X 18' chamber with 30 inch sound absorbent wedges on the interior walls.

Sound pressure levels were measured with equipment by Bruel and Kjaer; a Type 2203 Sound Level Meter, equipped with a Type 1613 octave filter set and Type 4165 condenser microphone. Calibration was made with a Larson/Davis Laboratories Type CA250 acoustic calibrator.

Shown in Figure 4 is the resultant noise criteria (NC) curve for the PT-90 playback cart machine and in Figure 5 are the results of the PT-90 record/playback machine test.

As can be observed in Figure 4 the PT-90PS has a NC16 rating because the highest penetration into the equal loudness contours is 16 at 1000 Hz. Likewise, Figure 5 shows that the PT-90RPS machine should be rated at NC12 because of the 12 reading at again 1000 Hz. The "A" weighted reading for the PT-90PS was 17 dB(A) and for the PT-90 RPS it was 15.3 dB(A).

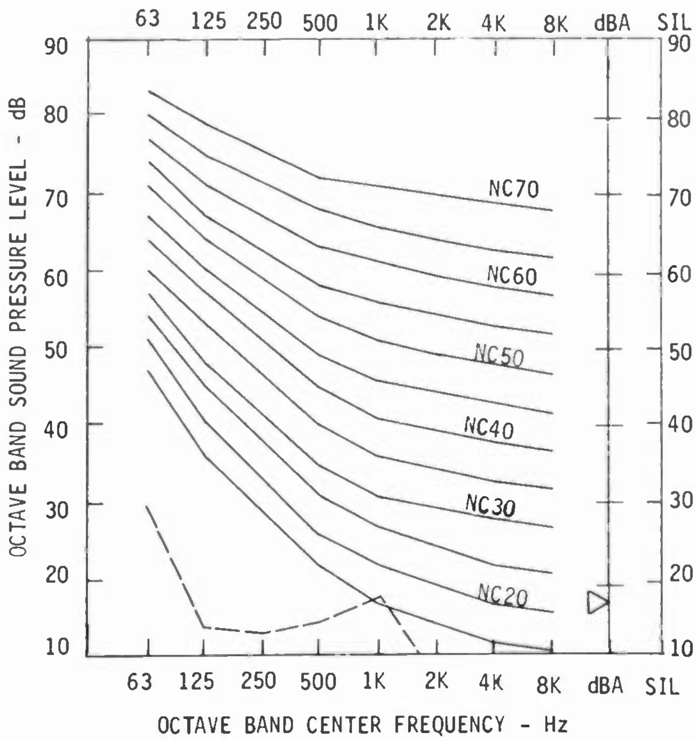

FIGURE 4. NOISE CRITERIA CURVE FOR PT90PS

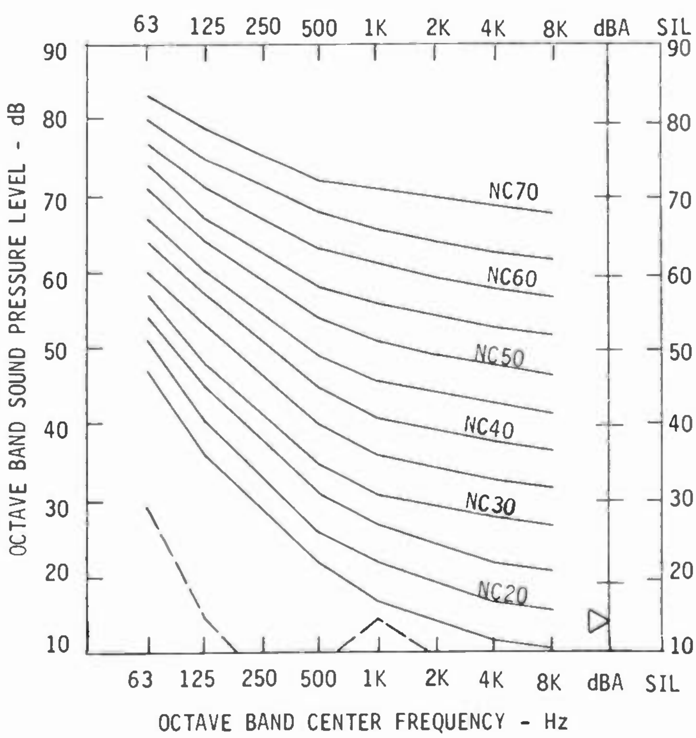

FIGURE 5. NOISE CRITERIA CURVE FOR PT90RPS

30 kW FM TRANSMITTER

A 30 kW FM transmitter requires forced convection cooling of the power tube. This thermal management is accomplished by a properly sized blower. Unfortunately the use of an air moving device of the size required in a large transmitter produces a high noise level. However, proper selection of this blower can reduce the offending noise level significantly.

Experience tells us that the noise sources from a blower are of two types: aerodynamic and mechanical. Aerodynamic noise comes from airflow either on the intake or exhaust side of the blower and vortex shedding that occurs at the impeller blade tip. Mechanical noise is only produced by a blower that has a mechanical deficiency due to bad bearings, rotating unbalance or mechanical interference. A blower with mechanical noise should either be repaired or replaced.

By far the most noise producing area of a blower occurs as vortex shedding at the blade tips of a rotating impeller. The faster the tip velocity of the impeller the more vortex shedding occurs and the noisier the blower. Tip velocity is a function of impeller diameter and impeller RPM. Thus, it is in the designers best interest to use a blower with the slowest tip velocity.

An interesting phenomenon about noise level from blowers is something called blade passing frequency. Each time an impeller blade tip passes a point in the rotational path an impulse is delivered to the air at that point. The frequency of these impulses can be calculated as follows:

Blade Passing Frequency (Hz)=

$$\frac{\text{(Impeller RPM)} \times \text{(No. of Impeller Blades)}}{60}$$

Typically the blade passing frequency is also the predominant tone frequency and the frequency of the highest sound pressure level.

During the cooling system design of this transmitter it was determined that 1200 CFM of air was required against 3 inches of water pressure to properly cool the power amplifier tube. Research of standard blowers revealed two blowers that could satisfy this requirement. Blower "A" used an 8 bladed impeller that was 14 inches in diameter x 3.25 inches wide turning at 3450 RPM. Blower "B" has an 8 bladed impeller that was 16.5 inches in diameter x 5 inches wide rotating at 1725 RPM. Calculations tell us that blower "A" has a tip speed of 210 feet per second (143 MPH) and a blade passing frequency of 460 Hz. Whereas blower "B" has a tip speed of 124 feet per second (85 MPH) and a blade passing frequency of 230 Hz.

Preliminary analysis indicates that blower "B" has the double advantage of slower tip velocity (less noise) and a lower peak noise frequency. This lower peak noise frequency should seem quieter to the human ear due to its lower sensitivity at low frequencies.

Consequently both blowers were tested for noise level and plotted on the noise criteria (NC) curve (see Figure 6).

As predicted the blade passing frequency of each blower was also the peak noise frequency. The noise criteria rating for blower "A" was NC70 and for blower "B" NC60 and the "A" weighted readings were 75 dB(A) and 65 dB(A) respectively. The lower blade passing frequency was advantageous to blower "B" as was expected and showed a 10 dB improvement.

From studies done on human hearing a 10 dB reduction in noise level will "sound" like the noise volume was cut in half. The speech interference levels (SIL) calculate out to 61 dB (SIL) for blower "A" and 44 dB (SIL) for blower "B". Referring back to Table 6 shows that the distance between a talker and listener using normal voice effort could increase from about 2 feet to greater than 12 feet. This is quite a comfortable change when working around a transmitter for any length of time. Thus, blower "B" would be the logical choice for low noise level performance.

Had there existed any sound level data for either one of these blowers, it would have been possible to calculate the sound level of the other blower. This can be accomplished by using the following Fan Law:

$$L_A = L_B + 70 \log_{10} \frac{DA}{DB} + 50 \log_{10} \frac{NA}{NB}$$

Where:

L_A= Loudness of Blower A (dBA)
L_B= Loudness of Blower B (dBA)
D_A= Impeller Diameter of Blower A (In)
D_B= Impeller Diameter of Blower B (In)
N_A= Speed of Impeller of Blower A (RPM)
N_B= Speed of Impeller of Blower B (RPM)

Assume we know that L_B= 65 dB(A), lets solve for L_A using the impeller size and speed given previously.

$$L_B = 65 + 70 \log_{10} \frac{14}{16.5} + 50 \log_{10} \frac{3450}{1725}$$

$$= 65 + (-5) + 15$$

$$= 75 \text{ dB(A)} \quad \underline{\underline{\text{CHECK}}}$$

The two examples above are typical applications of noise measurement and control methods required during the design of modern broadcasting equipment.

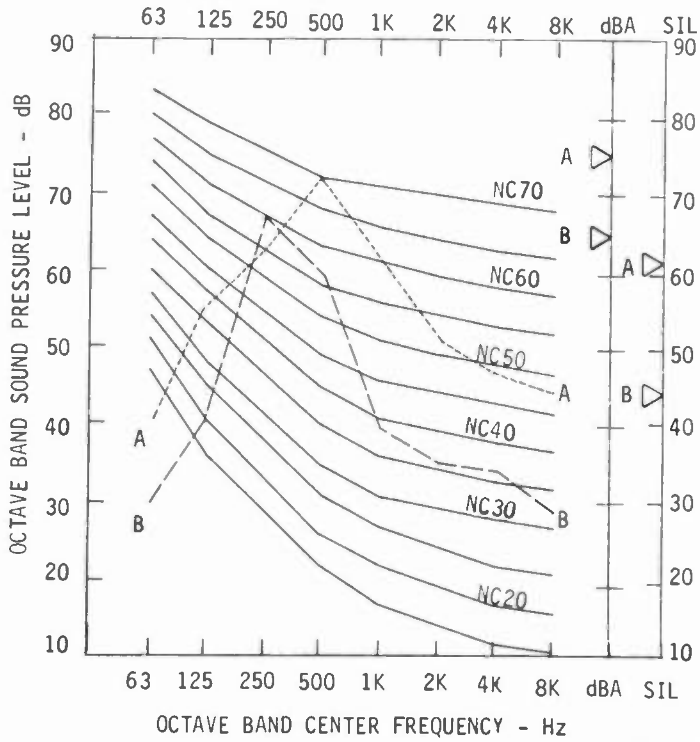

FIGURE 6. NOISE CRITERIA (NC) CURVES FOR BLOWERS A AND B

CONCLUSION

Understanding noise level measurement and control techniques are essential to both the designer and end user of broadcasting equipment. The designer can produce quality, quiet equipment with the proper use of noise control. The broadcaster will now understand the various measurement techniques and ratings when comparing competitive equipment for procurement. Finally the broadcaster and equipment manufacturer will hopefully now have a mutual understanding of the importance of noise measurement and control techniques.

ACKNOWLEDGEMENTS

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The author was selected as one of the top 20 young engineers of the year in 1986 in Illinois, and holds two U.S. patents for electronic and mechanical designs.

He has been employed with Broadcast Electronics since July 1982 as Manager of Mechanical Engineering. He was responsible for the overall mechanical design of the new PT90 Series Cart Machine and Broadcast Electronics complete line of FM transmitters.

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A DIGITAL DYNAMICS PROCESSOR FOR FM BROADCAST

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This paper describes the implementation of DSP technology to provide all the requisite signal processing functions applicable to treatment of musical program and speech for broadcast through a typical FM transmission system.

General outlines of functional signal processing blocks including compressors, expanders, and limiters is presented, as well as a method for detecting the audio signal level using a proprietary algoritm.

A description of a multi-band digital signal processor capable of performing all the classic signal processing functions, and many esoteric and arbitrary functions, is presented, along with an evaluation of the performance of the device used as the final audio signal processor in a typical FM transmission chain.

Future developments in digital broadcast equipment, including the need for enhanced interfacing and synchronization for an all-digital broadcast facility will be discussed.

In November, 1987, Mr. Rick McCollister of Recording & Music Group, an independent consultant for and on behalf of Valley International, Inc., in Nashville, TN, embarked upon an ambitious project to develop an audio dynamics processor capable of performing all the basic dynamic gain altering functions in the digital domain.

Among the objectives of this project were:
1.) Emulation of existing detector, or audio level sensing technology;
2.) Duplication of direct feed-forward processing configurations;
3.) Emulation of existing envelope modification circuitry action by synthesizing attack and release time constants;
4.) Determination of factors affecting audibility of the processing;
5.) Investigation of new and unique types of waveform manipulation possible in the digital domain.
6.) Development and adaptation of the technology for configuring useable product lines.

Criteria for DSP Selection

Of the digital signal processor VLSI chip sets available at the time of the initiation of the project, the one considered most suited for use was the AT&T WEDSP16-75. Among those characteristics deemed desirable were the device's speed, the presence of large (36 bit) accumulators, and an instruction set well matched to implementation of FIR filters and dynamic gain change algorithms. The ability to operate on serial data also enhanced the ease with which the device could be made to interface with all existing standardized digital audio formats.

Of equal importance was the availability of a comprehensive support software library, a DSP development system, and the enthusiastic support of the staff at AT&T, whose aid the authors wish to gratefully acknowledge.

Choice and Implementation of Detector Response

Since mathematical modeling of virtually any converter scheme, e.g., rms-to-dc, peak absolute value, "curve-fitting" envelope detection, etc. is theoretically possible in the digital domain, the designer had a number of options available. Based upon the same research which resulted in the development of the proprietary Valley Linear Integration, the choice was made to model the detector/convertor algorithm after the operation of the Valley circuitry.

The Valley Linear Integration Detection process possesses the unique property of being relatively immune to waveform complexity while maintaining a flat sensitivity response throughout the audio spectrum. The result is a noticeable improvement in operation over rms detection circuitry when applied in dynamics processing devices, particularly compressors and limiters.

During each sampling interval, the processor compares the values of the Left and Right channel data. The higher of the two values is selected and is used in the convertor algorithm. The convertor model then integrates the values by using a constant representing the attack time. The resulting value is applied to an accumulator representing the output of the convertor, then decremented according to a separate constant representing the selected value of the release time.

In this manner, the processor algorithm is made to produce a numerical value which corresponds to the output voltage generated by a traditional analog signal processor's control sidechain, as illustrated by the block diagram in figure 1.

In addition, the convertor algorithm generates both the L-R and L+R content of the signal so that the L-R value may be changed in order to enhance stereo separation.

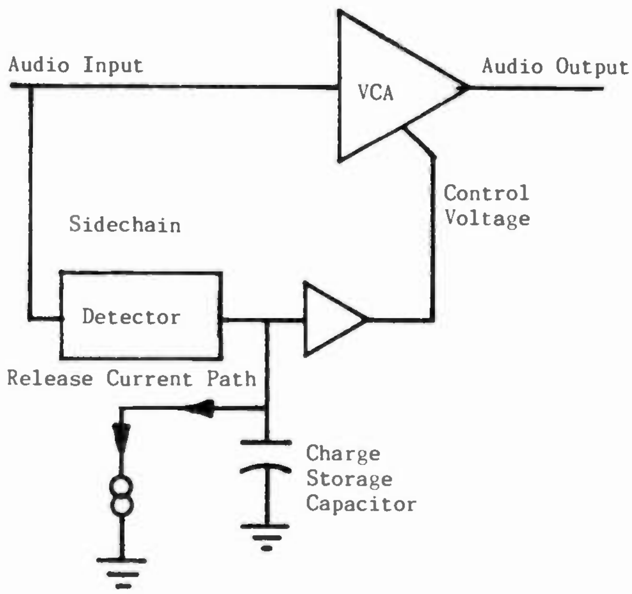

Fig. 1 Analog signal processor

Performing Dynamic Gain Manipulation

In place of the gain control element found in the analog signal processor, the digital processor must implement a multiply/accumulate algorithm which operates directly upon the numerical value of the audio sample on a sample-by-sample basis.

Unlike the voltage produced by the analog detector/convertor circuitry,the numerical output produced by the detector/convertor algorithm bears no direct relationship to the amount of gain alteration required to achieve a specific function, e.g., compression, limiting, or expansion in the digital domain.

The product of the detector/convertor is used to address a gain map residing in memory. Each specific mode of operation has its own unique gain map, thus there exists a separate map for each of the compressor modes and expander modes. Examples are shown in figures 2 and 3.

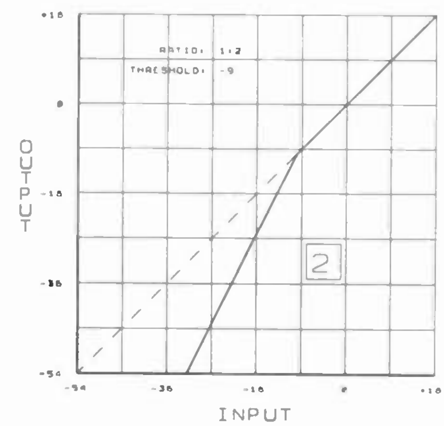

Fig. 2.

Expander Transfer Function

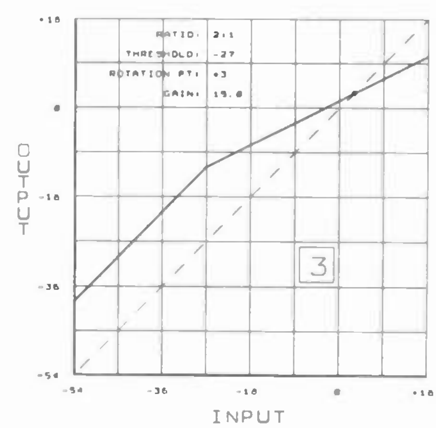

Fig. 3.

Compressor Transfer Function

Note that such parameters as Threshold, Ratio, and Rotation Point are integral parts of the map. These parameters are not directly variable, and must be controlled by manipulating other operations in the processing algorithm.

Listening tests conducted with varying degrees of resolution in the gain maps demonstrated that the size of the gain map, or the number of points on the transfer function used to describe the map, is inversely related to the distortion content of low level signals passing through the processor algorithm. A reasonable degree of resolution was obtained using maps of 16 kwords (32 kbytes). It then became obvious that the number of processor functions available to the operator of the system would be dependent upon the size of the available memory.

In order that memory expansion capability did not become the limiting factor in the usefulness and flexibility of the system, the designer developed a data compression routine which allowed each gain map to be reduced to about 10% of its original size for off-line storage. When a particular gain map is called up for use by the processor algorithm, the compression routine is reversed, and the map is regenerated so that it occupies a full 16 kwords.

Unique Types of Processing

After the audio waveform has been digitized, it exists only as a series of numerical values in the bitstream, and may be used as data in order to perform arbitrary functions. Of the vast number of possible operations which may be performed upon the digitized waveform, very few could be considered desirable. The unique architecture developed for the Valley processor allows for the implementation of any type of manipulation which can be described as a function of input level vs. output level.

In addition, other routines may be imbedded in the processor algorithm. One example is a type of digital "soft clipper" referred to as saturation control. In this process, the value of each sample is examined. If the sample value is near the high or low value extremes of the 16-bit full-scale value, it is used in a special calculation subroutine in such a manner that the waveform is approximately "logged" near its extremes,thus resulting in waveform distortion closely resembling soft clipping.

The limitation on waveform and dynamics manipulation in the digital domain, disregarding audible distortion, may be that of processing speed. If one has unlimited time to perform operations upon the audio data, many highly desirable results are possible, including noise elimination and reconstruction of missing samples.

In reality, no processor designed to operate in the broadcast signal chain can be afforded the luxury of relatively long processing times. The practical limit appears to be about 10 ms of propagation delay before monitoring of the transmitted signal by the on-air personality becomes problematic. The design approach used in the Valley processor results in nominal propagation delays in the 5 ms range, thus easily meeting the 10 ms criterion. In order to meet this requirement, the DSP chips in the processor must operate at a clock rate of about 15 MHz.

Further improvements in processor speed may not deliver the enhancements in performance one might expect; consider that a 25 ns DSP chip is available at the time of this writing, but no readily available, inexpensive external memory exists which could supply instructions to the device rapidly enough to execute a 16 bit X 16 bit multiply/accumulate every 25 ns, thus the device is limited to use of internal (on-chip) ROM and RAM at that speed, adversely affecting its flexibility.

It has long been an article of faith that dynamics manipulation in the digital domain would allow nearly unlimited use of processing without assessing the penalty of audible distortion in the waveform. This is simply not the case.

It is true that artifacts which are generated in analog processors as a result of nonlinearities in gain control elements and impression of control voltage onto the audio signal are absent when equivalent or similar processing is performed on the digital data. Any artifact which is a result of direct manipulation of the audio envelope or waveform, however, is still produced when that manipulation is achieved digitally.

Such dynamic processes as "zero attack time" limiting and clipping create harmonic distortion with components that might normally be out of range of human hearing if performed in the analog domain. In the digital processor, those artifacts become part of the data, and create aliasing by interaction with the sampling frequency.

This phenomenon can be addressed by various means, including high oversampling rates, allowing the processing to occur only within a narrow passband, etc. Further study of these artifacts and their control in real-time digital dynamics control is currently being planned.

A Multi-band Digital Dynamics Processor

The result of the research and development project to-date is the Valley DDP. Although the actual progression of design decisions which affected the final configuration of the product are beyond the scope of this paper, we shall briefly discuss the more important considerations and address a few topics of continued interest in the on-going development of the DDP and related devices.

The DDP is configured as a multi-band processor in order that it may alter the energy distribution in the processed program, allowing the user to create an unique on-air sound by altering the parameters of each band of frequencies independently.

The use of digital Finite Impulse Response (FIR) filters throughout the device assures phase integrity of the processed signal, and provides a relatively simple way to take the signal apart by splitting it into bands, then to reconstruct it without error. There are, however, some trade-offs associated with the use of the FIR algorithms. Lack of sharpness in the low frequency passband skirt is a problem, as is the inability to implement gain control within the FIR filter proper. Both these shortcomings relate to the peculiarities of the FIR algorithm, with its requirement for current waveform history and need to "look ahead" in the bitstream for upcoming data.

Each band processor card is a stereo device consisting of a DSP chip assigned to the FIR bandpass filter function, a DSP chip assigned to the dynamics processing function, sufficient memory to operate the DSP chips, and those peripheral communications and logic circuits necessary to support and coordinate each function. A diagram of a band processor equivalent signal flow is shown in figure 4.

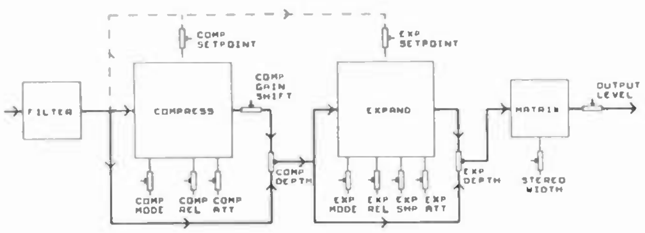

Fig. 4. Band Processor Signal Flow

Comprehensive metering is provided for each band processor, displaying information such as input level, band compression, band expansion, setpoint, and output level. In addition, the metering LED arrays are used to indicate the relative values of all the adjustable parameters in the processor algorithm, such as attack and release time.

The products of all the band processors are communicated to an output processor, which not only recombines all the passbands, but is also capable of performing further dynamic control over the broadband signal. Among the functions of the output processor are instantaneous peak level control and implementation of both the 75 us preemphasis, and the 15 kHz lowpass filter. The use of an FIR algorithm as the final lowpass allows the skirt response to be very steep, typically down 70 dB at 15,250 Hz, while maintaining linear phase response.

The output processor is also used to implement such specialized functions as high frequency limiting, saturation control, and the final formatting of the digital output into stereo or L+R and L-R.

The DDP mainframe contains up to eight band processors, a single output processor, and a system controller card, by means of which the operator configures and adjusts the device.

The system controller is a complete microprocessor-based computer optimized for control and communication with the DSP chip sets used in the band processors and output processor. The system controller also contains a set of interactive prompts and a menu-driven operating system which uses an LCD display to present information to the operator, and can be connected to a separate terminal or modem via an on-board RS-232 port.

The DDP accepts 16-bit linear PCM at a 50 kHz maximum sampling rate. Word clock and bit clock signals must be provided. For interfacing to analog signals, a companion unit, the DDP Analog Interface, is available. It includes the A/D and D/A convertors and provides all the necessary timing signals to source the DDP.

Results of On-Air Testing

The use of the DDP as the final processor both in an experimental transmission chain and in various Beta test sites has yielded promising results.

When applied to easy listening, light rock and classical formats, the advantages of digital processing have been quite obvious. When called upon to perform moderate amounts of compression and expansion using low ratios and relatively slow attack and release times, the DDP enhances fine details in recorded material, and provides a startling increase in loudness without adversely affecting clarity. Although any evaluation of the results must, unfortunately, be largely subjective, we feel safe in stating that the absence of artifacts generated in the analog signal paths of traditional processors allows digital compression to preserve subtle details in the processed material which become masked by analog processing artifacts.

Application of the DDP, in its current state of development, to high-energy rock formats has resulted in the subjective observation by some users at Beta sites that the device has a "busy" sound when performing large amounts (more than 12 dB per band) of aggressive processing with relatively short attack and release times. The consensus seems to be that the device will impart a great deal of loudness, but that the artifacts generated in the process become audible in the high frequency portion of the output. We shall not dispute these opinions, although our testing program using the DDP in an experimental low-power FM transmission chain, with a popular processor for comparison, demonstrated to our satisfaction that the DDP imparts comparable loudness while maintaining full frequency response and providing a significant increase in fidelity.

Although we endeavored to supply thorough, comprehensive instructions with the Beta units, and provided what we considered to be appropriate factory presets from which starting point the operator could adjust the 70-odd variable parameters, at that point in time we, in fact, had little more experience in actual operation of the device than did the engineers at the Beta sites. We are, frankly, unsure to what extent set-up procedure and/or inappropriate adjustment of parameters such as time constants affected the results of the Beta tests. We have been able to duplicate the "busy-ness" phenomenon, as well as we can understand it, and we feel it is a minor problem which can be corrected relatively easily.

At the time of this writing, there is at least one Beta unit in the field at all times, and data collection continues.

We remain quite confident that further refinements in the processing configuration will result in the ability to produce a louder and cleaner signal than can be achieved using currently available analog processing.

The Future of Digital Signal Processing

Given the flexibility of digital recording and production systems, and the premise that the cost of the technology is still decreasing, the advent of the all digital station is virtually assured. We feel strongly that the availability of more cost-efficient conversion packages (A/D, D/A) and the increasing popularity of the CD and DAT formats will soon initiate the trend.

A digital stereo generator is already in the first stages of development, and a moderately priced broadband digital compressor and expander unit, potentially suited for use as a microphone channel signal processor, has been introduced by Valley, International at the European AES Exhibition. The few tasks remaining are to provide a useable "master sync" system for radio broadcast facilities, inexpensive off-line storage, and affordable timebase correction for each program source. It is likely that these devices, if not already available, will soon exist.

OPERATIONAL FEATURES AND USER INTERFACE CONSIDERATIONS OF A RAM-BASED DIGITAL AUDIO WORKSTATION

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Abstract

AKG engineers saw an opportunity to improve the production methods used by broadcast engineers to make short recordings, such as commercials, through the application of RAM-based digital audio recording and editing. They interviewed production engineers at leading AM and FM radio stations, observed them at work, and analyzed their working methods and studio environments. As concepts began to crystallize, a preliminary Owner's Manual was prepared to serve as a design specification for a new product.

The product that has evolved uses modern digital audio processing, but is based on current working practice and has a familiar user interface. The resulting design is easy to use and will raise the quality and efficiency of the broadcast production studio.

Introduction

The arrival of new technology in any industry is usually an occasion for both rejoicing and suffering. Along with the genuinely useful and powerful advantages that it provides, new technology often exacts a painful price in terms of time lost while staff members learn how to use the new equipment. Sometimes new products are unnecessarily hard to use because ther designers, caught up in their own world of technology, have ignored the current working methods of their customers and have built awkward user interfaces. This tends to be especially true of products based on computer technology. Designers must anchor their products to the best aspects of current working practice, while they take strong advantage of today's powerful new technology.

The Typical Production Studio

Production studios are often in relatively small rooms, sometimes doubling as an on-air room. A centrally located mixing console is usually flanked by two turntables and racks holding auxiliary processors, cart machines, and a CD player. Directly behind the console and often suspended from the wall are a pair of monitor speakers. There are usually at least two analog open reel recorders, usually in their own free-standing consoles. A boom announcer's microphone hangs over the console.

Common Short Broadcast Productions

A production engineer is typically faced with a variety of projects each week. They might include:

Creating a concert commercial
Customizing the header, tail, or donut of an agency spot
Preparing a jingle for a local client
Creating a special occasion song by vocal replacement
Creating a public service announcement

Example: Creating a Concert Commercial

As an example, let's examine the production of a commercial for an upcoming rock concert at a local stadium.

Preparation. After reviewing the necessary details of the event, the production engineer prepares a script (if one doesn't already exist) for himself or another announcer. Next, he picks CDs or LPs by the artist from the station library and selects a few hits by the artist that will be instantly recognized by the target audience. He plans the order in which he'll use portions of the songs and how he will interweave the announcer and the songs into a fast moving, attention-getting commercial.

Recording. The next steps depend upon the complement of tape recorders in the production studio. In this case, we'll consider a studio with one 2-track and one 4-track machine. Cueing the 4-track to just beyond the end of the last production, the engineer enables tracks 1 and 2, cues the first song, and begins recording the stereo music bed. When enough of the first song has been laid down, he punches out of record and backs up the tape. Having cued the second song, he rolls the tape from within the first song and at the right moment (hopefully) starts the CD player and punches into record on the stereo music bed tracks. If he punches in early, or is off in starting the CD, he may have to begin over again and re-record the first song. If the transition from the first song to the second song was satisfactory, he will record as much material as he needs. Dropping out of record, he backs up the tape, sets record enable on track 3, and prepares to record the announcer.

Listening to the rolling music bed on tracks 1 and 2, the announcer will read the first part of the script, with correct timing and delivery, over the first two song excerpts. The announcer will continue reading until the first break in the script. At this point, the next one or two song excerpts will be cued up and recorded as before. The result will be a more or less continuous music bed, consisting of excerpts from up to seven or eight songs, running to the exact end of a 30- or 60-second spot. The rest of the announcer's script will be laid down, perhaps alternating between track 3 and track 4 to give more freedom in re-recording sections where the timing or delivery is off. Note that, due to the limited number of tracks, no such track leap-frogging can be done for the music bed sections.

Razor Blade Editing. In this scenario, it's pretty unlikely that the engineer will ever touch splicing equipment. This is because any editing that he might want to do would most likely involve only the announcer track or only the music bed tracks. Since they're both on the same physical tape, he really can't do any editing. Of course, he could have recorded the announcer on the 2-track, and edited on that tape. He could then transfer the edited 2-track material to the 4-track and add the music bed to the announcer track. But as soon as he's done this, the opportunity for editing the new material, or adjusting its timing relative to the music bed, is gone because now the tracks are again on one piece of tape.

Mixdown and Effects. Mixdown in such a production is quite straightforward. A good engineer controls levels and balance tightly as he records, so there's not much to fix-up during the mix. In some cases, a favored effects device may be used on the announcer. Cueing the 2-track to the end of the previous production, he prepares to run off a final stereo mix and does so, occasionally trimming the level and fading out rapidly at the end of the spot. He may repeat this several times making cart copies for on-air use, or the carts may be made from the 2-track master.

Comment and Analysis

While our example is typical, there is a lot of room for variation in technique among different engineers. There are also other types of productions that will lead to a different series of operations, some simpler, some more complex. Nevertheless, the following aspects of the production process are very common:

1. The production engineer is working fast. Our example project might easily be done in 30 minutes. Any operation that carries a high risk of failure (such as splicing or punching in to existing tracks) is not tolerated because the engineer can't afford the time to re-do things.

2. The production engineer is an artist. He is crafting a very short production that must have coherence, excitement, and its own tight, accurate rhythm. Because the engineer has limited tools, he

must bring a great deal of his emotional and physical energy to bear during the production. Watching an engineer working on one of these spots, one sees strong, rhythmic motion as tape recorders are started, turntables are cued and released, record buttons are pressed for punch-ins, and faders are moved or slammed up and down. The process uses somewhat unwieldy tools, but it is fast and it works.

3. Most errors of timing or content are fixed by multiple retakes. The engineer must re-record some sections many times.

4. The engineer gets almost all his working cues and information by listening. Visually, he is limited to reading tape counter values from the tape machine and watching for wax pencil marks slipping across the repro head. The equipment doesn't allow him to locate particular sound events by eye, but he is uncannily good at finding them by ear. Winding the tape forward or back against the heads, he can locate the right section by the high-pitched squeal of audio at 8 times the normal pitch and speed. Stopping the tape and grabbing the two reels, he can rock the tape back and forth until, from the almost inaudible growling, he has located a cue with an accuracy that is entirely adequate for the work at hand.

5. At the beginning, and indeed during the whole production process, the potential creation exists in two forms. One is the mental image of it that the engineer holds in his head, and the other is the developing recording arrayed across the invisible tracks and tightly coiled turns of the tape on the recorder. Due to skill and experience, the engineer <u>will</u> end up with a coherent product, but there is nothing in the visual field that will help to achieve that goal.

6. There are never enough tracks. In stations with a pair of 2-tracks, the engineers long for a 4-track; in stations with a 4-track, the engineers can tell you excellent reasons why they need an 8-track. They know that they could do more complex, more compelling projects and in less time with more tracks.

Opportunities for Improvement

While this working method has many strengths, there are some areas that definitely could be improved.

1. Too many operations will potentially fail because they damage material already recorded. Punching in and out and razor blade editing are powerful techniques, but they get used sparingly because of the difficulty of fixing up errors. An "undo" feature similar to that found in word processors would give these techniques their full range of usefulness by eliminating the risk.

2. It would be extremely helpful if one could adjust the timing between events on different tracks. No editor can take a razor blade, excise track 3 from 0:15 to 0:30 and slide it downstream by 1 second to correct a timing problem. The possibility of gaining this capability always produces smiles on the faces of production engineers.

3. 10-1/2" reels are big and slow to move. It's just plain tedious waiting for the transport to get you to where you want to be. A machine that responded immediately would be able to keep up with the engineer, not impose its delays on him.

4. Exact overall duration is essential in most short productions. The only way to keep track of the length of a production as it develops is to zero the counter at the beginning, run the tape forward to the end of the last material recorded, and read the counter. It would give the engineer greater control to be able to see the current duration at all times.

5. Production equipment is typically spread out between several tape recorders, mixing console, and other

gear that must be used. A more focused work environment would be preferable and would permit faster work.

6. Tape recorders must be cleaned, adjusted, and repaired. They involve many mechanical components that wear out. A system with fewer moving parts would increase reliability.

7. Good ergonomics require that, where possible, complex equipment should stimulate more than one of the operator's senses. Present equipment, as we have seen, only acts upon the engineer's hearing. A system that gives the engineer a good visual representation of his project and his tools would allow him to work with more confidence and more insight.

Key Design Goals for a Digital Sound Editor

After digesting our experience, the following is clear:

1. The user interface is tremendously important. It has to be clear, familiar, and uncluttered. All controls have to be of adequate size and heft, and must be rugged.

2. The product has to be fast and responsive.

3. The product needs a powerful video display in order to better link the operator to the work in progress.

4. While a computer is essential to the product, it shouldn't be allowed to get in the way of the true goals of the system. A keyboard and/or mouse are not adequate user controls for a unit intended to replace tape recorders and a mixer.

5. The product has not only to improve upon existing capabilities; it must also provide new ones. Furthermore, it has to be an open system that can grow over the years.

6. A system like this must be easy to learn, intuitive to operate, and should meet the user more than half way. It must build upon his existing skill base, and not require him to abandon his present skills.

Writing the Owner's Manual. It was decided to write the Owner's Manual before designing the product-- a reversal of the usual process. The evolving manual became a specification for the product: one that could be read, circulated to broadcast engineers, and refined as more insight was gained.

Implementation of a Digital Sound Editor

Physical Configuration. The Digital Sound Editor components are installed in a work stand whose size is compatible with a typical console-mounted broadcast recorder. The operator sits at a small work surface containing the controller and faces a 14" EGA color monitor. Under the EGA monitor is a small housing with two powered monitor speakers providing a stereo near field monitor. Under the work surface itself, and behind a sound-proofed "modesty panel," is a PC-AT compatible computer. Finally, below the controller, and just above the operator's knees, is a pull-out drawer with a conventional computer keyboard. In addition to soundproofing in the stand, the computer has been given a special power supply with a low noise fan.

The controller is the user's primary active interface to the system. It features 10 long throw faders that, under software control, realize a simple but versatile mixer. All recording, mixing, panning, and effects handling is done here.

The controller also implements the functions of the virtual multi-track recorder via an array of robust illuminated push buttons. Tape motion may also be controlled by a large rotary knob for scrubbing the "tape" across the "heads."

The PC-AT compatible computer houses the rest of the system's custom hardware. A high speed digital signal processor card handles DRAM memory management and all digital domain signal processing. This card is coupled to the PC-AT's system bus, allowing the PC to control the DSP Card, and permitting audio data transfer to and from the host's disc drives. Another bus allows for 16 bit audio data transfers to and from memory, and provides addressing for up to 268M words of audio data. The Memory Cards can each hold 128 DRAM's of either 1M or 4M size, and the system can support up to four Memory Cards. In addition, a second DSP Card can be installed to increase audio processing power.

The DSP Card also connects to one or more input/output modules. An input/output module for analog signals provides two inputs and four outputs, allowing for stereo inputs or effects returns and stereo outputs plus two effects sends. Input/output modules plug into half-height drive module spaces on the front of the PC's vertical case. An AES/EBU digital audio input/output module is planned as well, and other modules will be developed in the future.

The vertical PC case has space for up to six half-height devices, all of them accessible from the front. One will be taken for a floppy drive and another for the basic system hard disc. The remaining four can be allocated to other disc drives and input/output modules.

The system motherboard can accommodate a total of eight cards. The PC requires three slots, leaving slots for up to five Memory and DSP Cards.

The PC itself is capable of running user software as well, under DOS 3.3. All compatible software, such as word processors, databases, and spreadsheets can be used in the unit when it is not active as a sound editor.

Custom Controller Layout and Operation. The controller surface is divided into three sections, two of which will be very familiar to a typical engineer. The left half of the surface is a simple ten-input mixer. The right half of the surface contains tape motion control and location buttons. Above the tape location area is a small set of buttons dedicated to editing and menu control.

The mixer section has been configured for simplicity and ease of use. Two closely spaced faders on the left are dedicated to input monitor control. Each of these two has a single button above it which is used to mute the input signals entirely.

The other eight long-throw faders are used for controlling the multi-track playback signals. Remember that no audio signal passes through these faders. Instead they are "remote controls" for digital signal processing which occurs on the AKG add-in boards. Each of the eight faders has two buttons above it. The lower button is also a mute, while the upper one serves as a record enable function for that track. Though none of the buttons latch, each can be backlit by an easily replaced incandescent bulb which is switched on and off by the computer. These buttons also serve a purpose during editing which we will discuss later.

The tape motion and location controls will be familiar in appearance and operation, if not effect, to most engineers. Play and Stop do exactly what they say, although because the "tape" is digital, there is no mechanical start-up or slow-down time; response is instantaneous. Rewind and Fast-Forward can optionally behave exactly as with an analog tape machine, starting off slow, building up speed, and all the while playing the appropriate pitch shifted audio on all eight tracks. On the other hand, if the engineer releases the Cue button, Fast-Forward and Rewind become instantaneous tape movement functions, capable of stepping through a long production with just a few taps of the button.

Tape location functions are also instantaneous, regardless of the distance covered. There are two buttons dedicated to moving the tape to the "head" or to the "tail" of the production. There are also two user locations which may be set up with a single keystroke. Finally, for the common situation of fixing a faulty punch-in, there is a button which will return the tape to the last Record punch-in point.

Visual Feedback. None of the functions of the DSE would be as valuable or as easy to use if there was not some kind of visual feedback to confirm and reinforce their operation. One of the most fundamental kinds of feedback has already been discussed, i.e. the backlighting behind each button when that button is active.

Another fundamental piece of visual feedback is the tape counter or tape time indicator. For distance viewing, as from across the studio, the remote control surface contains a large, bright red LED display which shows the current tape position in minutes and seconds. The microprocessor on board the controller can also display diagnostic messages on the LED for ease of maintenance and repair.

Tape time is also shown on the color video display along with several other time indications. The time locations of the two user-programmable location points are shown, as well as the duration of the production and the current editing points, if any.

The color video screen is a rich source of other information as well. Besides the input/output metering which is a standard part of any recording system, several new visual structures have been created. The upper half of the screen contains what we call the "track envelope display." This gives a ten-second past and future visual perspective on the audio content of each track. It is not a sampling representation as one might think upon first view, but rather a peak level indication of the running amplitude of each track. It is presented in great enough detail to be able to discern individual syllables in a word or single notes in a musical instrument track. The display pans sideways underneath a fixed cursor in real time and is even able to follow the tape in fast wind modes.

This track display is not intended to supplant, but rather to supplement the listening skills of the engineer with a visual reinforcement of the audio. The track display is also able to give a new perspective on audio signals which are just about to occur, making the precise timing of cues, punch-ins and edits considerably easier than ever before.

Directly under the track display is a structure we call the "production overview." In contrast to the track display, the production overview gives a global perspective on the production, showing the entire duration of the piece in a compressed view. Like the track display, there is a horizontal strip which represents each track, but here the track is painted only in places where it has been recorded. During unrecorded spots on the track, the production overview shows that track as blank. The production overview is a useful aid during Fast-Forward, Rewind and other locate operations, because it gives the engineer positive feedback about his current location in the production.

External Connections. In addition to the analog and digital audio input and output XLR plugs the DSE has a number of other connections to the outside world. Chief among these is the external control/sensing connector which is conveniently placed on the back of the remote controller.

The connector is a 37 pin D-type, and it contains the following: six individual opto-isolated input channels which may be used for on-off kinds of input and six individual normally-open, relay outputs. These are also fully isolated from the system, in order to avoid ground-loops. Both the inputs and the outputs may be configured in system software providing both the ability to control the DSE from a remote location, and the ability to control other devices from the DSE. These inputs and outputs make it possible to integrate the DSE into an existing production studio in the same way as one would integrate other recording and playback machines.

The DSE can be configured with computer inputs and outputs to provide interfaces to printers, modems and other computers. These connections may also be used in conjunction with interfaces to SMPTE and MIDI. In short, the DSE may be fitted with the appropriate audio and data connections to make it the centerpiece of any small production studio.

Editing with the DSE

The DSE editing facilities are layered to provide easy learning for the new user and plenty of power and speed for the more experienced user. Although editing functions are mostly contained in video screen menus, there are buttons dedicated to the most important and most often used functions.

The only new idea which a first-time user must grasp is that editing is no longer confined to the whole width of the tape. Because all the audio information is represented digitally in DRAM memory, each track in the production is essentially on its own piece of tape. Each track may be freely cut, copied, spliced and rearranged by itself or in conjunction with other tracks. Tracks may be moved in time relative to each other. Stereo tracks may be moved in time-locked pairs relative to the rest of the production.

Once this concept is fully digested, the rest of the editing process is simple. The user selects which tracks will be affected with the buttons above the faders. Edit points can be marked with buttons dedicated to that purpose. Edit locations can be easily located with the "reel-rocking" method by using the scrub wheel on the controller. The engineer may audition the edit points with a single keystroke.

The desired edit operation may be selected from a menu of simply labelled selections like "cut" and "copy." The arrow keys on the remote assist with this process. Finally the edit is carried out by pressing "execute" on the controller.

This is not the end of the story, however. What if something has gone wrong, and the edit is bad, or the timing is off, or one of many other possible mishaps has occured? A single touch on the "undo" key will restore the production to its previous state. Undo works with EVERY operation that affects the audio, including punch-in recording and bouncing. Furthermore, undo may be toggled in order to compare the edited and un-edited states. The undo function makes it possible to attempt flashy or tricky edits without fear of losing time if the edit doesn't work out.

Example: Concert Commercial Using DSE

Let's go back and re-record the example given above, using techniques made possible by the DSE. We will assume that the operator is familiar with the DSE, but not an expert or veteran user.

Preparation. Preparation for the concert commercial is much the same. A script must be prepared for the announcer, and the production engineer must have selected music and prepared a mental image of the flow of the piece. Of course, with the DSE computer right at hand, the engineer who is familiar with a word processor might choose to prepare the script and an outline on the DSE. The engineer could also choose to keep an on-line log of the pieces on which he has been working.

Recording. Recording a music bed from a series of LPs and CDs is considerably simpler on the DSE because during the initial recording the engineer does not need to pay careful attention to timing or duration. It is no longer important to cue the LP or CD to exactly the right spot because the excerpt can be quickly trimmed once recorded on the DSE. Likewise, by recording the excerpts sequentially on alternate pairs of DSE tracks the engineer can precisely slip the timing of the different parts relative to each other after they have been layed in.

The announcer can also finish his job earlier. It is a simple matter to mix and match different phrases once they are recorded into the DSE. Another easy task is slimming the duration of that perfect take that came out two seconds too long.

Editing. Editing is now an option which the engineer can choose to make a more sophisticated or snappy production. First of all, editing with the DSE's electronic tools is many times faster than editing with grease pen and splicing tape. More importantly, there is never a danger of destroying or marring a part of the production because of a faulty edit. The engineer can make any edit, no matter how tricky, safe in the knowledge that the edit can be quickly and perfectly undone with a single keystroke.

Mixdown and Effects. Mixdown on the DSE is a straightforward process which is nearly identical to existing techniques. The DSE controller's mix facility allows precise control over track level, pan and two effects sends. The mixer output may be recorded by an external analog or digital recording device, or it may be recorded entirely within the digital domain to the DSE's own audio memory.

The engineer has one tool for mixing that traditional analog methods can never give. That is the ability to anticipate cues visually using the color track display that the DSE provides on its video screen. Using this display the engineer can actually see 10 seconds into the future of the production and anticipate upcoming sounds and appropriate fader movements.

Upgrades and Expansions

The future course of a product is always hard to predict because it depends so much upon changes in the marketplace, competition, user perception and satisfaction and many other factors. AKG engineers have expended considerable effort to ensure that the DSE can grow in useful directions.

The computer itself, a PC-AT compatible, is one of a family of personal computers which represent the largest installed base of hardware and software in the world. This implies a virtual guarantee that parts, repair facilities, and additional hardware and software packages will be available for many years to come. The design of the computer permits easy delivery of new software via floppy disks, and integration of new hardware via the plug-in card slots.

AKG currently provides several hardware and software options for the machine. One may choose between analog audio interfaces which perform any of the most popular sampling rates, and a digital interface which conforms to the AES/EBU digital audio transfer standard. AKG is also strongly committed to providing access to data interface standards such as SMPTE and MIDI.

The DSE system can utilize from one to four memory cards, with a choice of memory component size (as soon as four megabit DRAMs are available). This allows the possibility of budgeting a gradual expansion of the DSE over a period of several years.

AKG has also made the provision for the system to contain more than one signal processing card. An additional card may be added to perform other signal processing tasks such as reverberation and other effects. Of course, hardware upgrades such as additional signal processing cards also require software to control and utilize them. AKG is committed to providing software upgrades and enhancements both for audio signal processing and for control features such as fader automation.

The DSE has enormous mass storage requirements, both for temporary storage of productions and for permanent archiving. Every DSE system comes configured with a sizeable winchester hard disk, which may be used for temporary storage of one or two productions. AKG engineers are also closely examining a variety of newly-emerging mass storage technologies such as read/write magneto-optical disks and R/DAT data tape. One of these will be chosen as the primary permanent digital archiving medium for the DSE.

Sonic Characteristics and Specifications

The DSE is a multi-track recording and playback device which is capable of simultaneously recording 2-

4 tracks and playing back 8 tracks mixed into 4-8 outputs, depending upon analog and digital audio interface options selected. Total recording time varies from 4.4 minutes up to 70 minutes, depending upon memory option selected.

Sampling rate may be configured at the factory as 32 kHz (15 kHz bandwidth), or as selectable 44.1 kHz or 48 kHz (20 kHz bandwidth). Sample word size is 16 bits and sampling technique is linear PCM (like CDs). Analog sampling interfaces use four times oversampling and digital filtering as well as digital dither. Stereo inputs are time aligned, as are all outputs. Analog inputs and outputs are differentially, actively-balanced.

Intermediate digital signal processing results are kept to a precision of 32 bits. This provides important sub-mix headroom. Analog-style saturation logic is used in all digital mixing nodes to avoid harsh digital overload artifacts. Six peak level readings are derived from the digital mixing process for precise metering.

ANALOG AND DIGITAL TECHNOLOGY FOR AUDIO PROCESSING

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Abstract:

Because of its ability to accurately reproduce, digital audio is becoming more popular in the broadcast arena. However, in the field of audio processing, more needs to be accomplished than just accurate reproduction. Modification to frequency response, amplitude response or both is desirable in a broadcast audio processor. This paper examines some of the similarities and differences that can be expected as digital audio processing techniques are substituted for analog.

Digital technology and audio signals meet in many areas of communications. Examples of this are satellite links, telephone equipment, and mass storage devices. Most familiar are Compact Disk and RDAT. These applications are only the first teetering steps in the newborn field of digital audio. In the above mentioned Compact disk and RDAT, when analog is converted to digital, the signal becomes more robust. This is done so that the undesirable effects of the transporting medium, whether it be radio link, tape or disk, do not degrade the quality of the audio. In these examples the audio is not intentionally modified while in the digital domain since the goal is accurate reproduction.

In audio processing, however, the story is quite different. For example, filtering of the audio, whether it be low pass, high pass, band pass, or stop band, is a commonplace occurrence. There are filters that limit the spectrum occupied by the broadcaster, filters that pre-emphasize and de-emphasize, and filters that split audio into multiple bands for compression and limiting. Filtering is one of the basic tools used for audio processing. Since filters are so basic to the functions of audio processing it would profit us to understand the characteristics of digital filters and how they are similar and dissimilar to analog filters.

A look at high performance filters in both analog and digital topologies will show what differences in complexity and performance are observed. For the high performance example, an examination of the anti-aliasing filter suggested for the Multichannel Television Sound (MTS) system will provide a good case

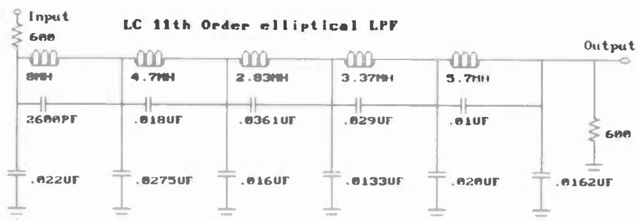

Fig.1. *Passive LC realization of MTS 15kHz LPF.*

study. As originally suggested, (Zenith p.29) this filter is an C11-20-73 type. This specifies an 11th order Cauer (elliptical) that cuts off at 15kHz and reaches its rated attenuation of 65dB at 15685Hz. The passband ripple is 0.177dB. A first approximation analog implementation is shown in figure 1, the frequency and group delay response is shown in figure 2.

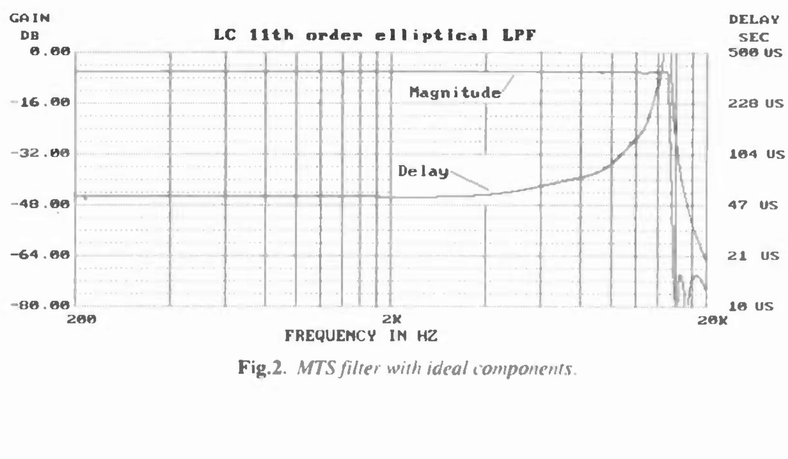

Fig.2. *MTS filter with ideal components.*

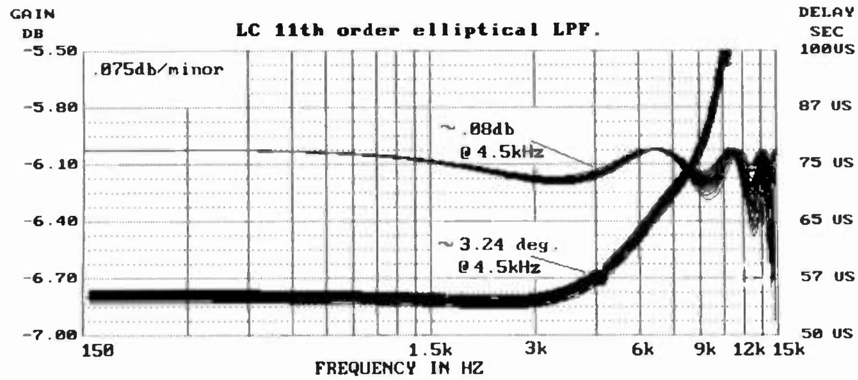

Fig.3. *MTS filter w/random 2% component changes.*

Unfortunately, even with very high quality real world components, the response shown in figure 2 is almost impossible to achieve due, usually, to inadequate Q of the inductors. Generally an additional equalizer[1] is needed to correct for the droop in passband response caused by insufficent inductor Q. The 11th order Cauer filter could alternately be realized as an active FDNR (Frequency Dependent Negative Resistor) filter. Using this topology, a filter could be constructed that would give good results for less money, if components are carefully chosen.

Since this paper is interested in the relative merits of digital versus analog technology and not on the specifics of design, we will use the passive version and assume that the passive version has components that have Q's adequate for the application. We will be dealing with computer simulations of both analog and digital filters. One of the big problems of analog filters is that even if the quality of the components are within limits, the value may not be. Even if by some fluke of nature, or by careful quality control, components close to design value are found, usually time and temperature will soon take care of that. Figure 3 shows what happens to the response of figure 2 after thirty sweeps of a worst case analysis program. In each sweep the filter components were randomly changed by plus or minus two percent of their correct values.

$$\text{Separation} := 20 \log \left[\frac{(\cos(\theta) + A)^2 + (\sin(\theta))^2}{(\cos(\theta) - A)^2 + (\sin(\theta))^2} \right]^{\frac{1}{2}} \qquad \text{Separation} = 30.857$$

Equ. 1a

$$\frac{10^{\frac{B}{20}}}{10^{\frac{C}{20}}} = 0.991 \text{ ratio L+R to L-R} \qquad B := 6.16 \text{ db} \qquad C := 6.24 \text{ db}$$

Equ. 1b

$$\theta = Tgd \cdot 360 \cdot f = 3.24 \qquad Tgd := 2 \cdot 10^{-6} \qquad f := 4500$$

Equ. 1c

EQU 1 a, b, c. *Calculating separation.*

Since the MTS filters are generally located in L+R and L-R matrix paths, any difference between the L+R and L-R low pass filters will manifest itself as a rapid degradation in stereo separation. To analyze what affect random two percent changes would have on stereo separation, we will assume that the MTS system is operating in equivalent mode (no compandor) to simplify the calculations. In this case the separation can be calculated by Equation 1a.(NAB p. 3.6-196) where A is the gain ratio of L-R and L+R, and where theta is the phase error in degrees of L+R and L-R. In this equation the maximum separation occurs as A approaches the value of one and theta approaches zero. Looking at figure 3 it is apparent that the ratio of gains between L+R and L-R could deviate from unity by a considerable amount at frequencies above 10kHz. To be generous, a value of 4.5kHz is chosen. In Figure 3 a log magnitude change from approximately 6.16db to 6.24db at 4.5kHz can be approximated from the graph, and a delay change of approximately 2us can be seen. Equation 1b converts log magnitude ratio to linear magnitude ratio and equation 1c illustrates the conversion of delay in microseconds to phase in degrees.

Equation 1a, with the parameters given by Equ. 1b and Equ. 1c, indicates that the best separation to be expected at 4.5 kHz would be approximatly 31 dB. It is easy to see that the separation frequencies above 9kHz would be considerably lower. With the compandor in the MTS system activated, the situation could become worse as any change in the L-R gain between the compressor and expander would be magnified by the expander. Therefore, careful attention to absolute component values and to how these component values will change over time and temperature is required to achieve a successful high performance analog design.

To examine how this filter might be implemented using digital techniques, the filter from figure 1 could be transformed directly

1 See Williams p..8-12, for further information on equalizer.

into a discrete time realization. This type of filter is known as an IIR (Infinite Impulse Response) filter.

"The response of an IIR filter is a function of current and past input signals and past output signal samples. The dependency on past outputs (i.e. recursive) gives rise to the infinite duration of the filter output response even when the input values have stopped."(DeFatta p.47) An IIR filter is analagous to the old trick of creating echo on a three head tape deck. The signal that comes

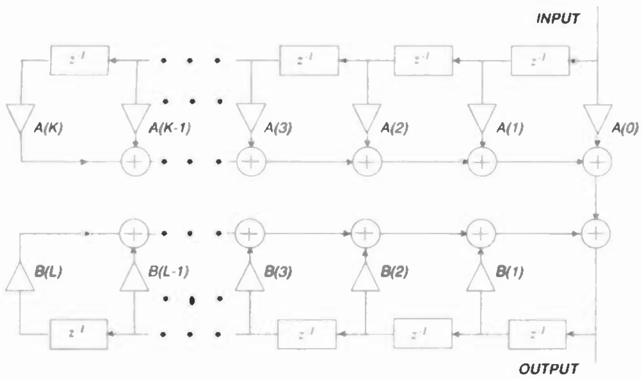

Fig.5. *IIR filter structure.*

out is a product of the original signal going in, plus the mix back of the play head. In the case of the IIR filter, instead of the time between echoes taking hundreds of milliseconds, the time between echoes are reduced to tens of microseconds. Additionally, there is the equivalent of multiple play heads all mixing in at different amplitudes with delays that are integer multiples of the shortest delay which are equal to the sample period. The different amplitudes of the mix corresponds to the coefficients that describe the impulse response of the filter. The time delays are generally one sample period in length. IIR filters, "when implemented in fixed point arithmetic, may have instabilities (limit cycles) and may have large quantization noise, depending on the number of bits allocated to the coefficients and the large signal variables in the filter".(Parks p.13) Figure 5 represents the Direct Form I method of implementing an IIR filter. Other methods are available to implement an IIR filter, and some are considerably more efficient in memory utilization. The blocks labeled z^{-1} represent

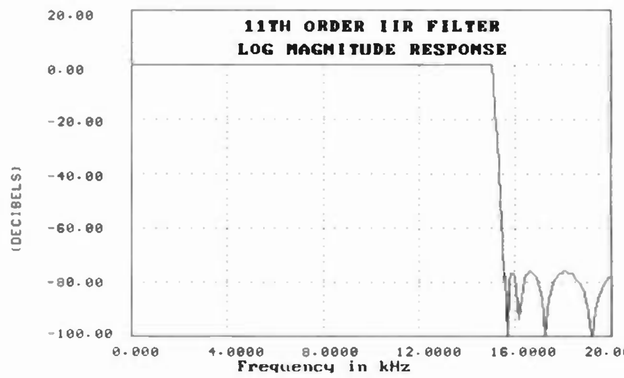

Fig.6. *IIR frequency response.*

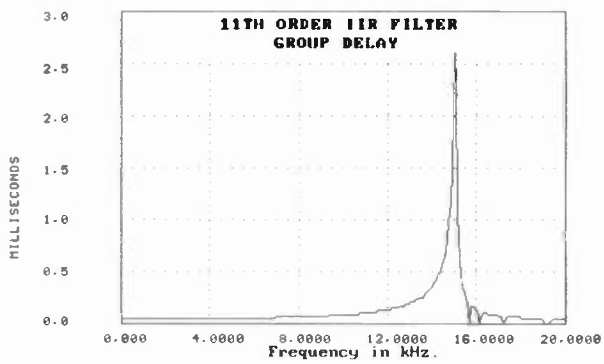

Fig.7. *IIR Group Delay Response.*

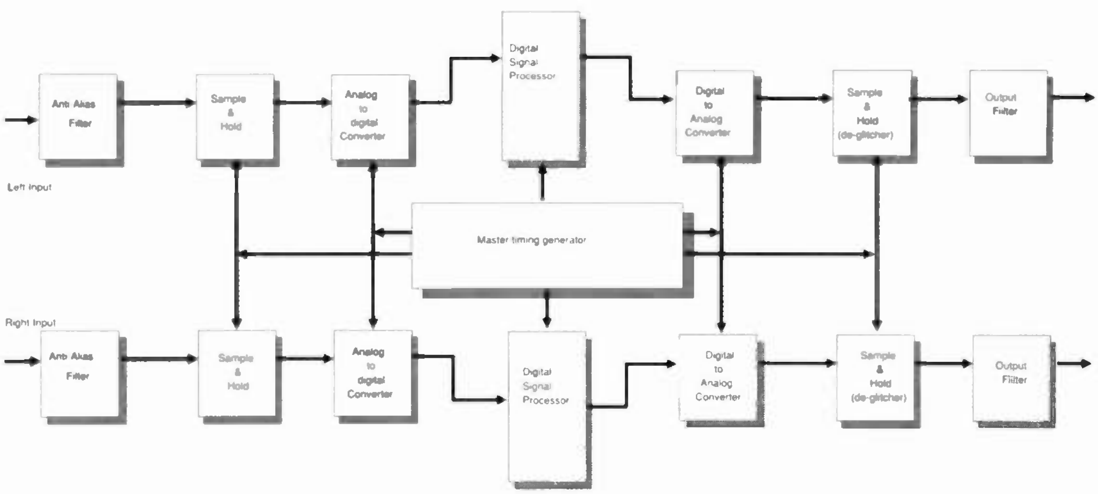

Fig.4. *Digital filter block diagram.*

$$y(n) = \sum_{l=0}^{L} A_l x(n-l) - \sum_{k=1}^{K} B_k y(n-k) \qquad (a) \qquad\qquad y(n) = \sum_{l=0}^{L} A_l \, x(n-l) \qquad (b)$$

EQU 2 *a). IIR filter b). FIR filter*

a single sample period delay. The blocks denoted by A and B represent a multiplication, or gain change, of input data to the A or B block by a coefficient located within that block.

Figures 6 and 7 depict the amplitude and delay response respectively of an IIR implementation of the before mentioned 11th order elliptical filter.

Notice that the characteristics of the analog filter, figure 2, and the IIR filter, figures 6 and 7, are very similar both in magnitude and delay responses.

As with the analog filter, there are areas where the response of the IIR filter can be compromised. The long term accuracy of the digital filter characteristic is dependent on mainly two things: the coefficients that describe the filter impulse response, and the sample rate at which the filter operates. Since the coefficients are constants and are contained in ROM, they cannot drift with time or temperature. The sample rate clock, if a well designed crystal type, is not likely to drift enough in frequency to make any noticeable difference. If the sample rate varies, the filter simply will scale up or down in frequency, with all other attributes remaining constant. For example, the above IIR filter was designed to have .177dB of ripple in the pass band, a minimum attenuation of 65dB in the stopband, a ratio of 1.04 between the highest passband frequency and lowest stopband frequency, a cutoff frequency of 15kHz and a sample rate of 44.1kHz. If the sample rate were moved to 88.2kHz, the cutoff frequency would simply scale up to 30kHz. The passband ripple, stopband attenuation and passband stopband ratio would remain unchanged.

Getting back to our discussion of separation,if both IIR filters in L+R and L-R, used identical sets of coefficients, or shared the same set of coefficients, and received their sample rate timing from the same master clock then any change in sample rate would not influence the separation since both filters would scale up or down in frequency identically. With a crystal controlled sample rate and two IIR filters, the ultimate separation could be very high and could be maintained over long periods of time. Figure 4 depicts a block diagram of how such a IIR filter could be realized in hardware. The move to a digital implementation of filters could dramatically improve the repeatability and long term performance of critical circuits.

The question a designer now has to ask himself is: Can I maintain the design goals of the system using the less expensive analog implementation, or is the use of digital technology indicated? Certainly, if components with adequate quality and tolerance can be selected, an analog filter could match the performance of its IIR digital counterparts.

There are some improvements that could be made to the digital filter design that would be difficult to match in the analog domain. As figure 7 indicates, it is easy to see that the delay characteristics of the IIR filter (and the LC analog filter in figure 2) peaks greatly at the higher frequncies. The effect of this is to delay the high frequency components by well over 0.25 millseconds with respect to the low frequency components. This will delay the 15kHz note by several complete cycles. There is a body of research that suggests that the ear is deaf to these types of delay, however, delays such as these can cause measurable changes in the sound of an audio limiter.

Another class of filter exists that improves the delay response while still giving good magnitude response. This class is known as the Finite Impulse Response filter (FIR). "If the output samples

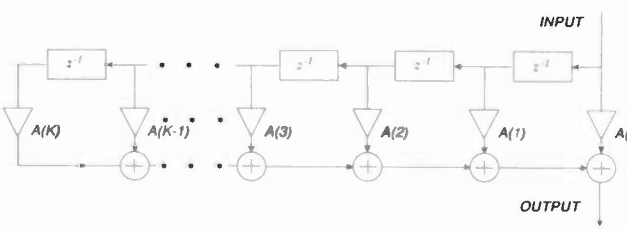

Fig.8. *FIR filter structure.*

of the system depend on the present input, and a finite number of past input samples, then the filter has a finite impulse response."(DeFatta p.49) "Digital filters with finite-duration impulse response have characteristics that make them useful for many applications. Equations 2a and 2b illustrate the most basic form of both the IIR and the FIR filter. If Equation 2a is considered with the B_k coefficients (recursive portion) equal to zero, then Equation 2a reduces to Equation 2b, which describes an FIR filter. Compare the IIR block diagram, Figure 5, with the FIR block diagram, Figure 8, and notice that the B_k terms in Figure 4 are the recursive elements of the IIR filter and that once removed you have the non recursive FIR filter.

It is possible to construct an FIR filter using analog techniques. One such example uses an analog bucket brigade line with taps at

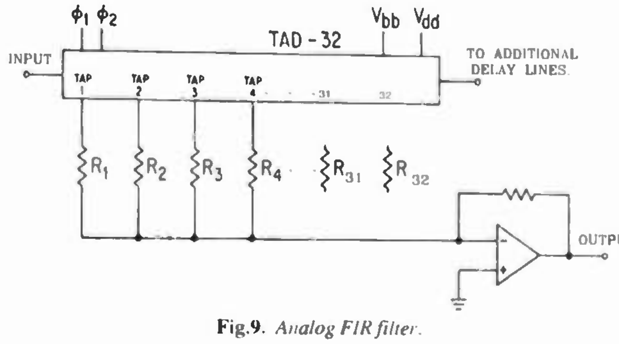

Fig.9. *Analog FIR filter.*

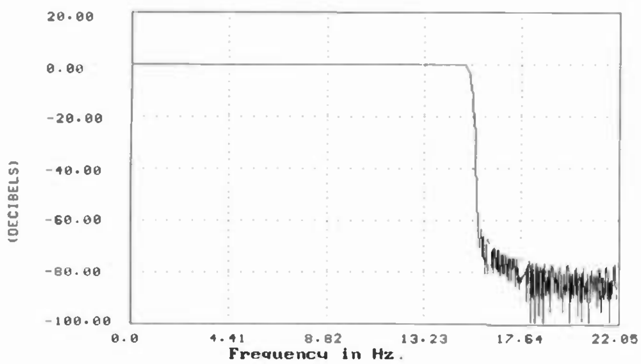

Fig.10. *Length 249 Kaiser window FIR, Magnitude.*

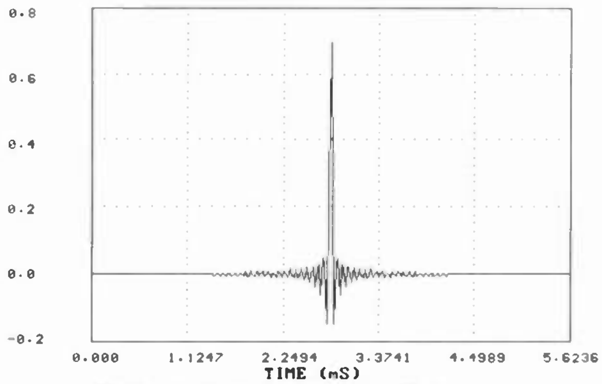

Fig.11. *Length 249 Kaiser window FIR, Impulse.*

```
ZAC                     ;CLEAR ACCUMALTOR
LT       MEM249         :LOAD T REGISTER WITH MEMORY LOCATION 249
MPYK     COEF249        ;MULTIPLY BY CONSTANT COEFICENT #249
LTD      MEM248         ;LOAD T, ADD TO ACCUMALTOR & STORE DELAY
MPYK     COEF248        ;MULTIPLY
LTD      MEM247         ;LOAD T, ADD AND DELAY
MPYK     COEF247        ;MULTIPLY
.
.
.
LTD      MEM003         ;LOAD T, ADD AND DELAY
MPYK     COEF003        ;MULTIPLY
LTD      MEM002         ;LOAD T, ADD AND DELAY
MPYK     COEF002        ;MULTIPLY
LTD      MEM001         ;LOAD T, ADD AND DELAY
MPYK     COEF001        :MULTIPLY
LTD      MEM000         ;LOAD T, ADD AND DELAY
MPYK     COEF000        ;MULTIPLY
RTN
```

Fig.12. *Example of FIR filter code for TMS DSP's.*

$$y(n)=h_0 x+h_1 x(n-1)+h_2 x(n-2)+\ldots+h_{42}x(n-42)$$

EQU 3 *Length 43 FIR convolution.*

each "bucket".(EG&G/Reticon pp.7-15 to 7-30) Figure 9 is a schematic representation showing resistor loading taps for the realization of a desired filter function. The heart of this design is the 32 tap bucket brigade from EG&G / Reticon. An audio signal is applied to the input and clocked through the chip one "bucket" at a time. At each bucket is a tap where the audio at that bucket can be brought out and mixed with the audio at all the other taps. The weighting of the mixing resistors corresponds to the impulse response of whatever filter is desired. The output of the filter is therefore the final mixed signal appearing at the op-amp output. The TAD-32 has a feed forward output for the addition of another TAD-32 to implement longer filters.

Taking the same filter parameters outlined in the preceding IIR example and applying them to an FIR filter results in a filter of length 249 when using the Kaiser window approximation. Figure 11 depicts the impulse response of this filter. The window is applied to the impulse response to prevent the Gibbs phenomenon from occuring due to truncation of the impulse response. (Remember this is a **FINITE** impulse response filter.) The Kaiser window is selected because it it somewhat more flexible[2] than the more common Hanning or Hamming windows.

A length 249 filter requires a delay line of 248 elements and 249 taps. Using the above described analog method would require eight of the TAD-32 components, and 249 mixing resistors.

Perhaps a better approach would be to convert the audio into digital form and use a Digital Signal Processor (DSP) chip to perform the required operations. Equation 3 illustrates the convolution used to generate the output signal. As we have seen in the analog FIR version,"if the filter were implemented with a tapped delay line z^{-1} would correspond to a physical delay element. However when a digital computer program is written to implement Equation 3, the boxes labeled z^{-1} correspond to storage of variables rather than any delay."(Parks p.140) A small portion of code from a Texas Instruments TMS 32010 signal processor is shown in Figure 12 illustrating the coding of an FIR filter program. Figure 10 shows the magnitude response. Figure 11 shows the impulse response for this filter. The delay will be found at the peak of the impulse response. In the case of our example FIR filter, the delay is approximately 2.8 milliseconds. It should be noted at this point that a length 249 filter would not execute within a single TMS 32010 DSP at sample rates suitable for broadcast audio. Either the more powerful TMS 320C25 or 320C30 chips could be used. or another possibility is to break the code into several modules running with multiple TMS 320C10's.

2 For further information on Windows see Parks pp.71-79

Let's take another common filter used in broadcasting that is substantially less demanding than the above discussed MTS filter. This filter can be found in FM stereo generators, and some makes

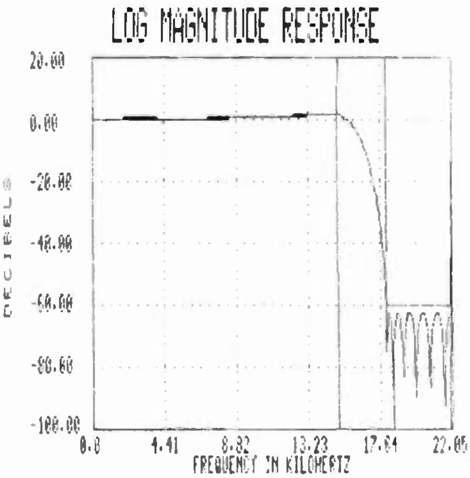

Fig.13. *Length 43 Parks-McClellan FIR, Magnitude.*

of audio processors. Its purpose is similar to the filter used in the above discussed MTS application in that it protects the pilot from interference caused by audio programming, prevents aliasing distortions, and protects the SCA region from interference. Generally, the filter should cut off at a frequency just above 15kHz and smoothly roll off to at least -60 below 100% modulation by the time it nears pilot frequency of 19kHz. Let's take a look at what types of problems we could potentially run into if we attempted to develop this filter using the linear phase FIR technology. Figure 13 shows a FIR filter using Parks- McClellan windowing that meets the above specifications. The filter is designed to operate at a sample rate of 44.1kHz. The length of the filter is 43 and the coefficients are shown in Figure 14. Total delay through the digital filter can be found by simply finding the peak of the impulse function in Figure 14 and multiplying that coefficient number by the sample period. From Figure 14 we see that coefficient number

```
PARKS-McCLELLAN ALGORITHM

MULTIBAND FILTER

FILTER LENGTH = 43

SAMPLING FREQUENCY= 44.100 KILOHERTZ

***** IMPULSE RESPONSE *****

16-BIT QUANTIZED COEFFICIENTS

H( 1) =-.823975E-03 = H( 43)
H( 2) = .259399E-02 = H( 42)
H( 3) =-.195313E-02 = H( 41)
H( 4) =-.552368E-02 = H( 40)
H( 5) = .500488E-02 = H( 39)
H( 6) =-.469971E-02 = H( 38)
H( 7) =-.222778E-02 = H( 37)
H( 8) = .994873E-02 = H( 36)
H( 9) =-.141907E-01 = H( 35)
H( 10) = .659180E-02 = H( 34)
H( 11) = .808716E-02 = H( 33)
H( 12) =-.243835E-01 = H( 32)
H( 13) = .267334E-01 = H( 31)
H( 14) =-.103455E-01 = H( 30)
H( 15) =-.252380E-01 = H( 29)
H( 16) = .538330E-01 = H( 28)
H( 17) =-.584717E-01 = H( 27)
H( 18) = .881958E-02 = H( 26)
H( 19) = .877075E-01 = H( 25)
H( 20) =-.185883E+00 = H( 24)
H( 21) = .227142E+00 = H( 23)
H( 22) = .802429E+00 = H( 22)
```

Fig.14. *Table of coefficents for length 43 FIR.*

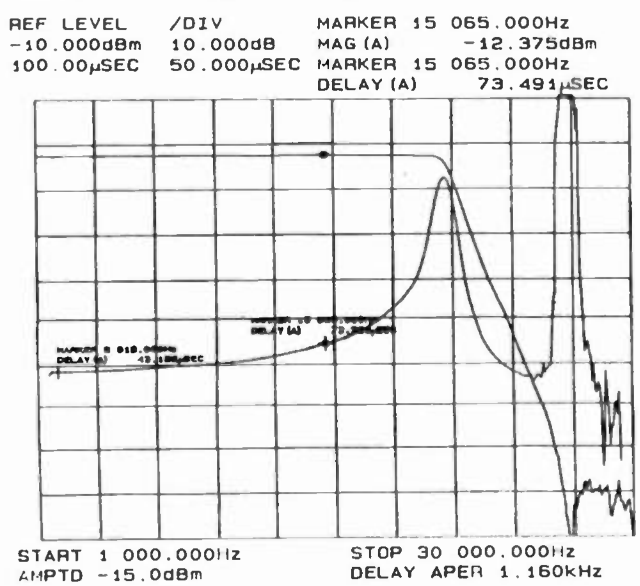

Fig.15. *Analog anti-aliasing filter responses.*

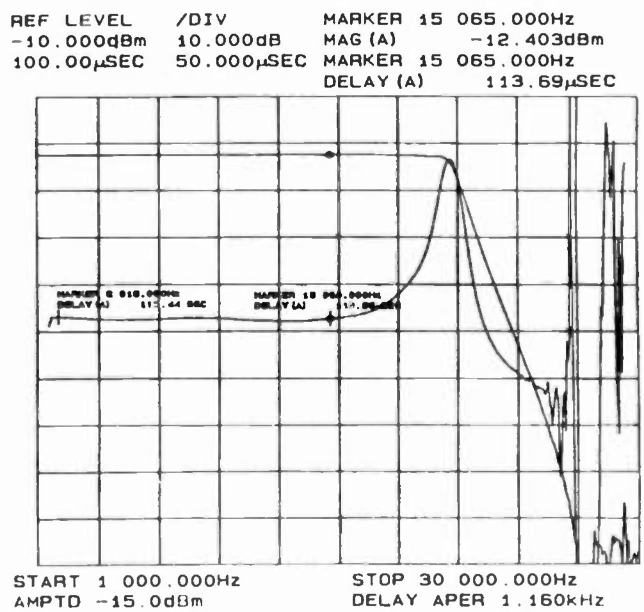

Fig.16. *Analog filter w/ group delay correction.*

22 is the peak of the impulse function, therfore we multiply 22 times 22.67 microseconds and find a total of 498.9 microseconds for the delay of this digital filter. But is this going to be the total delay when we view this filter as a system? We must also include the analog anti-aliasing filter that is ahead of the A to D converter and the reconstruction filter that follows the D to A converter. Figure 15 shows the delay response of a typical "off the shelf" anti-aliasing / reconstruction filter used for digital audio applications. Looking at Figure 15 we can see that the delay starts at approximately 43us and increases to over 250 us at 20kHz. Since our FIR filter cuts off above 15kHz, we are only interested in the time delay response through 15kHz. Judging from Figure 15, the analog filter varies by 30 microseconds from 2kHz through 15 kHz.. This variation in the analog anti-aliasing and reconstruction filter essentially will ruin the linear phase ability of the FIR filter. Several things can be done however to improve this situation.

An analog delay equalizer could be constructed to compensate for the region between DC and 15kHz. Figure 16 depicts the delay equalized to less than 5 microseconds of ripple. If the input analog anti-aliasing filter and the output analog reconstruction filter both used the filter described in Figure 14, then the total delay would be approximately 498 + 2(113) = 724 microseconds.

Another approach to the non-linear phase problem would be to incorporate a delay equalizer in the software, and still another approach would be to move the sample rate up in frequency.

Over sampling has merits in that it can greatly reduce the complexity of the analog anti-aliasing filter and the analog reconstruction filters. If the sample rate were to increase by a factor of two, then the analog filters that surround the digital technology could reduce in order dramatically. If the sampling rate is increased to 88.2kHz, then the analog anti-aliasing filter and reconstruction filter could have a response similar to Figure 17. Notice that the delay time curve in Figure 17 is dramatically reduced in magnitude as compared to Figure 15. In both figures 15 & 17 the delay curve is the lower curve. In Figure 15, markers are placed on the delay curve at 2kHz, (43 microseconds) and at 15kHz (73 microseconds). Over sampling thus simplifies the filter needs and group delay correction. What effect will increasing the sampling rate have on the DSP filter? If the passband and stop-band specifications remain the same, the only major change that will occur is the length of the filter will approximately double. The filter shown in Figure 13 is a length 43 Parks-McClellan FIR filter. This filter, if used in this over sampling example, would then become a length 86 filter. This means that the microprocessor now has twice the number of calculations to make and half the time in which to make them. The point is that a doubling of sample rate causes a four times increase in the raw processing power needed to implement the digital filter curve.

This is a good time to look at just how much raw processing power is enough for implementing DSP filters. "Causality refers to a system that is realizable in real time. A causal system is a system that at time m produces a system output that is dependent only on current and past inputs, n less than or equal to m, and past outputs,n less than m."(DeFatta) The FIR filters discussed above are causal systems. In order for these filters to operate in real time, they must be able to complete all of the calculations and data storage operations required within one sample period. At a rate of 44.1kHz, this gives roughly 22.67 microseconds. The length 43 filter discussed above requires 43 addition operations, 43 multiplication operations, and it must shift 43 data memory locations by one memory location each. In addition to these duties it must input from the A/D and output new data to the D/A and have some mechanism to detect the beginning of a new sample period. The processor must be able to carry out these operations within the single sample period. If the processor is able to just meet these requirements, then the filter should operate fine. If, however, a processor could perform the above calculation in only one quarter of the sample period, what effect does this have on the filter? None. The processor simply has spare time that it could fill by running through a dummy loop until the next sample period arrives or it could perform additional processing such as

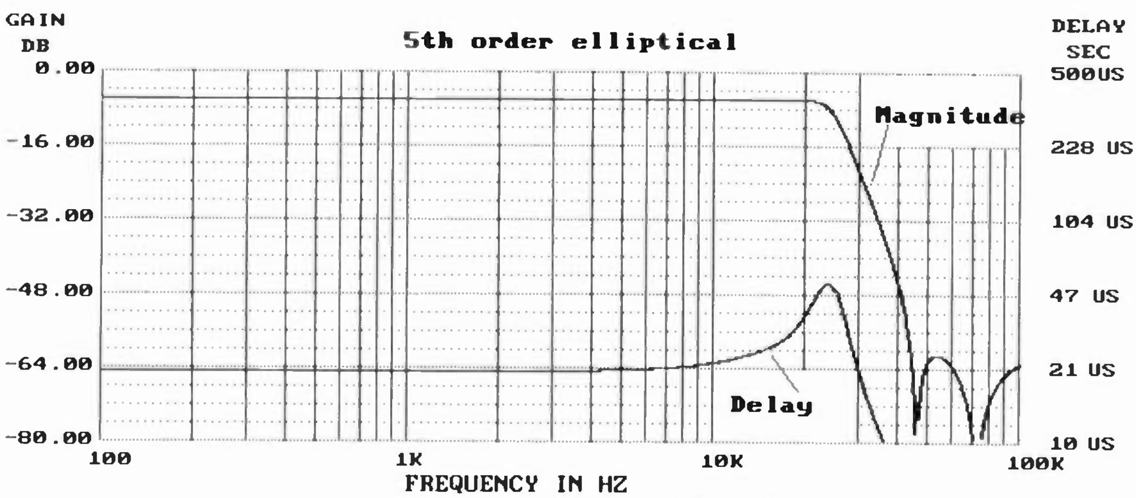

Fig.17. *Anti-alias filter for over-sampled FIR.*

AGC, limiting, or other filtering functions.

The following is a list of advantages of each technology that summarizes what we have discussed so far.

Digital filter advantages:

- **No drift due to component tolerance/temperature changes.**
- **Very sharp cutoffs achievable.**
- **Excellent tracking for stereo operation.**
- **Flat group delay responses (FIR types).**
- **Easy to change characteristics.**

Analog filter advantages:

- **Many times less expensive than digital filters.**
- **Generally more compact that digital hardware.**
- **Better serviceability.**
- **Can be realized with passive components only.**

Let's turn away from filtering issues now and examine some of the other important processing functions. Automatic gain control is another fundamentally important processing function. There are many good analog AGC's currently available that can give quite impressive specifications. Signal to noise ratios of AGC stages can, in many instances run from 80 to better than 100 dB while being able to provide an AGC range of 30 to 40 dB. If a digital AGC circuit were to be considered, one of the first stumbling blocks that could be encountered is signal to ratio versus AGC range. Sixteen bit digital audio systems have at best 96 dB of total dynamic range. If 30dB where allotted for gain reduction range and 10dB for headroom range, there would only be 56 dB left for signal to noise ratio. In actual use this number would be more like 45 to 50 dB once real life considerations are taken into account. A brief list of some of the critical factors are: "accuracy, aperture uncertainty, glitches, linearity, precision, quantization error and resolution of analog to digital and digital to analog converters." (Williams p.16-1). The mathematical precision of the processor is another factor. For instance, Figure 10 gave a brief piece of code for implementing an FIR filter on a TMS 32010 series processor. Once the data is taken in at sixteen bits, all further manipulations such as additions and multiplications are performed to 32 bit precision. The TMS 32010 is designed with a 32 bit accumulator to reduce the amount of error introduced while operating on a signal. If an eighteen bit system were considered, a 12 dB improvement could then be expected and our AGC would now have a whopping 57 to 62 dB signal to noise ratio.

There are alternatives to the above problem. One possible solution would be to pre-scale the input of a digital audio gain control with an analog gain element. The analog gain element could control itself using either feed forward, feed back or take control instructions from the DSP processor. This could in effect extend the range of a sixteen bit system. The point here is, that 16 bits may be adequate for simple reproduction of audio but for extreme manipulation of audio, a 16 bit system would be less than adequate. With the state of digital hardware as it is, (16 bit devices are plentiful and 18 bit devices are arriving on the scene) it would

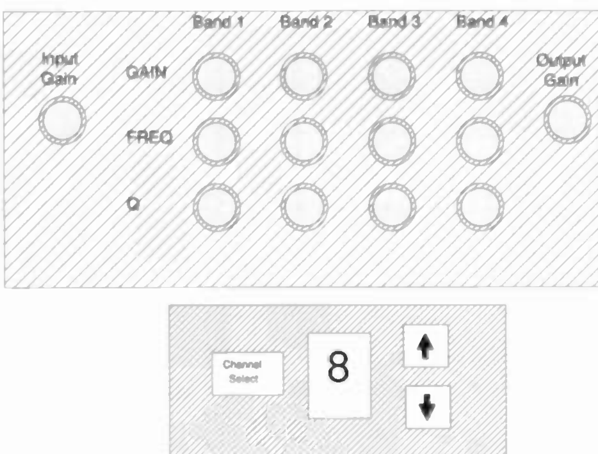

Fig.18. *Analog and digital userability.*

ıalog technology is still an important factor in the ɹigital products at this time.

ʌnother issue in this topic of analog and digital processing for ·oadcast is the issue of "user-ability". In the pure analog world most devices would have a separate control for every user adjustable parameter. For instance, a four band parametirc equalizer would have four gain adjustments, four frequency adjustments and four Q adjustments. Additionally you would find a separate input and output control. Digital technology has given us the ability to replace these controls with just 3 buttons and a alphanumeric read-out. One button could select which parameter to change while the alphanumeric read-out displays where you are. (Hopefully in a familiar language.) Once you have paged through the controls and found which one you want to adjust, you are now free to use the up and down buttons to arrive at the desired setting for that control. Figure 18 depicts the contrast between these two hypothetical systems.

The designer of such a piece of equipment has eliminated costly pots and knobs and in the process saved his company money. But has this process also eliminated the user friendliness. Equipment with just a few push-buttons here and there are becoming more common. The problem is that operating them is similar to being a one armed paper hanger. The opportunity for the user to glance at and distinguish where the control settings are has been lost. In addition the ability for the user to instantly or even simultaneously change several operating parameters has been lost. This push-button versus rotary control aspect is important in that it affects the efficiency of everyone who comes in contact with it.

Sometimes the reliance on push-buttons instead of knobs is justified by the designers of equipment to ease some design factor, such as remote control interface. The rationale being that contact closures are easier to interface. Some products implement the idea of "digital soft controls" very well. In these systems several pushbuttons and perhaps a pot or two are switched via software to implement many different functions. A small LCD screen is usually available to help guide the user to the appropriate menu. All that a user must do to use these systems is to familiarize himself with the various paths through the menu system. Even with the helpful menus, the ability to quickly set and ascertain control settings is compromised. However, with microcontrollers becoming less expensive every day, the user should be able to have eat his cake and eat it too.

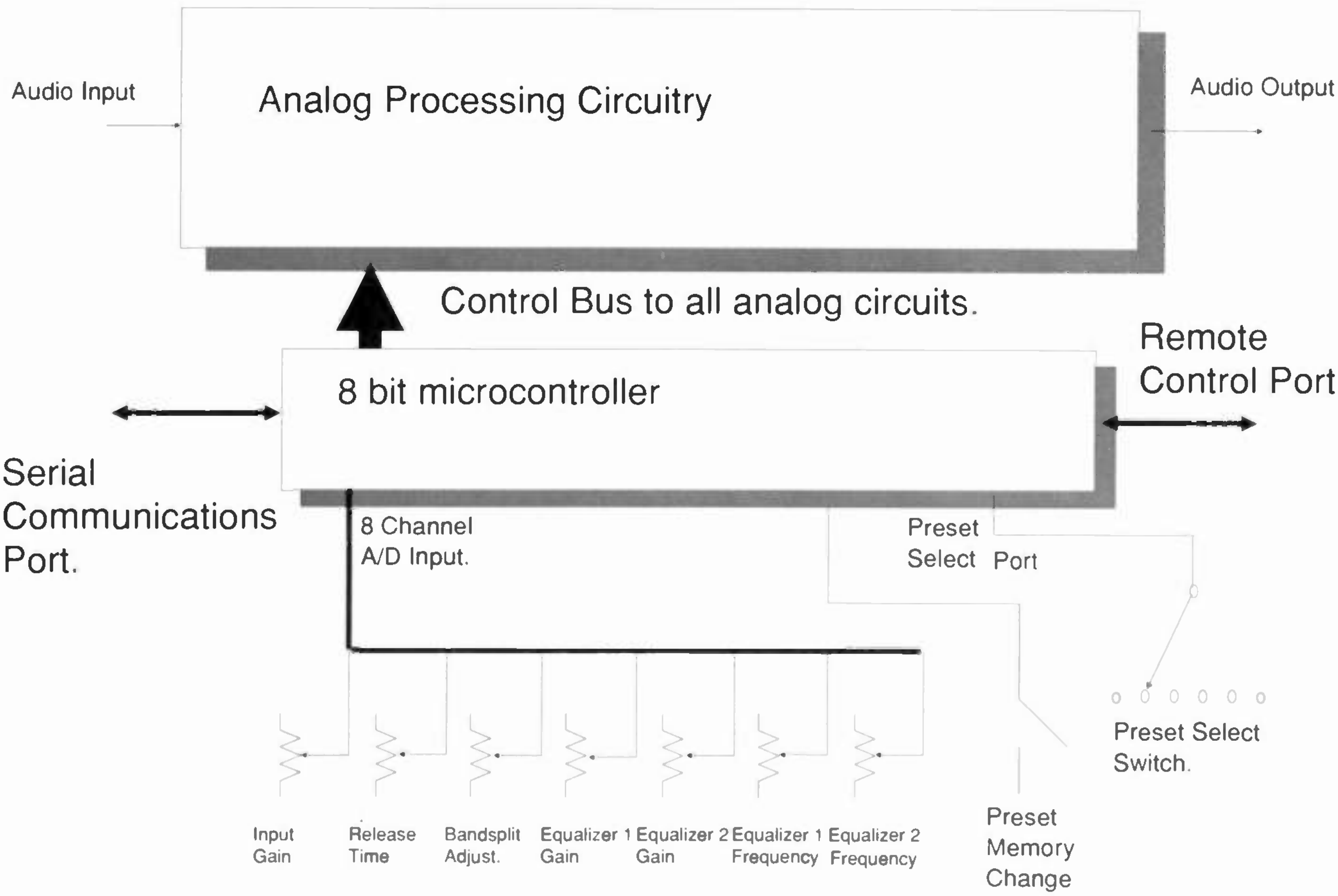

Fig.19. *CRL IPP-100 Programmable Mic. Processor.*

Figure 19 depicts the CRL IPP-100 studio microphone processor control arrangement. Even though all functions could have been implemented using the menu and soft key arrangement, the choice was made to interface front panel controls through the microprocessor in such a way as to correspond to controls found on fully analog models. In Figure 19 we see all the necessary controls implemented with potentiometers. These controls in turn drive a multiplexed analog to digital converter. The microprocessor is now able to perform several functions that are difficult to acheive in a fully analog system. In one mode, the microprocessor simply passes the analog control settings directly to the analog audio processing hardware. In another mode, the microprocessor can record the settings of each analog control for later use as a pre-set. And in another mode, is where the microprocessor allows only the use of the pre-set adjustments. With this arrangment the user has very little to learn about the functioning of the unit.

Digital technology promises us much, but there are still areas where analog techniques can achieve cost effectice results.

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AUDIO PROCESSING FOR NRSC

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ABSTRACT

In contemplating adoption of the NRSC recommendation for transmission preemphasis and bandwidth restriction, the AM broadcaster must consider the constraints of a system which would no longer have the enviable property of a frequency-flat modulation and overload characteristic.

This paper describes an audio program signal processor, specifically developed to meet the several technical challenges imposed by NRSC implementation.

THE NRSC SPECIFICATION

In January, 1987, the National Radio Systems Committee (NRSC), representing both broadcasters and receiver manufacturers, formally adopted a technical standard toward improving the quality of AM radio in the U.S. This standard, now called "NRSC-1," specifies transmission preemphasis, based on a 75-microsecond characteristic, with complementary deemphasis in the receiver to restore flat overall response. Figure 1 graphs the "truncated" NRSC curve, and shows deviation from true 75-microsecond preemphasis.

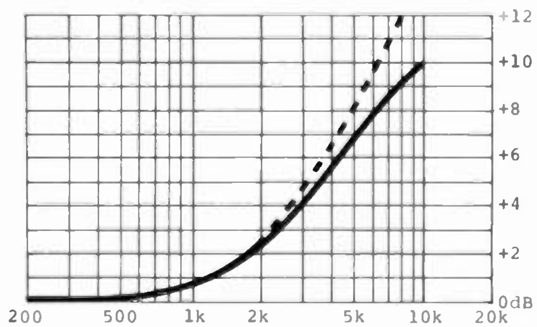


Figure 1

75-Microsecond Preemphasis (broken line), and NRSC Curve (solid line)

The NRSC standard also specifies a very sharp 10kHz audio cutoff. This cutoff should restrict transmitted bandwidth and effectively eliminate interference in second-adjacent channels. The audio cutoff specification is shown in Figure 2.

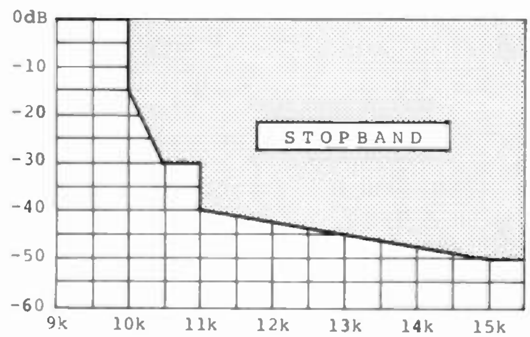


Figure 2

NRSC Audio Stopband Spec.

The preemphasis / deemphasis part of the specification was intended not only to improve perceived signal-to-noise performance, but also to establish a common point of reference for broadcasters and receiver manufacturers alike. It was anticipated that new AM receiver designs would be optimized for the standardized transmission characteristics, and broadcasters, in turn, could substantially reduce the often brutal high frequency boost considered essential for decent sound from the "obsoleted" narrowband radios.

IMPLEMENTATION CONSIDERATIONS

Preemphasis "Protection"

The NRSC preemphasis characteristic of Figure 1 is easily attained with a simple R/C network comprising one capacitor and two resistors. For a total cost of about one dollar, the broadcaster can add the specified NRSC preemphasis to his program signal chain. What this simple solution fails to address, however, is the resultant effect on real and perceived modulation.

If NRSC preemphasis is applied after the final program limiter, high frequency program energy, though already limited to a 100%-modulation level, would be boosted to an equivalent of 300% modulation. If, on the other hand, simple preemphasis is imparted before the final limiting device, the overall program level will "duck," as accentuated highs demand additional gain reduction at all program frequencies. Consequently, high frequencies must be dealt with independently to avert the modulation sacrifice implicit in either case cited.

Low-Pass Filter Considerations

To satisfy the 10kHz cutoff requirement, yet retain program clarity and "brightness," the low-pass filter response must be flat within its passband, and have as high a corner frequency as practicable, with precipitous rolloff beyond. Only the Cauer, or "elliptic" response, filter conveniently meets this demand. The specified stopband response (Figure 2) suggests a ninth-order filter of this type. Such a filter may be built from "passive" L/C components, or with "active" op-amp circuitry.

When presented with amplitude-limited complex program waveforms, any sharp cutoff filter will invariably exhibit a certain degree of output overshoot. This is particularly true when peak-limiting circuits square-off the program signal waveshape. Even filter designs which are fully phase-corrected with all-pass networks of similar order will exhibit these output overshoots. Much of this is attributable to the filter's normal and, in fact, desired elimination of higher-order frequency components which, themselves, help define the instantaneous peak value of the signal.

A ninth-order elliptic-function low-pass filter may easily overshoot 150% or more. If modulation is lowered to accommodate overshoot peaks, the average value of the program falls accordingly. These overshoots cannot simply be clipped; the harmonics generated by post-filter clipping would then violate the stopband specification.

Clearly, some form of filter overshoot compensation is required. Several ingenious techniques are in common use in FM broadcast equipment, which has similar filter constraints. These techniques may be borrowed, provided that they prove effective for the much sharper cutoff characteristic of the NRSC specification.

PROCESSOR DESIGN CRITERIA

In conceptualizing an audio processor for NRSC compliance, many factors were considered. Most could be lumped into either of two areas of primary importance:

1. The device must not only meet the letter of the specification, but it should comply with the intent, as well; that is, "AM improvement." Thus, aside from imparting the specified parameters to the program signal, the Processor must add no "sound" of its own, or alter the effectiveness of other audio processing equipment in the program path. Dynamic action should be transparent, and neither enhance nor degrade measured or perceived modulation.

2. The Processor must be convenient to install, and very simple to set up and use.

As of this writing, the great majority of AM broadcasting is monaural. For this reason, and for reasons of simplicity and economy in general, the Processor evolved as a single-channel device. To accommodate AM-Stereo, either for immediate or for future use, it was anticipated that steps could be taken in the manufacturing process to hold frequency and phase response to close tolerances. This would assure that any two units could serve as a matched pair for stereo sum-and-difference processing.

To make installation straightforward and as convenient as possible, the unit was configured as a post-processor; that is, it would simply connect between the existing audio processing system and the input to the transmitter. Though this position in the signal path is the only proper placement for the 10kHz low-pass filter, it is not suitable for a fixed preemphasis network for reasons already explained. This dictated use of some form of "adaptive," or variable, preemphasis if the function were to be successfully included at this point.

DYNAMIC CONTROL REQUIREMENT

The need for program-dependent preemphasis has been established by the placement of the Processor in the audio signal path. Because it feeds the transmitter directly, the Processor must be self-protecting against overmodulation from preemphasized high frequencies. This requires linear control of the boosted portion of the program spectrum, based on the energy therein.

There are probably more ways to remotely control the gain of an audio program signal than Carter's has pills. Variable-mu pentodes, light-dependent resistors, FETs, VCAs; these devices and countless others have been employed as voltage-controlled gain elements. Each has its good and bad qualities, from the standpoints of control range, signal distortion, noise, stability, complexity, etc.

Pulse Width Modulation

One sorely neglected technique for program signal gain control, yet one which has decided advantages over several of the more common methods, is pulse-width (or duty cycle) modulation. In the PWM system, the program signal is turned on and off, or chopped, at a rate several times the highest audio frequency; generally 100kHz or more. The ratio between the signal "on" time and the "off" time directly determines the effective gain reduction. A 100% "on" time would, of course, represent a 0dB loss; a 100% "off" time, an infinite loss. "On" values between 100% and zero yield corresponding signal reduction: 50% = 6dB, 25% = 12dB, 10% = 20dB, etc.

A big advantage of PWM gain control is its simplicity. The signal switch, which may be a junction transistor, a FET, or CMOS transmission gate, is either on or off. There is no "linear range" over which the device must operate, save the boundaries imposed by the peak-to-peak amplitude of the program signal. Balancing and distortion nulling is not required, and a very simple low-pass filter removes the high frequency switching components.

The only real limitation in PWM gain control is finite switching time. This limits the duty cycle ratio to a maximum of about 50:1, or 35dB worth of gain reduction. Though this is clearly insufficient control range for something like a console fader, a 30dB range is more than adequate for an automatic gain controller, or program "leveler."

Most audio levelers (compressors and limiters) are based on a feedback technology. The output of the gain stage is referenced to a threshold value, and as the signal reaches this value, an error voltage is developed which effects signal gain reduction. Amplification of the error voltage determines the slope of the input / output transfer function. This slope can assume any value from 2:1 (or less) for gentle compression of program dynamics, to 20:1 (or more) for absolute signal limiting.

Feedback gain control is typically characterized by a well-defined "knee," or transition from a linear to a controlled state, and a constant slope above this transition. Figure 3 illustrates these characteristics.

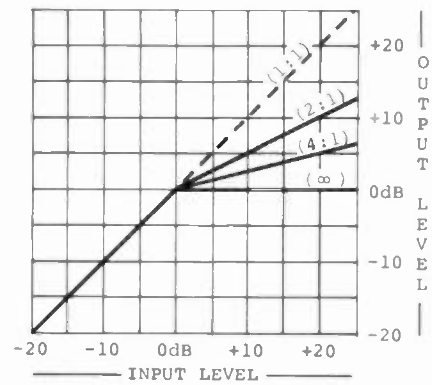


Figure 3

Feedback Leveler Transfer Function

Feed-Forward Gain Control

The predictability of PWM gain control makes it ideal for feedforward designs. These feature a gentle, less abrupt transition into the gain-controlled state, with a corresponding sonic "smoothness" and freedom from audible artifacts.

Because the feedforward method bases gain reduction on the amplitude of the input signal, and does not create and utilize an error voltage, the transfer function cannot be derived; rather, it must be "fabricated."

For the sake of explanation, let us assume a simple program peak limiter with a linear 1:1 relationship below its knee, and an infinite, flat-topped characteristic above. If the knee is arbitrarily set at one volt, we can plot the circuit gain required to hold the output at this value, once the input reaches and exceeds the one volt figure. Figure 4 graphs this relationship.

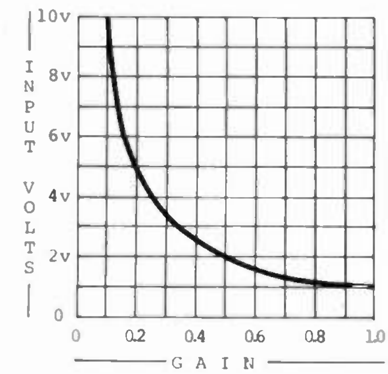


Figure 4

Input Voltage vs. Gain
For 1-Volt Limited Output

The curve thus defined has a hyperbolic shape, and this is the function which must be factored into the value of the input signal to effect the desired gain reduction. Fortunately, this is a simple matter with PWM; fortunate, that is, for the author, who couldn't factor the hyperbola mathematically if his life depended on it.

The pulse width modulator, which opens and closes the signal switch, is nothing more than a zero-hysteresis comparator. The input program signal is rectified, filtered and applied to one input of the comparator, a repetitive "ramp" waveform at the switching frequency is fed to the other input. The shape of this ramp determines the circuit transfer function which, in this case, must be hyperbolic. Figure 5 represents a single cycle of the requisite ramp waveform, and also shows the duty cycle of the signal switch as the hyperbola is intercepted by a DC control voltage.

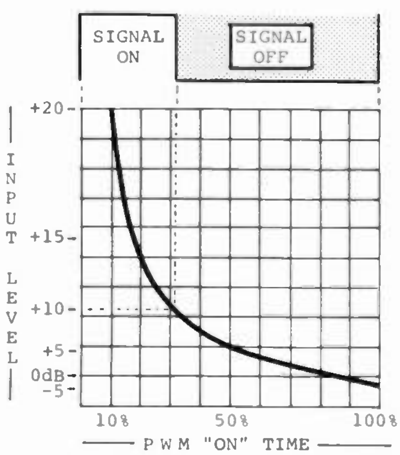


Figure 5

Ramp Waveform And 10dB
Attenuation Example

The control voltage is derived directly from the input signal; 10dB above the threshold value in this example. The switch "on" time of 31.62% would give a signal attenuation of 10dB, the figure required to maintain the output at the 0dB ceiling.

It may be noted that the ramp waveform in Figure 5 deviates from a true hyperbola at bottom-right. What this intentional deviation accomplishes is to stretch the transition into the limited state over several dB of input level change. The result is the "soft knee" which yields the sonic advantage mentioned before. The actual 150kHz ramp waveform is shown in Figure 6, the transfer function of the limiter in Figure 7.

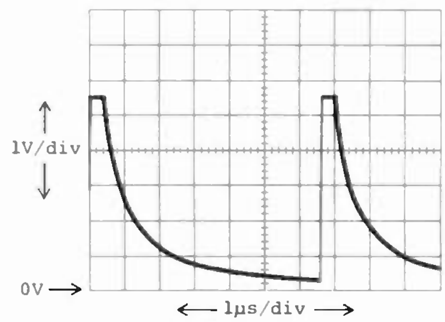


Figure 6

Repetitive Hyperbolic Ramp

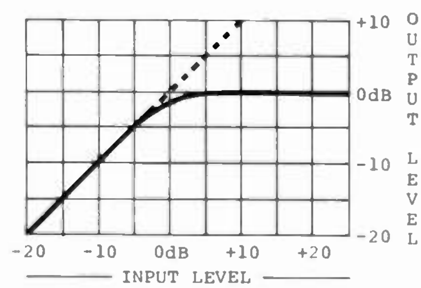


Figure 7

Limiter Transfer Function

Processor PWM Implementation

Though the program signal at the input to the NRSC Processor is assumed to be peak-limited, a feedforward limiter of the design just discussed was included, just in case it might prove of some utility at the transmitter site. This limiter can, however, be switched out of the signal path.

A second, nearly-identical feed-forward limiter circuit performs the adaptive preemphasis function. Rather than using a simple R/C network to obtain high-end boost, the program signal is split into two paths. One path passes through a high-pass filter with a pole at 8700Hz per the NRSC spec. When the output of this filter is summed with proper gain and phase back into the "flat" path, NRSC preemphasis results.

The second feedforward limiter is placed just after the high-pass filter. Its ceiling is adjusted such that the high-pass signal contribution cannot sum to more than the 100%-modulation level of the previously limited broadband program. This means that high frequency program components of low energy get full preemphasis, but high energy, high frequency peaks receive a temporarily reduced amount. Since the adaptive preemphasis limiter works only on high frequency program material, its time constants can be accordingly shorter. This factor, plus the "soft knee" transfer function, minimize audible artifacts of the varying, program-dependent preemphasis. The range of adaptive preemphasis is shown by the family of curves in Figure 8.

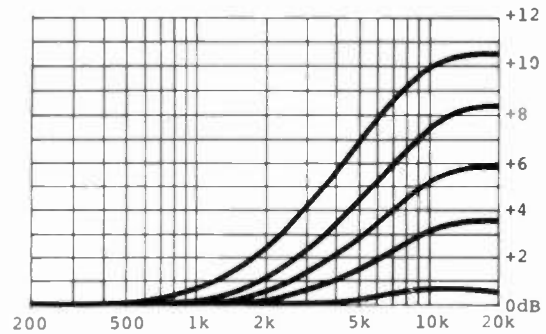


Figure 8

Range Of Adaptive Preemphasis

The 10kHz Low-Pass Filter

The 9-pole low-pass filter meeting the NRSC stopband specifications may be either an L/C or an active design. Since coil-winding is a Grade-A drag, the active approach was chosen. As a further testimonial to laziness, filter component values were calculated from tables in one of the better "cookbooks" on the subject, the Electric Filter Design Handbook by Arthur B. Williams, published by McGraw-Hill. The "cookbook" active filter, which is actually derived from classic L/C elliptic designs, is commonly known as the "FDNR," or Frequency Dependent Negative Resistance type. In Williams' book, it is also called a "GIC," or Generalized Impedance Converter. Whatever!

With 2.5% capacitors and 1% resistors, the calculated values gave a filter which met the NRSC stopband specification with room to spare. But to enable phase response matching between Processors used for Stereo, fine-tuning adjustments were included in the final filter design. Stopband response of the filter is plotted in Figure 9, upper passband response in Figure 10.

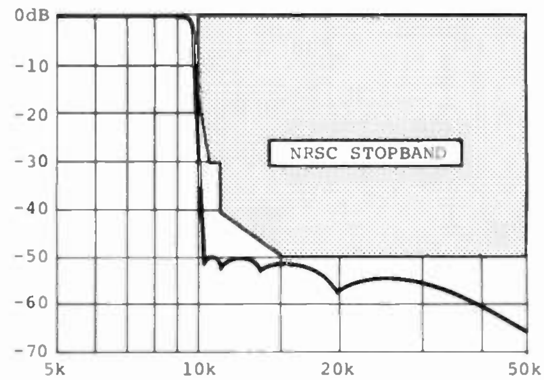


Figure 9

10kHz Low-Pass Filter
Stopband Response

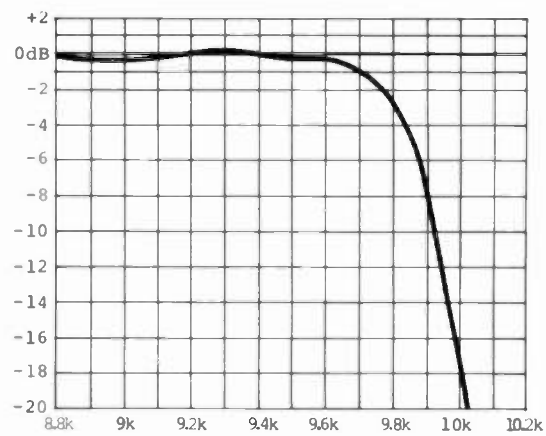


Figure 10

10kHz Low-Pass Filter
Upper Passband Response

Filter Overshoot Compensation

As anticipated, the sharp cutoff low-pass filter exhibited the usual output overshoots and "ringing" when presented with nearly anything but a clean sinewave. Of the various techniques which have been developed to cope with low-pass filter overshoots, the ones that really work are well protected by patents. Rather than risk a stretch in the slammer, effort was put into an alternative means of overshoot compensation.

A couple of the more popular overshoot compensation schemes permit the low-pass filter to generate the expected overshoots. These are then isolated, re-filtered, and somehow introduced back into the signal path to cancel themselves. Another technique in current use distributes peak clipping circuits among the several cascaded filter sections.

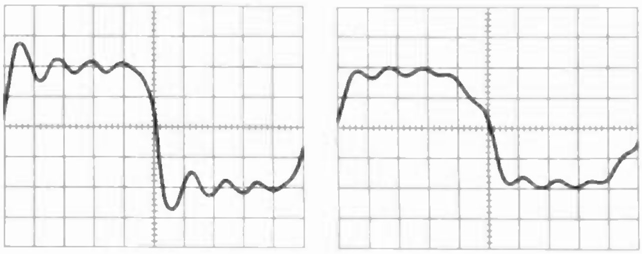
For the NRSC Processor, it was instead decided to address the property of the limited, squared program waveform which excites the filter and causes it to overshoot in the first place.

Overshoot of a low-pass filter can be calculated by its "step response"; the manner in which the filter responds to a given input DC voltage step function of negligible risetime. Program signals invariably contain similar step functions with fast-rise leading and trailing edges. These are among the properties of the program signal which can excite overshoots of sizable porportions.

The compensator developed for the NRSC Processor is placed <u>ahead</u> of the 10kHz low-pass filter. It so conditions the limited program signal waveshape that the filter has little or no tendency to generate any overshoots. The fast-rise components of the program signal are displaced in <u>time</u> to add to the <u>amplitude</u> of the limited signal. They are subsequently clipped, recovered, and re-added in opposite phase.

The overshoot compensator does not affect frequency response. It provides only first-order "static" phase correction for the low-pass filter, and high frequency program components can undergo as much as 180-degrees of "dynamic phase rotation" during compensator operation. This turns out to be inaudible, however, and a small consideration when compared with the much greater phase displacement within the low-pass filter itself.

Figure 11 shows the effect of the overshoot compensation circuitry. In oscillograph (a) at left, the filter was fed a 1kHz squarewave without compensation. An input squarewave amplitude of 4 divisions peak-to-peak comes out of the filter at almost 6 divisions p-p. In oscillograph (b) at right, the same squarewave signal was routed through the compensator circuitry.



(a) <u>Figure 11</u> (b)

Low-Pass Filter Squarewave Response At 1kHz; Uncompensated (a), Compensated (b)

For a more detailed explanation of this overshoot compensation technique, the reader is directed to U.S. Patent No. 4,737,725.

A final "safety" clipper verifies the effectiveness of this method of filter overshoot compensation. Though the clipping circuit <u>follows</u> the 10kHz filter, analysis of the output spectrum during normal operation with program material shows that the NRSC stopband is never compromised by the final clipper.

A Block Diagram of the complete NRSC Processor is shown in Figure 12.

<u>SUMMARY</u>

This paper has described an audio processor developed for quick, simple and effective implementation of the NRSC-1 standard for program preemphasis and audio bandwidth restriction in AM broadcasting. It has been suggested that an audio processing device such as this, which satisfies the NRSC-1 specification, will almost guarantee compliance with the more recently proposed "NRSC-2" standard. NRSC-2 specifies bandwidth constraints in terms of the transmitted (RF) signal.

Factors such as transmitter linearity, antenna bandwidth, etc. will determine whether, in fact, NRSC-2 compliance can be guaranteed by NRSC-1 implementation. This has proved the case in many, if not most, of the broadcast installations investigated.

Broadcasters who have implemented NRSC-1 are unanimous in its support, citing reduced "splatter" interference and better qudio quality, even on present narrowband radios. NRSC-1 is a first, very easy step toward improving AM radio.

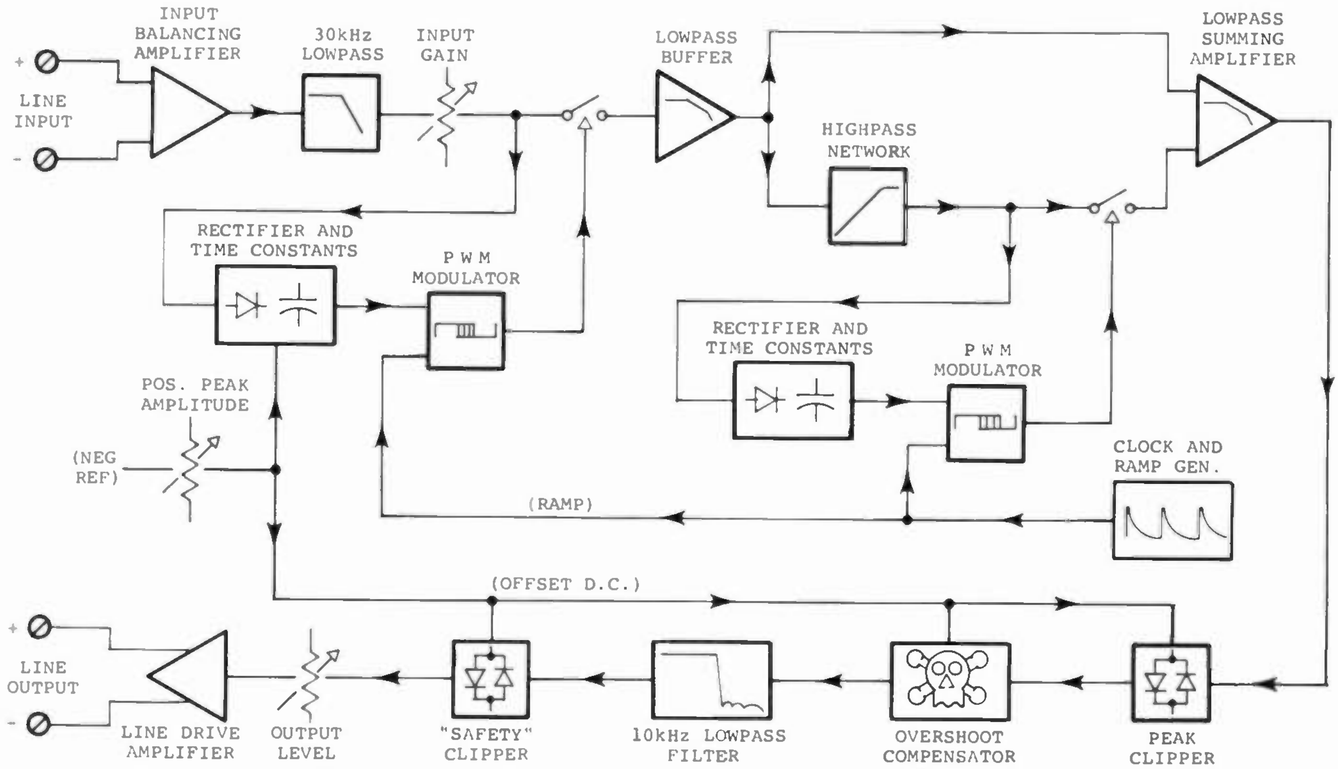
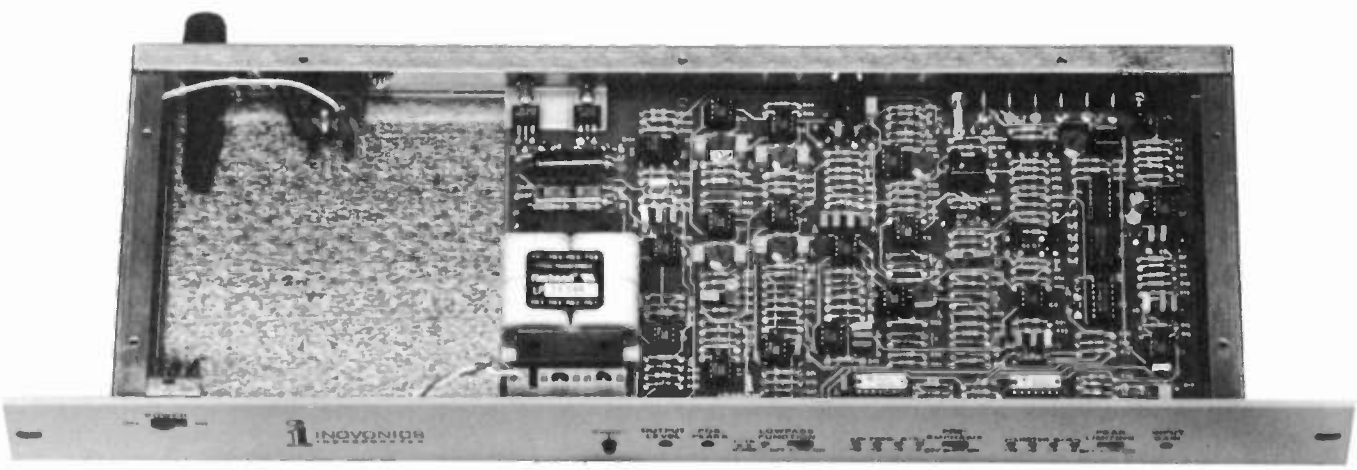


Figure 12
Block Diagram - NRSC Processor



Top View - NRSC Processor

THE AUTOMATED LIBRARY SYSTEM

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This paper deals with new concepts of television station automation using an automated tape library system as the central focal point for all station automation and operations.

For many years, television station automation has centered around the Master Control Switcher and individual machine controls. The Master Control Operators focus was the switcher and all of the elements that it controlled.

Centering the station operation on the Master Control switcher occurred because television stations used many sources including film, slides, tape, still stores and other inputs that had to be manipulated and timed by the Master Control operator.

In the past few years, film and slides have generally been eliminated as regular sources for station programming, commercial matter and public service spots. Tape has become the medium of choice for almost all origination within the television station.

Many of the older two inch tape cartridge players have served out their useful lives. For several years, television stations have been looking to replace these machines with a more highly automated, larger capacity tape systems.

The LaKart Corporation was formed several years ago to provide automation systems using existing cassette players with manual "insertion" of the cassettes. The concept of the automation system was to control not only the tape machines themselves but to provide automated switching, special effects, machine control and traffic system interface all in one computer system.

LaKart manual insertion systems are now operating in over one hundred stations and cable systems throughout the world.

Approximately two years ago, the LaKart Corporation embarked on a project to design a robotic library system to feed cassette tapes to a standard Lakart automation system. What resulted was the ALS, Automated Library System that we will discuss here in detail.

In the design of ALS, we not only looked at typical station automation with the Master Control Switcher but at the concepts we had instituted in the original LaKart product as well as the great increase in tape usage for on air presentation.

Basically, the design team started with a powerful central computer and "smart" machine controllers that could control all facets of a television station "on air" operation. The concept was to set out to make the Automated Library System the central operating system for the station, still having the flexibility of allowing the user to operate the ALS with a traditional Master Control Automation system if so desired.

To allow ALS to perform all the necessary functions, the design had to accommodate the following areas:

1. Interface with existing traffic systems to receive daily operating logs.
2. Control tape machines internal and external to the library.
3. Control routing and/or Master Control switchers.
4. ALS to be controlled by existing Master Control switchers
5. Control graphics and still store equipment.
6. Control satellite receive/send equipment.
7. Perform record and program delay functions.
8. Perform station logging and "upload" as run log information for billing purposes.

In general, ALS was designed to control any part of the present station on air operation and automate some areas that had not before been addressed in station automation.

When the design team looked at the total concept of an automated library system, they asked whether the library design was just to be a "bookmobile" dispensing previously recorded material or would the system be an all purpose library that not only stored short pieces of material, ie commercials and public service announcements, but played programs and recorded material. We then asked ourselves whether we should offer the ability to automate anything and everything that was associated with tape functions. We quickly answered in the affirmative.

In visits to many broadcasters, we learned that a great deal of time and labor was being spent recording incoming program and commercial material. There was no real automated system offered to cover this part of tape operations.

We believe that to be a true library system that we must have ALS cover all aspects of tape, both playback and record.

With these thoughts in mind, a two fold design approach was initiated. The first part covered ALS software to perform all of the record, playback and control functions required for the library. The second part was to design a robotics system that would handle any popular cassette tape format, both now and in the future, using unmodified cassette machines of the users choice.

The present system design will operate in the following tape formats; Beta and Beta SP, MII, U-Matic, S-VHS and D2. Other formats may be accommodated and the system can be modified in the field to change formats.

Multiple formats can also be operated within the library such as a mixture of Beta SP and U-Matic or Beta SP and D2. This would usually be done with a separate carrousel and machine rack for each format. The same robot system handles both formats.

In the software design, the already existing LaKart software controlled cassette and reel tape machines, performed switching functions and interfaced to traffic systems.

On top of those software modules, the design group added robotics control, library location control, cassette identity, library database, library purge functions and other library control functions.

Software features were designed and added including satellite and delay recording, satellite position control and automated compile from library tapes to an external machine for preparation of backup tapes.

The design of the robotics took many turns before a library and robot system were finally selected. Several different physical configurations were considered including a "video wall" with a linear x/y robot, a variation with moveable walls and a central robot with carrousels surrounding it holding the cassettes. The later method was finally selected for several reasons.

1. Robot base was stationary with minimum of travel.
2. System could be placed in a small footprint.
3. System could be designed to hold up to 1500 tapes in three carrousels and two tape machine racks holding up to twelve machines.
4. System would accommodate a removable loading cart.

The final configuration is shown in Figure 1 and 2.

The ALS robot is multi-axis with a multiple axis arm and gripping device, commonly known as an "end effector" in the robotics industry.

The robot rotates 360 degrees and has a vertical travel of 65 inches.

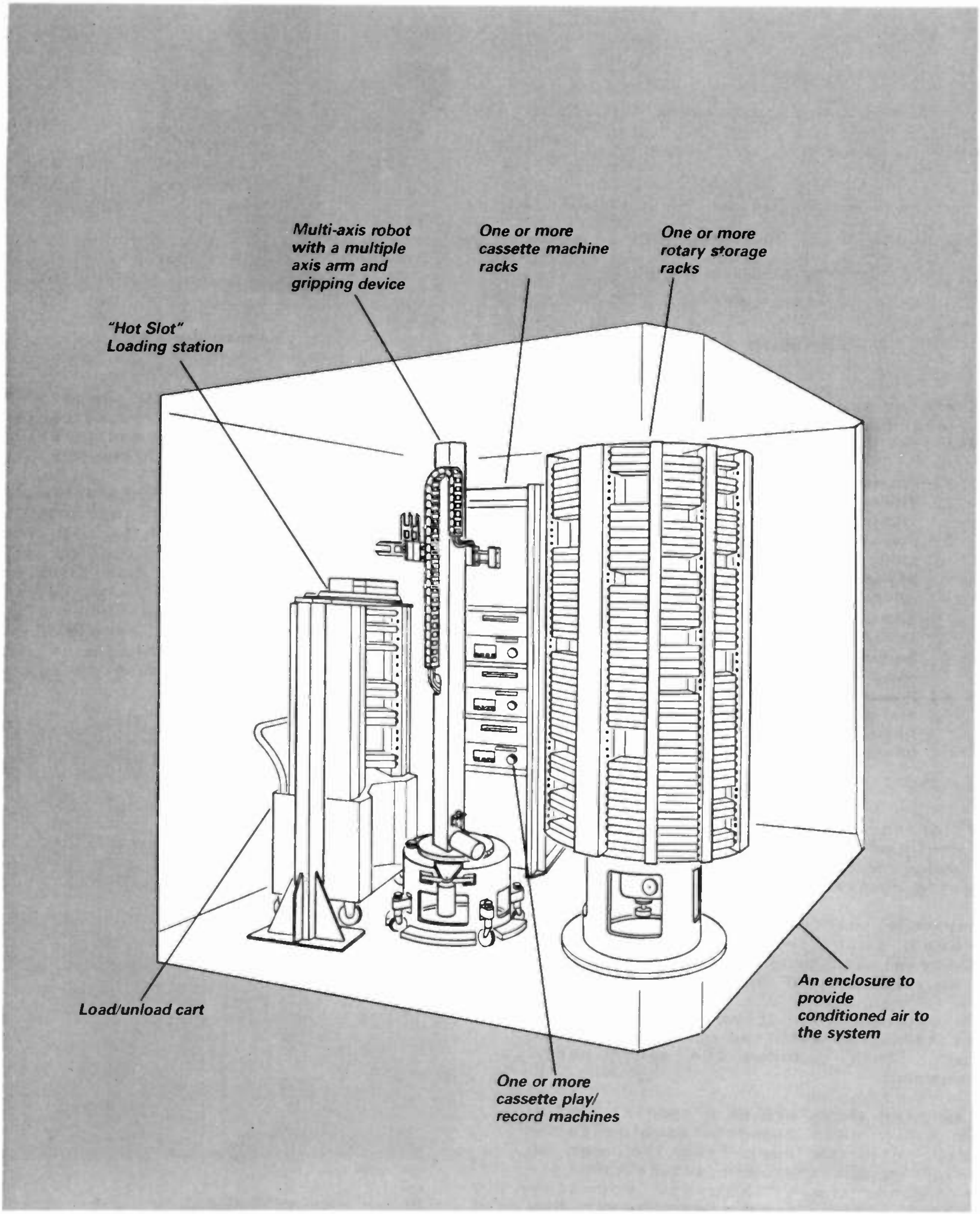
The real heart of the system is the robotic end effector that travels up and down and rotates on the central robot shaft The end effector has two tape cassette grippers to double the speed and reduce mechanical motion time in exchanging cassettes. One gripper retrieves a cassette from the carrousel. The robot moves to the assigned transport. The end effector rotates 90 degrees, retrieves the ejected tape, rotates back 90 degrees and inserts the new tape.

The end effector is shown in Figure 3.

The rotational and vertical motion of the robot is handled with DC servo motors. The entire operation of the arm-end effector is pneumatic.

Much consideration was given to the design of the end effector whether to use

Figure 1



Multi-axis robot with a multiple axis arm and gripping device
One or more cassette machine racks
One or more rotary storage racks
"Hot Slot" Loading station
Load/unload cart
An enclosure to provide conditioned air to the system
One or more cassette play/ record machines

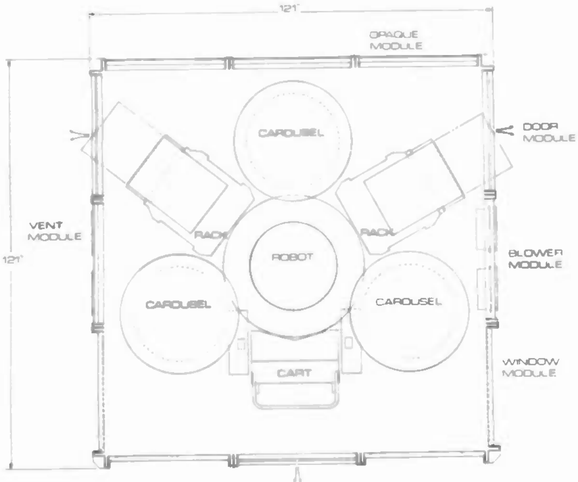


Figure 2

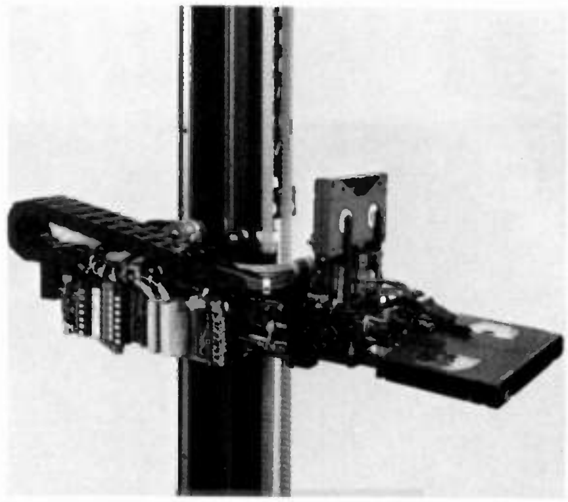


Figure 3

electric or pneumatic actuators. In the end, all factors pointed to the use of pneumatics.

1. No magnetic fields created.
2. Pneumatics proven in robotic industry.
3. Chance of total failure much less than electric components. Pneumatics start to leak and warn of failure. Electrical components tend to short without prior indication.
4. Pneumatics are less expensive and easy to replace in the field.
5. Pneumatic components are lighter and give the tape gripper a true "feel" to insure that the cassettes are handled with care.

The carrousels that surround the robot are designed to hold up to 500 cassettes. The amount varies by the type and size of cassette desired by the user.

A carrousel can be configured to handle several sizes of cassette so that commercial and program length cassettes can be accommodated in the system.

Each carrousel is driven by a DC servo motor identical with the ones used on the robot. This reduces the spare parts requirement.

The machine racks are of a special design that allow each cassette machine to be loaded into the rack from the rear on special guides that are substituted for the rubber feet that are normally supplied with cassette machines. The machines are locked into place. No further adjustment is required to replace a machine nor are any other modifications required of a stock cassette machine.

To further facilitate the installation of cassette machines, a small elevator is located on the rear of each rack so that a cassette machine can be pulled out, lowered to a comfortable position and either replaced or adjustments performed. This system also helps in the routine head cleaning that is required of cassette machines. A detail of the rack/elevator system is shown in Figure 4.

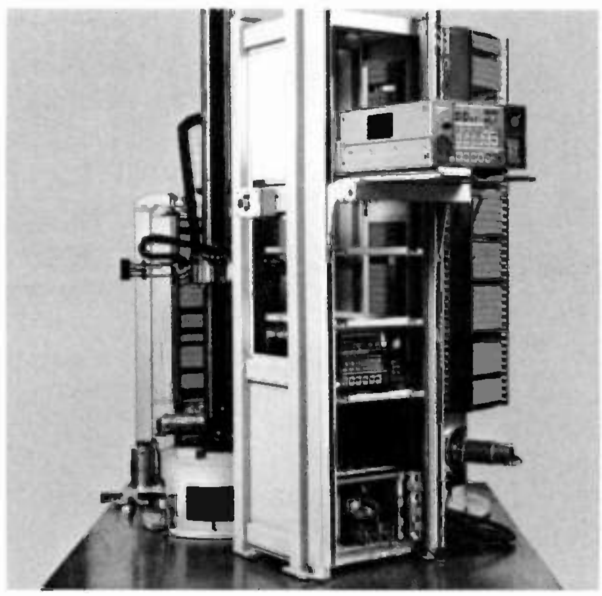


Figure 4

To load or unload cassettes into the library system posed a major problem in design and station operations. The first thought was to provide a fixed area of slots where cassettes could be introduced into the system. After discussing the concept with several potential ALS users, it became apparent that tapes would travel back and forth from the active library to the inactive library. This usually meant loading a cart with cassettes and moving them from one place to another. The design group decided that the loading and transportation cart should be one in the same to reduce labor and save time in the process.

The ALS loading cart will hold up to 50 cassettes of varying sizes. The cart can be loaded in the inactive library, wheeled to the ALS system and left for the robot to load and unload, depending on the tasks required. Several carts can be kept by the station to facilitate cassette movement.

The entire library system is enclosed in an attractive housing that can be fully air conditioned separate from the station system to insure a clean environment for the cassettes and robotics. See Figure 5.

The computer systems used to control the robotics and the rest of the library operation are based on Motorola 68000 CPU's with various enhancements added to allow for fast, multi tasking operations.

It is important to note that an great many tasks are being performed by the software in this system. Several tasks have to be performed simultaneously which requires high speed operations for software and hardware. For example, one time change in an execution log requires that the entire log be retimed. A operator cannot wait for seconds for this task to be done. The software must perform the task in milliseconds while other tasks are running.

Tape preparation and introduction into the library was designed to be as simple and quick as possible to reduce labor necessary to operate a large tape library on a continuing basis.

In tape preparation, SMPTE time code, audio and video are recorded on a cassette. For each cassette, a unique 8 bit reel number is recorded in the user bits of the time code. The reel number is entered into the tape preparation computer terminal at the time of recording.

Once the operator has made the recording, the beginning point of the video is located and the "mark in" key is activated on the terminal. This notes the time code location. The same is done at the out point. This information, together with the name of the commercial and the "house" tape number is entered into the database from the terminal.

Multiple cuts may be put on one cassette in this system, reducing cassette inventory and library size. This is done simply by the use of multiple in and out points providing one unique set of time code addresses for each cut on the cassette.

A bar code reader is run across bar code that has been applied previously to the cassette. The bar code number is also stored in the database. The bar code resides with the cassette for its entire life in the library system. Sequential, crack and peal bar code is used that is readily available and inexpensive. No bar code printing equipment is required.

At this point, the database has the following information concerning the material recorded on the cassette.

1. Name of material.
2. Tape house number.
3. Tape reel number.
4. SMPTE time code location for start/stop.
5. Cassette bar code number.

The cassette is introduced into the library system through the loading cart. The robot scans each cassette looking for the bar code number on the cassette. The bar code is matched with the reel number already in the database. A bin number is assigned by the computer and the cassette placed in that bin. The bin number is also placed in the database with the appropriate cassette reel number.

Each time a cassette is played, the bar code is counted to build a usage history for each cassette in the system. This history may be printed on a regular basis to review tape passes and remove high use cassettes before failure.

Typical operation of the library system would have a traffic system download the days log. This creates an execution log in ALS. This is the log that actually runs all the on air events.

The execution log not only has events within the library system but switches to other sources such as satellite, external tape machines, still stores, graphic generators, special effects and keys. All of this would be operated and switched by the ALS computer system.

A completed log of the days events would be saved for posting to the traffic accounting system or that information could be sent directly to the traffic computer for reconciliation.

Other operations within the library system include various record and compile functions as well a library management or purge functions.

For record functions, the system can perform a program delay of five minutes or more using two to four cassette recorders. This feature is very useful for delay of news or sports events, eliminating costly manpower that is required now to perform program delays.

Automated recording of incoming satellite feeds can also be done by the system. Data may be entered into a record scheduler for at least a week in advance. The recording will be started at the time scheduled including control of the appropriate input routing switcher to receive the right program material.

On a custom basis, software can be provided that will control a steerable earth station to receive the proper satellite for the requested recording.

After a recording is made, it is placed in the library and the database notified of its location. This means that incoming programming can be recorded and played back without any human intervention.

The software is provided with a powerful compiler system. This will take any number of events and compile them in sequence on another cassette or external reel or cassette machine. This process can be used as a backup for high revenue areas.

Both the main and backup machines can be started simultaneously by the ALS operation to insure that critical material is not missed.

With a tape library system in operation at a television station as described, manpower is materially reduced as many functions previously performed on a manual basis are automated. On air presentation with minimum error is achieved to eliminate costly makegoods of lost commercials.

The ALS system is designed for all sizes of television operations as the system can be sized from 500 up to a maximum of 1500 cassettes. Further, up to twelve tape machines may be accommodated. Only four are required for normal playback operation with 10 second throughput. Other machines can be placed in the system for various record and backup functions.

The system can also be designed to feed multiple channels from one library source. For example, four channels can be fed simultaneously using three machines per channel. The software is configured to handle separate execution logs for each channel.

In summary, the ALS Automated Library System is designed as a total television station automation system with the library at the heart of the operation. The traditional Master Control switcher can become obsolete if the full power of the ALS software is utilized.

THE DESIGN AND IMPLEMENTATION OF A THREE CAMERA STUDIO REMOTE CONTROL SYSTEM

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ABSTRACT

Starting in March 1988, WPIX in New York began operating its three Ikegami HK-322 studio camera's via a Vinten Microswift Mark II camera remote control system. Since the system went on line, it has been used seven days per week televising approximately four shows per day (a daily community affairs talk show and three nightly Independent Network News (INN) newscasts).

This paper will discuss the various technical, production and economic issues associated with the WPIX installation, with attention paid to the specific design specifications and the solutions implemented for proper on-air operation of this device.

BACKGROUND

WPIX had been seriously considering the use of studio camera remote control since 1984. We felt quite confident that the available technology was capable of reliably supporting the actual operation of studio cameras, but we needed to convince ourselves that the systems being manufactured were capable of controlling these operational functions without hindering the smooth "flow" of the production process. Unfortunately, we could not afford to experiment with remote control because our one studio is used every day to produce taped and live programming. We needed a system that could be introduced into the operation in a matter of a few days, with a minimal amount of disruption to our studio production schedule. In analyzing the camera remote control products being manufactured, we were looking for a combination of qualities - full operational capabilities, the ability to provide various modifications deemed necessary for our operation, sound mechanical and electrical architecture with quality construction, fast installation and affordability.

From the beginning, the decision to implement camera automation was a bit more than an equipment purchasing decision. There were concerns and questions about how it would effect the quality of our productions, and there were feelings and emotions which revolved around the fact that people were going to be replaced by machines. We needed to make sure that we were sensitive to the feelings and concerns of the people and departments involved in producing our shows, so it was considered quite important that all of their questions and comments be discussed and addressed.

It was apparent that in order for this proposal to succeed, the support and confidence of the senior officers of the company were needed. With this in mind, the engineering department put together a thorough proposal which sought to address the many concerns involved and pointed out the advantages and disadvantages of going with remote control.

Among the questions we had to resolve was rationalizing why we really wanted studio camera remote control - what it would do for us. It was important to address the point that we did not feel remote control posed any insurmountable risk to our live on-air product. We were careful to avoid any impression that this was an experiment, or an attempt at another form of automation that had some "potential". We communicated the confidence we had in the technology and the decision to proceed was a prudent engineering and business decision utilizing existing technology applied in a viable, efficient manner.

Before the purchase of our system, the president of our company, the vice president in charge of news, the executive producer of news, the vice president of engineering, and the engineering supervisors, all looked at various products being offered for studio camera remote control. The potential was understood. The unknown was the practical application, how would it integrate with our equipment and would it work in our particular production environment?

Since there were a number of people, departments and productions potentially effected by this decision, we decided to implement some team building techniques. Now one might ask the question, what does team building have to do with remote control cameras? The answer is simple - quite a bit! It was considered vital that from the day we decided to go with remote control, nothing was going to be a mystery.

The installation of the remote control cameras was going to be an open, participative process in which those involved would actually work together in designing, constructing, training, implementing and operating the system.

This included department heads, managers, maintenance engineers, operational engineers, directors, and talent. We operated on the premise that "no one had a monopoly on ideas" and we took any and all suggestions seriously, weighed them as a team, and came up with a rational approach to implementing a working system that is conducive to our particular television station. The result of this team building was a system designed by consensus and implemented with a sense of pride and ownership. Perhaps it sounds like a cliche, but this aspect of the project was what made it most enjoyable - working with people who cared, wanting to make it the best possible installation, and having real satisfaction that all had substantially contributed to the finished product.

ECONOMICS

One of the important components of any engineering decision is the associated economics. In recent years, the increased competition in the broadcast industry has caused a greater emphasis on overall station efficiencies, return on investment (ROI), and productivity. Broadcast Engineering managers have been asked to implement and utilize new technology to improve operating efficiencies in their station by decreasing costs while either sustaining or improving quality. Remote control cameras can improve quality while reducing costs and thus are an ideal system to consider from an economic viewpoint.

In our installation, the person doing video is also operating three cameras. Since each camera was previously manned, one person is now doing the work that previously required four people. This is obviously an improvement in overall efficiency and productivity.

A simple return on investment analysis quickly provides an economic justification for remote control cameras. Assume that the three cameras are manned for one shift a day, seven days per week. This requires twenty one individual shifts. Assuming efficient scheduling and a five day work week, it would take twenty one divided by five, or four plus people to man the cameras. If each person were paid $100 per shift, the cost would be $2100 per week which is over $100,000 per year. If an installation of remote control cameras cost approximately $200,000, it would pay for itself in a little over two years. With this sort of labor costs and equipment costs, it is easy to cost justify the ROI of remote control cameras based on labor savings alone. In some large market stations, the cost of manned cameras is higher than $100 per person per shift. In these cases the economic justification is even stronger. Even in stations where the cost is less than $100 per person per shift, the justification may still be valid if based on equipment life of 5 to 8 years. Naturally if there is more than one shift per day of studio operations, remote control cameras would show a higher ROI more quickly.

Another economic benefit of remote control cameras is they allow for consideration of productions that might have been too costly to do if the cameras had to be manned. For example, early morning news updates or late night news shows on weekends that might otherwise be too costly to produce, now become economically feasible due to the cost savings associated with camera remote control. Updates such as these could of course be done with unmanned cameras taking fixed shots, but there is then the resultant compromise of production values.

Presently we can control the cameras from the studio switcher, a Grass Valley 300. As we look to the future, there may be further efficiencies found by having the cameras controlled by a newsroom computer. Also, cameras at remote news bureaus could also be controlled from the station via a modem link, again resulting in cost savings.

In summary, the economic benefits gained by the installation and operation of remote controlled cameras are so significant that they should be reviewed by every broadcast engineering manager.

TECHNICAL

It was our feeling that the camera remote control products being manufactured by a number of companies were technically capable of controlling the various parameters necessary for proper operation of a full size studio camera. Specifically, the device had to pan, tilt, zoom and focus a full size studio camera with associated lens and teleprompter in a smooth, coordinated manner. In our particular case, the teleprompter monitors have 19" screens and are quite heavy. The overall weight of our cameras with lens and teleprompter is approximately 240 lbs., so moving the camera in a "smooth, coordinated fashion" was going to test the limits of any system. Additionally, the system should be able to perform these movements in a real-time, trimmable fashion from a control point or from pre-programmed recorded shots which could be easily recalled and accurately replayed at any time, while still retaining full operator control and/or override.

Since we have one studio that is used for a number of different productions, it was an important requirement for our system to have the ability to quickly and easily transform from an automatic operation to a manned operation. Optimally, we would want it such that a camera operator could walk up to each studio camera and flip a switch to take over normal manual control of that camera. This ability would allow us to have any combination of automated and manually controlled cameras deemed as necessary for the specific production. For example, if we were to do a live telethon from our studio, we might want to have two cameras manned by operators and one camera operated via remote control, or perhaps two cameras operated via remote control and one camera operated manually.

The ability to go manual at any given moment also provides a significant measure of backup. If the system were to incur some catastrophic failure, the ability to flip a switch and go manual meant that our cameras could still be operated as they had been before remote control. I cannot stress the importance with which we viewed this capability, and quite frankly, I don't believe we would have gone with remote control, had this feature not been available in a reliable, working fashion.

Careful consideration was given to the long term durability of the device given the daily stresses associated with using and moving cameras throughout the studio. We wanted a system that would stand up to the rigors of every day usage for at least ten years. Granted, this is a tough specification to prove when considering a new piece of equipment, but it suffices to say that we were seeking a system that had as design criteria, high quality components and construction, with a high mean time between failure, and minimal maintenance. It should, if possible, utilize fairly well known technology.

We were not looking to reinvent the wheel, but simply to employ a device that had been proven in use and was flexible enough to be modified for our particular situation. In essence, quality design with quality hardware applied in a user friendly manner.

Last, but most importantly, the system had to incorporate specific safety measures which insured that upon power up the cameras would not move from their position, and the cameras moved only when specifically requested to do so by an operator. Also, if the camera were to come into contact with an object, there would be a feedback circuit that would sense the obstruction and stop the motion of the camera.

WPIX System Specifications

As previously mentioned, there were overall system features which were prerequisites for considering camera automation. After it was determined that these features were available, we began talking seriously with manufacturers about our particular system requirements. WPIX chose the Vinten Microswift Mark II system because it satisfied all of the prerequisites we had established and we were able to come to financial agreement on system cost at a price that was acceptable to both companies.

We began talking with Vinten Equipment Inc. in early December of 1987 about how we envisioned the capabilities of WPIX's system. After a few meetings and a number of phone calls we agreed on a system which had the following technical specifications:

The Mark II system purchased is a four camera system populated to presently control three cameras. We have the ability to expand to a four camera system by simply purchasing the necessary servo control module printed circuit boards for a fourth camera. The system was designed to accommodate future growth.

The servo pan and tilt heads on which the cameras are mounted have 150 feet of control cable connected between the camera head and the servo control modules located by the operator control panel. This allows for full manual movement of our cameras and pedestals throughout the studio. Our installation did not necessitate the use of automatic tracking of pedestals, and in fact, for our particular studio, we felt that automatic pedestal tracking would prove to be a limitation. Our studio presently houses three different sets on three different areas of the studio floor. It was our belief that by being able to move the cameras manually to any position on the studio floor, we were taking full advantage of our available studio space.

The majority of basic camera movement functions including pan, tilt, zoom and focus are inherent to the system. In addition to the camera control system, we also purchased three Vinten Fulmar pedestals. One of the pedestals is equipped with remote controlled height adjustment which proves quite useful in operation. At the time of purchase we did not fully understand how much this feature would add to the creative aspect of the system. The height control function is used on almost every show and adds a unique dimension to smooth on-air height adjustment, especially when combined with long zooms. It adds a smooth, swooping perspective to what would otherwise be a simple zoom shot.

Vinten offered three types of joystick controller for the control of the multi-axis of the cameras. One is a spring biased unit, another is a so-called stiffstick which utilizes applied pressure to the joystick and the last is the fluid-damped joystick. We chose to incorporate the fluid-damped joystick because of its smooth, sensitive control and good sense of axis separation.

The WPIX system was designed to integrate with the camera control units of our Ikegami Hk-322's. To this end, we requested that the Vinten operator control panel have the ability to control the iris and pedestal level of each chosen camera, as well as the gain and set-up of each camera's blue and red channels. One of our technical directors, Joe Casazza, suggested that we incorporate a set of switches that would, depending on their position, allow the operator the choice of either operating the CCU's manually from the normal Ikegami CCU control panels (which are located directly to the right side of the Vinten operator control panel) or allow the CCU to be operated via the rotary controls located on the Vinten operator control panel. There are a number of benefits to having this flexibility. With the switch in the local position, the operator has traditional access over any of the three CCU's. He does not have to choose which camera he wishes to access on the Vinten control panel, and he has full access to all of the various functions of the HK-322 CCU panel. However, if we were to have a studio production which made use of a number of different lighting arrangements in conjunction with different camera shots, the ability to have the CCU levels tied into the memory of the Vinten system is quite powerful. In tandem with the scene file capabilities of the Ikegami cameras, we are able to recall scene files for specific set locations and then customize the color balance and levels for every shot stored in each of our three cameras.

For storage and replay of shots, the operator control panel has a numeric keypad and a window with a preview and active area for number illumination. The keypad allows the operator to number each pre-set shot before pushing a button called "Store P" for storing the previewed shot to memory. Once the shot is stored it flips up to the active position of the illumination area. If the operator were to trim or adjust the active shot with the joystick, a tag appears under the illuminated active number saying "Off Shot". This new, corrected position can be re-stored as the original number by hitting a button called "Store A" for store active. Since talent can shift in their seats, the "Store A" button is used often in practice. This function allows the operator to make necessary trim adjustments to the recalled shot and then re-store the shot under the same number without having to renumber the shot. This function is a significant time saver.

Pre-set shots are stored in RAM memory, and thus, could be lost if AC power were interrupted. Vinten has equipped the memory boards with batteries so that interruption of power will not affect the stored shots. The expected battery life is approximately one year, therefore, timely change of used batteries is important.

We have suggested to Vinten, and I am of the understanding that they are working on a floppy disk back-up system that would allow the user the ability to down load and up load stored shots from a computer disk.

For numbering shots, Vinten employs an architecture we particularly liked. Each camera has enough memory to store approximately 1500 shots. However, we were concerned with the practical application of a system that used a numbering system utilizing large numbers. Remembering what shot 749 is on camera 1 versus what shot 387 is on camera 1 seemed cumbersome. Vinten's approach simplifies this by creating 15 "pages" of 99 stored shots for each page (shots numbered 1 through 99). In practice, we apply this by saying that page 1 recalls the stored shots for our 7:30PM newscast for all three cameras. Page 2 is the page of stored shots for our 9:30PM network newscast and Page 3 is for the 10:00PM local newscast. We then can say that shot #1 on camera 1 is a close-up with DVE over the right shoulder on all three pages. It is easier for the operator to remember that shot #1 is the same type of shot for each of the three different shows.

Once shots have been stored, they can be recalled either in a cut mode or in a time controlled fade mode. On the operator control panel, the "Cut" button will cut from the shot that is presently on camera to the next shot in the shortest possible time. The published specification for the Vinten Mark II system for maximum pan and tilt rate is 60 arc degrees per second. In use, we have found this rate of speed to be more than adequate. A "Time Fade" shot is a shot that takes a pre-set time to transition from one shot to the next shot. A zoom is an example of a time fade shot. Time fades can also be used for rack focusing the lens and for adjusting the height of the pedestal on-air. Time fades are governed either by the time values stored with a pre-set shot (this is done when the shot is initially stored by hitting the "Stored Time Fade" button), or by the setting of a time fader bar located on the operator control panel. The time fader bar can be used in real time to either slow the movement down or speed it up while it is occurring. The time fader bar can also be used to stop the shot in the middle of its movement before it reaches its final destination. In either case, the actual time fade movement is profiled to produce very smooth start acceleration and stop deceleration for the on-air move. The precise repeatability of "Time Fades" assures an accurate move each time a shot is recalled.

There are two other features in our system which can be used in situations where the critical timing of a certain sequence of shots is necessary. The "Sequence" button causes shots to be automatically recalled in numerical order (1,2,3,4,5 etc...). If a shot position does not have a movement stored within its memory, the "Sequence" function skips over that number and recalls the next sequential number with recorded data. The "Link" button gives the operator the ability to construct his own sequence using any order of numbered shots. The operator links the out of sequence shots before hand, and when they are recalled they are played back in that predetermined order. This is very helpful for predictable, rehearsed situations, although we have not found in practice that it is all that necessary to use, due in large part to the overall ease of storing and recalling shots in general. In all of the above mentioned modes, the operator always has control over the trimming of the cameras.

The remaining control buttons are system software controls. The "Command" button allows the operator access to a number of "user defined movement" modes. For instance, the operator can set limits on the range of the pan, tilt, and height of the camera. Again, we find this a very useful feature. Some of the lights in our studio hang fairly low from the ceiling; by setting a height limit on the adjustable pedestal, we can insure that the camera will not bang into any lights. Similarly, the cameras are sometimes positioned close to a wall or door - the software limit capability allows the operator to set a pan limit which prohibits the camera from panning into the wall or doorway. Another user defined feature we use is setting the sensitivity of the joystick controller.

Since each operator has their own way of operating, the ability to customize the "feel" of the system to each user provides added flexibility. An example of this is the ability to change axis direction of the joystick. Some operators like to pan the joystick to the left and see the camera pan right (much like the manual operation of a studio camera); while other operators like to see the camera pan left when the joystick is moved left.

These might seem like minor features, but in practice, we have found that the more user friendly a system is, the more willing an operator is to accept it. This system's wide array of capabilities tends to answer many of the questions operators pose.

The last three technical features of WPIX's system are modifications which we requested and were developed by Vinten's engineers. When we started to lay out how we wanted the facility to be built and operated we came to the important realization that the Vinten system is not controlling the switching of video output from the cameras. We did not request that it control the switching of video since we have a Grass Valley 300 switcher which performs that function quite well. But, the problem we saw was that if the operator was looking at a color monitor in front of him for critical control of the video aspects of the Ikegami cameras, as well as the control of the movement of the three cameras, the video output from those cameras was not going to follow the operators commands to the Vinten control panel and thus would not appear on the monitor in front of him. The solution we came up with was to take a Grass Valley Ten X/L 10x1 switcher and have the video output from each camera routed into our Asaca 14" color monitor. The Vinten control panel would then operate in parallel with the Grass Valley Ten X/L and when an operator chose a specific camera on the Vinten control panel, that associated camera video would also appear on the color monitor. This was accomplished by Vinten through the use of binary coded decimal (BCD) closures.

For tally, we tied the Vinten system into the tally system of the Grass Valley 300 switcher so that a tally signal would illuminate on the Vinten control panel whenever a chosen camera was on-air; in addition, we also constructed a small switching matrix that illuminated tally lights over the individual camera monitors. The purpose of this was to reinforce to the operator which camera was on-air, thus reducing the chance of mistaken movement of an on-air camera.

The last modification we incorporated into our system was to integrate the effects memory or E-MEM capability of our GV-300 switcher into the Vinten camera control system. The purpose of this was to allow the E-MEM to store and recall stored shots from the Vinten. What this means is that the GV-300 video switcher can recall stored shots and control the movement of our cameras. We discussed this modification at length with Vinten engineers and laid out a plan that would incorporate some important rules for operation. First, we wanted a button to be located on the Vinten control panel which had to be activated by the operator in order for the GV-300 to control the cameras. This button is called "Auto Mode". The purpose of this is that we did not want the GV-300 controlling the cameras unless a specific command was given to it to control them. By the operator making a conscious decision to push and illuminate the "Auto mode" button, he is fully aware that the GV-300 will be controlling a given camera. There are Auto Mode buttons for each individual camera such that any combination of cameras can be operated by the GV-300. In addition, there is an overall system Auto Mode button which gives the GV-300 control over all three cameras. There is another button called "Auto Select". When activated in combination with the Auto Mode, the Auto Select function automatically parallels the camera chosen by the GV-300 E-MEM to the Vinten trim joystick. This allows the operator to trim the cameras chosen by the E-MEM without having to manually choose that camera on the Vinten operator control panel.

The last rule we incorporated into the E-MEM interface was an inhibit function for any camera chosen on the GV-300's program bus. We did not want the E-MEM to be able to access or move a camera that was on-air. If camera 1 is chosen on the program bus of the GV-300, the Vinten interface will not allow the E-MEM to automatically move that camera. This was done to eliminate the chance of the GV-300 operator accessing an incorrect E-MEM and inadvertently moving an on-air camera.

In practice, the E-MEM interface can be used in a talk show format when there are two or three people discussing a topic. This situation requires quick cutting between cameras covering the people talking. The E-MEM control of the cameras allows the GV-300 operator to choose which camera is moving much as he chooses a cross point on the video switcher. The result is a minimal amount of time expended on the positioning of cameras covering the action. The strength of this interface is that it allows the cameras to be moved without an operator manning the Vinten control panel. When an operator is at the control panel, he has complete override control of the cameras at all times.

PRODUCTION

The integration of the camera remote control system into the WPIX production environment was the critical component to the success or failure of the project. We could have purchased the best piece of technology available, but if it did not integrate successfully into our production environment, it was useless.

As mentioned previously, a team approach was used for all problem solving. We set out to design an operator's console that would be ergonomically correct for the operator, while at the same time capable of housing each piece of equipment necessary for performing the job. After many discussions and hours of planning, a semi-circular console was decided upon. The console would have the Vinten operator control panel in the middle, with the Ikegami CCU's and Master Control Panel (MCP) to the operator's right side; an intercom communications panel and routing switcher panel would be located on the left side of the console.

Since one operator was going to be operating a number of pieces of equipment, equipment placement had to be such that each control was well within an arms length distance. The semi-circular concept served this purpose well by allowing the equipment to wrap around the operator, thus keeping any reaching distance to a minimum. It was felt by our team that a highly ergonomic design would enhance the system's ability to easily integrate with the production environment. In reality, this has proved quite valid.

In January of 1988, two months before the Vinten system would be installed, we asked Vinten if they would be kind enough to set up a training class for our people. They graciously provided us five days of training with a full mockup studio with camera remote control system located at their Long Island, New York offices. Our purpose for doing this was two-fold. First, there were jokes and rumors going around about "roaming robots" etc., and it was our feeling that this would be a good opportunity to familiarize the operators with the technology while helping to demythologize the aura that seemed to surround the technology.

More importantly, the chance to train our operators and directors before the actual installation date allowed them an opportunity to understand how it ran, while also enabling them to ask questions and make comments. The resulting comments proved very helpful in smoothly implementing the system. The operators and directors came away from the training sessions with a respect for the capabilities of the system and a feeling of confidence that the system could easily handle the job of operating television cameras.

After the training sessions were completed, we took the knowledge we had gained and brought it back to WPIX to determine exactly how the system would integrate with each of our specific shows. One of our directors, Peter Pontillo, was instrumental in this effort. Pete helped in blocking out our entire studio for camera placement. We borrowed an approach used by the British Broadcasting Company. Each camera was color coded - camera one pedestal has red tape around its base, camera two has green tape, while camera three has yellow tape; then we proceeded to mark with appropriately colored tape, outlines of the base of each camera's pedestal on the studio floor. These color coded outlines are located throughout the studio, allowing us the flexibility of moving each camera to a number of locations throughout the studio. It is worth noting that the placement of the cameras within the boundary of the outline is fairly critical. Obviously, if a camera is off placement by a couple of inches, the recalled shots from that camera will be off by a noticeable factor.

To solve this situation, we put one thin strip of tape on the center area of each pedestal and one thin strip of corresponding tape on the floor. This allows for accurate camera placement, and resulting accurate recall of stored shots for each show.

In terms of manning on the studio floor, before remote control cameras, there were three camera operators, a floor manager, a teleprompter operator and a lighting director. Since the Vinten system went on-line, there is a floor manager, the lighting director and the teleprompter operator. The teleprompter operator moves the cameras to their specific locations for each show. We do not move the pedestals during live shows, but we do move camera positions in between sections of pre-taped shows.

In organizing how the directors would communicate with the new "video control" position (this is the name we have given to the position of operating the remote control camera's and camera video), an interesting development occurred that was somewhat unexpected. Prior to the installation of remote control, the producer's "rundown" for a show was on written sheets of paper that were prepared by the producers and directors in advance of the show's air time. If changes were made to the rundown just prior to air time, these changes would by overwritten onto the original version. The overall result would be a somewhat messy rundown that at times could be difficult to read, and as such, errors might occur.

With the introduction of the remote control cameras, we introduced computer generated rundown sheets. The new rundown sheet had specific columns delineating different types of information. One column is dedicated for camera number and shot number information written as C3/23. Every show has a rundown sheet prepared which lists necessary camera and shot numbers for each camera/shot of the show. This information is then given to the video control operator prior to show time, and is used to pre-set the proper cameras and shots. An unexpected synergy resulted from installing the remote controlled cameras. With the computer generating the rundown, we not only get a more organized rundown, and as such, a calmer, more organized studio control room environment, but we also get the ability to change the rundown information on a moments notice. If news stories come into the newsroom just prior to air time, we can easily edit the rundown information and print out new rundowns, which in turn are distributed to all production personnel. In essence, the remote control cameras in combination with computers in the newsroom have enabled us to better organize our news show productions.

Once the rundown for a show is completed, just prior to the show, the director will communicate with the video control operator and recall most of that particular shows camera shots. This takes approximately two to three minutes if the shots are already stored in memory. If it is necessary to store new shots, the director and video control operator rehearse the new shot, assign it a number and store the shot accordingly. For our more involved productions, such as community affairs talk shows or specials, approximately five to twenty minutes might be set aside for rehearsal and storage of camera shots.

A typical dialogue during a show between director and video control operator would be the following:

> Dir.: Joe, camera 1, shot 3 please.(The video operator does not respond verbally, but presets camera 1,shot 3)
>
> Dir.: Joe, camera 3, shot 24 please.
>
> Dir.: Joe, slowly tighten up on camera three...that's good.
>
> Dir.: Joe, set camera 1 for a ten second time fade from shot 3 to shot 14.
>
> Dir.: Ready camera 1 for time fade.
>
> Dir.: Zoom camera 1.

In practice, the directors find that by only having to communicate with one person, they have confidence that what they are calling for will in fact occur. In many instances, because the video control operator has a rundown sheet detailing all shots, the cameras are set to their positions before the director formally calls for the shot, and the director simply confirms that the camera and shot number are correct. Additionally, the director is assured that a difficult shot, such as a 10 second time fade zoom with specific pedestal height, will occur as rehearsed every time it is called for, precisely the way it was rehearsed.

Needless to say, as with all intricate systems, there are limitations. Camera operators can give creativity and nuance to a camera which perhaps exceeds the capabilities of remote control systems. On those occasions when the dynamics of the studio setting require the creative abilities of excellent camera operators, we can put our cameras into manual, and use the cameras as we have in the past. But most of our productions are fairly predictable, fixed situations which can be blocked out once and played back consistently night after night. It is this type of studio environment in which the concept of remote control cameras makes sense.

SUMMARY

In summary, there was a certain "leap to faith" needed in making the final decision to go with remotely controlled cameras. A good recommendation for anyone contemplating remote control would be to visit a facility which is presently using it on a daily basis and determine if it will work in your particular situation. Each station has its own studio production needs, so it is important to determine up front if remote control cameras can continue to provide you the type of coverage your station is accustomed to. There are usually tradeoffs, but in retrospect, the trade offs have been pleasantly balanced by the capabilities of the technology. Speaking on behalf of WPIX, we have been quite pleased with the results.

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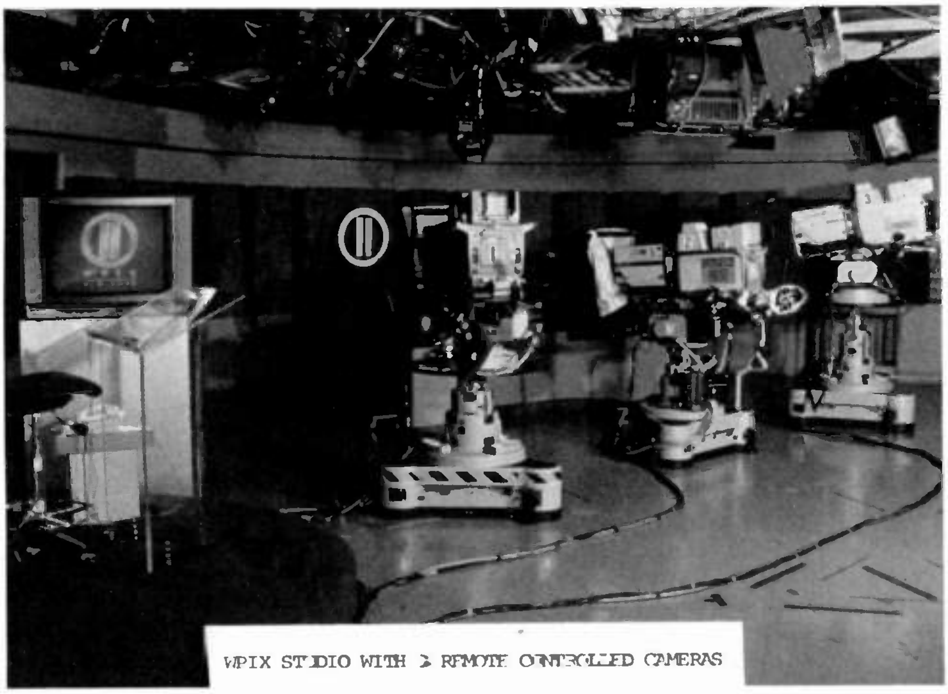
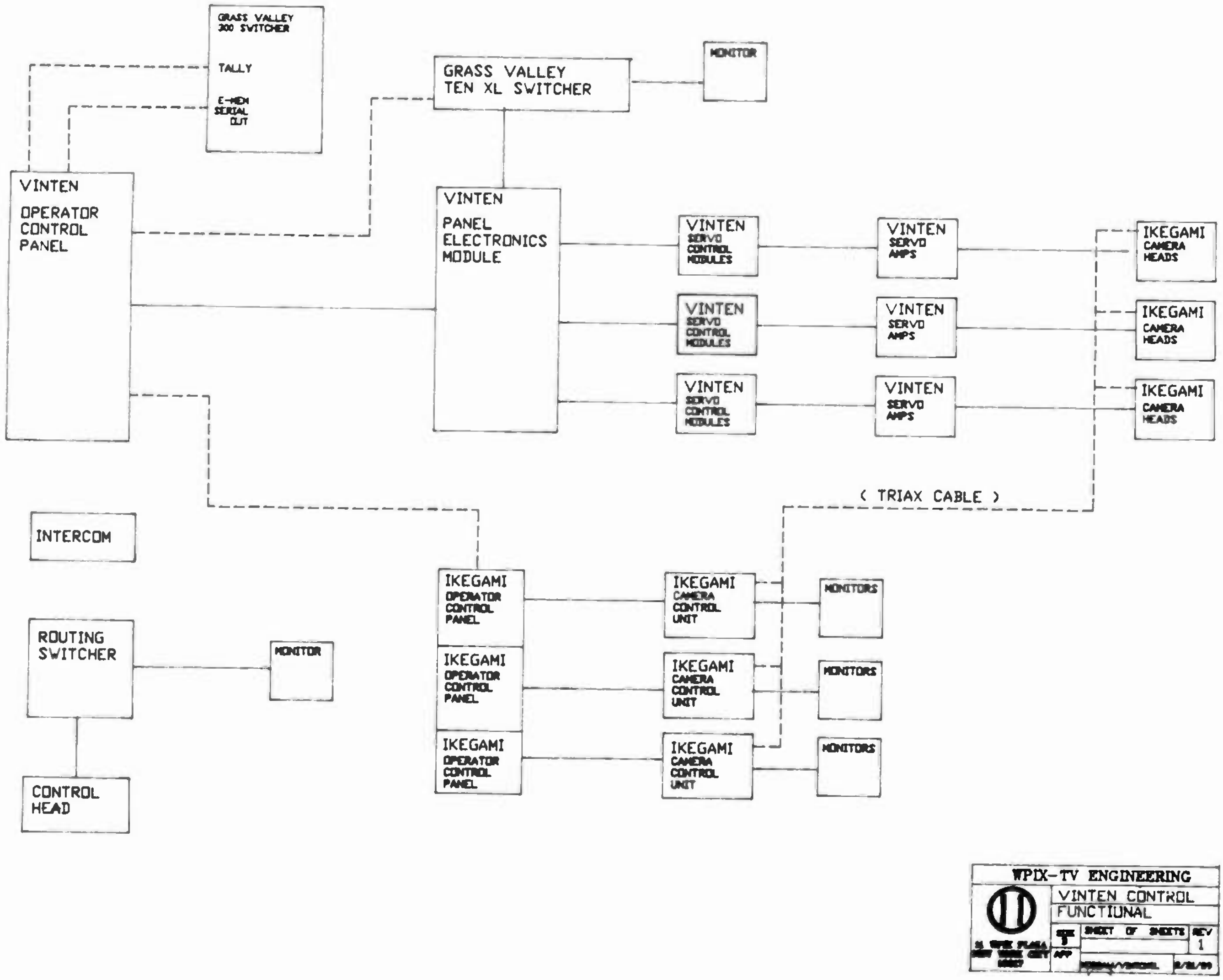


WPIX STUDIO WITH 3 REMOTE CONTROLLED CAMERAS

WPIX VIDEO CONTROL ROOM


GRASS VALLEY 300 SWITCHER
TALLY
E-MEM SERIAL OUT
GRASS VALLEY TEN XL SWITCHER
MONITOR
VINTEN OPERATOR CONTROL PANEL
VINTEN PANEL ELECTRONICS MODULE
VINTEN SERVO CONTROL MODULES
VINTEN SERVO CONTROL MODULES
VINTEN SERVO CONTROL MODULES
VINTEN SERVO AMPS
VINTEN SERVO AMPS
VINTEN SERVO AMPS
IKEGAMI CAMERA HEADS
IKEGAMI CAMERA HEADS
IKEGAMI CAMERA HEADS
(TRIAX CABLE)
INTERCOM
IKEGAMI OPERATOR CONTROL PANEL
IKEGAMI OPERATOR CONTROL PANEL
IKEGAMI OPERATOR CONTROL PANEL
IKEGAMI CAMERA CONTROL UNIT
IKEGAMI CAMERA CONTROL UNIT
IKEGAMI CAMERA CONTROL UNIT
MONITORS
MONITORS
MONITORS
ROUTING SWITCHER
MONITOR
CONTROL HEAD
WPIX-TV ENGINEERING
VINTEN CONTROL
FUNCTIONAL
REV
1

INTEGRATING NEWSROOM AND STATION AUTOMATION SYSTEMS

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Abstract

Automated TV broadcasting has been available for more than a decade and yet is installed at only a handful of stations. Will automation ever meet the optimistic expectations of its proponents? What requirements hasn't automation met that limit current acceptance? What is automation anyway?

A new system will be described which addresses these questions and offers hope that automation will finally begin to take its place as standard equipment in the broadcast facility. Network design, hardware and software for machine control and automation will be examined. The major attributes and criteria of automation will be evaluated.

History

To understand where automation is today, it is important to understand where it started. Operations (on-air and off-line) and news have each evolved automation concepts from their own perspectives without significant consideration of how their group fits into the overall facility goals.

The difficulty with automation begins with the radically divergent image the word conjures in each of us. There is no right or wrong definition for automation. There are, however, right and wrong answers to the question "What automation configuration makes the most sense for my broadcast facility?". The solutions are as different as the facilities.

Early developments were really nothing more than remote control of the same functions controlled by front panel buttons on the particular device. This rudimentary "automation" at least saved the engineer from having to physically move from machine to machine - cueing, playing, rewinding, etc. In early devices, this remote control was implemented in a hardwired fashion where dedicated lines were fixed to each individual function on the device. The total number of lines was naturally dictated by the number of controlled functions. In this parallel controller arrangement, adding control of just one more function in a device could become a non-trivial exercise.

Fortunately, the controlled equipment has evolved such that today, most devices provide a serial interface for remote control. There is no longer necessarily a physical equivalence - button to button - for remote control of the equipment. Now serial code identifies the specific function activated. Serial control requires more complexity at both the device and the controller; while creating the potential for more sophisticated operations and concurrently the greater possibility for error. While still controllable manually, computer control extracts the full potential and minimizes the errors.

Meanwhile, master control, as the logical center of operations began to provide more control. (Master control itself having evolved out of sequential development of: an integrated switching function with preset, master fader with automatic transition between line and preset, and preselection of event composition in a "preset programmer.") Utah Scientific developed a master control which properly set the routing switcher to get the desired source to the preview, preset and program buses. Simple relay control of machines was provided for "ready," "start," etc. Next, serial machine control allowed device control through master control. Finally, the manual "stacking" of events going to air was computerized.

Now, station automation has evolved to a higher level: traffic is ingested with all the data normally provided by the business service, from this a daily schedule is created, pull lists printed, machines controlled then routed to air, and discrepancy reports created. Limitations are subtle but significant: too slow to make late changes, can't rejoin aborted schedule, and doesn't include newsroom.

Across the hall, the newsroom has been evolving too. It began as did many businesses with word processors replacing typewriters. A nice improvement, but not enough. Next the terminals were tied together as a network and special interfaces were developed to read and file news wires. Networking enabled "text sharing" functions such as electronic mail, script routing, and system wide archiving.

In the communication/interaction intensive environment of the newsroom, tremendous efficiencies were achieved. Stories were better researched, better written and quickly updated. Editorial approval followed simply as completed scripts became immediately available for review through the network.

Automation at this point was limited to the newscast creation. Actually producing the show with automation required the next level of control: the production rundown schedule. Forms available on all networked terminals display such things as story order, story status (completed, approved, etc), story time, time to air and total show length. Any changes made at one terminal, are immediately reflected on all terminals. If a story is updated, all timing for the show is updated.

Now timing could be more accurately controlled by the producer. Creative choices are greater: swap stories, drop stories, add late developing stories. Clearly the on-air look improved. Significantly, with improved efficiencies due to automation, this was accomplished without adding staff.[1]

To support these visible changes called for more changes - and automation - behind the scene. Teleprompters are now electronic and fed directly from the computer script. And of course, as the producer juggles the lineup, the teleprompter scripts must deftly follow these changes. Closed captioning is also fed directly from the computer script and must follow changes in the same way.

The next frontier for the newsroom was integrating critical pieces of broadcast equipment. First was the character generator. Any news director using a newsroom automation system can describe late stories which made it on-air because of automation... but without supers. Now supers can be created from the news terminal and the playlist reordered as needed, just like the teleprompter. Cart machines can also be integrated so that the playlist within the cart machine dynamically follows any changes initiated at the newsroom terminal. Robotic cameras are sent to predefined positions as the newscast dictates, where camera commands are initiated within the newsroom rundown form. Still stores can be recalled in a manner similar to character generator supers.

Presently, NewStar can provide automatic scripting and device play lists. It does it all - except take it to air. And station automation systems exist which can start a tape machine and take it to air. But they haven't really integrated the whole station - including the newsroom - into a single efficient system. Operations and newsroom have come to the point of their evolution that now the two will meet.

<u>Criteria</u>

The experience of early automation adopters has led to a better understanding of what automation should and should not do. While some of the criteria - such as network transmission speed - are quantifiable, a majority are somewhat subjective. Nevertheless, the absence of these functions can lead to a situation worse than no automation at all. The major criteria are described below:

Cost Effective: Increases facility efficiency, improves on-air look, minimizes "make good", achieves cost justification.

Fast: Real time control is essential for accurately timed execution and schedule changes made either just before going to air or in response to on-air failure.

Modular: No two stations are alike, so tailoring should be allowed through appropriate module selection.

Flexible: To accommodate the wide range of applications.

Expandable: To accept system expansion as further budgeting or new equipment dictates.

Generic: Must be format independent, manufacturer independent and compatible with new equipment.

Efficient: Design must be streamlined for reactive/ responsive environment and cost effectiveness.

Expansive: Interfaces must be available for a complete range of devices from business services to satellite dishes.

Reliable: Failure of the automation is unacceptable. A single point failure should not take the whole system down.

Redundant: Critical components and functions must be backed up continuously.

Simple: Design simplicity results in reliable operation, less expensive installation, and easier operation.

Time Code Compatible/Frame Accurate Operation: Automation is capable of more accurate timing and must be able to utilize time code.

User Friendly: System must be straightforward to operate with direct, easy, error free access to devices and rapid execution of schedule changes.

No Programming: Programming must be complete including any special interfaces required. Automation is a means to an end where the end is improved efficiency not programming a computer.

Powerful Software: Every detail must be addressed before automation achieves value. Features such as conflict arbitration, resource allocation, disaster control, library management, and security are required.

Agreeable: Software operation must recognize that the engineer is in charge, not the automation system. Scenarios such as aborting a schedule then rejoining it must be direct and simple to achieve.

Creative: An agreeable automation system allows for greater creativity. It does not confine operations to strict and narrow procedures, but allows the previously undoable to be achieved gracefully.

Strategizing: Fundamental strategies such as: how close to air to load tapes, how long to hold tape tension, etc. must be built in and user definable.

Standard: Standard network protocols should be used to facilitate communication with the outside world.

Self-testing: Provision to self test for just failed and imminent failure conditions should warn users of the condition.

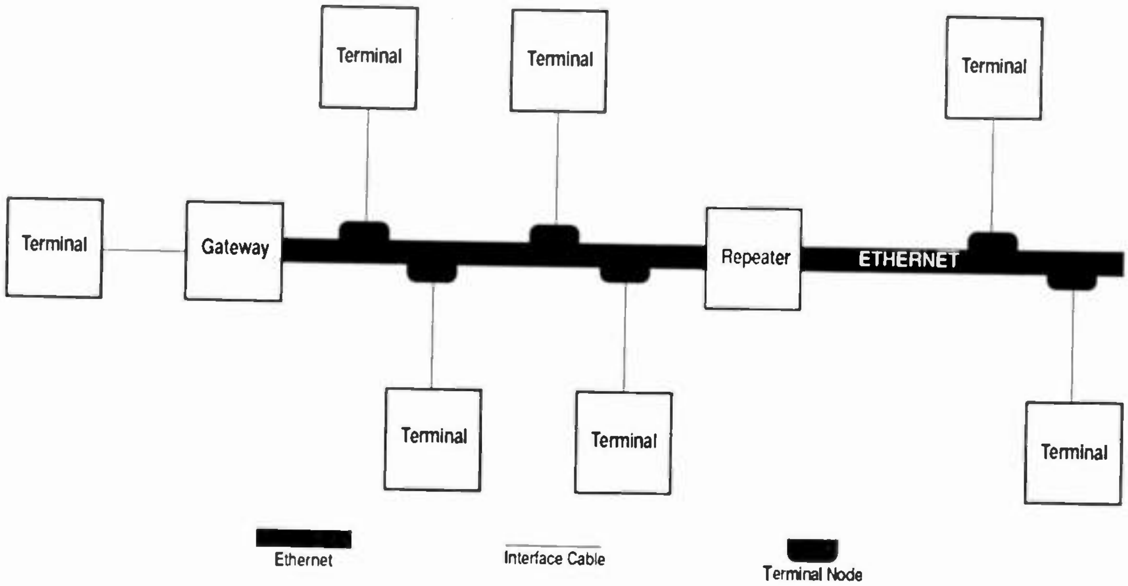

Figure 1. Standard Ethernet Configuration

Network

In considering the above criteria, the logical starting point is the network. This system is implemented on the Ethernet Local Area Network (LAN).

The simplified block diagram above shows all the major components in a typical Ethernet arrangement. The Ethernet itself is simply a 50 Ohm RG-11 coax cable. The terminal nodes tap into the ether and provide a 15 pin connector to interface to terminal devices. The "terminals" are varied: minicomputer, PC, "smart" terminal, printer, modem, mainframe computer, or in this case, an Ethernet-based machine control processor.

Repeaters are used for greater distances and gateways for interfacing individual networks together. More complete details on Ethernet can be found elsewhere.[2] In evaluating Ethernet for broadcasting, its most vital attributes are:

1) *Reliable Operation*: Ethernet is a widely accepted, well proven network. If one "terminal" should fail, the rest of the system remains operational. Its simplicity enhances its reliability.

2) *Speed*: Transmission rate is 10Mb/sec. While other networks may be perceived as faster, the application is a key factor. As seen in Figure 3, the only two "command issuing" devices on a typical broadcast system are the main automation processor and the NewStar newsroom automation system. All other devices communicate only to confirm command receipt, action taken, etc. In this low usage environment, commands can be transmitted/executed at frame accurate speeds. Consequently "next to air" can be changed up to the last moment.

3) *Expandable*: Additional equipment can be easily added without even turning off the system! Enough unique device addresses are available to satisfy the requirements of even the largest system.

4) *Simple*: Low cost of equipment and installation accrue from Ethernet design.

5) *Standard*: Ethernet is among the most common LANs in use today. Standard Ethernet devices and existing Ethernets within the facility can be easily connected to this automation system.

Token ring designs are often promoted for their high speed. However, for broadcasting (as described above) Ethernet is easily fast enough to allow frame accurate operations. Of more critical importance is the fact that with Ethernet, if a single device fails, the rest of the system still functions. In a token ring, one failed terminal takes down the entire system.

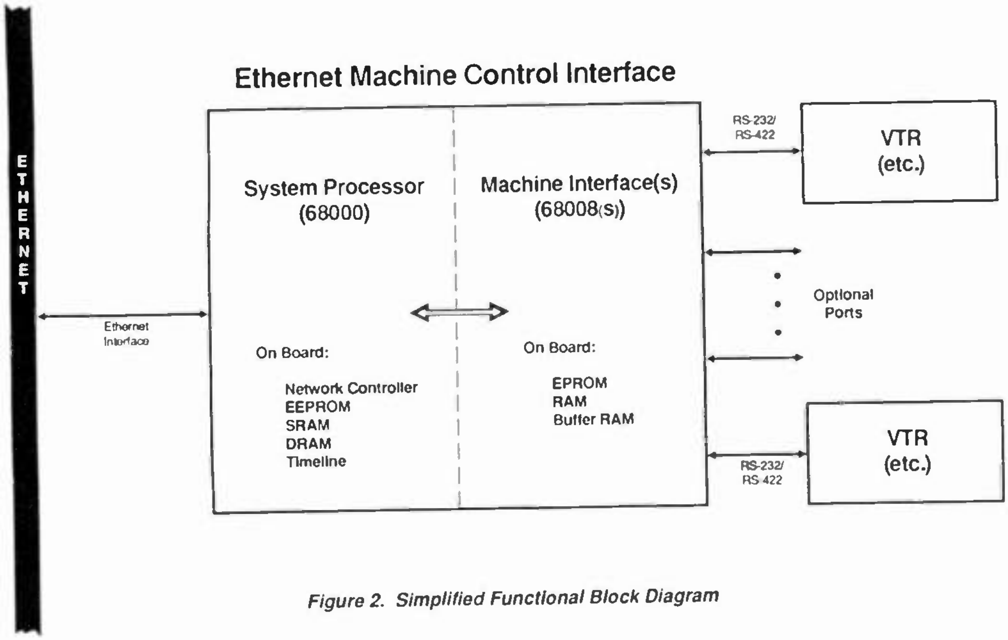

Figure 2. Simplified Functional Block Diagram

Hardware

The heart of the system is the Ethernet machine control interface. In Figure 1, this is represented as a terminal. Only one of several configurations is shown here in figure 2.

Distributed processing is vital for achieving the system's high performance. While transparent to the operator working at a system terminal, the multiple processors distributed throughout the network are at the root of much of the advanced performance.

The system processor card includes video sync and timeline for required timing. Non-volatile RAM is downloaded with "personality" for each unique device connected to the EMC. This non-volatile memory is also available for program and parameter storage, while DRAM is used for operating programs and network buffering. The Ethernet controller on the card handles network operations, allowing each device connected to an EMC to communicate with every other device connected via EMC to the Ethernet.

The "machine" processor cards are of modular design to provide control of one to four serial devices or parallel devices. Up to six of these cards can be housed in an EMC chassis. The "machines" interfaced could be virtually anything: tape recorder, telecine, character generator, routing switcher, etc. EPROM and RAM reside on the card for its own code and housekeeping, while separate RAM provides network buffering. Additional items such as relay/opto-isolator, relay driver, and delegation panels round out the requirements of a full function system.

The brain of the system is the main automation processor as seen in figure 2. System operation emanates primarily from this processor which typically consists of a Motorola 20MHz 68020, 2MB RAM, 80MB hard disk, 1.2MB floppy disk, two RS-232 ports, an Ethernet port, time code and real time clock. Due to its critical nature, this processor is backed up with a duplicate system implemented with a "heartbeat" technique which puts the backup on-line immediately if the main system fails. Standard non-proprietary components are used for reliable operation and user serviceability.

Typically four terminals are provided for: traffic, machine loading, master control, and off-line/miscellaneous. Additional terminals can be included as desired. For example, one might be placed in the News Production Suite. These terminals are in reality diskless PCs with memory, color monitor and full function keyboard. Rapid switching between the variety of display modes is one virtue of this design.

From the perspective of the EMC, the newsroom system is just another specialized RS-422 device interfaced to the network. However, news is unique in that it can issue commands and take control of other devices. From a system's perspective, it is more correct to think of news as one of the terminals. Each news terminal in fact has an individual address and therefore, network access equivalent to the dedicated network terminals.

All the equipment in the station from traffic computer to character generator to cart machine is now networked together, and accessible through any of the network terminals. One key to this shared resource concept is the routing switcher.

The main router, when integrated as part of this system, is automated not only to support master control, but also to automate off-line routing throughout the facility. Resource management, allocation and networked access begins with router control.

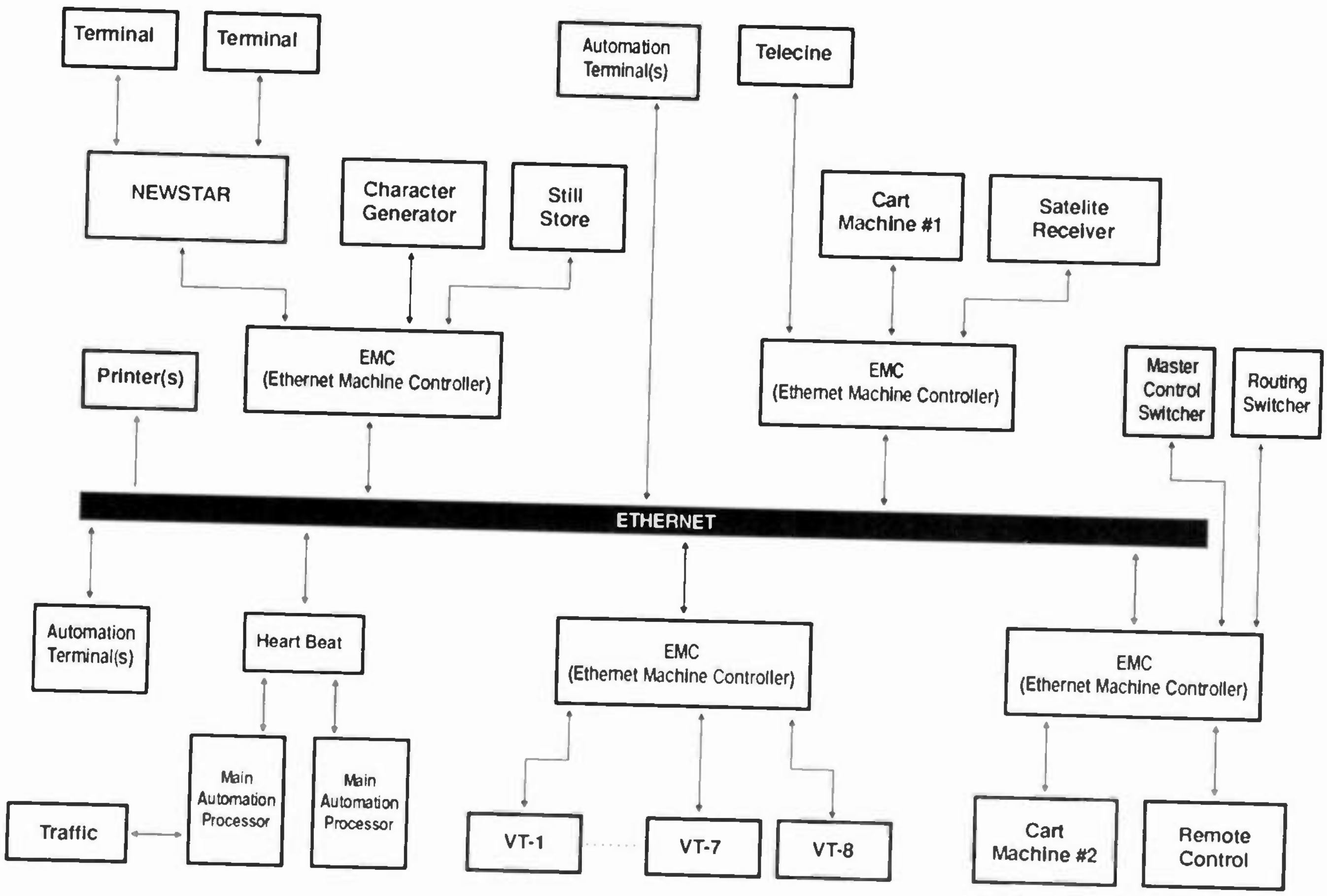

Figure 3. Typical System Configuration

Software

Automation operation cannot be easily quantified. The operational flow is as important as any individual task specification. A task must be executed at a precise time; where that precise time may be redefined just seconds before execution. Performance is measured by what operations it can do, how quickly, how easily, how accurately, how changeably; and done absolutely reliably. Slightly unreliable automation is oxymoronic.

This system is extremely comprehensive; offering a wide range of capabilities, for a variety of devices, in a multitude of circumstances. Yet, for all this power, it operates within the stricture that the operator is in charge - not the automation. This subtle, yet vital, attribute leads to a cooperative rather than adversarial relationship between the users and automation. This, in turn, fosters an environment where creative new solutions are found for old problems.

This system's combination of firmware and software provides for customization through data entry not user programming. No previous computer programming skill is required for effective implementation. Regulus, a real time operating system, is used, while programming has all been done in C.

The display oriented menu driven operation, in plain English, engenders operator confidence; in turn leading to easy use, predictable performance, and a dynamic response to the ever changing daily schedule.

Unlike slower systems of the past, this system provides dynamic, reliable operations through rapid communications. First, closed loop commands/ acknowledgements provide constant status of equipment coming up to air. Fatal errors are detected, before trying to take to air. Distributed processing and a fast network result in rapid schedule changes. On-air malfunctions are resolved with a take next or panic source as appropriate within just a few frames after malfunction (depending on the type of malfunction). Schedule changes to next on-air are possible less than a second before pre-roll start of the event.

Automation forces a reevaluation of TIME. In manual operation, time is flexible. If an event is longer or shorter than expected, "on line" adjustments can be made to get back "on time." The system expects everything to be precisely timed, preferably frame accurate. The blessing is improved on-air look from the clean transitions. The prerequisite is that event durations must be known accurately when they are entered into the days schedule.

Operationally, the focus is daily forms - both on the monitors and printed. The fields in a schedule are user selected depending on individual need. Typically, the most important information is condensed on the primary display: event type, start mode, time, duration, source, transition, ID. Displays are organized to supply important details at a glance. Operation is implemented to allow easy, unambiguous access to inspect and change events in the form: swap, add, delete, etc. Timing is automatically recalculated after such changes.

The system is bi-directionally interfaced to the traffic system. The upcoming days schedule is forwarded to operations for completion with details such as equipment used and transitions. The completed days log is annotated and returned to traffic for reconciliation.

A key feature in properly integrating news with automation is library management. Traditionally the library has contained commercial and programming material. News archived their own scripts. Now, however, automation and integration make all facility resources available to news. In investigating a story, the reporter can search the archive for any related stories. Both scripts and videos from these existing stories could then be routed to the reporter. A sophisticated library system is the first step in the process.

One hazard of providing anyone with a terminal access to the tremendous resource available, is creating simultaneous demand for the same equipment. This system manages the system by establishing priorities, making specific allocations and providing queues - all based on input such as time of day and requestors department.

It is also vital to control access to the system. We have implemented a security system based on user ID rather than terminal ID. This allows users with the highest level of access to address the system regardless of the terminal being used, while simultaneously preventing lowest level access users from disrupting operations.

Conclusions

By meeting the criteria listed, we offer the kind of complete solution needed for automation to gain wider acceptance. Just as it unifies operations, it also unifies station management by offering something for everyone.

For today's business environment, it offers good bottom line sense through improved efficiency and lower operating cost. For the chief engineer, it offers improved on-air look, better resource utilization, improved "disaster control" and fewer discrepancies. For the news director, it offers better writing, more creative flexibility, and more timely and sophisticated productions. Expect automation to become more widely used as it becomes the best strategy to meet everyone's needs.

[1] L. Sanders Smith, "Newsroom Automation Opportunities," *1988 NAB Engineering Conference Proceedings*, April 1988, pp. 225-232.

[2] Tyler North, "Ethernet in the Newsroom," *Broadcast Engineering*, January 1989, p. 58.

INCREASED VERSATILITY FOR THE ESBUS

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Abstract

This paper investigates the current status of the ESbus Remote Control Standard; and examines some possible routes by which its functionality may be enhanced, in order to meet continuing developments in the area of broadcasting equipment.

1. Background

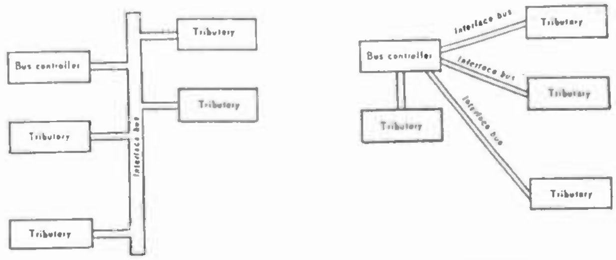
ESbus achieved the status of a formal standard with the publication of the primary documentation, jointly by the EBU and SMPTE, in December 1984.

The Standard has been adopted fully, also, since that date, by the OIRT - making it, in principle, the only broadcast remote control standard with worldwide recognition. An official 'observer' from the OIRT is a permanent member of the EBU Remote Control Committee (G/RC); and as Chairman of that group, I attend sessions of an ESbus-specific Committee of the OIRT.

Subsequent to the issue of the fundamental standard - which describes the principles and communications protocols of the remote control system - four further supplementary documents have been developed, which define fully control messages for network administration; and also for the remote control of Video Tape Recorders, Audio Tape Recorders and Telecines. A dialect for Routing Switchers has been prepared by an EBU/SMPTE joint sub-committee, and is presently passing through the stages of formal approval.

Full control of all functions of these machines is possible using serial transmission at the standard ESbus transfer speed of 38.4kb/s. Where particularly time dependant fuctions are needed, the bus can be structured in the point-to-point mode, rather than in the conventional multi-point configuration, with one tributary only on each bus. (See Figs. 1a & 1b) (Such an arrangement can also dispense with the need for the use of the POLL address, in certain cases, since each communications channel is unique).

Fig. 1 Bus Configurations



a) Multipoint b) Point-to-point

Extensive use is made of 'distributed intelligence' in the nature of the architecture - enabling 'advance notice' to be given to tributaries, of time dependant and time critical functions; and also to enable operational conditions and/or sequences to be preset in advance of execution of such functions.

In addition, the introduction of the 'timeline' concept has enabled full exploitation of this capability, by the synchronisation of time-critical functions across several tributaries.

These techniques, developed originally for use within the VTR message dialect, have been employed widely for complex editing purposes.

2. The development of modular equipment

In recent years there has been a growing move towards modularity within equipments formerly of monolithic construction. Discussions in both the SMPTE and EBU Committees have explored the need to recognise this migration, by accepting the potential of greater penetration of ESbus into areas which have previously been the domain of the system designer.

To turn a blind-eye to this fact will merely guarantee the consignment of ESbus to the broadcasting archives, and to the museums!

In the case of the Production Switcher - or Mixer, in European parlance - the functionality has been compartmentalised to such a degree that type-specific dialects must be considered for each architectural component of the mixer desk.

Whereas in former times a Production Switcher had been considered to be an integral entity, capable only of external control, it must now be seen as a number of discrete and defineable modules which interact with each other in order to achieve the desired effect.

It could, for example, be considered to contain a Router - for which a dialect already exists; a character generator; and a 'special effects' section.

This latter may, itself, be considered in a modular way, by specifying a specific effect in terms of a series of 'layered' images, each being defined during the rehearsal phases of a production; together with an appropriate degree of transparency between the layers. The 'dialect' for such a generator might then be reduced to a series of preset and numbered parameter calls, which define the layer or layers, the degree of transparency, and the transitions between them. Such calls could be accommodated by the addition of further messages within the Common Message set.

A certain number of pre-defined library layers and transitions might be provided within a particular proprietary product - in the same way as standard mixes and dissolves are held currently in production switchers; others could be introduced, for one-time use, during the rehearsal phase of a programme. (This may be compared with the pre-setting and use of a studio lighting plot).

3. Communication Channel Speed

In order to achieve adequate rates of data transfer under such circumstances, an extension of the present ESbus capacity is clearly needed.

Within the supervisory level protocol as presently constituted, a mechanism exists which employs the ESCape character (03h), and allows individually addressed tributaries to escape from the standard protocol, and to enter non-standard communications. This is accessed through the tributary's SELect ADDress. On receipt of a BREAK character, however, the tributary MUST revert to standard communications, and return to the ACTIVE state. This cannot therefore be taken as a realistic way of achieving the necessary data throughput on a regular and standard basis.

Alternative means of achieving high data transfer rates, by the extension of the basic ESbus protocol, are therefore under investigation, and this paper outlines some possible routes by which this may be achieved.

4. Should an overall higher bus speed be introduced?

It is now some ten years since the present data transmission rate of 38.4kb/s was proposed and adopted. This rate was chosen because it could fulfil, economically, the perceived needs of the interface at that time.

UARTs (Universal Asynchronous Receiver-Transmitter chips) were freely available for this speed, and at a price which was acceptable to manufacturers. The cost implications precluded the use of higher speeds; however, chip technology has developed considerably since then. Speeds of 10Mb/s are now common in the computer Local Area Network environment.

It must be recognised firstly, however, that the transmission speed of 38.4kb/s is adequate for the majority of the broadcasters' needs - certainly for those type-specific machines for which dialect provision has already been made.

Secondly, the adoption of higher universal speeds could have a limiting effect on the distances over which ESbus can be run within the studio environment - without resorting to repeaters or bridging techniques.

Distance limitations clearly become restrictive as transmission speeds increase. However such limitations might be considered acceptable in certain specific circumstances, and for certain type-specific devices - the Production Switcher to DVE link for example.

In determining a 'preferred' data rate for 'high-speed' communications in broadcast remote control, a number of broadcasting-specific factors must be considered. Some of these were discussed at a joint meeting of the EBU G/RC and SMPTE T14.10 Committees in London in September of last year.

The real, rather than the perceived, need must be investigated. What is truly required in terms of the message lengths and the time windows into which they must be slotted?

Too high a speed could impose constraints in a broadcasting environment, due to the inherent nature of the digital signal. Fast edges are a potential source of interference with both audio and video signals; and could also cause unacceptable degradation in communications services.

5. Terms of Reference

It might be worthwhile reminding ourselves of the terms of reference for the original work of our two Committees:

1. The system must incur only low additional cost.

2. The interface must be capable of being integrated within the type-specific equipment hardware and software.

3. The local network must be deterministic in nature - that is, it must have predictable maximum time delay in the delivery of control messages to their ultimate destination.

4. The system must employ to the full, the potential of distributed intelligence and local data storage.

Each of these points has been respected within the present definition of ESbus.

In particular, the local storage capability, and the intelligence distributed across tributaries have enabled complex functions to be undertaken with relative ease. The use of 'timeline' as a synchronising mechanism, and the ability to store program steps within tributaries, have overcome most of the potential difficulties of 'real-time' operation.

Above all the incremental cost to provide hardware with an ESbus control interface is small, both in terms of hardware needed per machine, and also in the 'distributed' cost of software developments. All type-specific machines share the same basic software development costs. The only additional costs incurred in the provision of a control system for a new type of machine is the implementation of the appropriate new message set.

In the development of a high-speed capability for ESbus, those original terms of reference must still be respected.

6. What are the Options?

What are the options which the Committee can consider?

1. It has been suggested, on occasion, that we should dispense with the lower layer protocols, (Supervisory and Electrical/Mechanical levels), and adopt a 'standard' from the world of computers and Local Area Networks.

This would, of course, be possible. It would allow the syntax and semantics developed for the message dialects to be retained fully; yet would enable alternative network access mechanisms - CSMA/CD (Ethernet); Token Ring; Token Bus; Buffer Insertion Ring etc. - to be employed.

One of the principal reasons for adopting a layered architecture is to permit this form of growth. Any layer, or layers, can be removed and replaced by others, as developments in technology occur.

2. In making a decision on which access mechanism to employ, the original terms of reference should still be borne in mind (cheap, deterministic, inclusive, distributed intelligence). Some LANs fulfil these requirements fully - others are lacking in some respects. However, as an overall objective this is a perfectly valid approach.

However, the major factor which opposes progress in this direction is that it invalidates, immediately, equipment installed to the current standard.

It is my personal view, that it is essential for continuity to be maintained; any solution to the speed problem must recognise and respect current practice. In this regard it is very similar to the introduction of colour television, and the emphasis placed on compatibility with the former black-and-white television services. The introduction of the NTSC system built on existing engineering experience, and standards; it extended the capability of the transmission mechanism in order to accommodate the additional information.

I believe that for ESbus to retain respect as a standard in the market-place, and yet to meet developing needs, it must remain a standard. Enchanced capabilities must be achieved by the introduction of additional mechanisms - not by replacement of those currently in use.

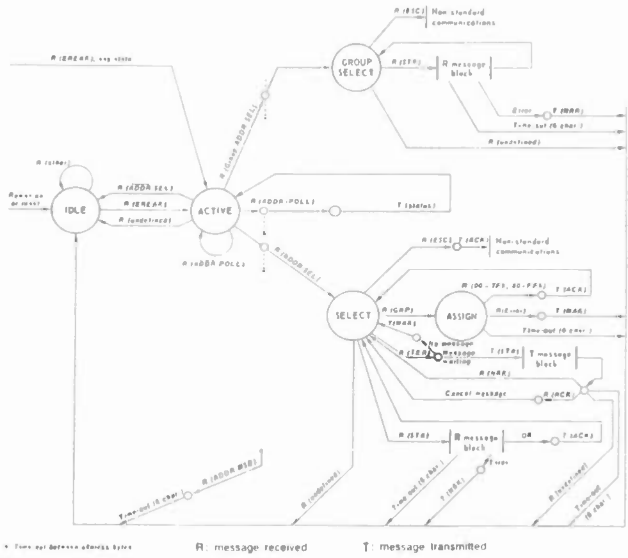
I do not, therefore, support the policy of direct replacement at the present time. It may, of course, be an appropriate move at some time in the future.

So what are the options? We have considered a couple alternatives, and I will attempt to describe these briefly.

7. Extension of the ESbus Supervisory Level Protocol

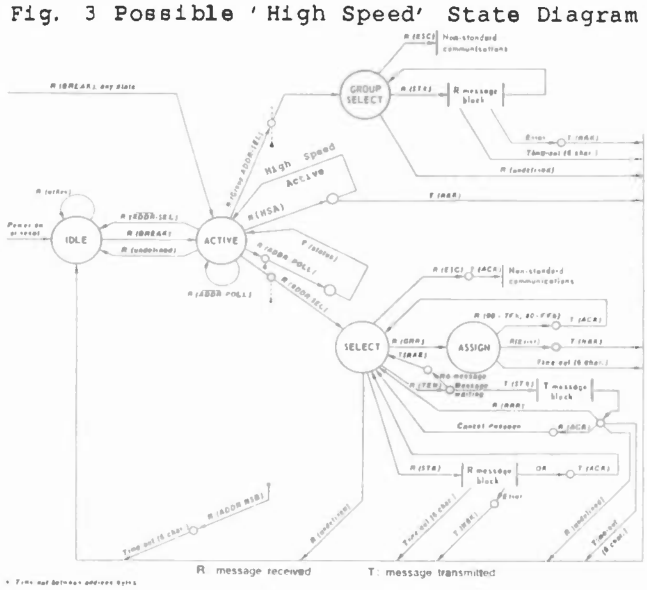
As was noted above, and as can be seen from the State Diagram in Figure 2, the current Supervisory level allows for a nominated tributary to 'opt-out' of the standard protocol, and enter non-standard communications.

Fig. 2 Existing ESbus State Diagram



By the use of an additional Supervisory level control character, it may be possible to invoke a higher speed of working for the overall local network.

Fig. 3 Possible 'High Speed' State Diagram



Consider, for example, a new control character (0Ah) - High Speed Active, (HSA). This would be issued by the bus controller following the issue of a <BREAK> synchronising character; that is, when all tributaries are in the ACTIVE state. Tributaries which are high-speed capable would enter the high-speed mode, but remain in the ACTIVE state. (See Fig. 3)

Once the bus controller has transmitted the HSA character it will itself change speed to the higher data rate for ALL functions - including <BREAK>. On receipt of a standard speed <BREAK> character (i.e. at 38.4kb/s), all tributaries would re-enter the 'standard' speed ACTIVE state.

Further higher speeds might be achieved by the use of additional nested HSA characters. Alternatively, higher speeds might be achieved directly by the use of additional HSA characters - (0Bh, 0Ch, etc.).

Tributaries incapable of performing at the higher speed would interpret the HSA character as 'undefined', and would therefore return to the IDLE state; or, alternatively, if they recognise the character, but are incapable of changing speed for any reason, they would transmit NAK (05h), and then return to the IDLE state. It would then be the responsibility of the bus controller and hence of the system designer to decide on what action should be taken.

A fundamental problem does exist with this proposal, of course, with non-high-speed-capable tributaries. When the network is in the high-speed mode, they can no longer take part in the activity of the network; they would not recognise the <BREAK> character at the higher speed. They will therefore remain in the IDLE state, until the bus controller issues a <BREAK> at the standard speed once again. They would also be incapable of raising the Service Request flag (08h) - SVC - during this time, and would effectively, therefore, be dis-enfranchised within the network.

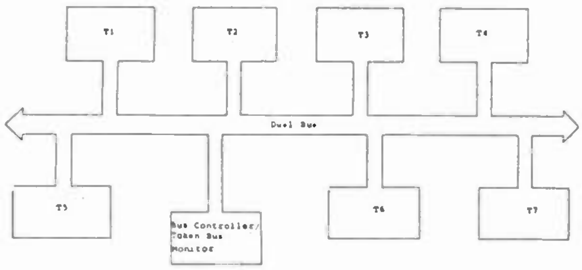
Perhaps a system of 'flagging' tributary addresses within the address allocation table held in the bus controller might be possible - indicating whether individual tributaries are high speed capable or not, and hence allow the bus controller to vary transmission speed accordingly, on a tributary-by-tributary basis.

However this could be somewhat risky, with the possibility of a tributary mis-interpreting character streams because of speed differences.

8. The nesting of Protocols

Another proposal under consideration is the possibility of the nesting of protocols running on the same bus.

Fig. 4 ESbus co-existing with Token Bus



Consider the network shown in Figure 4.

Seven tributaries are shown connected to a bus with a bus controller. Under normal circumstances, the bus controller would issue a roll-call poll to each tributary in turn.

However, if the system is now considered to be, for example, a Token Bus network with some of the attached devices - e.g. T3, T6, T7, and the bus controller - being peer tributaries, serviced by a token, it could be possible to run the system as a Token Bus network, and with the bus controller fulfilling the role of network monitor.

On receipt of the token, the bus 'monitor' would convert to the role of bus controller, and issue a <BREAK> to put the 'standard' speed devices (T1, T2, T4 and T5)into the ACTIVE state.

Following a cycle of POLL addresses to this pre-defined list of tributaries - with the bus controller responding to, and serviceing, any SVC service requests - the bus controller would relinquish the token, and the bus would once again operate in the Token Bus mode, serviceing the high speed devices.

This nesting of protocols has been the subject of some experimental work in other areas of activity , and does appear to be a workable option. However much detailed work would be needed before a proposal could be formulated.

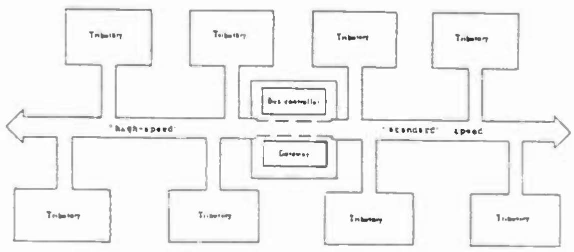
9. The Bus Controller as a Bridge or Gateway

As a variant of proposals 7 and 8 above, it would be possible to employ the bus controller itself as a 'bridge' between differing speeds of the standard Supervisory level protocol (i.e. effectively working in a point-to-point mode, but with different speeds on each individual bus - thereby separating the high and standard speed tributaries).

Alternatively, it might be possible to use the bus controller as a bridge-cum-gateway between protocols.

In this arrangement (see Figure 5), the ESbus definition would be extended to incorporate - to use the earlier example - a high speed Token Bus network, with the bus controller providing the interchange channel between the two access mechanisms. It would also provide the monitor function within the token bus environment, as well as fulfilling its native bus controller role.

Fig. 5 The Bus Controller as a Gateway



Either of these two arrangements would require the separation of high and standard speed tributaries. Messages could still be exchanged between the two bus systems, using the bus controller as the 'gearbox'. However, factors such as timeout must be catered for in the transfer of messages across the bridge. The bus controller/monitor must handle first-line responses at the protocol level - in particular when transferring messages from the high speed to the standard speed network. Delays caused by the polled half of the network could otherwise be unacceptable.

10. System Service Messages

In order to initialise any high-speed method of working, I would propose the introduction of an additional System Service level message, which could be introduced from a tributary, and directed at the bus controller.

A System Service level message 'High Speed Active' - with a logical parameter (ON/OFF), or even a numerical SPEED parameter - would permit the entry to, and exit from, the high speed mode. On its receipt, the bus controller would then issue the appropriate Supervisory level message.

11. Summary

At present the SMPTE Recommended Practice RP113, relating to the Supervisory Level Protocol of ESbus, is in its review period. This is an appropriate time therefore to consider making changes which could affect the capability of the standard for future use, should this be thought to be appropriate.

Other changes could be made purely by extension of the definition of the ESbus to include alternative bus mechanisms.

Whichever route is chosen ultimately, one fact remains overwhelmingly clear. The work undertaken over the past seven or eight years has achieved a standard with exceptionally broad flexibility. This must not be lost.

The message structure within ESbus is very sound. The carrier mechanism for these messages must protect existing investment.

Higher speed working is inevitable for the reasons outlined above; but this must not be seen to invalidate any of the earlier work, or equipment which is currently in service, or in development.

Any changes must be extensions in capabilities, and not replacements for the earlier work.

I have not attempted in this paper to describe all possible avenues of development. I have attempted to indicate a couple of ways whereby the capabilities of ESbus could possibly be extended.

Whichever route is selected by our joint SMPTE/EBU Committees, I am certain that ESbus will remain the first choice of broadcasters for many years to come, by virtue of the flexibility it offers.

I hope and trust that manufacturers will see the wisdom and advantage in bringing ESbus systems into the market-place at an early date.

INTEGRATED STATION AUTOMATION

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ABSTRACT

This paper highlights some of the operational and technical design objectives which must be considered if maximum commercial benefit is to be derived from the introduction of computer based control, automation and information management systems within a TV production and broadcasting system.

INTRODUCTION

Connectivity and communications between individual and often complex computer systems within any applications environment can be a minefield. When the environment is fiercely competitive commercial TV, involving live and dynamically changing programme schedules, together with intensive commercials traffic and late sales, fluent integration of computers, machines and men is essential. Today's TV system engineers face operational objectives and technical requirements relating to control and communications in TV stations and networks which can far exceed the complexity of the sound and vision system itself. System designers must therefore now be both television and computer literate.

Tackled properly and with an eye to the future, well-planned control and communication systems can contribute greatly to the efficiency and commercial success of a station.

Conversely, isolated and poorly integrated systems can severely dent profitability and growth. A single-vendor common architecture for at least the major elements of a station's business management and operational systems is, of course, the ideal solution, but one which at first sight seems unattainable - or is it?

Analysis of a typical TV station requirement for computer-based support systems shows that the applications fall into two distinct areas. Firstly, information management systems, often requiring multi-user access and manipulation of data such as programme screening rights, scheduling, sales, billing, media libraries, etc. Secondly, real time systems, commonly involved in network control, master control automation and, more recently, production automation. Clearly a multiplicity of applications from different vendors running on an equally varied range of hardware and operating systems can be connected together to form 'a system'. The connections between them, though, are often then forced to be relatively simple, inflexible and by no means as transparent as modern networking technology allows. Users are therefore restricted in their ability to access one system, for example a still picture library, from or through another, for example, the programme scheduling system.

As a world leader in both multi-user computer systems and networking, Digital Equipment Corporation (DEC), offers an ideal computing and communications environment within which a wide variety of powerful and proven broadcasting products are available.

The Quantel Digital Picture Library System employs DEC VAX computers as its Central Lending Library. BASYS utilise single or clustered VAX systems for extensible electronic newsroom automation.

The CATS Computer Aided Transmission System, manufactured by Connolly Systems, employs DEC Micro PDP-11 real time systems for master control automation, network automation and data transmission. Connolly CATSmanager software for programme library, scheduling, media libraries and logging can run on any DEC VAX computer.

By virtue of their common vendor architecture, all these systems can be easily interconnected utilising broadband communication techniques, such as Ethernet, to form an open and homogeneous system which is both flexible and efficient. Other applications and products can be easily added within existing systems or as additional functions available on the network. DEC's PCSA Personal Computer Systems Architecture can also be layered onto the network to provide PC users with the ability to provide and share access to corporate data held, for example, within the CATSmanager, news or stills libraries.

DESIGN STRATEGY

Having accepted that an open, flexible and extensible network should be the long-term objective for the management and control systems within the station or TV network, it then becomes necessary to keep these objectives clearly in sight when making isolated equipment purchasing decisions. For example, in making what at first sight might appear to be an engineering choice of a new master control or news production automation system, it is not sufficient to only satisfy the immediate direct user requirements. Many other departments may have creative, artistic, technical or corporate requirements of these systems which, if properly considered and implemented, yield a commercially better decision. Given a particular programme schedule for a station, significant contributions to the style, visual acceptability and corporate image come from its 'Presentation' or 'Continuity'.

In order to minimise programme post production costs associated with achieving a high standard of smooth programme continuity, it is often better to utilise a master control/continuity switcher which offers transitions over and above "crash-bang-take"! Many stations have in the past compromised in this area, largely on the basis of what a single operator can achieve by way of accurate, complex and artistic transitions in a live situation. With the advent of powerful computer aids in scheduling, continuity and master control automation, and with the enhanced effects capabilities of the switchers involved (e.g., GVG Master 21 with mix effects), new and cost-effective techniques can be employed to reduce costs and provide flxibility in programme packaging. Additionally, the quality of programme continuity and the on-air corporate image of the station can be defined, changed and improved by management and planning staff from a computer terminal in their office.

Pre-programmed and potentially very complex programme junction routines involving separate video and audio source selection, machine control, graphics, keying and digital effects can be achieved easily in a computer-aided environment. A good planning and scheduling system should be capable of defining this level of continuity detail. A good automation system should be capable of implementing it in both a recorded and live transmission environment.

CONTROL SYSTEMS

To fully support the stated aim of constructing an open and flexible system, the machine control systems for VTRs, ATRs, multi-cassette systems, stills, graphics, etc., which form part of the automation path should have extensive control, status reporting and information transfer capabilities. This may give rise in some cases to the automation system having two connections to the controlled device. One for real-time remote control, via which a guaranteed response time to a particular command can be achieved. The second for less time-critical transfer of data such as library enquiries or updating. Examples of this type of control structure can be seen in Figures 1 and 2 in relation to the Quantel DLS and Central Lending Library and to the multi-cassette system.

With regard to actual real time control and status reporting for the machines being controlled by the automation system, one of two 'structures' is normally employed. A 'star' connected system in which each machine has a separate connection back to a control port on the central automation controller. Alternatively, a 'bus' connected structure can be employed whereby machines, subject to their having the necessary interface capability, are connected in parallel across an RS-422 or co-axial bus.

In the case of star based systems, intelligent machine control interfaces normally share a common micro-processor bus with the central automation processor and its disc and tape peripheral controllers. This multi-processor architecture off-loads processor intensive tasks associated with communications protocol handling on to the interface processors and guarantees the response time of the overall system. This is particularly important in a live transmission situation. At the same time, it contributes to the overall communications bandwidth of the system and allows fast DMA (Direct Memory Access) techniques to be employed to transfer data, e.g., cart machine playlists and logs, through the automation system to and from the scheduling system.

The Connolly CATS system shown in Figure 1 is an example of a star structured system in which each interface can fully utilise the 38.4 Kbaud capacity of the RS-422 link to its respective machine without imposing any undue load on the central CATS processor. The central processor simply places control commands or requests for particular machine data in a shared area of memory and then reads the reply from another area. Very fast and predictable machine response times can be achieved in this way which, although perhaps under-utilised in tape replay automation, are essential in computer aided presentation of live programme material such as sports and news.

If a bus-based machine control system such as is used in the Connolly BASmaster system shown in Figure 2 is to meet the same design criteria as the star-based system, very careful consideration must be given to the choice of the bus and its associated protocols. The two most common system restrictions which can result from utilising a bus are, firstly, limitation of individual machine communication bandwidth by virtue of having to share the available bus bandwidth with other devices. Secondly, propagation delay can be unpredictable and response time can be slow if the bus architecture and protocols do not guarantee predictable access to the bus for each device and controller. Collision Sense Multiple Access protocols, as are commonly used on Ethernet data networks, can suffer from these limitations under heavy loading conditions. To counter these two potential limitations, system designers and manufacturers employ a variety of techniques including wide bandwidth bus structures such as Ethernet running at 10 MHz combined with time-line operation of machines. The latter, which provides the ability to instruct a device ahead of time to perform a particular function at a given time of day, is again a viable solution in an automated replay situation, but is not ideal for dynamic live situations.

The BASnet real time control network, Figure 2, which links the Machine Control Processors to the central automation controller, has a bandwidth of 1 MHz. It employs both a high level automation protocol and low level machine control protocols, depending on whether a machine is being controlled fully automatically, manually via the automation system, or via an assigned control panel on the BASnet network. Full 38.4 Kbaud bandwidth communication between 20 connected machines and the BASmaster central processor is achievable over BASnet with a predictable worst case response time of any machine to a live executive action on the BASmaster control panel of 0.1 second. This ensures that, even when time-line techniques are employed, real time live intervention and override of the automatic execution of a schedule is possible. Larger numbers of machines can be connected to BASnet, but, in keeping with the principles of good local area network design, it is a good idea to separate bus traffic by the use of area bridges or network gateways in order to maximise performance within each operational area.

ON-AIR DISPLAYS AND CONTROLS

In a pressured on-air situation, it is vital to present data relating to both the on-air schedule status and contributing machine status clearly and accurately. A simple colour display of the running schedule with only high level status of machines, 'OK' or 'NOT OK', indicated by colour seems to be the widely acceptable method. Detailed reporting of individual machine activity and status is best displayed separately on a dedicated status screen or control panel. User flexibility in the way in which displays are formatted may be an important consideration for some stations.

Consideration should also be given to the best way to make the latest version of the on-air schedule visible to other potential users both within and outside the station. Here again, different users will have different requirements. Planning, scheduling and sales staff need to be able to 'window-into' the on-air schedule to obtain information or make changes. This is best achieved by pulling back a copy of the on-air schedule from the on-air system over the business Ethernet, viewing and/or editing it within the editorial systems, and returning the new-version schedule to the on-air system, where the operator can then link to it at a convenient point. There is not normally a requirement for second-by-second on-air event monitoring via this route of enquiry. If, however, it is deemed necessary, techniques can be employed which provide a pseudo real-time display by using time-line principles in reverse with forced screen updates whenever an event is taken on-air or when an edit is made via the master control on-air terminal.

If an absolutely accurate real-time display of schedule status is required, for example, in other studio control rooms, it is preferable to feed the real-time event stack of the automation system via conventional video distribution amplifiers to monitors as required. Alternatively, a copy of the event stack could be provided as a dedicated data feed from the automation system. In CATS-100 and -200 network automation systems, this data is available as a vertical interval data signal which allows all stations on the network to decode a real-time event stack display and which also assists contributing outside broadcast units to accurately comply with network timing requirements.

Operator intervention of various types needs to be possible both during fully automated transmission and periods when the automation system is being used as an 'aid' for live programmes. The ability to edit the schedule from a single terminal/ keyboard in the knowledge that all other contributing machine schedules will be updated immediately is paramount. Basic functions of hold, take next, take 'n', add event, insert event, delete event, skip event, etc., should, ideally, be supplemented by more complex macro functions such as select a specific alternative schedule, select a pre-programmed emergency schedule or routine, show available promos to fill/drop in order to meet fixed time events, etc., etc. Once a really open communications and control system has been combined with a powerful real-time automation system, there really is no limit to the amount of assistance that the system can provide to the 'man in the hot seat'.

LOGISTICS AND INFORMATION

When examining the business objectives for installing computer based management and automation systems, a variety of motivating requirements are apparent. Achieving a better on-air look than the competition, longer transmission hours with no increase in head count, reduction of operating costs, ability to sell late (really late!), one man operation even during peak times, are just a few of them. Speed and accuracy of information transfer through the system and thence the station should be implicit in all of them, because from this stems overall business efficiency.

A detailed quantitative analysis of both information flow and product (programme material) flow through the station needs to be done at regular intervals to ensure that the computer, communications and control systems are constantly tuned to maximise their overall contribution to efficiency and profit.

In order to effectively conduct its part of a station's daily operation, each department needs to be sure that it is working with absolutely up to date information. Data bases such as programme contract libraries, tape libraries, still picture libraries, are therefore best held as centralised resources connected to the business network so that all users and departmental computer tasks are always working with the same current information. Schedules, for example, are better held as a linked index into a relational data base holding programme and tape data, rather than as a fully detailed schedule in a file. This ensures that, whenever any programme or tape details are confirmed on arrival into the station or changed as a result of further in-house post-production, every schedule affected automatically reflects the change. Similarly, by imposing some constraints on long-term scheduling, namely to schedule a programme, it must normally first be entered into the programme contracts library, it is possible to guarantee total compliance with screenings, contracts and corporate programming policy.

CONCLUSIONS

An open and well-integrated network of broadcasting management and automation systems can make the vital difference in efficiency and profitability for a station. A fully compatible and ideally uniform single vendor architecture capable of satisfying the current needs of the station and at the same time facilitating future growth is ideal. Focus on speed and accuracy of information transfer, avoid systems which require duplication of data entry, commit yourself to systems which help all the station's operational staff to improve the quality and consistency of the on-air appearance, but above all:

- find out what everyone in the station wants of the system

- find out what everyone in the station actually needs of the system

- make the best possible technical and commercial design and purchasing decisions

- be a perfect engineer and manager at all times!

Acknowledgements

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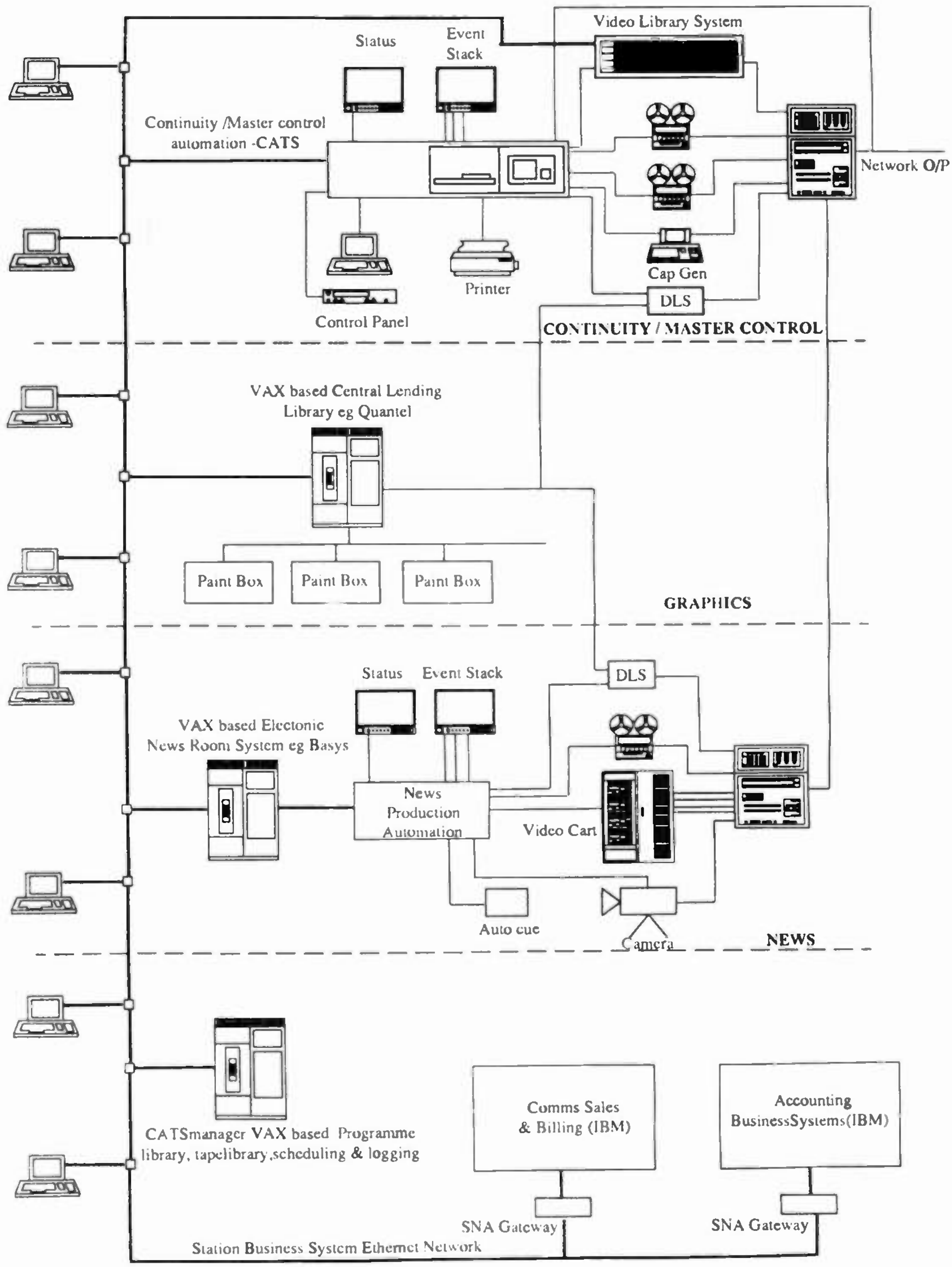
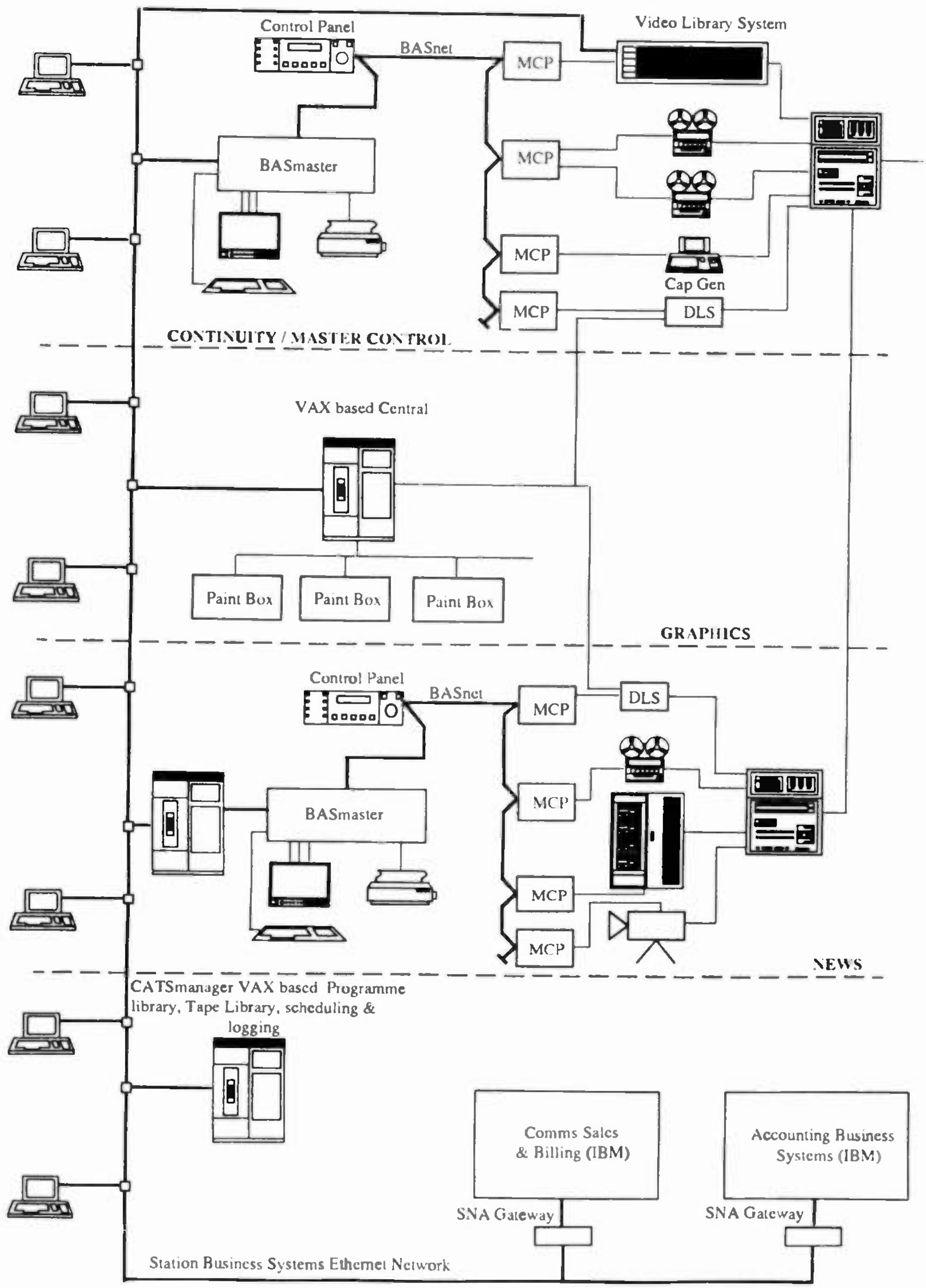


Integration of Typical Star Structured Automation System Within a Station

FIG. 1



Integration of Typical Bus Structured Automation System Within a Station

FIG. 2

NBC OLYMPIC GRAPHICS AND ANIMATION

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Abstract

In preparing for the sizeable task of televising the 1988 Summer Olympics Games, NBC assembled a unique and impressive array of the finest graphics systems available.

This paper will present an overview of these systems and how they were used to achieve our goals.

Overview

In addition to the athletic records set during the 1988 Summer games, a number of new records for broadcast were set. This was the largest Olympics ever televised with over 9000 athletes from 160 countries participating in 237 events. This was the first all stereo Olympics. NBC covered this event with 96 unilateral cameras, 24 mobile units, over 150 tape machines, and the largest broadcast center for a single event in television history.

The event lasted 16 days and NBC provided 179 1/2 hours of Olympic coverage, of which over 75 percent was live.

The Olympic Graphic facility had numerous production and operational requirements:

1. Information display of data concerning athletes, countries, events, and results.

2. Operation of the graphic systems must be as artist friendly as possible.

3. Database interlink between the graphics display devices and existing informational databases.

4. Real time display of event clock and finish information.

5. Program signature to provide consistant identification of the 179 1/2 hours of NBC Olympics programming to the viewer.

In order to generate the highest quality graphics possible, a number of goals were defined in the early stages of planning.

1. Graphics should be created and distributed in a component video format whenever possible.

2. Graphics must be consistent in both appearance and technical quality.

3. Information graphics should be generated as quickly and accurately as possible.

To achieve these goals it was decided that all graphics, from animations, to bumpers, to results, to clocks should be created by NBC artists using NBC equipment. This equipment would be controlled by NBC staff and NBC computers when played back in Seoul.

At the International Broadcast Center, NBC constructed the network's third largest facility. Occupying 55,000 square feet of space, it contained four large multisource edit rooms, eleven smaller edit rooms, two studios and two control rooms.

In the broadcast facility and at remote locations throughout Seoul, impressive arrays of state of the art graphics equipment were installed.

The IBC contained 12 Quantel Sports Cypher character generators, 3 Quantel Paintboxes, 1 Quantel Harry cell recorder with a Paintbox, and 6 Quantel 6031 component still stores. 3D animations were created on a Silicon Graphics IRIS system with software by Wavefront. The Grass Valley Group Kadenza and Kaleidoscope systems were used for layering and effects in graphic preproduction. An Abekas A-64 was used for both layering and recording of animations. The graphic systems were linked by a Grass Valley Group Horizon routing system.

At the venues, 9 Quantel Sports Cyphers were used to provide a consistent "look". They used the same graphic styles as those in the IBC.

Also, Dubner 5-K character generators, preprogrammed with fonts identical to those of the Cypher were used to provide a direct interface to the event clocks provided by Swiss Timing and Omega.

A freelance staff of 56 talented people manned the graphic operation: 29 character generator operators, 9 Paintbox artists, 4 still store operators, 2 graphic editors and 2 Wavefront animators.

New Systems

Quantel Sports Cypher
At the 1988 Summer Olympics the new Sports Cypher System was formally introduced to live television broadcast and production.

In the spring of 1987, NBC Sports had met with Quantel to discuss the evolution of a new character generator. This system would display any logo or cameo created on the Quantel Paintbox. Its fonts were anti-aliased. Any object on a page could be manipulated in 3D-space, in real time, to create complex animations. The system provided NBC Sports with features which were never before available for live television.

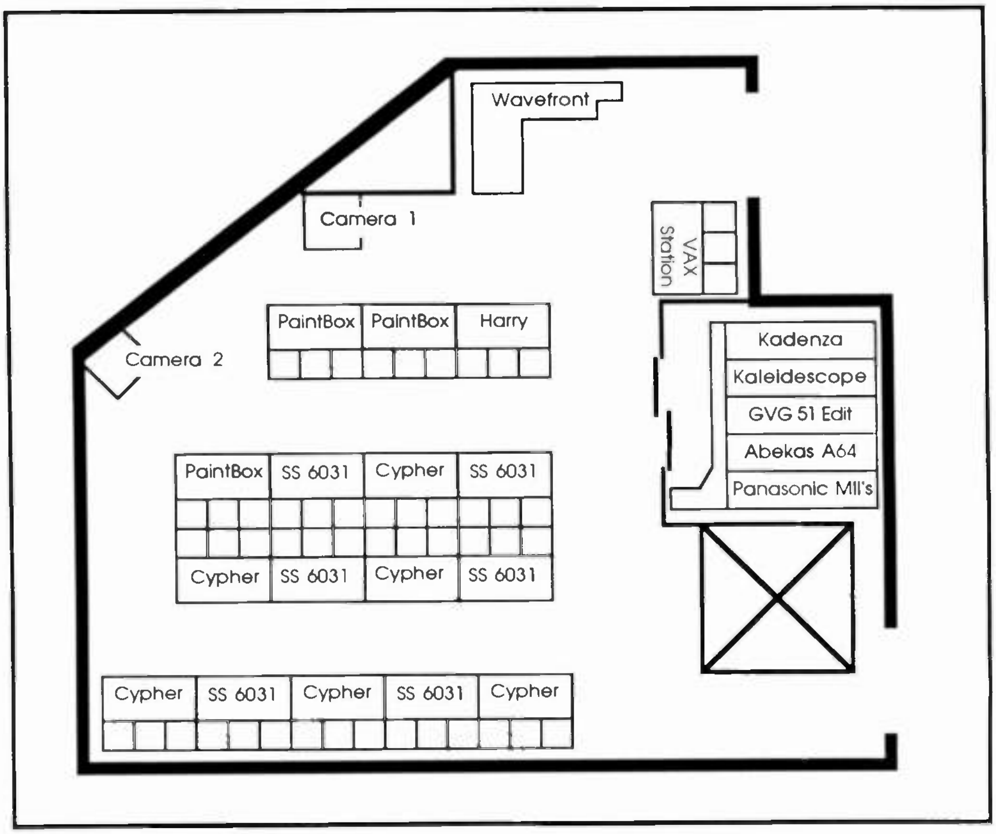
In addition the system would be linked via NBC computers to the Swiss Timing Scoring Systems and the Korean Games Information System (GIONS).

Grass Valley Group Kadenza
The Grass Valley Group Kadenza was also used on-air for the first time at the 1988 Summer Olympics.

This system is a 4:2:2 digital switcher. Its ability to take almost any video format as an input, NTSC, RGB, YUV, etc. and combine these various inputs together in real time made it an extremely useful tool in the graphic facility.

Graphic Facilities

The graphic room (Fig. 1) contained 6 Sports Cyphers, 3 Quantel Paintboxes, 1 Quantel Harry Paintbox, 2 Panasonic CLE 400 Color Cameras, 6 Quantel 6031 Still Stores, 1 Wavefront 3D Animation System, 1 Grass Valley Horizon Router, and a graphic edit room.



International Broadcast Center
NBC Graphics Facility

Figure 1

The graphics edit room contained 1 Kadenza switcher with 5 channels of Kaleidescope, 1 Grass Valley Group 51 Edit System, 1 Abekas A-64 digital disk recorder and 3 Panasonic MII VTR's.

All devices in the graphic room were connected to each other via the Horizon YUV router (Fig. 2).

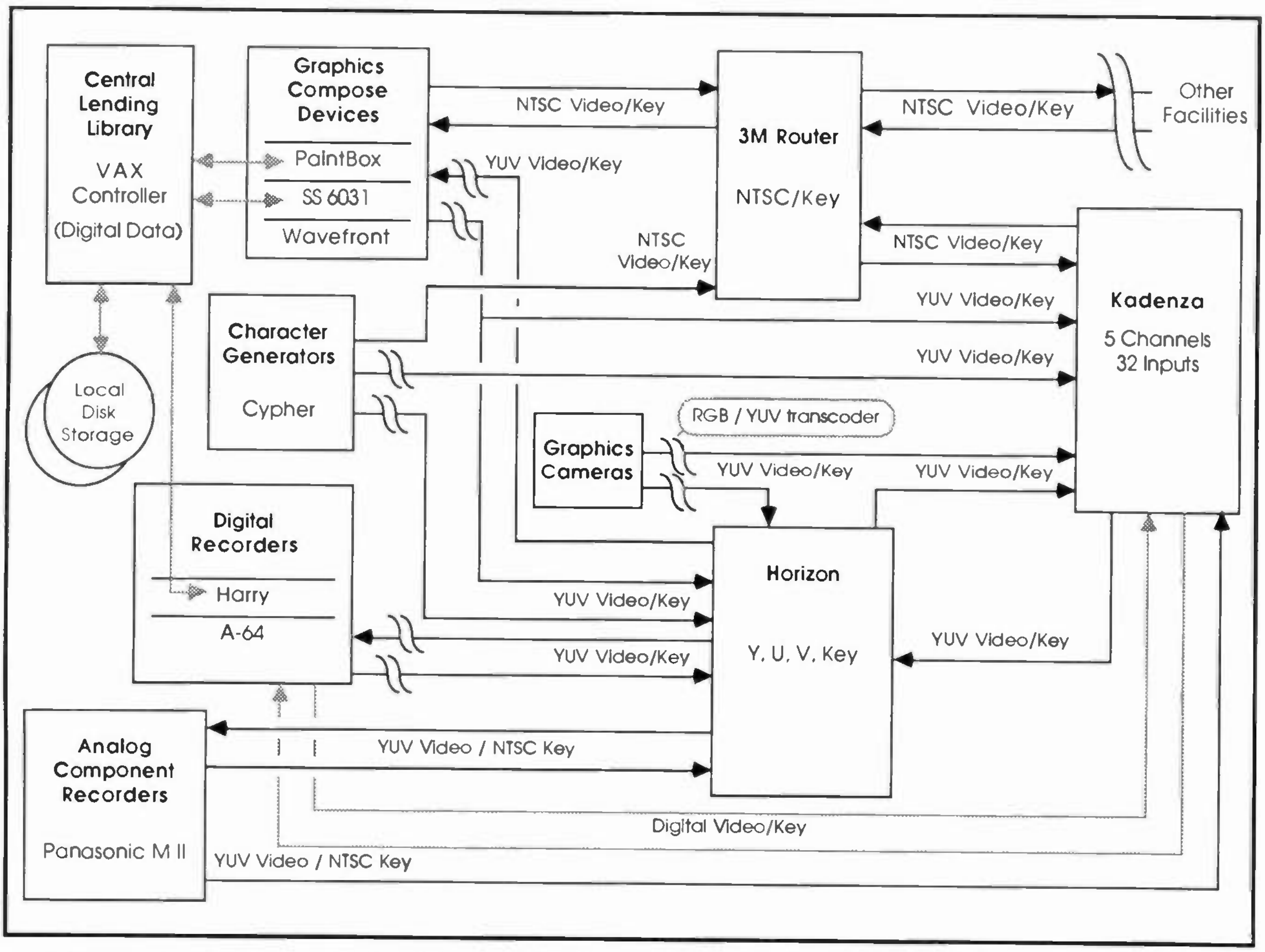
Every device in the graphic room was available to the 2 control rooms, 4 large edit suites and the 11 small edit suites via the NTSC 3M Router.

Full keying capability was available to any user from all graphic devices providing a key output.

Still stores provided 2nd page video for key shape. Cross over of video and keys was possible for all devices.

The graphics editors had full machine control of both the Abekas A-64 and the Kadenza as well as assignable GPI control of still stores and character generators. These could be locked out to prevent accidental changes while on-air.

Still stores were equipped with component RGB keyers at their input to key Cypher graphics over stills without the need for a switcher.



NBC Olympic Graphics System
Video Flow Diagram

Figure 2

All Quantel Paintboxes and still stores were digitally connected to a Quantel central storage library.

The Wavefront System used for 3-D creation consisted of Silicon Graphics IRIS 3130 workstation, a CS-12 rendering engine, a PC containing a Targa board for video input and a Diaquest controller for the Abekas A-64. All equipment was connected via Ethernet. Rendering on the CS-12 could be in progress while creation was taking place on the IRIS. Recording of rendered images was done in analog component to the Abekas A-64 or Harry.

Component Panasonic M-II recorders were used extensively for record and playback of graphics.

DEC MicroVAXes were used to interface the Swiss Timing results systems at the venues, and the Korean Games Information System (GIONS) at the IBC, directly to the Quantel Sports Cyphers.

Conclusion

Many issues had to be considered when deciding which graphics systems would be purchased and how well they would complement each other. The ability to create new graphics, animations, and logos, etc. quickly and efficiently was critical. Graphics created on one system had to be compatible with all systems.

NBC had to determine first, what our needs were, and then how these needs would be met, halfway around the world in Korea.

The selection of equipment was very difficult, not only because of our needs, but also because we had to choose from the many excellent graphics compose and manipulation systems available on the market.

ISSUES IN ELECTRONIC GRAPHIC INTERFACE TO NEWSROOM COMPUTERS

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INTRODUCTION

Increasingly, television newsroom operations are turning to computer systems which assist in the preparation of news broadcasts. Originally, these systems were newsroom tools only, which did not connect with the actual production and operational functions necessary to put the news broadcasts on the air.

Recently, newsroom computer systems developers have been extending the capabilities of their systems to include the operational functions. Broadly, these areas include prompting, closed captioning, robotic camera control, videotape playback, lighting cues, video switching, and control of electronic graphics equipment.

In this paper, we will focus on some technical and operational issues raised by the interface of electronic graphics interface to newsroom computer systems.

EQUIPMENT INTERFACE

Of the various types of electronic graphics equipment which are potentially interfacable to newsroom systems, character generators and still stores are the most common.

Operational Integration

It is important to consider the level of operational integration with graphics equipment provided by a given newsroom system. It is insufficient to simply specify that the newsroom system will interface to graphics equipment without defining the level of integration. Different manufacturers offer different levels of integration.

Still Stores

First, we will consider the case of electronic still storage systems. The connection between still stores and newsroom computers can be categorized in terms of four increasing levels of operational integration:

At the lowest level of integration, no physical data connection exists between the still store and the newsroom system. The newsroom computer system is simply used as an offline database for information about the images in the still store system. The information is used by news producers in the formulation of program rundowns.

While this level of integration is an improvement over the use of index cards, there are still some potential problems. Each time the actual images in the still store are modified, the database must be separately updated. In the real world, there's no guarantee the two will match, causing inevitable production problems.

Still stores which have an internal database capability and a computer interface port provide the possibility of increased integration - The newsroom computer can poll the still store for its actual contents.

In the first two cases, the information about the order of stills needed in a broadcast must be manually re-entered into the still store, based on the program rundown. At the next highest level of integration, we have the newsroom system generating the sequence list for the still store on the basis of the program rundown, and transferring this information to the still store.

At the highest level of integration, the newsroom computer uses rundown and script information to call up and change stills in real-time. Last minute changes in the rundown order are therefore coordinated with changes in the order of stills.

Character Generators

As in the case of still stores, different levels of integration exist between newsroom computers and character generators. The basic difference is that instead of cataloging and calling for elements which have been pre-produced (stills), character generator interface involves the transfer of text information which generally originates in a scripting database.

At the minimum level of integration, newsroom computers might generate hardcopy lists of required supers for use by the CG operator in preparing for the broadcast.

At a higher level of integration, the supers are transferred to the CG disk memory based on the program rundown and news scripts, but this is done as a block transfer before the broadcast and a CG operator is required to cope with the real-time operations.

At the highest level of integration, the newsroom computer changes the supers in real time, according to the rundown and script information. Two different approaches can be used for this: Either the text for the supers is transferred in real-time, or screens previously recorded on the CG disk are called-up in real time. Optimally, the newsroom computer rearranges the supers with last-minute changes in the running order.

CG Interface Considerations

Different strategies are used to interface newsroom systems with character generators. In general at the physical level, a serial interface is used, either RS-232, or RS-422. At the protocol level, character generators have been limited to three basic interface types: These are keyboard emulation, off-line entry, and real-time polled update.

Keyboard Emulation refers to a serial protocol in which the host computer simulates the keystrokes of a CG operator to control the character generator. In single channel CGs, keyboard emulation directly affects the online video output, and concurrent manual operation of the CG is usually not possible. Newsroom computers which operate CGs via keyboard emulation must have software drivers which handle most aspects of text formatting, such as text position, color, and font selection.

Off-Line Entry. With off-line entry, the CG handles more of the text formatting chores. A host computer port and special off-line entry protocol operates separately from the keyboard port. Delimited text from the newsroom computer is positioned into fields on a format page and then the result is recorded at an address on the CG disk. Manual operation can proceed concurrently, albeit at the penalty of a degradation or slowdown because of the need to share the CG CPU and disk. Also, offline entry ports do not necessarily provide the newsroom computer with real-time control of the CG, which, as stated previously, is necessary for on-air control, the highest level of integration.

Real-time Polled Update differs from the previous two protocols in that it is the character generator which initiates the transfer of data from the host computer. When a CG page containing special fields is recalled, the CG polls the host computer for the current information for each field, and puts the information on the screen in the proper format. This mode of operation is particularly useful for updating sports scores or other statistical information on the fly.

Interface Limitations

One reason the interface between newsroom computers and character generators has not reached its full potential is because most current character generators have been primarily designed for manual operation, as apposed to an automated interface.

Another issue is that CG control protocols are generally incompatible. The sequence of character codes necessary to obtain given visual result varies greatly between character generators of different manufacturers, and even between different models made by the same CG manufacturer.

A third problem is that with the commonly-used keyboard emulation protocols there is generally no feedback from the character generator to the newsroom system that a requested operation has been performed successfully. For example, the newsroom computer might have no way of knowing that the text just transmitted for the first line of a lower-third identifying super is too long to fit on the screen.

Few character generators are currently available which possess the ability to intelligently handle such text formatting and feedback considerations.

Proprietary Interfaces

As a result, newsroom computer systems which have succeeded in interfacing with electronic graphics equipment have done so by developing highly specialized and proprietary software drivers for whichever character generator is used. The "intelligence" of the interface resides in the newsroom system.

The high programming manhour investment necessary to develop and debug CG interface drivers gives rise to limitations in the number of different character generators supported. The various newsroom system vendors have preferences based on their experience with specific character generators. Newsroom system vendors who truly understand the complexity of developing software drivers for character generators will be very cautious about saying they can interface to CGs outside of their experience. Any less caution should be viewed with skepticism.

Certainly, this can pose a major problem if the character generator you've just purchased or already own is not on the list of those supported by the newsroom system supplier. Interfacability to the newsroom system has therefore become a one of the criteria for selecting a character generator.

Standardized Interfaces to CGs

To eliminate some of the complications of CG interface, it would seem logical to develop a standardized protocol for communication between host computers and character generators. Very little has been done in this area. At the RTNDA Wire Standards Committee meeting in December of 1988, it was suggested that perhaps a proposed standard for data transfer to computerized teleprompters could be extended to graphics equipment - However, the specifics are far from being proposed.

Also marketing economics of both newsroom system vendors and character generator manufacturers do not favor the establishment of a standard protocol. Newsroom system vendors have a major investment in their proprietary CG interface drivers, and CG manufacturers would question the return on their investment to implement changes in their systems.

There may eventually be a role for intelligent interface "black boxes" which sit between the newsroom system and the CG which provide the necessary intelligence to implement a standardized interface at both the hardware and protocol levels.

OPERATIONAL CONSIDERATIONS

Another consideration in evaluating the character generator interface is the question of what types of CG screens used in news broadcasts are generated and updated by the newsroom system.

Types of CG Screens Supported

Various types of CG graphics used in news programming can be generated by interface to host computer systems. These include lower third supers, full-screen information pages, bumper text, weather statistics, sports scoreboards, polling results, school/business closings, and election displays.

Unfortunately, newsroom systems don't necessarily offer all of these categories in the CG interface. The most commonly offered CG graphic is the lower third super, which accounts for approximately 90% of the CG graphics in a typical news broadcast.

The CG interface, particularly one including lower third supers, does result in improved efficiency. However, unless total spectrum of CG screens required can be generated by the newsroom computer, it is still necessary to utilize a CG operator to handle the residual items.

GRAPHIC DESIGN ISSUES

With all of the emphasis given to the operational aspects of the newsroom computer to CG interface, it is all too easy to forget that the ultimate goal is the visual communication of information. It is necessary to recognize the importance of basic graphic design concepts as the driving force in the appearance of the CG screens, as apposed to the technical restrictions of data driven displays. Ideally, one should not have to design the look of the supers around the limitations of the newsroom computer CG interface.

CONCLUSION

In conclusion, the interface of electronic graphics equipment to newsroom computers provides many operational advantages and economies. Despite any present limitations, in the near term future, advances in newsroom software will no doubt continue to extend the level of interface to electronic graphics equipment. In the meantime, engineers and managers who are contemplating the installation of such systems should actively question potential newsroom system suppliers to understand exactly what capabilities the electronic graphics interface will or will not provide.

FILM STYLE CREATIVITY AND DIGITAL POWER IN VIDEO ANIMATION

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Abstract

Although the traditional video edit suite has been well-established for decades, many animators prefer to stay on film - even for video projects - often completing the entire project on film and transferring to video at the end of the line.

Recently, component digital video-based systems have been developed which bring to video animation, not only the advantages of instant digital picture processing, compositing and opticals together with the power of integrated digital effects, graphics, rotoscoping and high-quality digital audio - they also bring to the video environment the natural and flexible film-style editing techniques so much loved by so many creative people.

FILM-STYLE CREATIVITY AND DIGITAL POWER IN VIDEO ANIMATION

Before we start, let's first think in general terms about how animations are produced.

VIDEOTAPE BAND 1

You can paint on cells.
You can shoot and move models frame by frame.
You can use computers to generate 3-D images.
You can use video painting systems like the Quantel Paintbox.

VIDEOTAPE BAND 1 ENDS

And I am sure that you can think of other methods which I haven't mentioned. But, however you produce your animations, there comes a time when the animated character or object has to be integrated with the other elements which bring a video production to life: the compositing of backgrounds, shadowing and lighting effects, interaction with critical sound cues and so on.

This is the area on which I want to concentrate. First of all, let us have a look at some real productions.

VIDEOTAPE BAND 2
No voice-over
2 Video Items - Nesquick and Smarties
VIDEOTAPE BAND 2 ENDS

I'm not here to discuss the merits of the various character animations or how they were generated - I want to discuss how they were meshed with the rest of the production.

What both of these examples have in common is that they were both composited in a Quantel Harry Suite. So first of all, I shall explain the elements of a Harry Suite.

VIDEOTAPE BAND 3

The Quantel Paintbox pen and tablet control system is well known throughout the industry.

The Quantel Harry also uses this method to control a hard-disk based video picture editing, compositing and digital optical effects system.

Because they use the same control system, the Harry/Paintbox combination has quickly become established as a powerful interactive graphics system.

This interactive control system has now been extended to embrace control of video and audio tape recorders

digital video effects with Quantel's Encore HUD

and digital audio for video with Solid State Logic's HarrySound.

All from the same tablet, pen and monitor control system.

VIDEOTAPE BAND 3 ENDS

Let's see how this system provides the film-style creativity and digital power that the title of the paper outlines.

The first animated example showed traditional cell animation over a live shoot. What were the problems encountered.

35mm SLIDE (RABBIT CHARACTER)

Well, the basic character has no highlighting, shadows or whiskers. This is because the animator, Oscar Grillo, delayed these important decisions until he could take full advantage of the powers of the Harry Suite.

35mm SLIDE (DARKENED ROOM)

For example, the blue cast in the darkened shot gives the picture a very individual personality, and the shadow cast by the animated character needs to be as convincing as possible. CAL Video Graphics used their Harry/Paintbox/Encore combination in the following way:

35mm SLIDE (RABBIT SHADOW MATTE)

First of all they took the shadow matte, which had been generated separately, and used Encore's Defocus feature to soften the edges. Then, to match the colour tone of the room, Paintbox was used to pick off a suitable sympathetic shade from the table leg,

35mm SLIDE (RABBIT WITH SHADOW)

and this colour was keyed into the live action through the matte to produce the shadow.

Paintbox was also used to mute the rabbit's overall colour in the darkened sequences, but then you'll notice some highlighting in this shot. This was not painted on frame by frame.

VIDEOTAPE BAND 4
Rabbit Highlight Matte
VIDEOTAPE BAND 4 ENDS

A separate matte was used to retain complete flexibility in colour choice right up until the final compositing operation.

35mm SLIDE (RABBIT WITH SHADOW)

What this meant was that the animator could keep his options open all the way along. His choice of colour tones for alternating mute and bright sequences, highlighting and shadows were not hard decisions until he saw the whole composite image come together in the Harry Suite. He had plenty of chance to use the maximum interaction in picking colours from the live action shots, continuously retouch both the animation and the live shoot as the composite image was built up, and use precisely the right technical setups to suit the compositing task in hand.

To illustrate this last point, let's have a look at the rabbit's whiskers.

VIDEOTAPE BAND 5
Rabbit Whisker Matte
VIDEOTAPE BAND 5 ENDS

The whiskers are very fine lines, and required a critical keying setup which did not suit the rest of the rabbit image. So the whiskers were matted in a separate series of cells. This sequence was coloured and keyed in a separate pass, using a special keying setup that suited the whiskers.

VIDEOTAPE BAND 6
Sequence from Nesquick tape
VIDEOTAPE BAND 6 ENDS

The other animation sequence was the Smarties promo.

In this promo, the animated characters were generated on a 3-D graphics system. Here are some of the original pencil sketches:

VIDEOTAPE BAND 7
Smarties Pencil Sketches
VIDEOTAPE BAND 7 ENDS

Individual elements from the 3-D system were rendered separately into Harry. The six individual elements were the space creatures, the galaxy, the purple planet, the spaceship, the comet shower and the master blue smartie. Here's one of the original elements:

VIDEOTAPE BAND 8
Space Creatures
VIDEOTAPE BAND 8 ENDS

Mattes for these animation elements were quickly generated automatically within Harry.

The Harry operator started with the Galaxy level, which was keyed over a paintbox background to create extra depth.

35mm SLIDE - (COMPOSITE STILL FROM SMARTIES)

Then the planet, the comet shower, the spaceship and finally the space creatures were added. Using this element-independent approach gave great freedom in the sequencing and editing phase as any of these elements could be time-shifted and faded in and out individually - for example, the spaceship.

VIDEOTAPE BAND 9
Tube fading in and out
VIDEOTAPE BAND 9 ENDS

The tube constellation was drawn on Paintbox and keyed in next,

35mm SLIDE - (ANOTHER COMPOSITE STILL INCLUDING TUBE)

and the four blue smarties were copied, cut and pasted in Paintbox from the master blue smartie generated on the 3-D system.

A serious problem arose after a great deal of time had been spent compositing, when there was a request that the blue colour of the smarties be changed at the last minute. At least it would have been serious if the job hadn't been done on Harry. First, Harry's keyer was used to generate a black and white mask from the blue of the troublesome smarties. Then Fettle - Harry's colour-correction system was used to bend the blues in a digital copy of the clip. Using the mattes the new blue smarties were keyed through the original clip, and the job was done.

This correction took less than two hours - rather than involving the re-compositing of the entire promo from the smartie laying stage.

Before we move on, let's have another look at the final result:

VIDEOTAPE BAND 10
Smartie Promo
VIDEOTAPE BAND 10 ENDS

These are just a few working examples of the numerous applications of digital power in the Harry Suite.

Finally, I would like to return to the matter of film-style creativity. Those who like working on film tend to like working on Harry. Many of the techniques are the same. For example, the visual presentation and operational routine of cutting edits and making dissolves is very similar to film:

VIDEOTAPE BAND 11
Harry Videola Sequence
VIDEOTAPE BAND 11 ENDS

Except that unlike film dissolve decisions in the Harry Suite can be reviewed straight-away.

And this film-style philosophy is extended to audio within the Harry Suite. Here you can see the Harry video clip on the left and the six sound reels or tracks on the right of the HarrySound menu.

VIDEOTAPE BAND 12
HarrySound Sequence
VIDEOTAPE BAND 12 ENDS

Not only has the Harry operator all the digital power of new technology at his disposal, but this is presented to him in an easy to understand and easy to use format, which he operates in a very similar way to a film editor using a flatbed videola and sound editor.

This has important spin-offs in relation to animation. Tight syncing of sound effects and dialogue lip movements are crucial in making a convincing animation sequence, and the Harry/HarrySound combination makes light work of these tasks in the video environment.

Before I close, I would like to thank Sally Kneel and Terri Hylton of CAL Video Graphics for their kind assistance in writing this paper, and I would also like to thank Oscar Grillo and Martin Lambie Nairn for their permission to use the animated sequences I have shown you today.

To finish off this presentation, I would like to play a tape. The main character in this tape is not animated, but the reason I want to play the tape is to show what can be achieved in post-production in a Harry Suite.

VIDEOTAPE BAND 13
Short Clip of Lady + Matte
VIDEOTAPE BAND 13 ENDS

The lady who is the central character in this video was shot against a blue screen, and for all intents and purposes can be thought of as an animated element in the post-production process. All of the special effects in the piece were created after the blue-screen shoot, and it doesn't take a great deal of imagination to see how these effects can equally be applied to the post-production of animated sequences.

VIDEOTAPE BAND 14
Doll's House Tape
VIDEOTAPE BAND 14 ENDS

NEW TRENDS IN WEATHER GRAPHICS, IMAGES AND HARDWARE

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This past year has brought some subtle but significant changes in the area of weather graphics. On the one hand, the cost of hardware for creating and displaying weather images has dropped significantly in price, thanks to the Amiga computer by Commodore. New Amiga units with enhanced capabilities are now available for under $10,000 and the result has been that smaller stations, independents and even public television stations have been able to enter the world of real-time weather graphics. We see this trend accelerating especially over the next year or so. Some of the new capabilities available on Amiga weather graphics units will be discussed.

There has also been a sharp increase in the use and desire for higher resolution and more colors so that weather graphics would be of the same quality and could have the same style and look as many graphics used for news and sports. As a result, the trend toward high resolution, high color, weather graphic images is increasing. Expected trends along with methods of transmission, equipment and new types of weather graphics that can be produced in this mode will be discussed.

The evolution of weather graphics over the past 25 years can be compared to the two graphs that you saw on the tape. One showed the acceleration of growth in world population. There was the slow increase in population through the middle ages and then the rapid acceleration over the last hundred years or so as disease, famine and plagues have been eradicated. The second graph has a similar parabolic shape; it shows the growth in computer modems in the use in the U.S. While these graphs have nothing directly in common with the parabolic acceleration in the complexity and use of weather graphics, the nature of the curves is, I believe, similar. During the 1950's and 60's, weather presentations on television evolved slowly. They then increased a bit in the 1970's but as late as 1980, the old fashion weather board remained the main backdrop for presenting national and local weather.

Two new graphics appeared in the 70's. One a black and white satellite photograph, the other a color radar display.

The black and white satellite pictures allowed viewers to see for the first time the intricacies and complexities of the atmosphere as shown by the detail in cloud photographs. Likewise, local radar displays showed the complexity and detail of precipitation patterns to the public for the first time but these two developments were merely the tip of the iceberg. The major changes in weather graphics and displays have come in the last seven or eight years, with the introduction of lower cost computer power including the personal computer.

The spectacular developments that have occurred since 1981 in the presentation of weather information and forecasts include the development, refinement and sophistication of colorgraphics equipment displays and the application of ingenuity and creativity to develop dozens of new ways to portray and display the weather to peak and excite viewer interest.

Simply by reviewing the tremendous changes that have occurred in our graphic offerings at Accu-Weather, we can appreciate just how sophisticated television viewers have become in their understanding of weather phenomenon and weather patterns, and how much more they are looking for and will be expecting in the future. Surveys continue to show that viewers are hungry for weather information. Likewise, in the past few years, we have seen a significant increase in the weather sophistication of News Directors, General Managers, Weathercasters and Engineers.

When Accu-Weather first began offering graphics six years ago, they were limited compared to today's selection. Yet, they more than satisfied the wants and needs of television Weathercasters, Meteorologists, News Directors and viewers. Now, we offer over 1500 graphics every day and the number and variety is increasing weekly. Let me give you some examples.

We initially offered national radars twice daily. These are composites of over 100 individual radarscopes across the United States. In response to client demand, we began offering these national radars every hour. Then, clients indicated they wanted regional radars. As a result we divided the country into six regions and began offering, in addition to the national radar composite, six detailed regional color radar composites every hour.

These radar products, while outstanding portrayals of the underlying national weather service radar data, suffer from the faults in all government radar output. For example, these are gaps in radar coverage because of an insufficient number of radar sites and because of attenuation of the radar signal due to terrain shadowing or city skylines in certain areas.

As a result there are many areas in the U.S. where the radar representation is unreliable and inaccurate. In some places there simply is never any precipitation shown on National Weather Service radar reports.

In order to overcome this, two years ago Accu-Weather developed an enhanced radar called RADARPLUS™. Since radar displays are simply convenient ways of looking at precipitation coverage and intensity, we had our mainframe computers decode and integrate them with more than 1000 surface reports of precipitation, coverage and intensity, from across the United States and Southern Canada every hour. Through a proprietary software package we were able to integrate these surface weather observations into the hourly radar displays to show a more accurate representation of precipitation coverage and intensity; and, the display looks smoother, more organized and more attractive. In all likelihood, five or ten years ago, there would h ave only been mild interest in such a new product but today increasing numbers of television stations are using RADARPLUS. The demand did not stop there.

This past winter in response to further requests, we offered as an option a new version of RADARPLUS that depicts the kind of precipitation with white areas indicating snow, orange areas indicating sleet and freezing rain and green areas indicating rain. A further example of the broadening demand for weather graphics is shown in lightning graphics. Now, for the first time in almost 20 years there is a brand new way to show precipitation and severe weather, and that way is by showing graphic images of lightning occurrence, frequency and intensity. R-SCAN has set up a nationwide of lightning detection equipment that is used to monitor and detect each cloud and ground lightning flash and then relay that information instantaneously into our computer--providing the precise time, location polarity and peak current, all within seconds of each lightning hit. By collecting, decoding and formatting that data, it is possible to make available ready-for-air lightning display graphics, on a local, regional or national basis. These lightning graphics can be seen for the first time here at the 1989 NAB, and they will be in a number of different formats. There will be regional and national lightning displays, which show the frequency and intensity of discharges over different periods of time; for example, going back 30 minutes, an hour or three hours. In the future we will see graphics that superimpose lightning displays over radar and satellite images.

You might ask at this point how many different ways can you sense and show the weather? For instance, why is there any interest in real-time lightning? Lightning data is an outstanding short range thunderstorm forecasting tool. Lightning data complements and expands the value of radar and satellite imagery and provides information that can be used to control risk and reduce lightning related losses. Lightning is one of the major causes of weather related deaths in the United States. In some cases, lightning data may show thunderstorms developing before there is any indication of them on radar. In other situations, there may be no indication on the radar scope or on the satellite image of thunderstorm activity, yet the lightning graphic may indicate that there is severe and dangerous electrical activity already underway. Lightning data will be a valuable forecasting tool to your meteorologists. In fact, there was a storm in the gulf of Mexico this past hurricane season that developed rapidly from a tropical storm into a hurricane without much advanced notice; however, those forecasters access to the real-time lightning data clearly saw this development coming 12 hours in advance of official advisories because of a sharp increase in lightning activity around the eye of the storm.

Now turning our attention to weather graphics systems, some subtle but significant changes have been seen during the past year. On the one hand, the cost of hardware for creating and displaying weather images has dropped sharply thanks to the Amiga computer by Commodore. The Amiga Weather Graphics System was first introduced at the NAB last year. This system, which is used for receiving and displaying weather graphics is a result of software designed by Associated Computer and Accu-Weather programmers. You will be able to see second generation systems on display at the NAB this year. They sell for under $10,000, including hardware and software, and they contain many of the capabilities found in graphics computers costing $30,000 to $60,000. The changes wrought by the Amiga have, in the past nine months, been dramatic. Many smaller stations, independents and even public television stations have been able to enter the world of real-time weather graphics. In fact, within the last nine months, we have sold about 30 of these graphic units which has allowed these stations to go directly on-air with sophisticated weather graphic images. The viewer can't tell the difference between this $8000 Amiga and $50,000 sophisticated graphics computer. We see this trend accelerating especially over the next year or so. The Amiga Weather Graphics System already contains these capabilities: paint, color movies, automatic log-in, fast-frame looping, sophisticated wipes, automated graphic access and show production. Not only will the Amiga graphics system show all of the capabilities inherent in weather graphics but they also have these capabilities: Characater Generation, Graphics Manipulation and Animation, Graphics Library, Teleprompter, Tape Library Management, Tape Editor, Timing and Scheduling, Remote Controls and we anticipate over the next six to 12 months, the following additional capabilities may become available.

Interestingly, not all of the sales of these Amiga systems are going to small market stations, independents and public television stations but, in fact, 20% to 25% of the sales are going to medium market stations that want these units for two reasons. One reason is as a separate unit for weather so that the more costly, more sophisticated unit can be used by the News, Sports or Production Department. A second reason is as a backup in case their LiveLine III or primary graphics system goes down.

Of course the Amiga Weather Graphics Units are four bit, 16 color, medium resolution machines.

On the other side of the coin are the high grade units such as LiveLine IV and V, Artstar, Dubners, Vidifonts, Cubicons, Chyrons and Quantels that currently produce high resolution graphics with 256 to 16.8 million colors or more. These units are typically used for more nonweather applications than weather applications, although in the top 15 or 20 markets they are often used for the creation of weather graphics. However, in recent months there has been a sharp increase in the use and desire for higher spatial resolution images and more colors in weather graphics, so they will be of the same quality and can have the same style and look as the graphics used for news and sports. Sensing an increasing demand for these higher quality more expensive on-air graphics and believing that this is the long range trend for the future, we have developed a package of high spatial resolution, high color resolution weather graphics to meet this need.

Some other weather companies have offered these high resolution graphics for a while now.

For example ESD has been offering 256 color high resolution satellite images for more than a year. Weather Central has been offering certain weather maps and charts customized for the LiveLine IV and V as well. Sensing that this is the wave of the future, we have spent a good deal of time and resources in R&D over the past two years in working toward a complete offering of high spatial resolution, high color resolution graphics images and you saw some samples of these on the tape. Interestingly, all of our high resolution color images have been created on the Macintosh Computer. Not only will these Macintosh-created graphics interface with the LiveLine IV and V and some other high resolution graphic equipment but they can also be received and displayed on-air by a new MacWeather graphics unit that you will be able to see on the floor of the convention for the first time tomorrow. The MacWeather computer may follow the trend established last year by the Amiga; that is, it doesn't have all the capabilities of a LiveLine V but it doesn't carry the price tag of a LiveLine V either. These units come in at under $25,000, and so demand will be spurred by cost for a high end weather unit.

In developing and delivering high resolution weather images there are a number of complicating factors that any weather service company needs to deal with. First of all, many stations want to maintain a similar look from news to sports to weather. As a result, they may not want to use the map background that the weather service company is offering. Thus, it is necessary to offer graphics with map backgrounds as well as without, so the station can simply overlay the clouds, fronts, and data or onto the base map created by the station. This allows the station to "customize" the weather section to their look, appearance and motif. However, this apparoch does require that the dimensions of the maps be standard. There is also the question of transmission time. A typical 256 color image takes up 200 to 300 kilobytes compared to 50 kilobytes for a 16 color medium resolution image. As a result, the transmission time at 2400 baud would be six to nine minutes compared to only one to one and a half minutes for medium resolution. To overcome this problem and to minimize communications costs 9600 baud modems with effective of our 16000 baud from data computers are strongly recommended to reduce transmission time to two to three minutes.

Also the quality of the image has to be maintained by using the high color resolution to accentuate the weather message and look, not to over complicate the image. Pouring on color without purpose can frustrate the higher technical capability that higher color resolution can give.

As I stated earlier, the revolution in weather graphics continues both in the type of weather images that are available for display and also in the type of receive and display graphic units available. For medium resolution 16 color images it is the Amiga Weather Graphics Unit and this will continue to be a popular item for the smaller market and independent stations and increasingly for public television, whereas the Macintosh basic MacWeather Unit will, I believe, be one of the exciting hardware developments of 1989.

WORK OF THE BTSC MODULATION MONITORING COMMITTEE

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ABSTRACT

Broadcast television stereo has been with us for nearly five years, and BTSC has proven to be a system that delivers high quality stereo sound to the television audience. The fact that around 25 percent of U.S. television households have MTS reception capability attests to the viability of the BTSC transmission system.

There remains one unresolved issue in BTSC operation: the specific response characteristics of stereo modulation monitoring devices. Not all stereo television modulation monitoring devices produce the same indications when stimulated with the same program signals. The BTSC Modulation Monitoring Committee was formed to address these disparities. This is a progress report on the work of that Committee.

Background

The Broadcast Television Systems Committee of the Electronic Industries Association (EIA/BTSC) proposed a method to transmit and receive multichannel television sound (MTS) to the Federal Communications Commission in December, 1983. The BTSC MTS system was selected after considerable research and testing, and it supports not only stereophonic sound, but also a second audio program channel and a non-public channel for professional use, although only stereo has been widely implemented at this time. In many respects the BTSC system is similar to the FM stereo broadcast system used in the United States, but there are some substantive differences. The most radical difference between the FM stereo and television stereo transmission systems is the processing of the television stereo L-R or difference signal with noise reduction companding. This processing significantly improves the signal-to-noise ratio of the difference channel, thereby overcoming the "Achilles heel" of the amplitude-modulated subcarrier approach to multichannel sound broadcasting. It also creates a non-linear transmission system that requires some special considerations not required by the linear FM sum-and-difference system.

The BTSC system was approved by the Federal Communications Commission in early 1984 and is the de facto standard for television stereo transmission in theUnited States. Although MTS transmissions became legal in April, 1984, the first BTSC stereo transmission actually occurred in late July, 1984, and regular television stereo transmissions on a wide scale did not begin until 1985. Today, five years later, about 99 percent of the United States television viewing audience is within the coverage area of at least one television stereo signal, and about 25 percent of U.S. television households have stereo television reception capability, attesting to the success of the BTSC system.

An Unresolved Issue

There is at this time an unresolved issue concerning BTSC stereo operation. This is the matter of the specific reaction times and ballistic characteristics of modulation monitoring devices for BTSC stereo. Modulation level requirements for stations transmitting BTSC stereo are defined in FCC Office of Engineering and Technology Bulletin No. 60, Revision A[1], which specifies FM deviation limits for (1) the main channel or sum signal, (2) the combination of the stereophonic sum and difference signals, (3) the stereo pilot signal, and (4) total deviation of the aural carrier by the composite MTS signal. In keeping with current deregulatory philosophy, OET-60A does not specify any method for measuring those parameters.

The Recommended Practices

Comprehensive engineering and operating information for the BTSC system is found in EIA TV Systems Bulletin No. 5, *Multichannel TV Sound System--BTSC System Recommended Practices* (July 1985)[2]. The *Recommended Practices* specifies measurement of BTSC transmission parameters with meters, and additionally recommends peak flashers to indicate instantaneous peaks of both the main channel signal and the sum of the main channel and the stereophonic subchannel, while leaving the question of the ballistic response characteristics of these meters and peak indicators unresolved. In its discussion of modulation metering, that document states that the major concerns with respect to TV stereo overmodulation are distortion of the audio program and crosstalk among BTSC channels, and not, as in the case of FM radio, interference with adjacent channels.

Until final monitor design specifications are formulated, the *Recommended Practices* offers interim guidelines adapted from the former FCC aural modulation monitor rules, as they existed before their elimination in 1984. These rules specified a modulation meter with ballistic response such that modulation peaks of duration between 40 and 90 milliseconds are indicated to 90 percent of their full value. Upon signal removal, the indicator must fall from full reading to 10 percent of full value within 500 to 800 milliseconds. These ballistics impart a semi-peak indicating characteristic. A peak indicating device, or peak flasher, was also specified to detect modulation peaks of shorter duration than the meter could. The peak flasher was required to respond correctly to tone bursts at repetition rates from one to ten bursts per second with these two burst compositions: (a) ten consecutive cycles of a constant-amplitude 10 kHz sine wave; and (b) five consecutive cycles of a constant-amplitude 1 kHz sine wave. This specification corresponds to an integration time of about two milliseconds.

While the parameters that should be monitored when transmitting BTSC stereo are well-defined, the exact method of doing that monitoring is not. To borrow a sentence from IEEE/ANSI 152-1953 (R-1971), *Recommended Practice for Volume Measurements of Electrical Speech and Program Waves*[3], the document that standardizes the vu meter, "The measurement of complex and nonperiodic audio-frequency electrical signals cannot be expressed in the ordinary electrical terms of current, voltage, and power." The complex and aperiodic nature of such signals causes the indications they produce in a metering device to be dependent on the response characteristics of that device. Audio broadcasting professionals are aware of the differences observed when a vu meter and a peak program meter are driven with identical audio program signals. The ballistic characteristics of the respective meters produce very different reactions to the same program material. The vu meter is an average-responding device with a relatively long integration time and approximately equal rise and fall times. The peak program meter is a quasi-peak reading device with a much shorter integration time, a shorter rise time, and a much longer fallback time than the vu meter.

The ideal response characteristics of a metering device depend on its intended use. The former FCC modulation monitoring rules were basically developed for the FM broadcast service, where an overriding consideration is the adjacent channel interference that may be caused by overdeviation. An FM station may deviate its carrier ±75 kHz, and its allocated channel is but ±100 kHz wide, leaving a 25 kHz guard band on either side of the occupied spectrum. This generates a requirement for rather tight control of peaks to protect adjacent broadcasters.

Television stereo broadcasting is a very different situation. There is 250 kHz of spectrum between the TV aural carrier and the upper channel limit. Modulation of the television aural carrier with a fully loaded BTSC baseband fills a lot of that spectrum, but rather serious overdeviation is required to occupy spectrum beyond the channel limit, and audible distortion will occur in most receivers before that point is reached. Serious overdeviation in the downward direction will interfere with the TV station's own visual signal. The BTSC stereo system takes advantage of this relatively large amount of spectrum, in that transient peaks are generated by certain elements of its encoding system. Specifically, these transients are overshoots generated by the noise reduction compressor and the sum and difference low pass filters required by the BTSC transmission system. Such overshoots are very brief in nature, being less than 5 milliseconds in duration, and they contain insufficient energy to be audible. Because of the spectrum space available to the aural signal, neither do they cause any harm.

The absence of FCC regulations specifying the nature of MTS monitoring devices leaves the door open to concepts other than the traditional meters and peak flashers defined in the old modulation monitoring rules. Using modern techniques, it is possible to fully indicate every modulation peak, no matter how brief. As we have seen previously, however, the transient peaks generated by the BTSC encoding process are too brief to be audible, and offer very little interference potential. These inaudible transients are processing artifacts and not a part of the audio program. If modulation levels are set in consideration of such transients, the average level will be lower than that which would be realized using peak indicators that "ignore" the transient artifacts of BTSC encoding. It may also result in a mono sum signal that is lower in average level than the signals of mono broadcasters using modulation monitors which were designed in compliance with the former FCC rules, raising the issue of loudness compatibility between broadcasting stations. Changing channels should not result in the need to adjust the volume control, requiring the relative loudness of the mono sum signal to be equal to that transmitted by other stereo or mono stations.

The Modulation Level Subcommittee

Observed disparities between modulation levels of television stations using the BTSC stereo transmission system, and between stereo and mono broadcasters, has generated a certain degree of confusion among television station engineers. To address these questions, the EIA Broadcast Television Systems Committee with the cooperation of the National Association of Broadcasters formed the Modulation Level Subcommittee. This group is comprised of representatives of television broadcasters and the manufacturers of TV stereo monitors and encoders. The first meeting of the Modulation Level Subcommittee in June, 1988, produced agreement on these considerations for specification of the ballistic response of TV stereo modulation monitoring devices:

- audible distortion must be prevented,
- relative monophonic loudness parity should prevail when switching between channels,
- all stereo monitors should yield similar results when used to monitor the same stereo signal.
- there should be no harmful interference to other signals or to the broadcaster's own signal.

It was agreed that of FCC OET Bulletin No. 60 should be amended as necessary to reflect these conclusions. It was also agreed that the total composite stereo modulation level (less pilot) limit of 50 kHz on peaks of frequent recurrence can be exceeded not only by certain test signals but also by certain real-world audio input signals, notably in some cases where either the left or the right channel is excited with certain audio signals. It has been observed that in such cases total sum plus difference modulation can cause aural carrier deviations in excess of 60 kHz (less pilot), and that the limit should be increased to reflect these occurrences.

A task force composed of representatives of monitor and encoder manufacturers and broadcasters was appointed and charged to define the desirable characteristics of modulation monitor peak indicators and to develop a recommendation for the necessary amendment to OET-60A, and to report back to the full subcommittee when their work was completed. The task force produced a test demonstration in which four different BTSC modulation monitors were driven in parallel by each of three different television stereo generators. After sweeping the stereo generators to assure their compliance with BTSC performance standards and calibrating each monitor using the method approved by its respective manufacturer, several tests were run. Tone burst tests were performed which included standard monaural and modulated stereo subcarrier tests. It was generally agreed by participants in these tests that with certain exceptions, reasonably close correlation between monitor peak flashers was observed. Audio program tests were then conducted, using both compact discs and actual television program audio as sources. Program audio tests revealed that monophonic correlation between monitors fell within 1 or 2 dB, while stereo correlation was appreciably wider, often varying 3 dB or more. Efforts to determine the exact nature and cause of discrepancies among stereo peak flashers with program audio signals were inconclusive.

Conclusion

Although the work of the Modulation Level Working Group is not finished, some conclusions can be drawn from the work done so far. Stereo modulation monitors generally agree in their indication of the mono sum signal, but their indications of stereo and total composite signals differ substantially. It is felt that a two millisecond integration time constant for a monitor's peak flasher, or a two-millisecond "holdoff time" before peaks are responded to, is in order, and that the composite stereo modulation limit (including pilot) should be relaxed from 55 kHz maximum aural carrier deviation to 65 kHz.

The rise, fall, and integration times of stereo television modulation monitoring devices must be selected carefully to attain the desired objectives. The goal of these design and operating criteria is to provide the basis for uniformity in the monitoring of television stereo modulation, and to maintain loudness parity among television aural signals. It should also eliminate the degree of confusion that exists among television station engineers about exactly what constitutes compliance with the FCC's modulation limits.

References:

1. Federal Communications Commission Office of Engineering and Technology, *OET Bulletin No. 60, Rev. A: Multichannel Television Sound Transmission and Audio Processing Requirements for the BTSC System*, Washington, February, 1986.

2. Electronic Industries Association, *TV Systems Bulletin No. 5: Multichannel TV Sound System--BTSC System Recommended Practices*, Washington, July 1985.

3. Institute of Electrical and Electronic Engineers, IEEE/ANSI 152-1953 (Rev. 1971), *Recommended Practice for Volume Measurements of Electrical Speech and Program Waves*, New York, 1971.

A NEW APPROACH TO TELEVISION AND FILM POST PRODUCTION AUDIO

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Abstract

With the ever increasing requirement for multi-facet production and postproduction environments, the need and justification for specific task audio consoles is becoming limited to major production and post-production houses. To remain competitive, a facility should encompass as wide a variety of customer needs as possible. This paper describes a unique approach to the problem utilizing proven equipment able to respond to these changing needs -- a new postproduction audio console, together with enhanced automation systems, designed to satisfy a wide variety of tasks.

Highlights include television postproduction, Dolby matrix film and film re-recording. A complementary moving fader automation system integrated within the audio console proper completes the system.

THE CONSOLE

Introduction

A stereo television postproduction console and a single operator film re-recording console have many similarities given the types of tasks encountered in current facilities. Any console designed to fulfill these tasks has to perform several functions, each being obvious to the user. The design criteria was to provide a useful and sophisticated system which was transparent in operation to the user. Such exercises as intensive patch re-routing, multi-pushbutton switching, and complex operator moves are minimized by clever console routing. Valuable features such as Dolby matrix monitoring on switchable 4-track and 8-track video post working are accomplished by a unique film, television and multitrack architecture.

Facilities

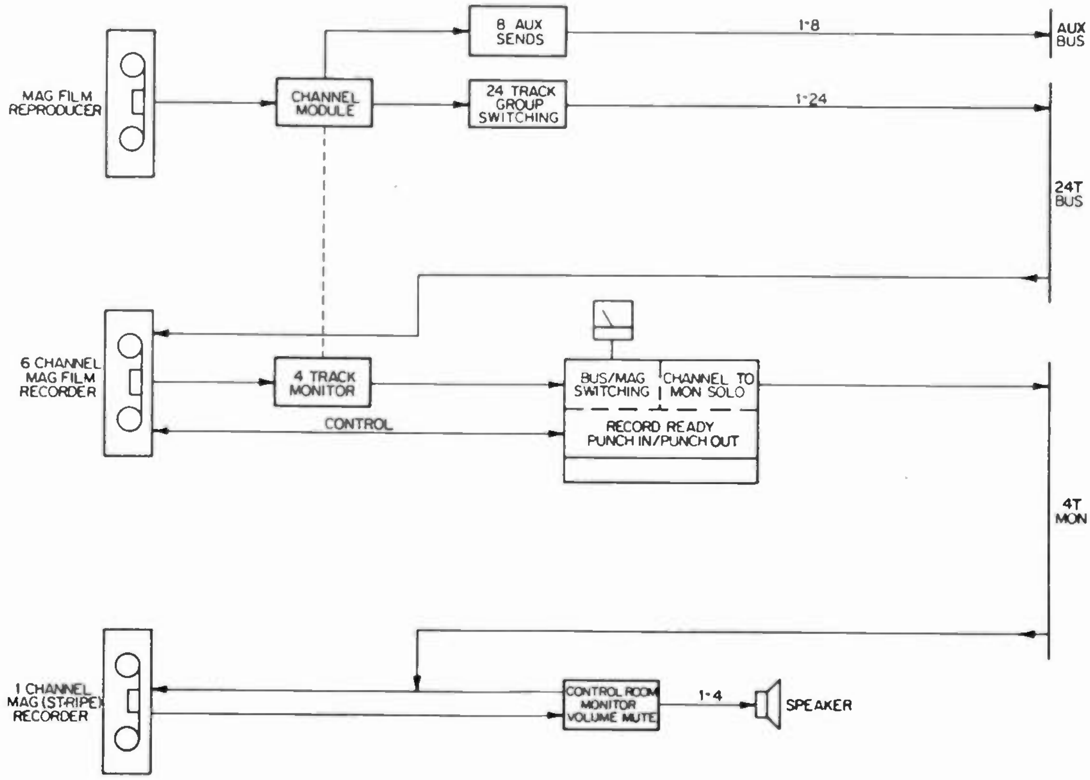
After careful study in conjunction with several U. S. facilities, the following target facilities were specified:

- 24-track multitrack operation with provision for up to two 24-track machines
- 8-track stereo television operation
- 4-track mono television operation
- Independent master recorder and stereo monitoring
- Alternative program to recorder and monitor to recorder facilities
- Master machine control
- Alternative mix balance to monitor via "Trim" provision
- Individual or paired additional sweetening via specialized utility output
- Music and effects separate feed
- Dolby DS4 matrix with solo interrupt
- 4-channel LCRS monitoring
- Stereo derived outputs
- Mono derived outputs
- Film, television, and multitrack architecture
- LCR pan provision
- LCRS quad pan provision
- Switchable bus panning

Basic Console Set Up

As a stereo television postproduction and single operator film re-recording console, full use is made of existing NEVE V Series console architecture with a separate console section dedicated to postproduction operation.

The key to NEVE's system lies in the dedicated master mode status section. Here the primary signal paths are chosen, each being suited to the application selected. An operator selects the required mode for the task in hand; the console architecture directs the signal flow into recognized patterns.



Basic Flow Diagram, Fig. 1

Monitor Master Status

The primary configuration for console architecture is set up using the Monitor Master Status controls. These control the global commands to the console and provide the necessary switching to effect the major modes of operation.

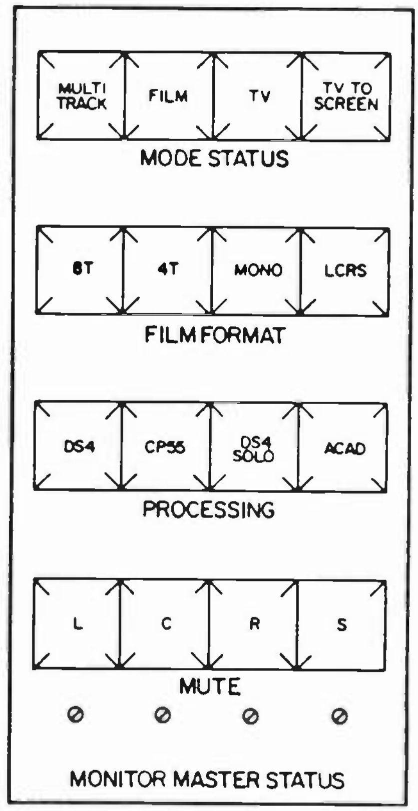
The following controls are provided.

1. Multitrack

Console operates as a standard NEVE V Series multitrack console with the post-production facilities having no effect to the operation of the system or in terms of degradation of the audio by having to pass through the electronics associated with the postproduction facilities. Up to 48 busses are accessible.

2. Film Format

Signals to and from the console are available from multitrack and 8-track busses with returns from tape machines or dubbers being returned directly to the monitor matrix at unity gain. Feeds to full-coat machines are independent of console monitor, and console monitor can be arranged in several formats according to the position of the master film format switching. Up to four separate loudspeaker monitors can be selected.



Monitor Master Status, Fig. 2

3. TV

The eight busses normally associated with the multitrack mixdown are assigned as feeds to the record device which may be either 4-track or 8-track. Returns from the record device are brought back by A,B,C,D monitoring facilities with separate signal paths for master record machine and control room monitor.

4. TV to Screen

In mixing stereo TV to a full size film screen, the alternate speaker feeds available to soffit, close-field, and TV can be sent to speakers normally mounted behind the projection screen. This allows the producer to hear a theatrical mix for full emotional effect.

When the console is in film operation, four alternative film formats from mono to LCRS can be selected, each selection automatically rerouting the appropriate signal path.

6T -- allows 6-track (actually up to 8-track) feeds to and from tape machines mixed into mono, stereo or LCRS mode through the console.

4T -- allows 4-track film recorders to be routed to and from the console in mono, stereo or LCRS working.

Mono -- enables monomix monitoring of 4-track (3-track) sources.

LCRS -- provides discrete or matrixed 4-track monitoring to left, center, right, and surround.

Having selected the film style required, matrix processing, where used, can be monitored. With Dolby DS4 matrixing, it is useful to be able to switch in and out of circuit the complete encode/decode package and/or monitor the effect of the decoder only.

For soloing, again both matrix and non-matrix signals can be selected.

Academy interjects customer-supplied Academy filter into mono feeds to console monitor (center channel).

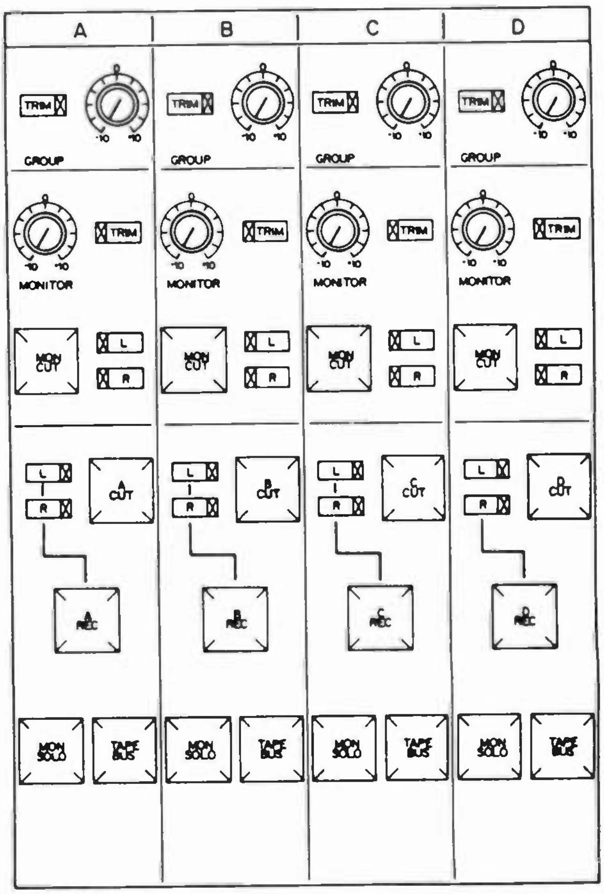
Turning to TV Operation

In TV-style postproduction, the individual A,B,C,D elements, whether in mono (4-track) or stereo/mono (8-track), can be mixed in any combination to the final signal format. Two distinct signal paths exist; one normally is associated with the console and control room monitor, the other with the 2-track (or mono) master recorder. The signal path is such that sources from the console 4/8-track main outputs are sent to their respective record tracks at a nominal fixed level which may be trimmed +/-10dB in stereo pairs indicated as group trim on the panel.

The return from each tape machine's track or pairs of tracks is routed via a tape bus switch relay that can flip on an individual or master basis to either the record bus or machine playback, after which the signals are split to either the monitor path or 2-track record path. Again, +/-10dB trims are available for user adjustment.

Hearing the Mix

Common to both film and TV operation is the A,B,C,D element monitor matrix.



Monitor Mix Panel - Fig. 3

Controlling both the film and TV mix components to the monitor, a maximum of eight tracks may be mixed in four groups to the console monitor and, in TV mode, to the record machine. Each of the four signal paths are identical and are controlled globally by a master status section included with the local controls on the same panel. It is intended to offer two versions of the panel: Version A will utilize pushbutton controls similar to that used elsewhere on the NEVE V Series console; Version B will use a combination of pushbutton and paddle switches familiar to West Coast film users. Paddle switches will be available for use in the tape bus switch and master record functions. All other buttons will remain pushbuttons and perform identical functions in both types of configuration.

1. <u>The A,B,C,D Elements</u>

Each of these signal paths can be addressed as individual elements (i.e., music, dialogue, effects, etc.) in either mono or stereo formats Separate paths exist for both monitor and program paths.

<u>Monitor Path</u>. Monitoring of each of the A,B,C,D groups is independent of the main signal path and includes a cut switch that may be preset to cut (mute), all elements of bus A, or individual elements (up to four -- normally LCRS) by means of associated inhibit switches. The stereo or mono mix from this bus may be altered in level from its normal by means of a series trim control inserted by its own switch having +/-10dB of range.

A monitor solo switch enables individual busses or combinations of busses to be monitored without affecting signals to a master record machine.

<u>Program Path</u>. A parallel signal feed from the tape bus switch is also provided. This signal path contains no level adjustment, but it does incorporate individual cut switches that can isolate unwanted signals. Local record in/out switches with record isolate functions for either the left and/or right channels is also provided as part of these facilities.

2. <u>Master Status Controls</u>

In addition the following master status controls are provided:

<u>4-Track/8-Track Monitor Mode Switching</u> Both stereo and mono monitor and machine sources are available in either mode with the pan controls on input modules operating between the A/B and C/D busses in 4-track, whereas each A,B,C,D bus becomes a stereo bus with pan in 8-track.

<u>Monitor Direction.</u> Selection of either monitor mix or machine mix to the control room monitors.

<u>Machine Direction.</u> Selection of either machine mix or monitor mix to the 2-track machine.

Finally, an insertion via the utility bus on any or all of the A,B,C,D busses is available such that an M & E may be easily configured from any bus arrangement, with the utility bus capable of being a stereo M & E feed. In addition, by repatching the utility bus equalizer together with, for example, bus A, EQ may be added to an individual bus. This makes for speedy sweetening to an individual track when dialogue is either no longer available or is difficult to reconstruct.

<u>LCS Panning</u>

For situations where 3-track or 4-track panning is required, console or rack-mounted panners may be specified that provide both joystick and rotary panning facilities. In the case of the rotary facilities, the following panning arrangement is provided:

- o LCR 3-track pan
- o Divergence panning between mono and wide stereo
- o Single-channel surround

All inputs and outputs to panners are accessible on the console patch bay, the facilities being inserted as needed into the console signal path.

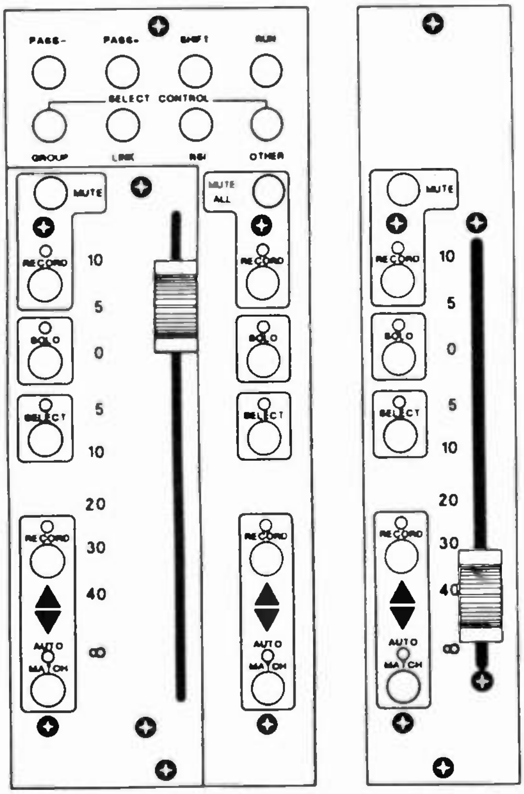
AUTOMATION

In parallel with the availability of the NEVE console is a new moving fader automation system, Flying Faders.

Flying Faders is NEVE's fourth generation console automation system designed in conjunction with Martinsound Technologies of Alhambra, CA. Both on line and off line facilities are included within the system architecture, together with a sophisticated machine interface controller allowing master and slave operation. Some of the many facilities included with the system are:

<u>Total Mix-Memory</u>

Pass after pass can be made without thinking about whether or not you like the way things are going. If at some point you decide to go back to an earlier pass and have another mix attempt, you can call up any pass from scratch memory, ready for instant playback, without keeping anything to disk.



Stereo Master and Global Controls *Channel Fader*

Flying Faders Controls, Fig. 4

Boundless Mixing With 1/10th dB Accuracy.

Play can be initiated anywhere on the tape, and Flying Faders will play back your moves and mutes for that point. There are no limits and no From and To times are required. On recording, Flying Faders scale is divided into 4,096 digital steps for essentially perfect analog to digital and digital to analog conversion of the most subtle moves. In addition, all levels are stored to 1/10th dB accuracy, anywhere along the scale.

Channel Module Automation

The Channel Mute, Monitor Mute, EQ In and Insert buttons can be directly interfaced to the Flying Faders with the existing channel strip buttons in complete control of each function. Automation of external cart starts, track enables, PEC/Direct switching, etc., can also be provided, all under subframe accurate automation control.

Global and Local Control of Faders and Mutes

A variety of fader and mute modes can be selected either as a total change or independently on a fader-by-fader basis. For the faders there are Replace, Relative, Auto Match, Safe, Isolate/Preview, and Trim. For the mutes there are Replace, Add, Safe, and Isolate/Preview. Each mode is designed to work smoothly and intuitively, with tallies mimicking the accepted multitrack tape machine status modes.

Master or Slave Operation

Flying Faders can run as a SMPTE Slave and will automatically start and stop mixing according to incoming timecode, even if it is vari-speeded code. For full transport control, Flying Faders can be interfaced to popular machine control systems and is supplied with an Adam-Smith 2600 synchronizer with full machine control.

High Resolution Color Displays

Together with the major facility capability of Flying Faders is a computer standard VGA color display. Lists and mix information are displayed in a graphic form with pull down menus and pop up displays controlled either from the built-in keyboard or from a supplied 'mouse'! Special HELP pages guide users through complex operations when necessary, eliminating the looking up of operator handbooks and the like.

Flying Faders Displays, Fig. 5

Summary

The new NEVE series of consoles represents a straight-forward and exciting approach to postproduction using the regular V Series console as a main frame.

The facilities offered are in demand by the postproduction and single-operator film industry and represent a good compromise within the capabilities of our V Series console.

Such important considerations as local pan control access to operator is achieved through ingenous use of signal routing already available within the NEVE V Series frame.

Acknowledgement

I would like to thank Gary Rotta, Photo-Magnetic Sound Studios, New York, NY, and Gregory Davis, Rupert Neve Incorporated, Bethel, CT for their assistance with this project.

GROUP DELAY CORRECTOR FOR IMPROVED TV STEREO PERFORMANCE

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Introduction

This paper describes a passive circuit that is intended to provide group delay equalization for a TV aural transmitter when operating through a notch diplexer. The circuit introduces group delay correction in the IF section of the aural exciter and after up converting to the operating channel, the group delay precorrection effectively corrects the adverse group delay existing in the output diplexer circuit. The end result is improved TV stereo separation. In addition, precorrection of the the FM bandpass can allow the use of lower cost, single cavity notch diplexers.

Statement of the Problem

Many TV engineers have noticed an increased level of distortion when making performance measurements on an aural TV transmitter at a point before the notch diplexer and after the notch diplexer. This is shown in Figure 1 where the measurement points are indicated as point A and point B.

Under monaural broadcasting standards, when the carrier deviation is limited to 25 kHz, the demodulated audio THD distortion was modest and usually tolerated since there was little the station engineer could do in the way of equipment adjustment to correct the situation. Frequency response was only slightly effected.

With the introduction of TV BTSC stereo broadcasting, however, the increase in THD distortion and stereo separation errors became borderline or not tolerable. The focus of attention was then on the error producing unit, the notch diplexer.

The notch diplexer is a passive device and under first consideration it may seem strange that it can introduce non-linear distortion and stereo separation errors. The basic problem is that the FM stereo signal is very sensitive to the notch diplexer group delay and amplitude response over the occupied bandwidth of the signal.

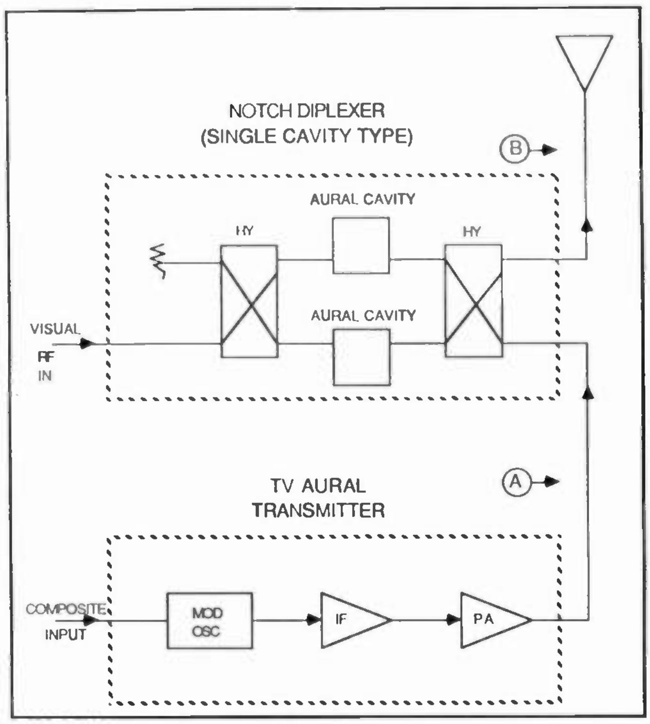

FIGURE 1.

TYPICAL TV AURAL TRANSMITTER OPERATING SYSTEM SHOWING MEASUREMENT POINTS A AND B.

The group delay and amplitude response of a single cavity diplexer is shown in Figure 2. (The phrase single cavity diplexer is intended to mean a single aural notch cavity is used on each branch of the diplexer.) The response curves shown in Figure 2 are typical of a notch diplexer optimized for minimum aural reject power. The bandpass is somewhat narrow and the group delay is steep. Fortunately, the response curves in Figure 2 show a high degree of symmetry which then makes it possible to entertain the idea of precorrection.

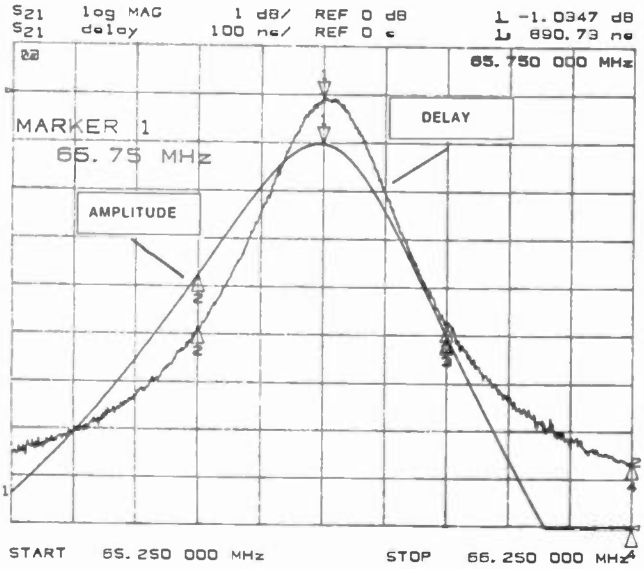

FIGURE 2. TYPICAL SINGLE CAVITY DIPLEXER AMPLITUDE AND GROUP DELAY

It is the intent of this paper to describe a precorrecting circuit to provide the necessary compensation and to further explain the sensitive nature of an FM signal to group delay errors.

Correction Concept

The group delay equalization concept as applied to an FM modulated signal is essentially a feed forward correction scheme. The phrase "feed forward" is used here to describe the technique of generating correction signals early in the RF line up of a transmitter for the purpose of correcting distortions occurring later on in the system.

A feed forward system is not automatic, that is, the system does not have a closed loop around the transmitter such as an AGC power control loop or a frequency control loop. Instead, feed forward correction signals are operated open loop without feedback and are manually adjusted for optimum operation and left to operate in this manner. The circuit described here falls into this category.

The amount of correction achieved with an open loop, feed forward circuit can be quite effective and many systems operate with feed forward correction schemes. For example, the visual transmitter uses a variety of feed forward compensation circuits in the form of group delay, differential gain and differential phase precorrection systems.

FM Group Delay Correction

The circuit proposed here is an adjustable group delay circuit that has been adapted for use on the aural transmitter. The equalization technique of adjusting the circuit to produce the inverse curve of that existing in the output system for correction is the same as that used on the visual transmitter.

There is however, a significant difference between group delay correction on a FM system versus an AM system. The visual transmitter has the option for correction at baseband (video) or at the modulated RF level depending upon the region of adjustment, that is, the single sideband region or the vestigial sideband region.

The aural transmitter, however, must incorporate group delay correction at RF or IF to compensate for group delay distortions occurring in the output diplexer system. Group delay correction at baseband on the stereo composite signal is not effective for equalizing group delay occurring at RF. The reason for this is that the occupied bandwidth of an FM signal increases or decreases as a function of baseband signal level. If the baseband signal level is low, the deviation is low and only a few significant sidebands are generated which generally fall within the acceptable group delay curvature area. Figure 3 shows the occupied bandwidth when the carrier is deviated 25kHz and how it compares to a typical notch diplexer group delay curve.

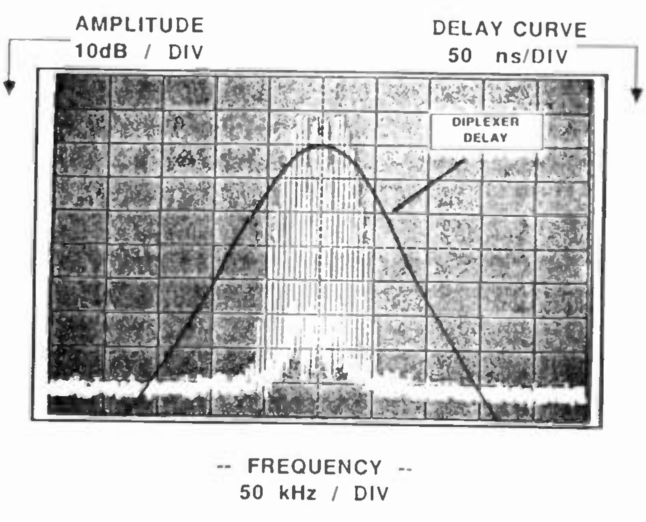

FIGURE 3. FM SPECTRUM, 5 KHZ TONE MONO, 25 KHZ DEVIATION

When the baseband signal level is increased, the FM deviation will increase and a number of additional significant sidebands will be generated which will begin to extend beyond the acceptable group delay curvature region. The result is that distortion starts to occur. Figure 4 shows the increase in occupied bandwidth when the deviation is doubled from 25kHz to 50kHz. The additional sidebands can now be seen to extend into an area of unfavorable group delay curvature.

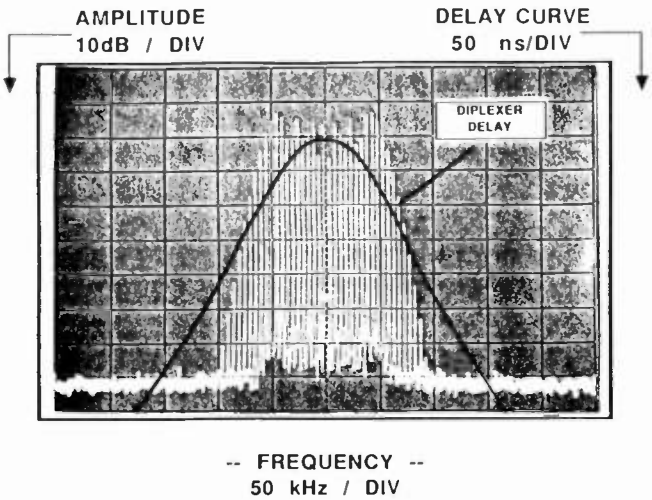

FIGURE 4. FM SPECTRUM, 5 KHZ TONE MONO, 50 KHZ DEVIATION

In addition, as the baseband high frequency content increases, ie., when switching from a mono to a stereo signal, the significant sidebands extend even further outward increasing the distortion. Figure 5 shows the spectral content of a TV stereo signal.

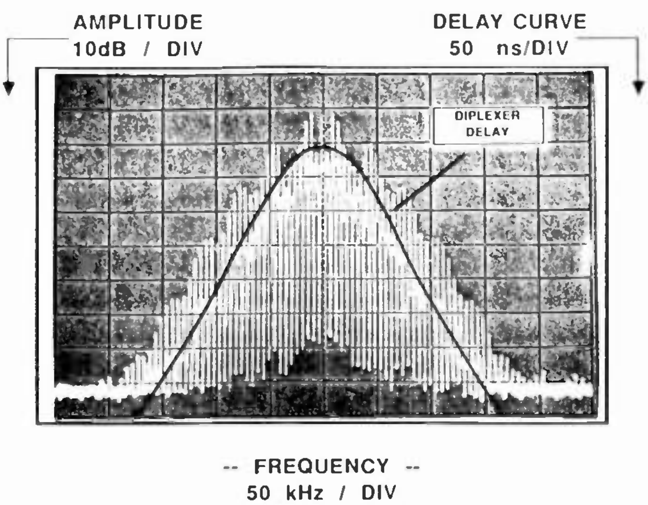

FIGURE 5. FM SPECTRUM, 5 KHZ TONE
TV STEREO WITH PILOT
NO SAP OR PRO
55 KHZ DEVIATION

Any attempt at group delay equalization of a stereo baseband signal would require a very complex circuit that senses both signal amplitude and frequency content, and makes adjustments accordingly on a dynamic basis. Simply put, there is not a direct relationship between the amount of group delay correction injected at baseband versus the amount of group delay correction achieved at the output of the RF system for a FM modulated signal.

Cost Effective Approach

Group delay correction in the notch diplexer is possible but expensive since it requires large RF components which can lead to increased insertion loss at a high power point in the system. To avoid the high power implementation problem, an adjustable group delay circuit was designed for the low power IF section of the aural exciter to provide the inverse group delay curve necessary to equalize the output. The corrector location in the aural transmitter system is shown in Figure 6.

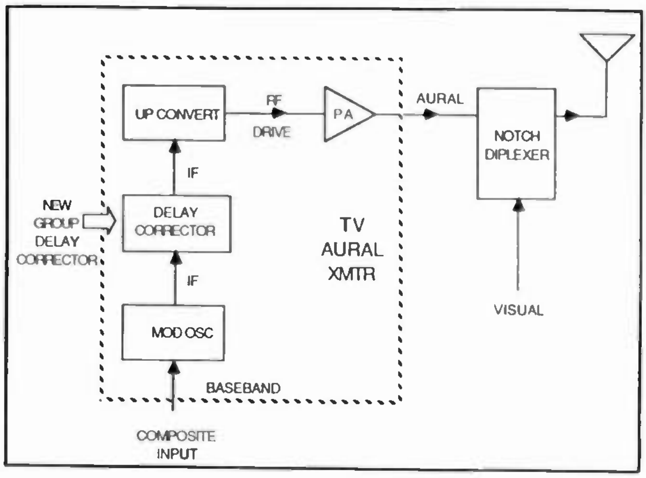

FIGURE 6.
LOCATION OF AURAL GROUP DELAY CORRECTOR IN OPERATING SYSTEM.

Equalization at IF frequency is effective because the up-conversion circuit uses a signal mixing process rather than a frequency multiplying process. In this manner, the precorrected sidebands are passed directly to the output for compensation.

Circuit Design

The circuit configuration is shown functionally in Figure 7. The circuit uses a quadrature hybrid with two high-Q resonators to achieve the necessary all pass condition. This is achieved by noting that a quadrature hybrid will reflect the input signal at port 1 to reject port 4 when the two output ports 2, 3 are either open circuited or short circuited.

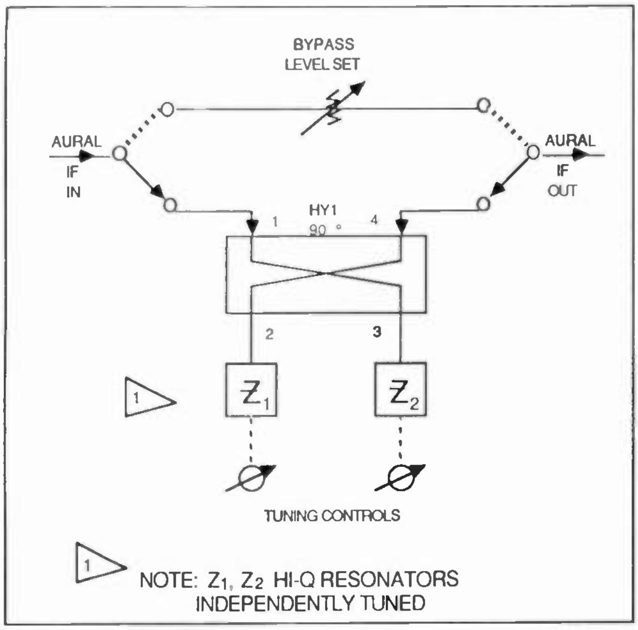

FIGURE 7. AURAL GROUP DELAY CORRECTOR FUNCTIONAL BLOCK DIAGRAM

The resonators, shown in Figure 7 as Z1 and Z2 provide this condition and also change the phase. The combined signal reflecting off Z1 and Z2 appear at reject port 4 as a vector sum with a substantial amount of group delay and minimal amplitude loss.

A bypass circuit with an adjustable attenuator is also provided to match the insertion loss of the group delay corrector when it is switched to bypass mode.

Results

The measured response of the group delay corrector is shown in Figure 8. This is a plot output of a network analyzer set to sweep over the aural IF and to display both amplitude and group delay curves. The lower curve shown in Figure 8 is the group delay curve which can be seen to effectively have the inverse curve shape to that of the notch diplexer shown previously in Figure 2.

It is interesting to note that the delay curve is negative which is different than the standard delay equalizer which introduces positive delay into the system. Equalization is then realized by operating on the decreasing edge of a positive curve.

In the hybrid equalizer proposed here, the two resonators are configured to provide the proper phase curvature to obtain a negative inflected delay curve. This does not mean that the circuit introduces negative time delay but rather the required phase curvature is changing very rapidly over a small frequency interval.

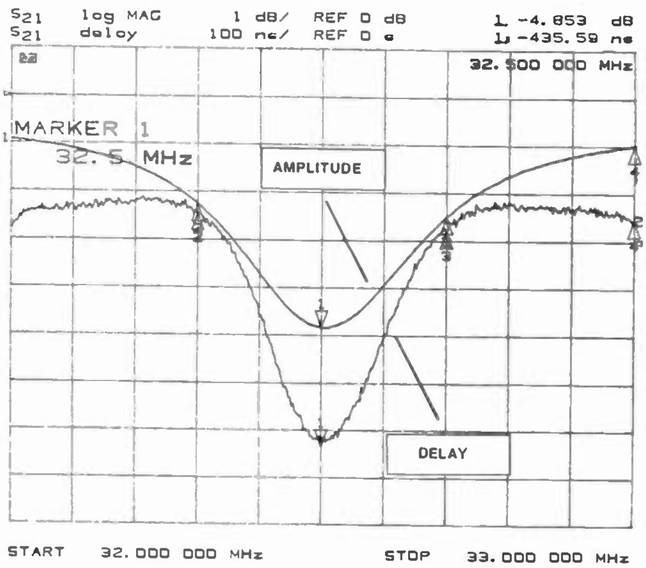

FIGURE 8. AURAL GROUP DELAY CORRECTOR MEASURED RESULTS

Figure 8 also shows the amplitude response. An ideal circuit without any circuit losses would have a flat response without a dip. The response dip shown in Figure 8, however, is very useful because it also provides a first order correction to the notch diplexer amplitude response.

It should be noted that the amplitude response is a "soft" function while the group delay is a strong function in terms of providing most of the required equalization of a FM system. Also, part of the amplitude response will be lost when operated through a saturated class C amplifier. However, new solid state transmitter designs now feature PA stages that are operated near to class B and part of the amplitude equalization can be passed on to the output for correction. This has a tendency to help correct AM synchronous noise.

The performance observations of the physical circuit are further supported by a computer model setup on Touchstone™*. Figure 9 shows the calculated results which match the measured results very closely. The computer model was used to find the circuit Q value to obtain the proper curve shape.

* It is a trademark of EEsof, Inc.

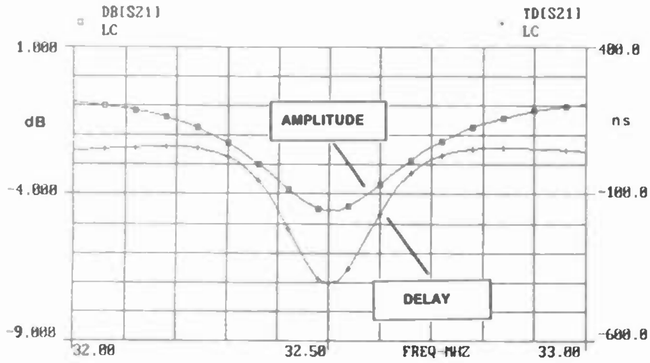

FIGURE 9. CALCULATED AMPLITUDE AND DELAY RESPONSE OF AURAL CORRECTOR CIRCUIT

Figure 10 shows the overall equalization result when the delay corrector is switched in and out of the circuit. The upper curve in Figure 10, shows the overall system delay response from exciter IF to diplexer output. It is essentially the diplexer delay response since it contributes nearly all of the delay errors.

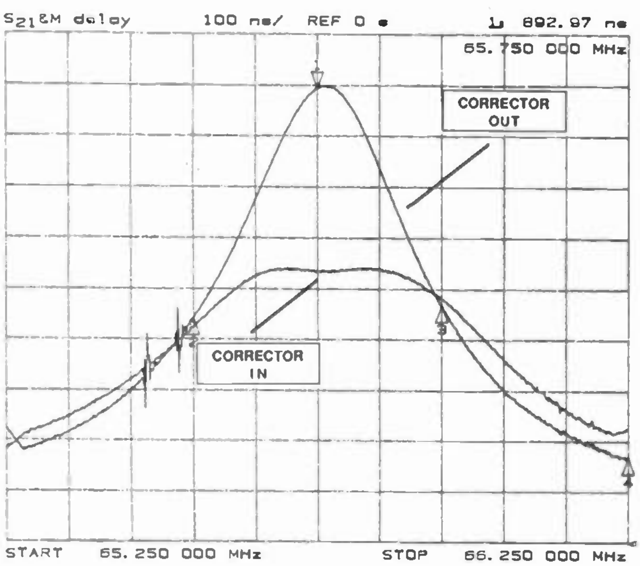

FIGURE 10. OVERALL TRANSMITTER DELAY WITH AURAL DELAY CORRECTOR SWITCHED IN AND OUT

The lower curve, labeled corrector in, shows the equalized result from the exciter IF input to diplexer output. This curve shows that a significant amount of equalization has been achieved over the occupied bandwidth of a stereo signal.

Data from the curves in Figure 10 was read off the network analyzer and is presented in Table 1.

TABLE 1
Corrector Results

Freq. in (kHz) Relative to Carrier		Group Delay (ns) Corrector In	Group Delay (ns) Corrector Out	Amplitude (dB) Corrector In	Amplitude (dB) Corrector Out
-300		-214	-595	-2.1	-4.54
-250		-166	-529	-1.7	-3.77
-200		-111	-452	-1.0	-2.72
-150		- 52	-348	- .5	-1.78
-100		- 10	-223	+.15	- .26
- 50		+ 5	- 84	- 0 -	- .26
- 0 -	STEREO BANDWIDTH 360kHz	- 0 -	- 0 -	- 0 -	- 0 -
+ 50		+ 5	- 32	- .13	- .36
+100		- 4	-153	- .43	-1.26
+150		- 16	-300	- .90	-2.5
+200		- 53	-426	-1.53	-3.83
+250		-109	-522	-2.33	-5.16
+300		-172	-591	- 3.2	-6.46

TV Stereo Improvement

The general improvement obtained with the delay corrector is about 30% to 50% reduction of wideband harmonic distortion and approximately .2dB response improvement over composite frequencies up to 50 kHz.

The most significant area of improvement is stereo separation which is shown in Figure 11 with the corrector switched in and out of the circuit. The curves generated in Figure 11 are from measured data through the system with the response characteristics shown previously in Figure 10.

TV STEREO SEPERATION
WITH AND WITHOUT GROUP DELAY CORRECTOR

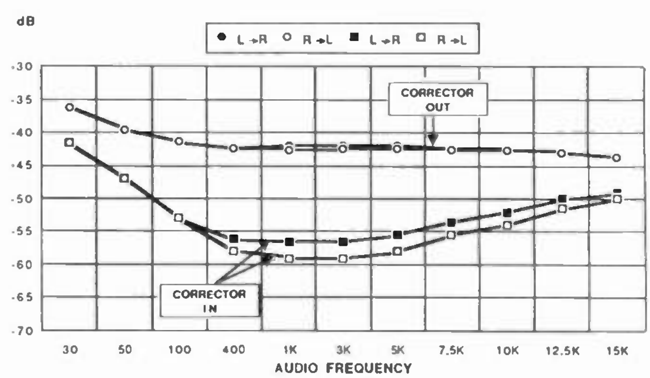

FIGURE 11. TV STEREO SEPARATION IMPROVEMENT

Figure 11 shows over 10dB stereo separation improvement can be obtained over the mid band audio frequencies.

The lower curve in Figure 11 shows L into R, and R into L separation which is very similar to the curve obtained when testing the stereo encoder and decoder back to back. The upper curve in Figure 11 shows the loss of separation due to the notch diplexer group delay. The measurements were made using BTSC equivalent mode stereo signals.

It should be noted that the degree of stereo improvement obtained with a group delay corrector is related to the level of performance in the basic system. For example, if the stereo generator separation is -55dB and the diplexer reduces it to -42dB then the aural delay corrector can correct the performance back to about -52 or -53dB for a 10 to 11dB improvement. If the stereo generator separation is -45dB and the diplexer reduces it to about -36dB, then the aural delay corrector can bring it back to about -42 to -43dB for a nominal 6 to 7dB improvement. In other words, if the upstream distortion is too great to where it begins to equal or exceed the distortion caused by the diplexer group delay response then, of course, group delay correction will have little effect.

Adjustability

The aural group corrector is designed for easy field adjustment. Two tuning controls are provided to allow slight stagger tuning of the resonator frequencies for the intent of changing the shape of the delay curve. The curve can be made narrower or wider or even saddled to resemble an over coupled circuit.

The adjustment can be done blind, that is, without using a network analyzer or other group delay measuring set for displaying the group delay curves. Instead, the aural transmitter is set up for routine stereo separation measurements and the group delay corrector is adjusted for the best separation value while observing an audio analyzer dB meter. It has been found in practice that it is best to use two audio frequencies, one low and one high, i.e., 400 Hz and 8 kHz, and adjust the delay corrector for the best overall compromise value.

Conclusion

The aural group delay connector proposed in this paper is a simple, passive device that is very effective for equalizing the group delay of a notch diplexer. Equalizing the notch diplexer delay provides a substantial improvement in TV stereo separation and provides the station engineer with the means to adjust his system to high quality standards.

Features of the aural group delay corrector are listed below.

1. Correction takes place in the RF sidebands where FM group delay correction is effective.
2. The corrector operates at low power in the IF stages of the exciter.
3. The corrector is adjustable to equalize a variety of notch diplexer curves.
4. The corrector allows use of a single cavity type notch diplexer.

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DIGITAL AUDIO DATA COMPRESSION—A PRACTICAL SOLUTION

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Abstract
Solid State Logic Ltd and Audio Processing Technology Ltd report the development of real-time music coding hardware which compresses high quality 16 bit PCM (Pulse Code Modulation) digital audio samples to 4 bits. The coding process is based on sub-band ADPCM (Adaptive Differential Pulse Code Modulation) techniques and is capable of compressing digital audio data to 128 kbit/s without audible degradation (Fs-32KHz). Apart from its lack of sonic degradation, the system boasts high error immunity, low hardware costs both for the encoder and decoder circuits, and a low coding delay. The coding hardware is suitable for a wide range of uses, including satellite and land-based broadcasting, ISDN (Integrated Services Digital Network) and numerous other low capacity high quality audio transmission and storage applications.

Introduction

While linear Pulse Code Modulation (PCM) has proved a popular means of representing high quality audio signals in domestic digital audio, the main disadvantage of its use lies in the enormous digital bandwidth associated with binary code. For example, using a linear 16 bit PCM system sampling at 32kHz, the basic binary bit rate will lie in the region of 512kbit/s per channel.

The bandwidth problem of PCM is particularly acute in transmission environments, where the transmission costs are proportional to the channel capacity. As a result, a number of alternative music coding schemes have been developed which require significantly lower operating bandwidth for these circuits. The most popular of these digital audio circuits operate at bit rates of around 256-400kbit/s, and include error protection overheads [refs. 1,2,3,4]. However, for many potential low capacity digital audio applications - for example in the ISDN, satellite and broadcasting fields - these levels of compression are still inadequate.

The excessive data rates of existing music compression schemes have been due primarily to their adherence to relatively simple digital companding or Adaptive Delta Modulation techniques (ADM). These systems exploit little of the natural redundancies associated with the sound signals of interest, unlike their speech coding counterparts. The situation has remained, partly because of the higher sampling overheads involved and, until recently, the absence of high speed Digital Signal Processing (DSP) hardware.

A medium-complexity candidate scheme which appears particularly suited to high quality audio coding and which does exploit the considerable natural redundancies of audio is sub-band ADPCM. A high coding efficiency is ensured in this system as it not only incorporates the benefits of digital companding, but also takes advantage of time and spectral redundancies by using linear prediction and sub-band coding.

The purpose of this paper is threefold;

1. to present the concepts involved in sub-band ADPCM and the nature of the signal redundancies which are exploited to provide a transparent 16 to 4 bit compression.

2. to discuss the final coding specification.

3. to describe our audio coding evaluation hardware, which incorporates stereo encoding and decoding sub-band ADPCM circuits and which has been designed primarily to allow potential users to assess the coder for their own application areas.

Characteristics of Audio

To extend the coding efficiency beyond that of linear PCM, the utilisation of known characteristics, or redundancy, within the digital music signals is essential. Music, and audio signals in general, exhibit a diversity of characteristics. The sub-band techniques described use those redundant characteristics as a means of compressing the final digital audio bit rate.

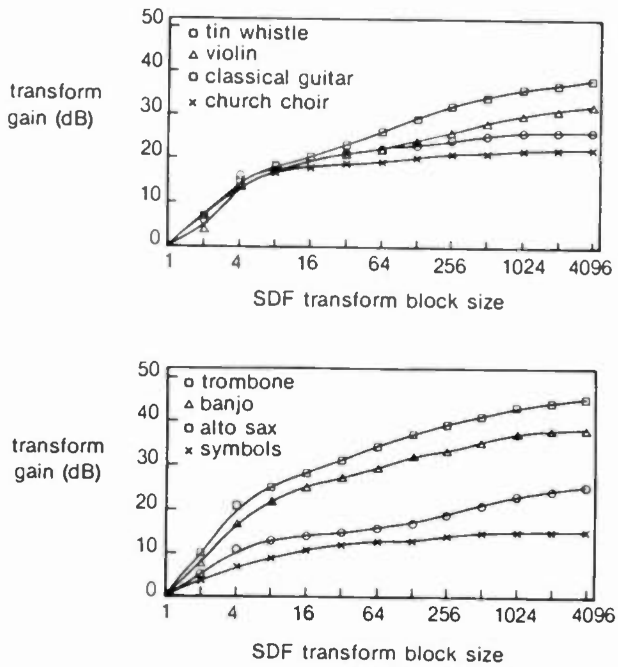


Fig 1. Symmetrical discrete Fourier transform gain versus block size

The effectiveness of the sub-band ADPCM approach can be estimated by considering the average Symmetrical Discrete Fourier Transform (SDFT) gain with transform block size for the sounds listed in fig.1. The diagrams clearly illustrate the considerable short-term coding gain. The SNR (Signal to Noise Ratio) is brought about by the exploitation of such redundancies. For example, the peak transform gain over PCM for a trombone signal approaches 45 dB. A 4 bit per sample coding allocation for a trombone signal would therefore provide a short-term coding SNR of approximately 24+45dB (ie. 69 dB) [refs. 5,6,7].

The foremost requirement of hi-fi music coding is the maintenance of a high coding transparency. This implies that the quality, bandwidth and distortion/noise levels of both original and coded music should not be subjectively different. A process relying on the inherant redundancy in music to maintain signal quality might not prove satisfactory for non-redundant signals. Fortunately, many signals of this class already incorporate specific perceptual redundancies to compensate for this (eg. noisy signals which will invariably mask coding error). This is also true for transient signals, which are exceptionally tolerant because of, amongst other things, temporal masking.

Sub-band ADPCM Audio Coding

A major limiting factor of low-bit rate PCM for audio coding, is the occurrance of audible noise or distortion in monophonic signals. With these schemes even the processes of digital companding and fixed spectral emphasis cannot adequately suppress this at word lengths below 11 or 12 bits [ref. 2]. While split-band coding schemes potentially face the same problem, the exploitation of redundant features of music signals allows sub-band ADPCM to operate succesfully at very much lower word lengths.

In the following sections we relate the operation of the coder to the characteristics of music, and compare this, where possible, with the deficiencies of digital companding. This will provide an indication of the reasons for the excellent performance of the system, which codes audio to 4 bits per sample, as opposed to 12 or more with PCM, with essentially no loss in quality.

Sub-band Coding

Split band techniques, such as sub-band coding, are used primarily to exploit spectral redundancies within the music spectrum. The mechanism of sub-band coding is to split the signal into a number of independent bands and to vary the accuracy of the quantisation in each band according to the input signal energy. The effect of this process is to allow the high energy regions of the music specturm to be coded more accurately than PCM, giving a lower coding noise floor. This improvement is commonly referred to as the sub-band gain.

Increasing the number of sub-bands improves the coder's ability to resolve the finer components of the signal spectrum, raising the overall sub-band gain. However, complexity, inter-band leakage, sub-band delay and an adequate energy classification procedure tend to offset the SNR advantage which may accompany an increasing number of bands.

An important subjective by-product of sub-band coding is the reduction in the perceived noise modulation over PCM. Since the music signal is split into several frequency bands prior to quantisation, modulated quantisation noise developed at each quantiser is constrained to that band and cannot interfere with signals in any other band. As a result, noise masking by the modulating spectral component is much more effective due to the reduced noise bandwidth.

Backward Adaptive Quantisation

Incorporated into each sub-band, backward adaptive quantisation provides a near optimal range match over a much wider dynamic range than is possible with instantaneous or block companding, and without the need to transmit gain information. The backward range matching is particularly effective for adaptive quantisers with large numbers of levels. By maintaining a constant SNR, adaptive quantisation exploits the masking properties of the human auditory system. As we have already mentioned, this process is very much enhanced with its incorporation into a sub-band coding structure.

Linear Prediction

An alternative approach to the exploitation of the spectral redundancy in the frequency domain is the application of linear prediction prior to quantisation. The validity of linear prediction for speech coding has been well documented. Our studies have indicated that substantial prediction gains exist for the vast majority of music sounds. A significant advantage of linear prediction is that its efficiency rises dramatically with an increase in the signal periodicity or spectral purity (ie. the ability to directly attenuate, prior to quantisation, those signals which normally promote audible noise modulation). The combination of linear prediction and sub-band coding thus avoids the need to resolve the spectrum in order to preferentially code the resonant components. By keeping within 4 sub-bands, the stationary characteristics of the sub-band signals are such that backward adaptive prediction has been found to provide an almost optimal performance. This is particularly true for sinusoidal type signals whose predictability is largely unaffected by sub-band decimation.

The overall effect of linear prediction, therefore, is to attenuate predictable signals, of which the most subjectively critical are highly resonant spectra. In the presence of noise-like signals, the prediction process is incapable of providing any coding gain. Fortunately, the significance of this effect is eclipsed by the inherent noise masking which accompanies these signals.

16 to 4 Bit Sub-band ADPCM Audio Coding

The 4 bit per sample coding system presented combines the processes of sub-band coding and ADPCM, to form what is known as Sub-band ADPCM. This is a very efficient solution to the problem of high quality audio compression. The diversity of redundancy removal processes involved not only reduces its sensitivity to uncharacteristic signals, but collectively takes advantage of the best properties of each.

The system which has been implemented using real-time hardware is illustrated in fig.2 and, for this illustration, compresses a single digital audio channel sampled at 32kHz to 128Kbit/s. This scheme uses a 4 band QMF (Quadrature Mirror Filter) tree structure where each band incorporates a backward adaptive quantiser and predictive block. An adaptive bit allocation minimises the short-term coding distortion across the audio band by continually re-distributing the sub-band quantiser levels according to the signal energy in each band.

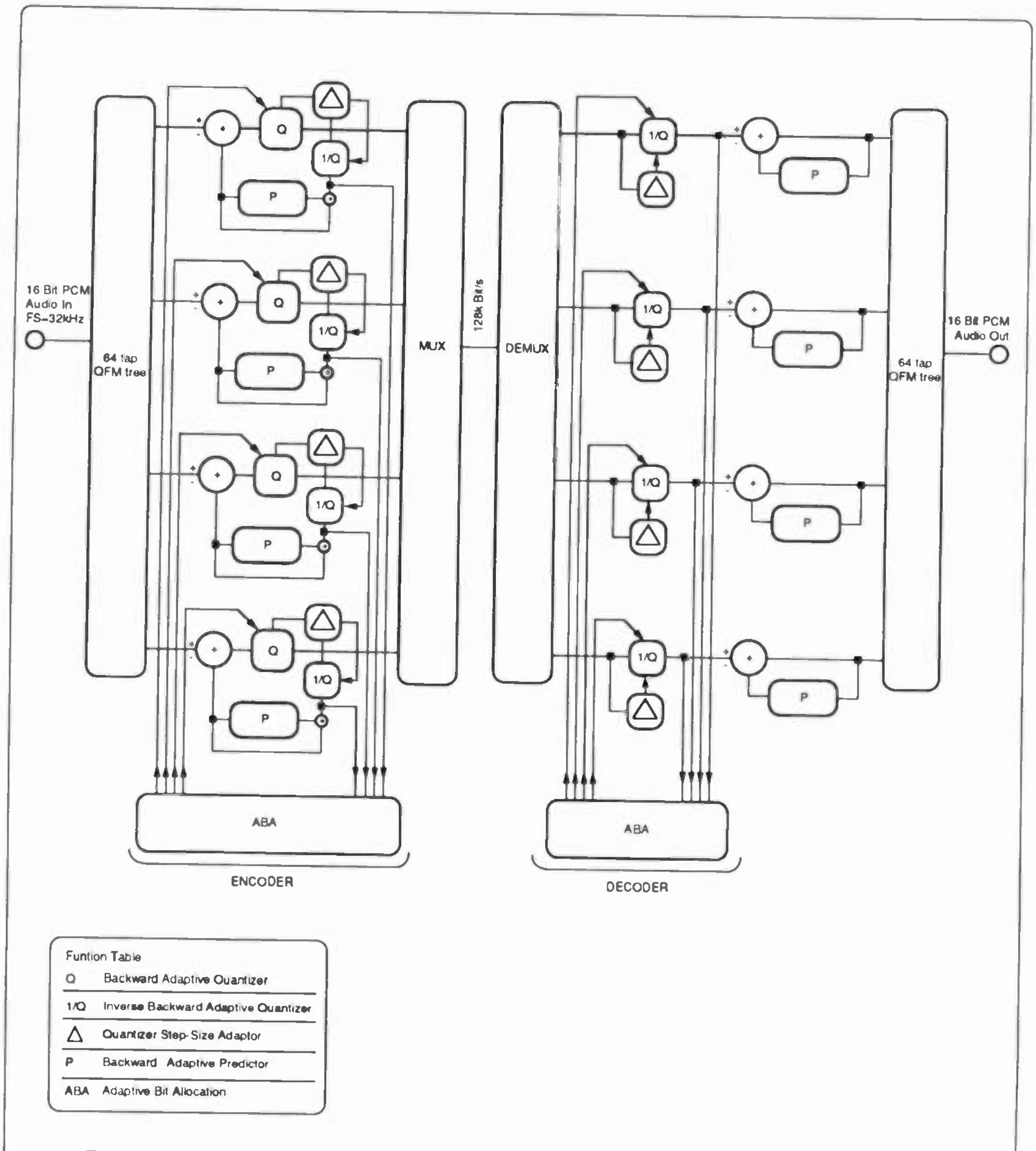
By allowing all parameter adaptations within the encoder and decoder blocks to follow in a backward mode, no specific side information is transmitted between them. The effect of this is to improve the immunity of the system to transmission errors and to reduce the coding delay across the encoder-decoder process.

Experimental Real-time Hardware

Early real-time implementations of each sub-band coder were undertaken using AT&T DSP16-55nS 16 bit fixed point signal processing chips. The object of this was both to optimise the coding specifications and sound quality and to establish the level of hardware complexity associated with these schemes. The main features of the DSP16 processor for which it was chosen are given below:-

55nS instruction cycle time
FIR tap operation in one cycle
Two 36-bit accumulators
512 words of on-chip RAM

Fig 2.



4 Band Sub-Band ADPCM Music Coder 128kbit/s

2048 words of on-chip ROM
Both serial and parallel I/O ports
Low-power CMOS technology

The DSP16 also facilitates the computation of fixed tap FIR stages in just one instruction cycle. This feature is very important for sub-band coders whose sub-band sections may consume up to 40% of the instruction cycles in every sampling period. The extended precision accumulators were also particularly useful for the sub-band filtering computations, enabling an increased coefficient resolution without risking overflow. The result of this increase in precision was that the residual noise levels of the coder were subjectively the same as the original 16 bit PCM digital audio.

Signal Processing Hardware

Real-time stereo versions of the coder were optimised using a purpose-built development board. The function of this board was to process 16 bit digital audio derived from either a compact disc player, a digital audio tape deck or a custom built ADC/DAC front end. The board incorporates two DSP16 processors which communicate with a common 16 bit parallel I/O bus via their own parallel ports. The basic modes of operation for this DSP board are illustrated in fig 3. In the "simulation" mode each processor encodes and decodes the samples from either stereo channel. In the "transmission" mode, however, samples from only one channel are encoded on one DSP and the compressed bits passsed onto the second DSP for decoding, duplicating a half duplex serial transmission environment. The board may be interfaced to existing sources of digital audio and has been tested using both a compact disc player and digital audio tape deck. Fig. 4, illustrates a typical connection to a CD player. In this configuration, 16 bit digital audio from the disc is re-directed in real-time via a CD interface adaptor, to the dual processors where each sample is coded to 4 bits, reconstructed and finally output to the players DACs. Using this system, subjective evaluations of the coding system are made directly by listening to the compact discs analogue output in the normal way. Subjective comparisons are also facilitated on the development board using a keyboard input for switching between the original digital audio and that ouput from the DSP16s.

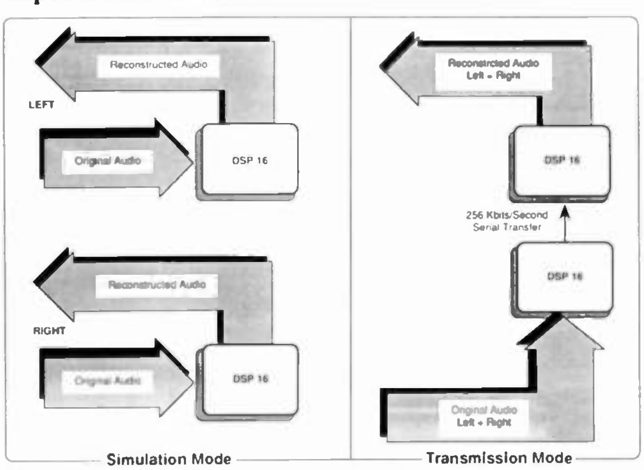


Fig 3. Signal processing hardware

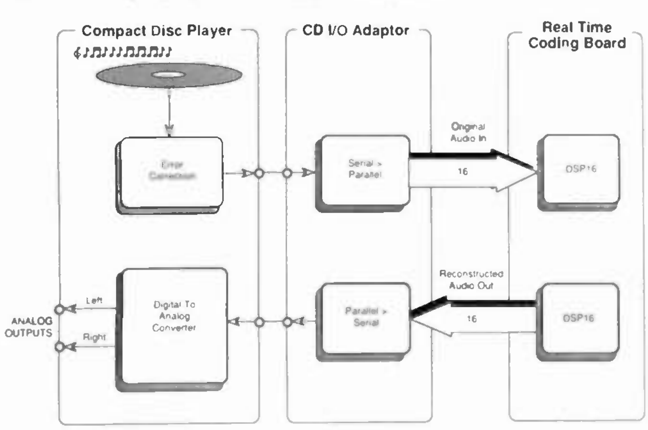


Fig 4. Test configuration with CD player

Subjective Evaluations

Informal subjective evaluation of the 4-band ADPCM coder implementation were undertaken at sampling frequencies of 32, 44.1 and 48 kHz and using 16 bit PCM digital audio. Source material was derived from both high-quality analogue record decks, compact disc players and digital audio tape decks.

The results of a typical ABAB subjective test using the real-time hardware at 32 kHz sampling frequencies are given in fig.5. The score table for this test is given in fig. 6.

+3	A better than B
+2	A slightly better than B
+1	A virtually identical to B
0	A same as B
−1	B virtually identical to A
−2	B slightly better than A
−3	B better than A

Fig 5. ABAB 7 point score table

		4 Band Coder		
		SNR (dB)	speaker average score	phones average score
Popular				
1. Peter Gabriel 'So'	(Bass, Guitar)	23.1	0.0	0.0
2. Thomas Dolby 'Flat Earth'	(Elect' percussion)	25.2	−0.1	0.0
3. Phil Collins 'Face Value'	(Alto Sax, piano)	32.7	0.0	0.0
4. Joan Armatrading 'Track Record'	(Percussion, symbols)	27.0	0.0	0.0
Miscellaneous				
5. CD Test Demo, Diapason	(Indian Harp)	30.9	+0.1	+0.1
6. CD Test Demo, Diapason	(Percussion-Drum)	43.8	0.0	0.0
7. CD Test Demo, Diapason	(Provence music)	31.0	0.0	0.0
8. Live PCM-F1 recording	(Tin Whistle)	49.2	+0.3	+0.1
Classical				
9. Denon Audio Technical CD	(Orchestra)	34.5	0.0	0.0
10. Denon Audio Technical CD	(Piano)	45.1	0.0	0.0
11. Denon Audio Technical CD	(Concerto)	29.8	0.0	0.0
12. CD Test Demo, Diapason	(Organ & Orchestra)	35.1	+0.1	0.0
Folk and Jazz				
13. CD Test Demo, Diapason	(Big Band)	28.2	−0.1	0.0
14. CD Test Demo, Diapason	(Banjo)	32.2	+0.1	0.0
15. Denon Audio Technical CD	(Jazz, piano, sax)	31.6	0.0	0.0
16. Live PCM-F1 recording	(Trombone)	53.9	0.0	0.0

Fig 6. Subjective evaluation score table

An Evaluation System for 16 to 4 bit Stereo Audio Compression Using Sub-band ADPCM

This evaluation system was conceived to allow potential users to assess for themselves the feasbility of the system within their own applications. The hardware incorporated enables one to quantify;

a) the complexity, size and cost of the final system
b) the system performance under any combination of operating conditions
c) the level of ancillary equipment required for a practical system

The evaluation unit consists of a stereo encoding board, a stereo decoding board and other auxilliary timing interface boards (fig.7). These are housed in a single 1U rack mounting cabinet complete with power supplies and the relevant audio and digital I/O connections terminated at the back panel. The encoding and decoding circuits have been designed to operate synchronously with a user interface and may therefore be treated on the whole as passive 4 bit ADC and DAC units.

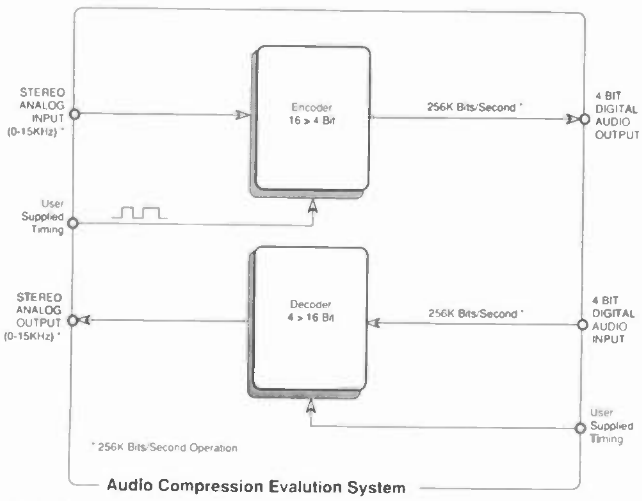


Fig 7.

Encoder Board

The encoder circuit is designed to receive analogue stereo audio signals, convert these to 16 bit PCM, compress the 16 bit words to 4 bits and then to output the 4 bit digital stream to the 'digital out' on the back panel (fig. 8).

The anti-aliasing filters are inter-changeable hybrid packages and are available for either 7kHz, 15kHz, 20kHz or 22kHz audio bandwidth applications. The sampling frequency of the ADC/encoder circuits may be operated up to 48kHz for stereo, and this is determined solely by the user word clock supplied to the unit (the word clock being one eighth the sampling frequency in each stereo channel).

After filtering, the analogue signals are sampled, converted to a linear 16 bit PCM format and fed into a single stereo encoder signal processing chip (fig.8).

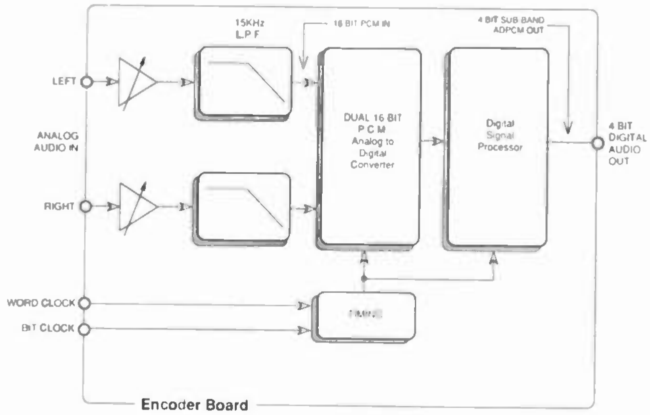


Fig 8.

The Digital Signal Processor (DSP) forms a 16 bit sub-band ADPCM word for every four PCM words input to the encoder. These sub-band ADPCM words are therefore output in a serial format at a quarter of the PCM input rate. For example, at a sampling frequency of 32kHz the total stereo bit rate output from the encoding DSP is 256kbit/s. The compressed data is clocked out by a user supplied 'bit clock' (256kHz) and the left and right stereo words are time multiplexed and synchronised with a user supplied 'word clock' (16kHz for stereo 32kHz PCM operation). The relationship between these clocks is illustrated in fig.9).

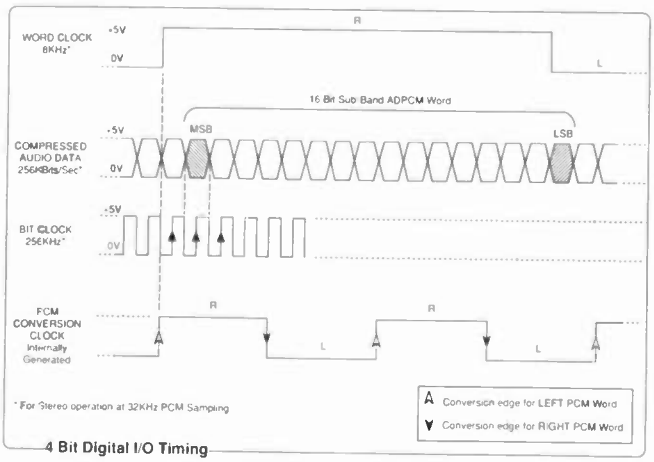


Fig 9.

Decoder Board

The decoder board receives the 4 bit codes via the 'digital in' connection and converts these firstly to 16 bit PCM audio using the decoder DSP and then to analogue through the dual DAC (fig.10).

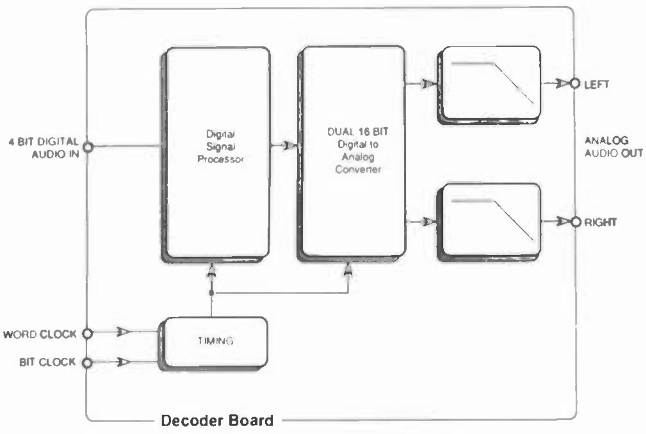


Fig 10.

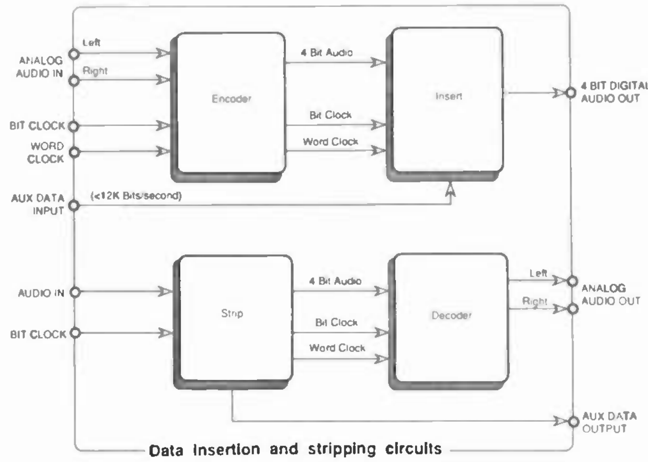


Fig 11.

The compressed words are input to the decoding DSP using the same serial multiplexed format as the encoder digital ouput. Again the data is clocked in using an external bit clock and synchronised with a word clock (fig.9). On receiving each 16 bit serial sub-band ADPCM word the decoder outputs four 16 bit PCM words to the DAC circuit. The converted samples are then reconstructed and fed to the analogue outputs. As with the encoder, the decoder conversion period is related to the rising and falling edges of the user supplied word clock.

Optional Features

Not included in the basic version of the unit, but which may be easily added, is the facility to inset auxiliary data into the compressed audio bit stream. This may be achieved by over writing the LSB (Least Significant Bit) of each 16 bit sub-band ADPCM word. The facility is necessary either when separate word timing is not available in the transmission/storage system, or where control information (for example satellites) might be required further along the system. Under exceptional transmission error conditions parity information could also conceivably be inserted at this stage.

The data insertion and stripping circuits are illustrated in fig.11. In this example both auxiliary data and the stereo left/right word timing are derived from the data stream at the decoder side. By dropping the LSB of each 16 bit sub-band word and allowing for framing overheads, a maximum of about 12kbit/s may be transmitted along with the stereo audio in this way. A slight loss of audio fidelity would inevitably accompany the inclusion of this facility.

Summary

This paper describes the techniques and practical implementation of a sub-band ADPCM coder for the compression of high quality audio signals. Using this system, digital audio compressed to 4 bits per sample is subjectively indistinguishable from the original 16 bit PCM audio for sampling frequencies down to 32kHz.

The performance obtained through the system at this level of compression clearly demonstrates the very high coding efficiency of sub-band ADPCM in comparison to existing digital companding and ADM schemes.

Subjective evaluation of the coder has been undertaken using both computer simulation and real-time hardware coding implementations. The latter being effected on AT&T DSP16 Digital Signal Processors.

A real-time evaluation unit has also been developed which incorporates separate encoding and decoding modules. The modular design of the unit allows prospective users to establish the feasibility of the sub-band ADPCM system for a wide range of application areas.

Acknowledgement

The author thanks Mr Roger Woods for his help in preparing this paper.

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A UNIVERSAL CONTROL NETWORK

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Videomedia
Sunnyvale, California

A PROBLEM DEFINED

Controlling various types of transports, even those manufactured by the same manufacturer, has, until now, represented major problems and expense to end users as well as the manufacturers and software developers. It would be ideal if there were a common language by which all transports could be controlled. Most of us have need to control transports from various different sources or control points. We would like to be able to control any type of transport on our editing system and, perhaps on occasion, use the same transport or group of transports with our graphics system or simply have a single remote control panel for all devices in house without having to buy another expensive interface.

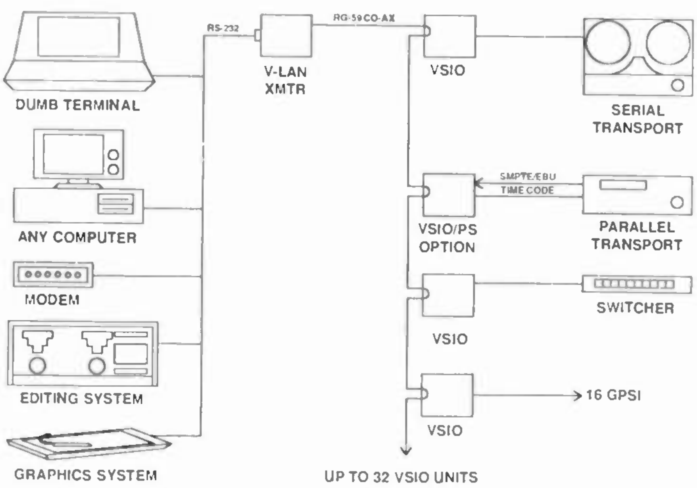
Many end users have several different brands of editors in their system and discover they must purchase additional expensive interfaces or drivers to get double duty from their VTRs. The situation is frustrating, expensive and unnecessary. After all, shouldn't there be a common interface for all types of transports available from all manufacturers of editing systems, graphic workstations, automated playback systems and remote controls?

The solution to this dilemma is not likely to come from the VTR manufacturers. After all, a Sony VO-5850 U-MATIC recorder doesn't require the sophisticated remote control functions of a Sony BVH- 3000 or an Ampex VPR-3. Even the basic operating philosophies of these transports are diametrically opposed. Ampex uses a "time-line" system; Sony, Panasonic, JVC and Hitachi do not.

V-LAN

Universal Control Network

UNIFORMITY, THE ULTIMATE CONTROL CONCEPT.



- Up to 32 devices may be controlled independently, as a group, or a sub-group.
- Instant interface to all popular transports.
- Extremely cost effective.
- Highly reliable.
- Simple RS-232 control from any computer, terminal, modem, editing and/or graphics control system.
- Remote any mixture of transports up to 4000'.
- Single co-axial cable connections (control may be routed via conventional patch panel).
- Frame accurate with SMPTE/EBU time code.

In addition to differences among serial controlled VTRs, the problem of a universal type of interface is compounded by the wide range of parallel controlled VTRs. Each VTR has subtle differences in search ballistics, phase of SMPTE time code, how effective capstan bumping can be achieved and countless other variables.

AN ELEGANT SOLUTION

With these thoughts in mind, Videomedia set out to develop a system by which any manufacturer may interface to any and all current VTRs. In other words, All transports LOOK the same to the controlling device and only one simple driver and one single serial port is necessary to control any and all transports. The system is called V-LAN. The V-LAN is a local area network with an extremely simple protocol that offers complete frame accurate (with SMPTE or EBU time code) control of up to 32 transports (or other devices) via a single co-ax cable.

Whenever transports are to be controlled in an environment where frame accurate editing or synchronizing is necessary, "real-time control" becomes an important issue. A key feature of the V-LAN architecture is its ability to effectively handle all "real-time" activity within the system. Since this cannot possibly be accomplished with a single microprocessor, V-LAN makes extensive use of distributed intelligence. Each device placed on the V-LAN to be controlled has its own control processor.

Manufacturers of editing, graphic, animation and station automation equipment have their lives greatly simplified in that the software developer needs to write only one single driver to have access to all VTRs. Videomedia has placed the V-LAN protocol in the public domain. This means that any software developer, manufacturer or even talented end users is free to use the V-LAN protocol at no charge. The software development expenses previously required to control multiple transports has been eliminated.

V-LAN can even be controlled from a "DUMB TERMINAL" with no program at all. Obviously this would be extremely slow and cumbersome, but it does illustrate the simplicity of the command structure implemented. The V-LAN language is not computer type dependent, nor is it dependent upon specific programming languages. It may be used with virtually any device that has a single serial port on it. The V-LAN may even be remote controlled via a modem and incorporated into virtually any software program.

The V-LAN protocol is complete in that virtually all parameters of a given transport are controllable. The user does not have to be concerned with transport ballistics, SMPTE/EBU time code reading, capstan override, drop-frame and/or non-dropframe algorithms or even mixing totally different types of transports on the system. The characteristics of all types of transports are compensated for by the V-LAN control module associated with the specific transport.

GENERAL DESCRIPTION

THE V-LAN LANGUAGE

One of the V-LAN design criteria was to develop a protocol whose syntax would easily convey the command meaning. The V-LAN TRANSMITTER module converts simple two byte ASCII codes and transmits these codes to the appropriate V-LAN RECEIVER module. For example, issuing "PY" places a selected VTR into PLAY, "ST" = STOP, "FF" = FAST FORWARD, etc.

The intelligence of the command language protocol eases control program debugging tremendously. For many custom applications requiring transport control, debugging the system can become the largest expense of the project. A readable control language eases the burden by eliminating the need to constantly translate commands and refer to manuals to discover what is actually being sent. Since the V-LAN language is a simple ASCII character based language, a simple "dumb terminal" can monitor the serial link and serve as an active display of previously issued commands and responses.

V-LAN HARDWARE

A V-LAN RECEIVER module must be used for each transport that is to be controlled. The V-LAN RECEIVER connects to the remote connector of the transport. Each V-LAN RECEIVER placed on the LAN must be specified as to type of VTR and NODE address (1-32). All V-LAN modules (TRANSMITTER and RECEIVERS) are identical from a hardware point of view. All modules interconnect via a single BNC RG-59 co-ax cable. The total length of this cable may be up to 4000'. Each V-LAN module has a looping input for the V-LAN cable. No modifications of the VTR are necessary. A list of supported VTRs is at the end of this article.

The VSIO-PS SMPTE time code reader is an available option for a V-LAN module. This unit is a piggy-back board that is housed inside of a receiver module. It accepts SMPTE time code output of the associated VTR and converts it for use on the V-LAN. If a VTR has an internal SMPTE reader and the data from such is available through the serial port of the transport, the V-LAN will utilize that SMPTE code and not require a VSIO-PS option. The users program may request a time code reading from any VTR on the V-LAN at any time.

The Videomedia V-LAN transmitter module accepts control data from virtually any device equipped with an RS-232 or RS-422 serial port. The transmitter module interprets simple two byte ASCII commands and controls all V-LAN receiver modules via the coax LAN. This method allows for easy expandability in as much as additional transport control may be added to the LAN at any time by simply adding another V-LAN receiver module and a single co- axial cable. From a hardware point of view, a transmitter module and a receiver are essentially identical.

A V-LAN module contains two indicator LED's labeled numbers one and two. These lamps provide status helpful in determining if everything is connected and operating properly. The number one LED indicates whether or not a module is receiving its proper synchronization signal. For the transmitter module, the LED will come on when proper video sync is applied to its sync input jack (RCA connector on upper right hand side). On a receiver module, the number one LED will light when the LAN is properly connected.

Under proper operating conditions, the number two lamp on all modules will blink on and off about once per second. If this lamp fails to blink, the module may be defective or power isn't getting into it. V-LAN receiver modules for parallel type transports receive power from the transports remote connector. Modules for serial type transports are shipped with an external power supply.

The video sync signal is only needed by the master V-LAN TRANSMITTER module if frame accurate control is required. When using the sync input, the signal should be composite video, black burst or composite sync. Whatever signal is chosen, it must be stable. It should be taken from a video generator, or time base corrector. The V-LAN module is small in size (5" x 5 1/4" x 1 1/2") and extremely reliable (no wiring harnesses and fully socketed).

TECHNICAL DATA REGARDING LAN ACTIVITY

GENERAL LAN OVERVIEW

The V-LAN LAN links small, economical self-contained Z-80 microcomputers together for demanding real-time process control applications. These include, but are not limited to, video tape editing, animation, and station automation. The communications protocol has deliberately been kept simple to maximize throughput while minimizing processing overhead. Short messages are sent and acknowledged in less than one millisecond. Over 400 bytes of information can be transferred within 30 milliseconds. There are no restrictions whatsoever on message content. The V-LAN also distributes timing information to maintain precise synchronization throughout the system.

Physically, a V-LAN network can consist of up to 32 V-LAN modules connected together with lengths of RG-59 coax. Standard BNC connectors are used throughout. The coax simply loops through the V-LAN modules exactly as though they were video monitors. A 75 ohm terminating plug is placed at the physical end of the coax, which may be as long as 4000' length total. This physical connection scheme is both simple and economical. It lends the V- LAN all the virtues normally associated with closed circuit television: exceptional noise immunity with almost no line loss, distortion or spurious RF emission.

At the most basic level, data on the V-LAN is represented as an amplitude modulated asynchronous serial bit stream with a low representing a mark and a high representing a space. Data transfer occurs at 153.6 Kbits/sec with each character consisting of 8 data bits plus an odd parity. This all works out to yield an impressive 72 microseconds per byte peak data rate -- the typical maximum sustained rate being only 10 to 20% less than this.

In the parlance of local area networks, the V-LAN is a half- duplex, bus type utilizing a hierarchical organization to insure that collisions are avoided. One V-LAN module is designated the LAN master, others are referred to as slaves. All synchronization and message transactions originate from the V-LAN master. Slave nodes may transmit messages only in response to queries directed uniquely at them from the master. This restriction insures that two V-LAN modules never attempt to transmit on the LAN simultaneously.

With each slave V-LAN module is associated a unique number: its node address. Each physical module may, in fact, be assigned more than one logical node address, provided that all node addresses are unique and no duplication of node numbers exist on the LAN.

Every data message transmitted by the V-LAN master begins with a byte containing the destination node address in the lower five bits. The remaining three bits indicate the type of message to follow (i.e. byte or string) and whether or not the receiving slave is expected to send back a response.

Whenever an error occurs or the V-LAN master needs to access a different slave, it inserts before the message data, a break condition (i.e. a constant high level) for approximately 100 microseconds. This break signal "wakes up" sleeping slave V-LAN modules so that they will compare the following node address with their own.

As soon as a slave node recognizes its node address, it echoes the node address byte back to the master and prepares to receive the rest of the message. Other slave modules simply ignore the message and go back to "sleep" at this point. Break must be sent again by the master to rouse them.

After the V-LAN master receives the go-ahead signal from the selected slave, it transmits the body of the message. This is either a single data byte or a string length byte followed by the string body. In general, there are no restrictions on message data; however, null strings (those of length zero) are not allowed. Instead, a zero string length byte implies that 256 string data bytes follow. If the application requires null strings, they can be transmitted by reserving a <u>byte</u> message (say, zero) to represent them.

To insure data integrity, a checksum byte is transmitted following the message data. The checksum is simply the modulo 256 sum of 0E5 (hexadecimal) and all the bytes in the message, including the node address byte. A more powerful cyclic redundancy check would at first seem preferable to this basic scheme. However, given the V-LAN's high data rate, it would take too long for most microprocessors to calculate the CRC in real-time, while sending or receiving the message.

If the receiving node finds that the checksum byte it received matches the one it calculated, it echoes the node address byte again to indicate the message body was received intact. In the unlikely event the receiving node detects a discrepancy, it echoes instead a negative acknowledge (NAK) code to the transmitting V-LAN module, which, in turn, re-sends the message.

Distribution of Sync Information

A V-LAN master broadcasts synchronization information at regular intervals in the form of <u>System Timing Tick</u> bytes. System Timing Ticks are always preceded by a break (spacing line condition) of at least 100 microseconds duration. In video applications such as tape editing, ticks are sent within the first 500 microseconds of the video frame. The most significant bit of timing tick bytes must be set to distinguish them from node address bytes. This still leaves 128 different timing ticks -- many more than are necessary to represent each NTSC and/or PAL color frame with a different value.

Slave modules automatically stop sending data when more than approximately 95% of the time to the next system timing tick has elapsed. They do not start again until the tick is actually received. This insures that no slave module is transmitting at the moment the V-LAN master needs to send a timing tick -- assuming, of course, that the master is sending them at a known, constant rate. The V-LAN master itself may transmit other information right up to the very instant of the tick.

An interesting consequence of all this is that system timing ticks may interrupt byte and string messages. When this happens, the interrupted message continues after the tick right where it left off before it.

USERS PROGRAMMING PROTOCOL

COMMAND STRUCTURE

All of the commands recognized by the V-LAN transmitter are two byte ASCII codes, with optional data following. Spaces are ignored. A carriage return marks the end of the command. All location numbers entered are to be in standard SMPTE time code format (HH:MM:SS:FF). The maximum number allowed in this format in the NTSC system is 23:59:59:29. Colons are not required when numbers are entered, and less than eight digits may be entered, with the higher order numbers assumed to be zero. Absolute frame counts from 1-99 are also allowed, as well are negative numbers. Numbers sent must be in ASCII form, which entails that numbers be offset by 30 hex.

After a command is received and processed, data will be returned to the user. This data can be either requested information, or an error message. A carriage return (CR) will mark the end of the response. If no data is returned, just a CR will be sent.

Possible error messages returned are: ERROR - indicates bad command format, COM ERROR - the master could not communicate with the selected node, DUR ERROR - the duration for the edit sequence is invalid.

NODE SELECTION

The node setting command lets the system know which of the current nodes is the default. On power up, node one defaults. All commands sent which don't specify another node themselves, act upon the default node. When doing an edit, the default node is always the recorder.

When in an edit sequence, the present node # is always the record deck. Once an edit is started, no motion commands should be sent to any deck in the system. Doing so will cause all the VTRs to abort their edit sequence. If a deck aborts an edit because it senses a problem, it sends an error and all the other decks also abort the edit sequence. When status is requested from the deck that aborted, the returned status will be ERROR#, where # is a two digit error code. The status is removed once a motion command is send to the node with the error.

THE V-LAN COMMANDS (ALPHABETICAL ORDER)

AUTO INC. **AI #, #=BCD auto increment value.** The number entered here is the number of frames that the inpoint of the record deck will be changed after performing an edit.

CLEAR NODE **CL** The in, out, and duration values for the present node are set to zero.

DEVICE TYPE **DT** A device type code is returned indicating the type of interface connected on the present node. The first three characters tell which type of interface it is. The last two digits are the version number.

ECHO OFF **EF** Turns the echo off. This is how the system will power up. Echo should be left off when using a computer to control the system.

ECHO ON **EH** Echo is turned on, meaning all characters sent to the controller and line feeds are sent back also. Useful when using a terminal.

EDIT ERROR **EE** Edit error is returned by the current node selected using the ND# command. The returned range is + - 0 to 99 frames. A non zero number before the edit point is an indication of bad control track, bad time code, bad sync reference, or an error in capstan bumping. Three data bytes are returned: (-01), (01).

EDIT STATUS **ES** Edit status is returned. This will indicate the stage of the edit. The possible codes are CUEING, EDITING, ABORTED, and DONE.

EDIT SYNC. **SY #, #= 0,1,or 2** The edit sync. command selects the synchronization mode used for tape speed override. Record lock mode (#=0) locks the sources to the recorder. This is the default mode of operation. Source lock (#=2) uses the source as the edit sync reference. Useful on some machines when time base correctors are not used, or when unstable video is likely to feed the recorder during preroll. System mode (#=1) is used only for Ampex 1 inch machines.

END CONDITION **EC data, data=command to be executed.** The command tells what to do at the end of an edit. If nothing is set, all decks just pause. By giving the command 'EC PF', for example, the deck will perform the edit again and again, until stopped. If the auto increment register contains a value, then a continuous animator is created. A simple command such as stop will cause the current node deck to be stopped. If a continuous loop is created, it is easily stopped by sending another EC command, but with no data.

ENTER IN **EI** The current location is entered as the inpoint.

ENTER OUT **EO** The current location is entered as the outpoint.

Fast Forward **FF**

Fast Rewind **RW**

GOTO IN **GI** The deck searches to the inpoint that was previously stored.

GOTO LOCATION **GT #, #=BCD location**

GOTO OUT **GO** Search to the outpoint.

GOTO PREROLL **GP** The deck is sent to the in point minus the preroll selected.

LOC CONT **LC** Continuous location.

LOC REQ. **LR #, #=optional node number.** BCD location is returned. 12 bytes are sent. If the number is positive, the first byte is a space. If negative, the first character is a '-'. If the timecode on the tape is drop frame code, the colon before the frames value will become a period.

LOC.+ STATUS **LS #, #=optional node number.** 12 bytes of location and 12 bytes of status are returned .

NODE ADDRESS **ND #, #= 1-32** The current node address is set. This node number specifies which deck subsequent commands will be sent to. When editing, this selects the record deck.

PAUSE **PS** The current deck is toggled between play and pause. If a guaranteed pause is required, use the shuttle command with a speed of zero.

PERFORM **PF #, #=other nodes to sync with the recorder.** Perform the edit on the presently selected node. Tracks should be set up ahead of time. Edit length is the duration held in the recorders duration register. This duration is rippled to all selected sources. Other nodes may be synchronized to the recorder by entering their node numbers after the PF. When the recorder gets to its in point, all other decks will be at theirs also.

PLAY **PY** The current selected deck is put into play.

POSTROLL SET **PT #, #=BCD postroll**

PREROLL SET **PR #, #=BCD preroll**

READ DURATION **RD** The duration is returned.

READ IN **RI** The in point is returned.

READ OUT **RO** The out point is returnes.

REHEARSE **RH #, #=other nodes to sync with the recorder.** Same function as the perform command, except that no actual editing is done. The record deck enters the EE mode on the selected tracks . This command, or the review command may be used to created multiple deck sync rolls.

REVIEW **RV #, #=other nodes to sync with the recorder.** Same as the rehearse command, except the record deck doesn't go into selective EE at the edit point.

SET DURATION **SD #, #=BCD location**

SET IN **SI #, #=BCD location**

SET OUT **SO #, #=BCD location**

SHUTTLE **SH # , # = +-0..9** Shuttle the deck at a speed dependent upon the number entered. Speed 0 is paused.

STATUS ENCODED **SE #, #= optional node number.** First, 12 bytes of BCD location are returned. Next, a byte is sent that indicates whether or not the deck is actually reading valid time code at the present time. An ASCII space character means it is not, while a '*' says that valid code is being read. The last byte is encoded VTR status.

HEX CODES FOR ENCODED STATUS BYTE:

80 - BLANK
81 - POWER OFF
82 - STOPPED
83 - THREADING
84 - UNTHREADING
85 - WAIT
86 - PLAY
87 - PAUSE
88 - STICK
89 - SHUTTLING
8A - CRUISE
8B - BRAKING
8C- SEARCHING
8D - FAST FORWARD
8E - REWIND
8F - READY
90 - EDITING
91 - LOCAL
92 - EJECTED
93 - RECORD
94 - VAR. PLAY
95 - CALIBRATE
97 - REHEARSE
98 - REVIEW
99 - TONE
FF - OFF LINE

01 - EXECUTION ERROR
03 - SEARCH ABORT
05 - ILLEGAL SEARCH RQ.
07 - SERVO ERROR

02 - CONTROL TRACK ERROR
04 - ILLEGAL COMMAND
06 - SYNCHRONIZATION ERROR
08 - NOT READY TO SEARCH

STATUS REQ. **SR #, #=optional node number** A twelve byte ASCII status string is returned indicating machine status. If the deck had an error, the word error is returned followed by a number indicating the type of error. Possible status messages are the following: Power off, Stop, Threading, Unthreading, Wait, Play, Pause, Shuttling, Braking, Searching, Fast Forward, Rewind, Ready, Editing, Local, Ejected, Record, Var Play, Calibrating, Rehearse, Review, Off Line.

STOP **ST** The current deck is put into stop. Tape is unthreaded, standby off.

TRACK SELECT **TS V12A** Select the track to be used on the next edit. V=video, 1=audio1, 2=audio2, A=assemble edit. The codes need not be in any specific order.

TRIM DURATION **TD #, #=BCD trim value**

TRIM INPOINT **TI #, #=BCD trim value**

TRIM OUTPOINT TO **#, #=BCD trim value**

LIST OF VTRS CURRENTLY INTERFACED

BRAND	**MACHINE**	**NOTES**
Sony	BVU Series	Can use optional VSIO-PS.
	BVW Series	No VSIO-PS required.
	BVH Series	No VSIO-PS required.
	5850, 5800	Must use VSIO-PS.
	7000 Series	Requires Sony BKU-701. Can use VSIO-PS.
	9000 Series	Requires Sony BKU-701. Can use VSIO-PS.
JVC	CR-850, 600	Can use optional VSIO-PS.
	BR-8600 VHS	Must use VSIO-PS.
	BR-6400 VHS	Must use VSIO-PS.
	8250, 6650	Must use VSIO-PS.
	KR-M800U	Must use VSIO-PS.
	KR-M860U	Can use optional VSIO-PS.
	BR-S810U	Must use VSIO-PS.
	BR-S610U	Must use VSIO-PS.
Panasonic	MII series	No VSIO-PS required.
	AG-6500	Must use VSIO-PS.
	AG-7500	Must use VSIO-PS.
	AG-6300	Must use VSIO-PS.
	AG-7300	Must use VSIO-PS.
	AG-7100A	Must use VSIO-PS.
	AU-300	Must use VSIO-PS.
RCA	HR-2	Must use VSIO-PS.
Ampex	VPR 3,6,80	No VSIO-PS required.
	ARC-40	Must use VSIO-PS.

MANAGING ROUTING SWITCHER GROWTH IN A MULTIFORMAT WORLD

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Abstract

Improvements in television technology are significantly changing the routing and distribution requirements of broadcast and teleproduction facilities. New switching systems are increasingly being installed to handle component formats, RS232/RS422, digital video, etc. Integrating these smaller matrices with the main house routing switcher, however, often requires a major rewiring of the entire facility. An inherent flaw in the control logic of standard routing switchers is held responsible. A virtual matrix control system can relieve this problem by mapping the relationship between all switching levels in software. The practical benefits of this alternative control method are explored.

Introduction

The routine day-to-day recording, monitoring and maintenance needs of every teleproduction facility presuppose the ability to select from any number of available signals. At least some of these switching and distribution needs can be met with a combination of patchbays and distribution amplifiers. However, complex and extensive switching networks can be established and changed more readily with a routing switcher.

A routing switcher is capable of distributing a source signal to any desired destinations within the matrix. This provides the best possible utilization of available hardware. Once a routing switcher becomes the central interconnect point for all equipment, uniform timing relationships and signal levels can be established throughout a facility. Equipment changes and facility expansions become greatly simplified.

There are operational, as well as systems engineering advantages. Control panels are usually installed in many areas throughout a facility, enhancing overall flexibility. Increasingly, these panels can be programmed to meet unique requirements. Software based features now include the ability to assign inputs or outputs to specific buttons, the use of alphanumeric names instead of numbers to improve user intelligibility, and "protection" options to prevent unwanted switches from occurring.

Commonly used setups can often be defined on system terminals, and executed as salvo switches. Many systems include a combination of local control panels, external computer control for automated time event switching sequences, and even dial-up telephone control via DTMF codes. The control and distribution possibilities of a routing switcher matrix have become an indispensable design consideration in all but the very smallest facilities.

Configuring a Matrix

Routing switcher systems are typically manufactured as distinct video and audio matrices, capable of being controlled together. To achieve an AFV switch, it is necessary to wire the video and audio signals to corresponding input/output locations on the router frames. In certain situations, it may be necessary to switch the video signal without changing the audio, or vice versa. These special case AFV situations require control panels with the ability to independently "split" or "breakaway" the video and audio source feeds .

Cameras, still stores and character generators are only a few of the many video only devices in common use. Audio counterparts, having no video component, can also be easily identified. If these single level sources are wired into corresponding "spare" input positions in an AFV matrix, breakaway switching must be used to prevent unintended switches from occurring. In practice, the probability of errors occurring is too great to permit this wiring scheme to be reliable.

Installing video only or audio only devices in a switching matrix requires dedicating a portion of all router frames for single level switching. This is accomplished by omitting input or output cards, if possible, or by terminating unneeded crosspoints. While this practice is quite effective, it has the unfortunate side effect of reducing the available frame capacity.

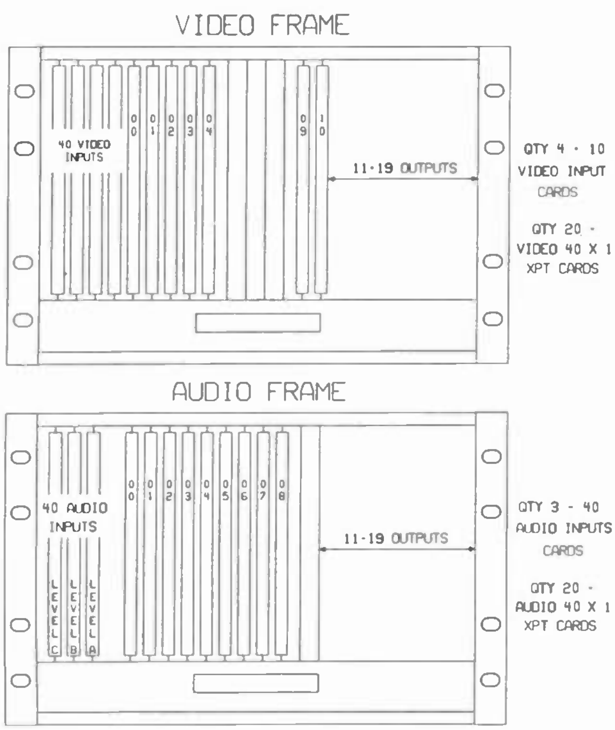

FIGURE ONE:

EACH VIDEO AND AUDIO FRAME CAN HOUSE UP TO 20 OUTPUT CARDS. IN THIS EXAMPLE, OUTPUTS 00-04 ARE AFV SWITCHES, 05-08 ARE AUDIO ONLY, 09-10 VIDEO ONLY, AND 11-19 AVAILABLE FOR EXPANSION. AS CONFIGURED, THE MAXIMUM USEABLE CAPACITY OF THE SYSTEM IS 17 VIDEO AND 18 AUDIO OUTPUTS.

Considerable attention must be paid to the likelihood of both single level and AFV expansion. Unless allowances are made, unexpected equipment changes may require a major rewiring and shuffling of switch cards to achieve the desired matrix. For this reason, some excess frame capacity is necessary to allow for future growth, and some must be reserved to accommodate single level switching.

During the last five years, most router upgrades were prompted by the need to add new audio switching levels to a video/audio one matrix. The reality of network stereo programming was a major stimulus. A SMPTE time code level, too, became an essential element in many systems to meet growing editing, automation and duplication requirements. Generally, these system expansions could be undertaken without a major rebuild.

In some cases, upgrading amounted to little more than adding plug-in cards to existing frames wired for two, or even three audio channels. Even when new frames were required, little rewiring was necessary. Since most of the equipment requiring video and three audio channels was already installed in the routing switcher, the matrix wiring had already been optimized for multi-level switching.

For new installations, it was merely necessary to order enough frame capacity for any needed combination of video, audio 1, audio 2, and time code signals. To be sure, few engineers specifying 128x128 frames really expected to fully populate them. A one-time expense, excess router frame capacity would help to maintain long term flexibility and provide insurance against a premature rebuild.

A New Class of Signal Requirements

Routing switcher expansions today are much more difficult to manage. On the one hand, facilities are increasingly faced with the prospect of routing signals which, to varying degrees, are simply incompatible with their existing video and audio grid. On the other hand, advanced planning must take into consideration both new uses for equipment that is already installed, and buying patterns for entirely new classes of equipment.

SWITCHING SYSTEMS TODAY

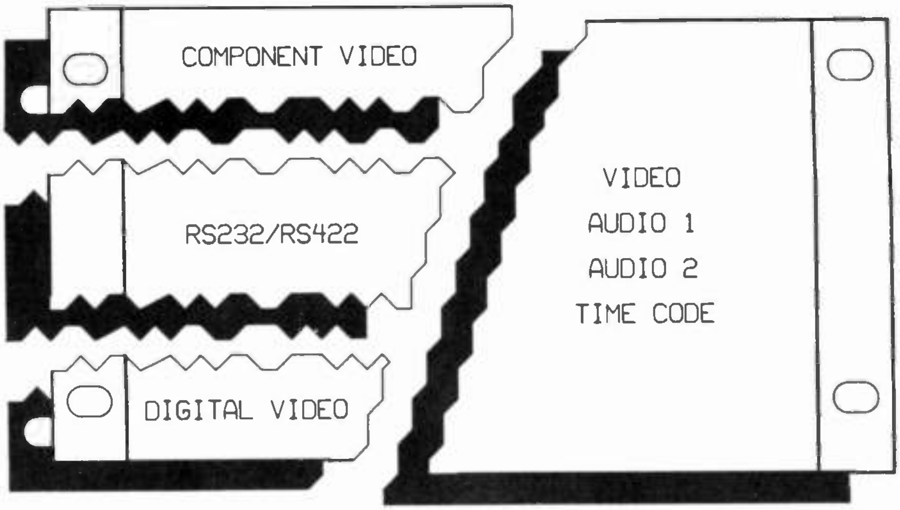

FIGURE TWO:
NEW SIGNAL FORMATS REQUIRE MOST FACILITIES TO SEPARATE THEIR SWITCHING INTO SEVERAL UNRELATED SYSTEMS.

Routing needs must be reassessed in light of the technological advances which have improved video signal quality. The most radical proposal for improved picture quality entails the complete replacement of all established video standards with a new high resolution alternative. The 30 MHz bandwidth necessary to realize HDTV specifications significantly exceeds the performance parameters which dictated most NTSC/PAL/SECAM router designs. Though wideband routing switchers are now available, it is legitimate to question why all video crosspoints should be upgraded to meet a limited requirement.

Other improvements have focused on preserving the quality of existing signals through new recording technologies. The Betacam and MII component video formats do not impose new bandwidth demands, (something that is not always true of high resolution RGB signals), but do require the simultaneous switching of three video signals. Most routers in the field can accommodate these requirements only by slaving two additional video frames together. Few manufacturers offer any practical way of utilizing remaining frame capacity for systems in the field. Fewer still can reconfigure a single chassis to route the four channels of audio these tape formats provide.

Many of the newer VTR's now provide the convenience of a separate, dedicated monitor output capable of displaying machine ID, diagnostic or time code data. For this reason, it is no longer even safe to assume that a single video path is adequate to meet the switching requirements of composite video. Moreover, production techniques have evolved which necessitate switching of video key as well as program video signals.

The D1 and D2 digital video formats require an entirely new breed of routing switcher. At the present moment, this means a multipin parallel data transmission path, though serial digital switching systems are believed to be under development. The D2 format has been promoted as a candidate for direct replacement of the Type C VTR, by retaining significant levels of compatibility with existing routing switcher systems. For this reason, we can anticipate D2 switching requirements in both the analog and digital video domains.

Machine control functions provide another area which can be managed through a switching matrix. Now that RS232/RS422 serial control ports have become standard features of tape machines and related support equipment, it is practical to establish a central machine room with the entire house inventory available to edit suites as required.

The need for special purpose routing equipment to handle these new signal formats should be obvious. It is equally clear, however, that new switching levels cannot be casually added to an existing routing matrix. Assessing the full impact of these new routing switcher systems on a facility, requires a look at how these new video and serial control products are currently being used, and how this is likely to change.

The Impact of New Technology

The technical prowess of these new signal formats has already been conceded in the marketplace. By now, nearly all major facilities have augmented their inventory with some component or digital video equipment. Yet, even the most ardent proponents of change will acknowledge that gradual, rather than wholesale, replacement has been the rule.

Many facilities have substantial inventories in proven formats (e.g., Type C and Umatic). Until this hardware is further advanced on the aging cycle, there is little incentive to abandon the security of reliable products. In many cases, a substantial investment of parts and service training is also at stake. Uncertainty regarding which of the competing formats to buy (Betacam, MII), the absence of necessary support equipment (e.g., a D2 production switcher), high cost (D1), and lack of market penetration and standardization (HDTV) will also be a factor in maintaining the status quo.

At first, new equipment can be introduced to meet limited and specialized needs. Independent centers of component video have already been established for newsgathering. Digital hardware, similarly, has been confined to postproduction or graphics suites. Yet, as inventories of these products grow, and the new capabilities of this hardware are more widely known, new techniques and procedures will be introduced to make these resources more widely available throughout the facility.

Inevitably, albeit at different rates, new technology begins to intrude beyond the confines of the dedicated centers. Interformat editing, and graphics or special effects transfers, may play a leading role in promoting a high level of shared technology. Composite video and analog audio signals, the "stuff" of which main house routing switchers are made, re-emerge as the obvious bridge between the new and old technologies.

Adding new switching levels to the master grid offers the same promise of operational simplicity and flexibility. However, once a system has been established, adding new equipment means rewiring all matrix frames to realign the relationship between signals on each switching level. Based on the prevailing pattern of technological acceptance and replacement, it may no longer be possible to avoid a rebuild by devising a wiring scheme flexible enough to accommodate future requirements. Rather, what is required is a more flexible routing switcher control system.

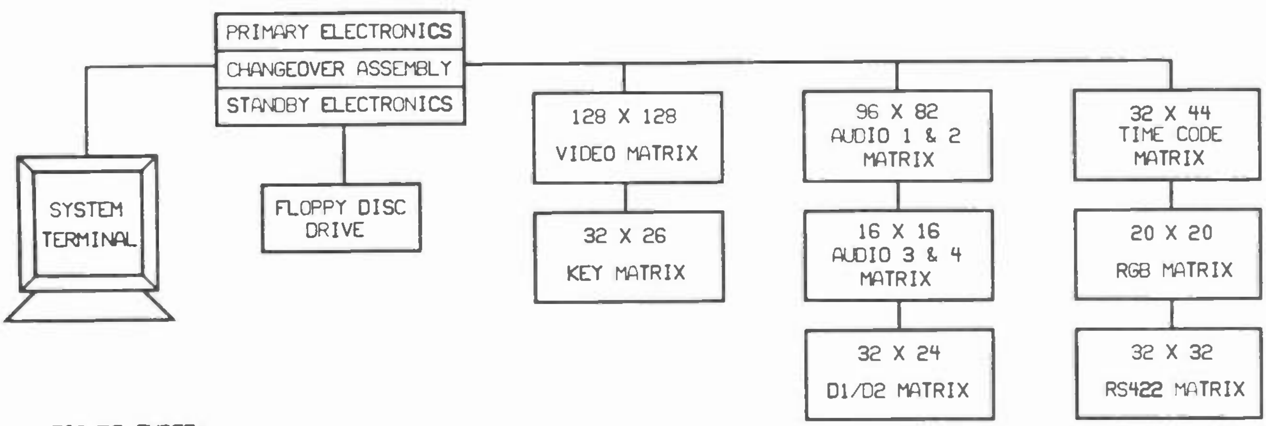

FIGURE THREE:

BLOCK DIAGRAM ILLUSTRATES HOW ANY COMBINATION OF LARGE OR SMALL ROUTING SWITCHERS CAN BE SMOOTHLY INTEGRATED INTO AN 8 LEVEL MASTER GRID.

Advances in Routing Switcher Control Systems

Conventional switching logic defines crosspoints based upon their physical location on a switcher frame. A sensible alternative would entail using the advanced abilities of a microprocessor to refine routing switcher control logic. Ideally, switching crosspoints should be selected on the basis of their logical and operational relationship to a given device, rather than how each switching level is connected on a frame. In short, it must be possible to completely wire the routing switcher via software!

This can be accomplished by creating tables of source and destination names, and then mapping the crosspoints associated with each name. Since a given name could refer to a single level or AFV device, the control system must consult these tables before executing any switch. (See Figure Four) . The term "virtual matrix" has been used to describe this type of routing switcher system.

A virtual matrix may be defined as a universe of input or output devices containing crosspoints on any physical connect point across one or more control levels. A system satisfying the following criteria would meet the design objectives of a virtual matrix:

1) The system shall be based upon parameters that users set on-line via a system terminal. These parameters should include matrix size, number of control levels, source and destination names, and virtual matrix switching tables.

2) The system shall be capable of single level switches, multilevel switches (i.e., AFV), breakaway switches, and "OFF" switches to prevent unintentional breakaways.

3) The switching matrix shall be defined in terms of source and destination names. Crosspoints associated with source or destination names may occupy any physical input/output connect point for each available control level.

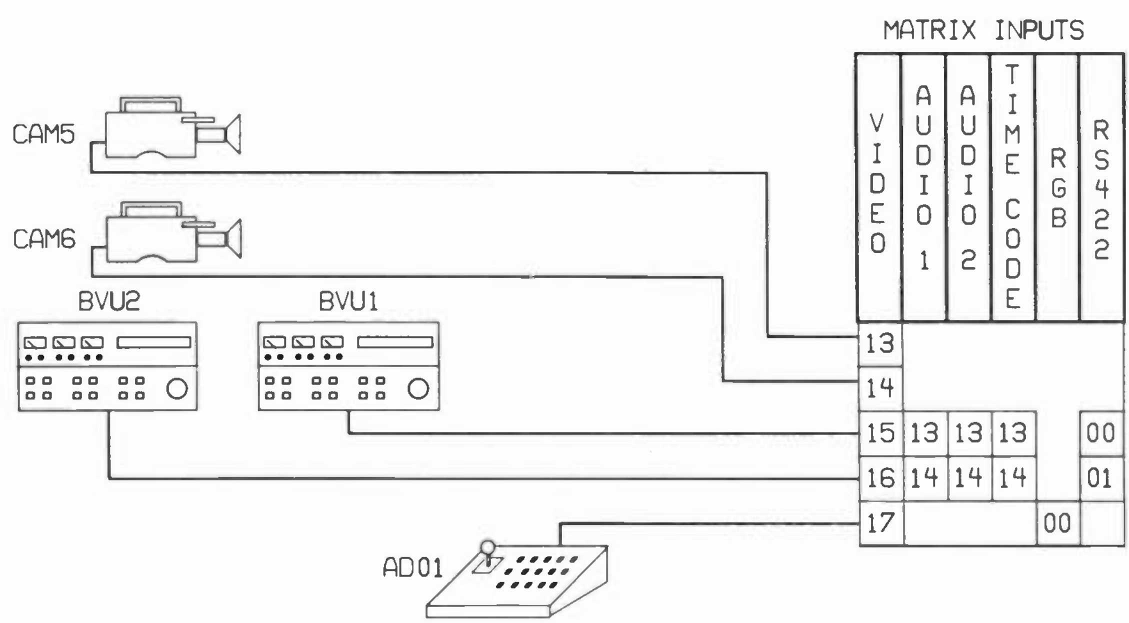

FIGURE FOUR:

IN A VIRTUAL MATRIX, ALL SOURCE OR DESTINATION DEVICES ARE NAMED, AND ASSIGNED TO SPECIFIC PHYSICAL INPUT OR OUTPUT CONNECTIONS. AN OPERATOR SWITCHING BVU1, FOR EXAMPLE, AUTOMATICALLY SELECTS AFV CROSSPOINTS ON FIVE SWITCHING LEVELS. CAM5, ON THE OTHER HAND, IS DEFINED AS A VIDEO ONLY SWITCH.

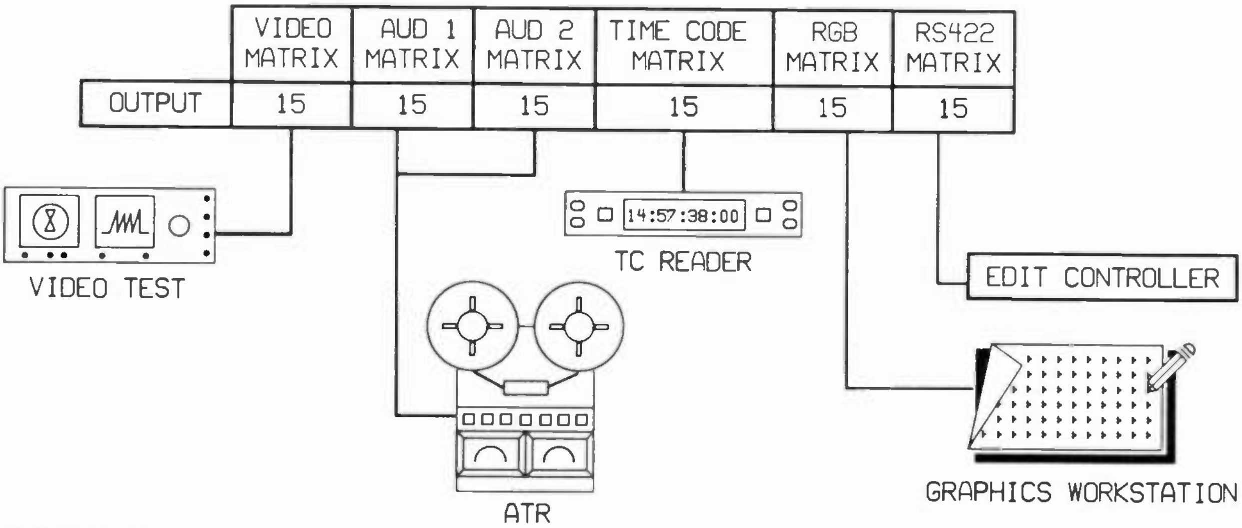

FIGURE FIVE:

IN A VIRTUAL MATRIX, EACH SWITCHING LEVEL OF EVERY OUTPUT CAN BE CONTROLLED INDEPENDENTLY. IN THIS EXAMPLE, PHYSICAL OUTPUT 15 HAS FIVE SEPARATE USES. OUTPUT SWITCHES COMBINING VIDEO-16 • AUD1-20 • AUD2-20 • RS422-15, FOR EXAMPLE, ARE ALSO POSSIBLE.

4) It should be possible to include a physical crosspoint in the definition of more than one device to conserve router capacity and simplify the operation. Confirming switch tallies should indicate this pre-defined "breakaway" condition.

Certain operational features are essential to implementing and maintaining a virtual matrix. A password to prevent tampering with the matrix definitions is essential. Provisions should be made for system backup via floppy disk, or documentation via hardcopy printout. Finally, the system should be protected against power line disturbances, be capable of detecting and rejecting erroneously or improperly formatted switch messages, and provide standby electronics to ensure operational redundancy.

The Benefits of a Virtual Matrix

Routing switcher systems fully implementing a virtual matrix control architecture have been installed since 1987 (see Pensinger 1988). Users of these systems have included post-production facilities, program originators, television stations, news bureaus and industrial concerns.

Simplicity of operation, enhanced flexibility, improved utilization of frame capacity, ease of expansion, cost savings, and reduced rewiring downtime are among the reported benefits. Although the examples offered have tended to show large system applications, it should be noted that even a video/two channel audio matrix becomes a functionally larger, and more versatile switcher when every input and output connection can be utilized (Compare with Figure One).

Teleproduction facilities will probably continue to depend upon many small, special purpose routing switchers to handle their signal flow requirements. Virtual matrix control technology offers a proven method of re-integrating these switchers into a new master grid.

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DIGITAL VIDEO: CONVERTING BETWEEN DIGITAL FORMATS

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ABSTRACT

The standardization of exchange parameters for digital video systems has made possible very high quality video production. Fully transparent processing is expected in a digital studio. However, conversion between the digital component standard (CCIR 601) and the proposed digital composite SMPTE Recommended Practice (SMPTE T14.22/082) is subject to processing which will make the conversion less than transparent. Quantization errors, decoding artifacts, loss of active picture area and reduction of bandwidth occur during digital format conversion. Several solutions exist which minimize the impact of processing. Compromises must be made.

INTRODUCTION

The standardization of exchange parameters for digital video systems has made possible very high quality video production. Component digital suites utilizing the CCIR 656 standard for signal interchange maintain noise free, wide bandwidth component signal quality. The signal suffers little or no loss in quality during recording and mixing processes. Providing the signal remains in digital form, multigeneration performance of recording devices is excellent, providing improved performance over their component analog counterparts.

The introduction of the composite digital tape recorder brings the advantages of digital recording into the analog composite studio. Composite digital recorders provide superior performance over their analog counterparts in terms of Signal-to-Noise (SN) ratio and signal stability. Although intended to operate in an entirely analog environment in most studios, the recorder makes the digital composite signal available for dub purposes and for interconnection to other digital composite devices.

The digital component and digital composite signals are incompatible. In order to use equipment of different signal type, the signal is converted through some sort of format translator. This can be accomplished by converting the signal to analog and using existing codecs, or while the signal is in digital form through a digital format translator. Unlike analog processing, fully digital processing will maintain improved multigeneration performance and high levels of transparency.

THE AMPEX DST-300

Ampex has developed the DST-300 Digital Standards Translator to meet the needs of digital studios. The DST-300 provides conversion of component digital video to digital composite and composite digital video to digital component. Both functions are packaged in one 3U chassis. The DST-300 is outlined in this paper.

THE DIGITAL FORMAT CONVERSION PROCESS

Conversion between digital component and composite signals entails some of the same processes as conversion between analog component and composite signals. The processes are somewhat modified because of the digital nature of the signals. However, new issues arise which require attention.

When converting from digital component to composite, the digital component signal is scaled and the color difference signals are filtered and quadrature modulated exactly as the analog component signal. The color difference signals are matrixed for modulation along the proper axes. When converting from digital composite to component, the digital composite signal is decoded to baseband and scaled. The decoding method can be multi-dimensional and adaptive, exactly as the analog composite signal. Encoding and decoding are much more controlled and precise in the digital domain, particularly when the signal is sampled at four times the color subcarrier frequency (4Fsc). Scaling is also more controlled in the digital domain but is subject to visible errors, most notably banding. Blanking is different in the two formats. The digital composite signal has more blanking both vertically and horizontally.

The digital component luminance is up-sampled from the component digital sample frequency of 13.5 MHz to the composite digital sample rate of 14.3 MHz (4Fsc). Similarly the digital composite luminance is down-sampled from the higher rate to the lower rate. The color signal sample rate is changed down or up at half the rate of the luminance. These processes are carried out in real time through a multiple phase interpolation filter. These processes are not required for analog signals since they are time continuous.

1. Encoding and Decoding

Encoding and decoding of the video signal involves mixing and separating the modulated chrominance signal from the luminance signal. The digital composite sampling grid is tied to the subcarrier frequency with a specific, fixed phase relationship along the I/Q axes in NTSC and along the U+V/U-V axes in PAL. Filtering and modulating a digital subcarrier is simplified if the sampling grid is related to subcarrier. In a digital format translator the coding and decoding may take place in the area of the circuit which operates at 4Fsc, although this may also be done at 13.5 MHz.

In NTSC, the 4Fsc sampling grid is phased along the I and Q axes. If the circuit operates at 4Fsc, modulation of I and Q baseband signals in digital becomes a simple process because the modulating subcarriers have only four possible values. Demodulation is a simple matter of demultiplexing the subcarrier pixels and inverting the alternate samples. Extracting the chrominance signal with a bandpass filter is simplified because of the nature of digital Finite Impulse Response (FIR) filters.

Digital video signal processing makes encoding and decoding more controlled and less subject to drift and noise errors when compared to purely analog methods, allowing the use of more sophisticated processing techniques. Fundamentally the same processing takes place on the video signal whether analog or digital, that is, encoding and decoding. Encoding requires filtering to band limit the baseband chrominance signals. Multi-dimensional filtering of the entire signal is employed while decoding to avoid visible cross luminance and cross chrominance distortions that result from one dimensional filtering.

There are two fundamental techniques in decoding. The first is to apply two separate filter algorithms to the composite signal in which luminance and chrominance are separately extracted. The second is to apply one filter algorithm to the composite signal to extract one of the components (luminance or chrominance) and subtracting this component from the composite signal to get the other (complementary) component. Both techniques are subject to errors in the form of familiar cross chrominance and cross luminance artifacts. The system application will dictate which decoding technique to use. Hybrid techniques are also useful. The major concern of any decoding technique will be accurate separation of the composite signal to baseband with maximum artifact suppression.

The luminance/chrominance separation process has been thoroughly studied. Two and three dimensional comb filters exist which provide quite accurate separation of signals by taking advantage of the repetitive nature of the video signal. These comb filters can be made to adapt the mode of operation based upon analysis of the spectral content of the area being decoded.

2. Sample Rate Conversion

The conversion of a digital data stream from one sample rate to another is a complex process. There are some simplifications which can be made in the case of conversion between the two digital format sample frequencies. First, the stability of the sampling grid is related to either horizontal sync or to burst. Second, the horizontal position of the

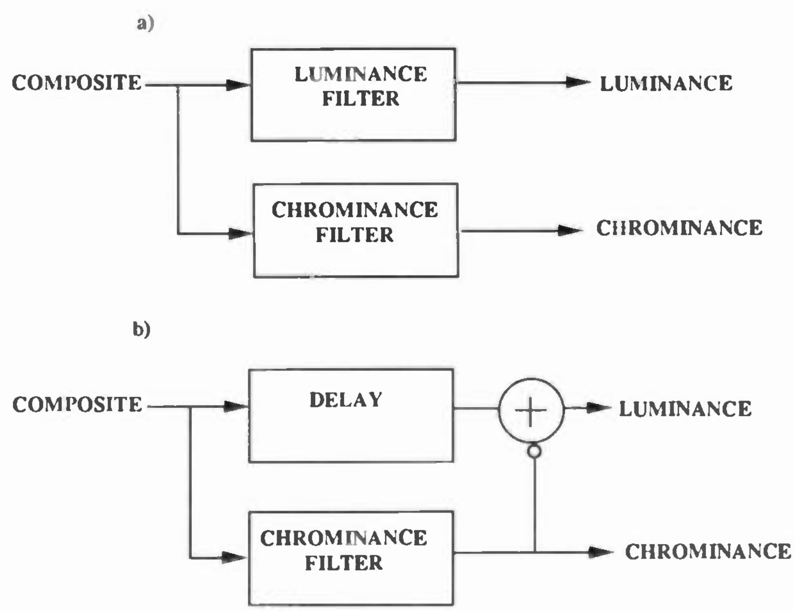

FIGURE 1 a) **Non Complementary Decoding**
b) **Complementary Decoding**

samples on a line is related to the 50% point of sync for digital component signals and to the burst signal as specified by RS 170A for digital composite signals. The interpolation filter provides the appropriate phase offset between input and output samples in order to position the output sample in the correct horizontal position. The phase offset changes for each consecutive sample in the data stream.

The interpolation filter for NTSC digital composite input provides 35 luminance output samples at 14.3 MHz for every 33 input luminance samples at 13.5 MHz. This requires 35 filter phases which yields an effective sampling rate of 472.5 MHz.

The chrominance is also interpolated to the appropriate higher or lower frequency but at half the rate of the luminance. The baseband (unmodulated) chrominance is interpolated rather than the modulated chrominance spectrum since the passbands are low and less critical at baseband frequencies and slight errors in the phase of the filter will manifest themselves as unnoticeable horizontal offsets rather than noticeable hue errors.

3. Bandwidth Limitation

The bandwidth of the baseband luminance and color signals is modified in making the conversion from one digital format to the other.

The higher sample rate of the digital composite signal allows more luminance bandwidth. The digital composite signal is sampled at four times subcarrier and as such in both NTSC and PAL has a wider luminance bandwidth specification. The color difference signals are both limited to composite bandwidth specifications of 1.3 MHz for U, V and I signals and to 0.5 MHz for Q.

The conversion from one digital format to another requires some compromises in bandwidth and ripple specification. Clearly a signal which originates in one format cannot be reproduced exactly in the other.

4. Active Line/Field Width

The digital composite signal is a sampled representation of the analog composite signal with a nominal Zero Subcarrier-to-Horizontal (0 SCH) blanking interval. The data stream could be applied directly to an analog-to-digital converter to recover a broadcast-ready analog signal. The blanking interval, including horizontal sync, burst and vertical sync is sampled and provides all timing and phasing information for the digital signal. The digital composite signal specifies a wider active line than its digital component counterpart. Vertically, the digital composite signal conforms to the standard of blanking the last half of the last line of field 1 and the first half of the first active video line of field 2.

The digital component signal is a multiplexed signal containing luminance and chrominance samples interleaved in a 27 MHz data stream. The signal cannot be directly applied to a D/A converter. The blanking intervals are not sampled; horizontal and vertical sync information is not contained in the video signal itself but rather in a sequence of data words inserted into the data stream at the start and end of each video line. The first sequence designates the end of the active video line with three reserved words and a timing byte. The second sequence designates the start of the active video line with the same three reserved words and a different timing byte. There is no difference in line width for 525 or 625 systems. The entire last line of field 1 and first active video line of field 2 are full active lines.

5. Quantization and Scaling

The digital component luminance signal has 220 quantization levels and the color difference signals have 225 quantizations levels. Luminance values have some negative headroom (-7 IRE units) but sync is not sampled.

The digital composite signal is a complete representation of the analog composite waveform; sync, burst and blanking are all sampled. Black to peak white luminance has 140 quantization levels and demodulated chrominance has 136 quantization levels.

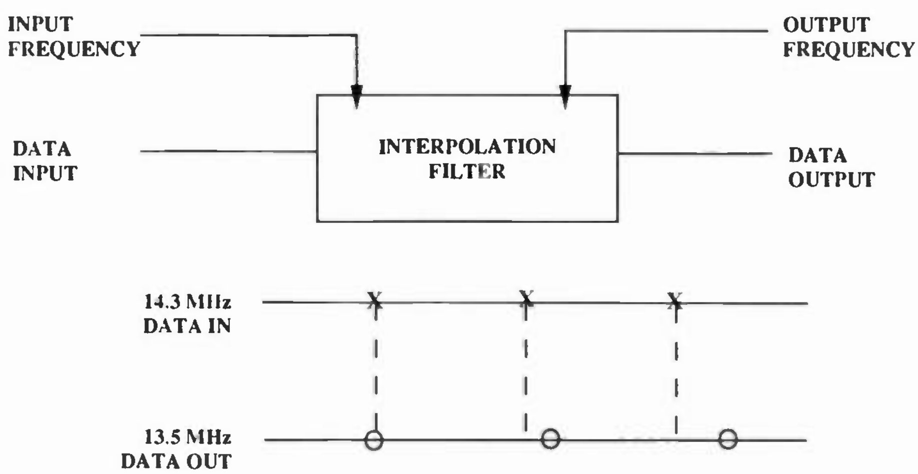

FIGURE 2 SAMPLE RATE CONVERTER

The conversion process requires scaling to match a given input range to the output range. The luminance is scaled between 140 to 220 quantization levels and the chrominance is scaled between +/-68 to +/-112 quantization levels.

6. Matrixing

The component digital signal contains luminance and the color difference signals C_b and C_r. These signals are scaled versions of the B-Y and R-Y signals. The scaling equations are:

$$C_b = (0.564) * E(b\text{-}y)$$
$$C_r = (0.713) * E(r\text{-}y)$$

The digital composite signal is sampled along different axes than C_b and C_r. As mentioned previously, digital NTSC video is sampled along the I/Q axes and digital PAL video is sampled along the (U+V)/(U-V) axes. A matrix is used to convert baseband signals from one format to the other.

SPECIFIC ISSUES ASSOCIATED WITH THE DIGITAL FORMAT CONVERSION PROCESS

Digital format conversion incorporates several processes. Each step is prone to error. The errors can be minimized or eliminated, although the solution may be a compromise.

1. Encoding & Decoding

The extraction of chrominance from the composite requires careful analysis. The bandpass filter covering the chromiance spectrum will faithfully extract low frequency chrominance signals. Horizontal chrominance freauencies cause subcarrier to deviate from is center frequency, leading to the misinterpretation of chrominance as luminance dots. This cross luminance can be reduced by the use of a wider bandpass filter at the expense of misinterpreting some legitimate luminance frequencies as colored rainbows. Rainbows (cross chrominance) and dots (cross luminance) are artifacts of composite decoding when a one dimensional filter is applied to a signal containing one dimensional frequency deviations.

The composite signal can be analyzed horizontally, vertically and temporally for spectral content. The repetitive nature of the composite video signal distributes the chrominance and luminance spectra into certain bands of the three dimensional video spectrum for most normal picture material. High order filters can be applied to the signal to extract only specific frequency ranges within the signal in order to improve performance of the signal separation. However, a given order of filter will only be appropriate to lower orders of spectrum. That is, a two dimensional filter will be appropriate for horizontal luminance and chrominance energy but a one dimensional filter will lead to errors in this instance. A one dimensional filter will be appropriate for a signal with only vertical energy and no horizontal energy in the luminance and chrominance. Three dimensional filters are appropriate for signals which contain both horizontal and vertical energy but will lead to errors when applied to signals with temporal information.

The ideal decoder should in fact utilize all three methods of decoding by analyzing the picture content and applying the appropriate filter to the signal adaptively. Inexpensive digital line delays have made adaptive two dimensional comb filtering cost effective and frame delays for temporal filters are becoming more economical.

	PASSBAND	PASSBAND RIPPLE	PASSBAND GROUP DELAY	START OF STOPBAND	STOPBAND ATTENUATION
LUMINANCE	5.5 MHz	±0.05 dB	±6 nS	6.75 MHz	-12 dB
	5.75 MHz	±0.1 dB			
COLOR DIFFERENCE	2.75 MHZ	±0.1 dB	±12 nS	3.375 MHz	-6 dB

FIGURE 3 DIGITAL COMPONENT SIGNAL SPECIFICATIONS

	PASSBAND	PASSBAND RIPPLE	PASSBAND GROUP DELAY	START OF STOPBAND	STOPBAND ATTENUATION
NTSC	5.5 MHz	±0.2 dB	<10 nS	7.16 MHz	-20 dB to -30 dB
	6.0 MHz	+0.0/-1.0 dB			
PAL	6.0 MHz	±0.2 dB	<10 nS	8.86 MHz	-20 dB to -30 dB
	6.5 MHz	+0.0/-1.0 dB			

FIGURE 4 DIGITAL COMPOSITE SIGNAL SPECIFICATIONS

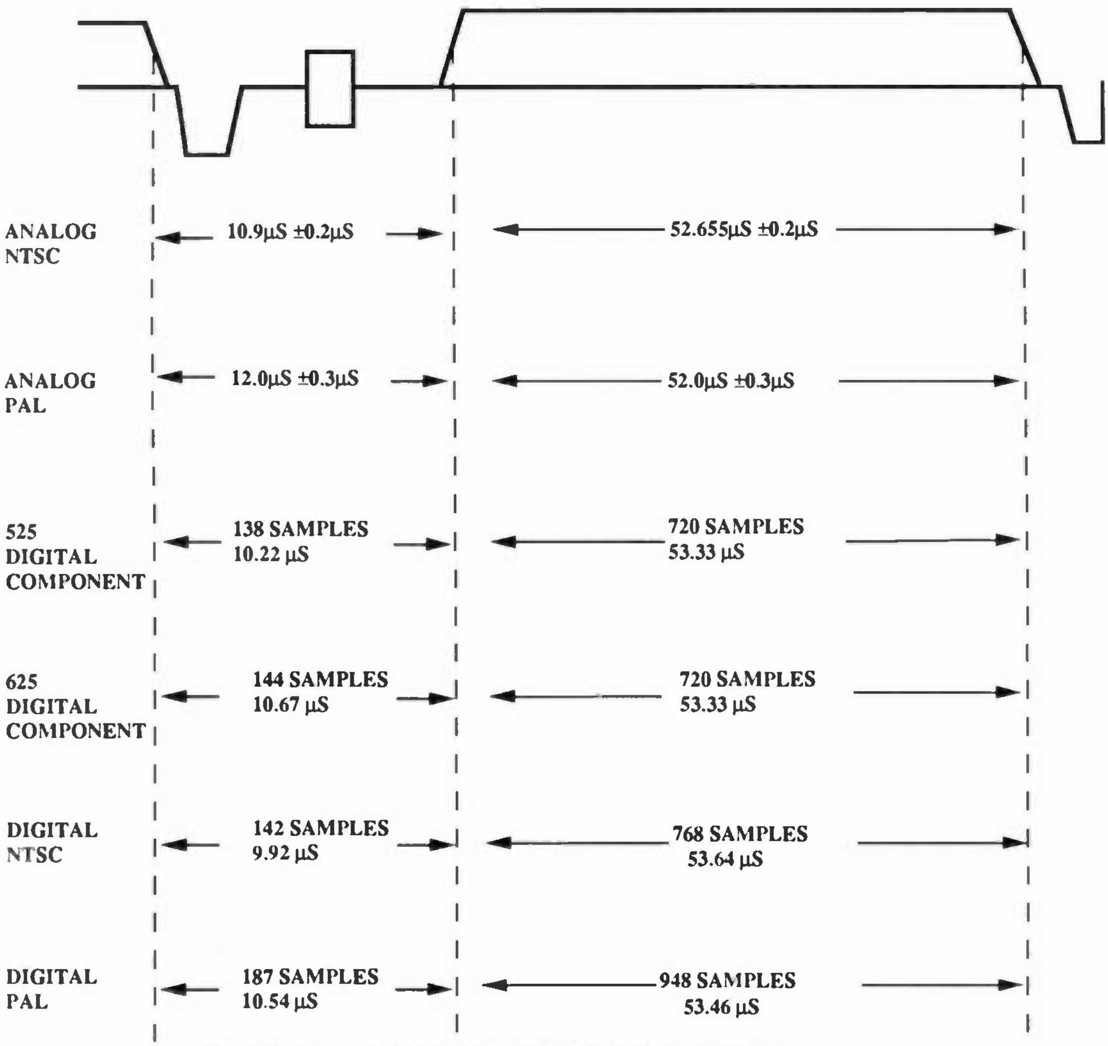

FIGURE 5 ACTIVE LINE WIDTH COMPARISON

1.1. Complementary versus Non-Complementary Decoding

Current decoder designs will create artifacts of some type given typical picture content with horizontal, vertical and temporal frequencies. These artifacts can be minimized using the aforementioned decoding techniques. However, if a decoded signal is re-encoded the artifacts present in the baseband signal will be re-encoded as well. One dimensional and two dimensional artifacts are related to the original chrominance signal and contain subcarrier rate motion. If decoded artifacts are re-encoded with subcarrier that is in exact phase lock with the original subcarrier the artifacts will be masked. The signal potentially could be re-encoded to the original state, ie. no new artifacts are created. This can be insured by sending the color frame information of the original signal to the encoder modulator. If the color frame is not sent, the encoder may re-encode a signal at the original phase or it may be off by 180 degrees (or 90 or 270 degrees in PAL systems). The re-encoded artifacts, with their changing subcarrier component, will be added in antiphase to the new subcarrier signal and will produce low frequency beats in the signal at 15 Hz (NTSC) or 6.25 Hz (PAL).

One way to prevent artifact propagation is to use a complementary decoding algorithm and preserve the color frame sequence when decoding the composite signal. Artifacts will exist and be perceptible in the baseband decoded image but will not grow on successive generations and will not require the loss of signal content associated with non-complementary decoding. Of course, the signal should be re-encoded using the original color frame information to lock the subcarrier to the same phase as the original signal.

Color frame is identified by the phase relationship between the subcarrier burst and the horizontal synchronizing pulse. The composite digital signal carries this information in the blanking interval and it can be readily decoded. The digital component signal, however, carries no subcarrier or color framing information. Proposals exist for inserting the color frame information into the blanking interval. The DST-300 sends the information on a separate BNC. No provisions currently exist which write this information to storage devices such as the D1 format recorder.

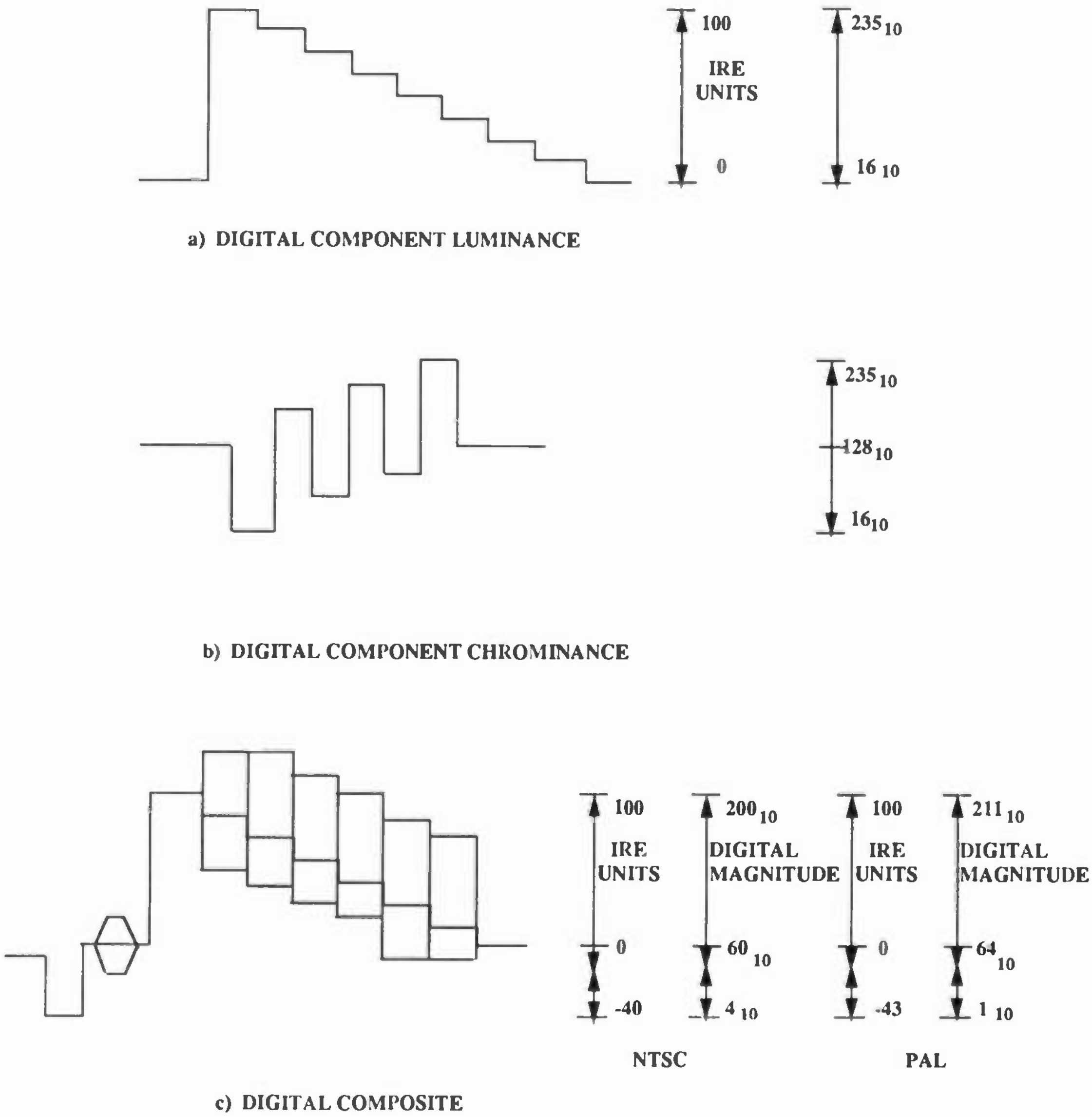

FIGURE 6 DYNAMIC RANGE OF THE SIGNALS

If the color frame of the decoded signal cannot be stored the decoder should not utilize complementary decoding. As previously discussed, the re-encoding of artifacts with incorrectly phased subcarrier leads to objectionable errors. A non-complementary decoding technique should be utilized which eliminates resolution in the spectrum that leads to the most objectionable errors. The DST-300 allows the user to select the correct decoder for the task.

1.2. Encoder Prefiltering

Artifacts can be virtually eliminated at the encoder using "clean" encoding techniques. Typical 2D adaptive decoders cannot properly separate portions of the diagonal luminance content from the composite signal. Non-adaptive comb and bandpass filters also fare poorly under such circumstances. Prefiltering the luminance and chrominance at the encoder is one solution. For a given decoder/encoder system, the same filter is applied to the baseband video at the encoder as is applied to the composite video at the decoder. Potential artifacts are virtually eliminated. The compromise is that resolution is lost. If a 2D filter is applied at the encoder, some diagonal luminance and vertical chrominance resolution is lost. Unmatched decoder filters will allow some residual artifacts to pass through the decoder. If a 3D filter is applied at the encoder, a 3D decoder will yield artifact free images but some temporal resolution will be sacrificed. An adaptive technique could correct this but the filter and the adaptive algorithm applied at the encoder should be matched at the decoder.

"Clean" encoding is one solution to the problem of decoder artifacts. If the loss of resolution is acceptable then it makes sense to perform this encoding on the first generation conversion from digital component to digital composite. The DST-300 allows the user to turn the "clean" encode function on or off as needed. Industry wide cooperation is needed to standardize the filtering technique.

1.3. Encoder and Decoder Delays

Decoding and "clean" encoding introduce delays in the video path. The delays may be lines or even fields and frames depending upon the technique used. If the user wishes to have the output digital composite signal timed to the external reference the input signal is advanced to compensate for the delay in the processing electronics. This is facilitated in the DST-300 by providing an advanced reference output. The advanced reference signal can be used to back up the timing of the source machine. In some instances the source may not be able to advance itself from reference. It may be possible to delay the output of the digital format converter to the nearest whole frame. This is possible only if whole frames of delay are tolerable in the system.

2. Sample Rate Conversion

Interpolation of samples at one sample rate to another rate is prone to phase error. To preserve bandwidth and minimize phase error the interpolation filter requires many taps. High order filters provide the best approximation to the digital component and composite bandwidth requirements. The filter has sharp cutoff at the edge of the passband to maintain the anti-alias requirement of the digital system.

The Interpolation filter provides the means of creating output samples at points horizontally which lie in between input samples. As stated previously for NTSC, there are 33 input samples at 13.5 MHz for each 35 output samples at 4Fsc. More taps in the filter and wider data paths help insure spatial accuracy. However, this should be balanced against the impulse response of a long filter. Very long FIR filters will cause excessive ringing around sharp vertical transitions.

The high performance expectations of digital video has created new ways of analyzing the performance of a piece of video equipment. Specifically, the promise of endless transparent generations of processing has prompted users to apply stringent tests to all digital video equipment. Some test signals are inappropriate for digital system testing and will lead to a false perception of the performance of a system. As just described, a wideband interpolation filter will ring according to its impulse response when tested with a digital impulse such as a sharp vertical transition. It may be necessary to tradeoff bandwidth to reduce ringing when out of band transitions are encountered. The DST-300 has a switch to select a less disturbing impulse response when non anti-aliased pictures are converted.

This concern exists in a more general way throughout the digital video world. For example, some color bar test signals now available for digital component systems have non anti-aliased transitions. Recording and playback devices introduce no distortions because they do not process the spectrum of the signal. However, any signal processing device (such as a digital format translator) that filters the signal will respond to the non anti-aliased transition with the impulse response of the filters in the system. A wideband filter designed to pass the full required spectrum of video will ring in response to this color bar transition, whereas a lower bandwidth filter (such as a simple bilinear interpolator with a gaussian characteristic) will not ring in response to an impulse. The observer may conclude that the lower band system actually responds better to a highband test signal, when in fact the opposite is true because the test signal is not providing the correct input spectrum.

3. Bandwidth Limitation

Both composite and component formats have areas of compromise in regards to bandwidth of the luminance and chrominance signals. The digital composite signal allows for luminance up to 6.0 MHz (NTSC) or 6.5 MHz (PAL) to be represented. The extra bandwidth is to insure frequency response and group delay are transparent with respect to video bandwidth after many generations in the analog composite . The signal is limited to 5.75 MHz when being converted to digital component and although this does not remove useful picture content some out of passband ringing is introduced if the luminance contains high frequencies.

The color difference signals in the digital component format are both wide band signals with the passband extending to 2.75 MHz. This allows excellent color detail representation. The signals are limited when converting to composite. U, V and I signals all limit the color passband to 1.3 MHz and the Q signal limits the passband to 0.5 MHz. Conversion to composite removes some color detail. This is recognized as soft chrominance edges, particularly with digitally generated graphic input.

The requirements of luminance and chrominance bandwidth imposed by the digital video formats does not allow much opportunity for total bandwidth preservation in either form. As described, component luminance signals are slightly lower bandwidth than composite luminance signals, and composite chrominance signals are lower bandwidth than component chrominance signals.

One possible improvement is to allow equi-band I and Q signals in NTSC, thus eliminating the loss of bandwidth incurred in the Q signal. Another is to modulate full bandwidth (2.75 MHz) color difference signals to preserve the high color definition of the digital component signal. Yet another proposition is to use specialized circuitry which enhances chrominance edges when they are decoded from the digital composite signal in order to simulate the lost chrominance resolution. The DST-300 provides correct NTSC encoding or equi-band IQ encoding.

The luminance signal in the component digital format has less bandwidth than the luminance signal in the composite digital format. The relative loss of bandwidth is necessitated by the fact that the sampling frequency of the component signal is lower than that of the composite signal. There are no exceptions that can be made to the bandwidth limitation requirements since any high frequencies above 5.75 MHz in the component signal will cause alias distortions.

4. Active Line/Field Width

The digital component signal can accommodate all of the active area of a legal composite signal and more. The digital composite signal can theoretically accommodate all of the active area of the digital component line area but in doing so exceeds the limit of analog composite active line width. Furthermore, the composite signal cannot accommodate the extra half line of video at the top and bottom of the raster.

The composite signal active line width is clearly in conflict with the digital component line width. The digital composite standard allows for more active line area than the broadcast standard and thus for studio use it may be useful to encode the entire digital component active line. This violates the composite blanking requirement but for studio applications it preserves the maximum line content. Similarly, the first line of field two is supposed to be clipped out by the vertical blanking circuit of an encoder, but it may be useful to pass video in that interval in studio applications. The DST-300 allows the user to preserve as much of the active picture content as possible. Of course, if wide blanking is necessary the system should provide a smooth blanked edge to protect against ringing. The only other area of conflict between digital formats, the last line of field one, must be removed by the encoder since the vertical blanking interval and vertical sync pulses begin at this time.

5. Quantization and Scaling

Some operations change the quantization level of pixels and lead to errors. If a digital signal is processed by filtering of any type, or if the signal is changed in gain in any way, artifacts will appear in the signal after processing. Remapping of pixel values to new quantization levels in a system with a finite number of quantization levels leads to visible errors. This remapping causes spurious frequencies to appear in the signal. For example, when converting 8 bit YUV signals to RGB, the effective chrominance range decreases because the number of quantization levels drops from 225 to 220. If a linear chrominance ramp or blend is run through this conversion and the resultant signal is truncated to 8 bits, the output would appear to have low frequency bands superimposed on the signal. This is one of the unavoidable artifacts inherent in digital signal processing. Quantization errors become a concern in the matrix conversion process or in the interpolation filters. These functions remap input signals to new quantization levels and sum the remapped numbers.

Inherently, quantizing an analog signal leads to errors since not all of the amplitude levels of the signal can be exactly represented in real digital processing systems with a finite number of bits. The theoretical analog signal with infinite Signal-to-Noise (SN) ratio has an infinite number of quantization levels. Quantization is by definition an approximation, and the approximation is in error by an amount inversely related to the number of bits used to represent the signal. This error manifests itself as a reduction in signal to noise ratio of the digital video signal.

The SN ratio is set to a level defined by the number of bits in the output signal. Typical adding hardware used in filters allows for 1/2 LSB rounding of a signal as it is processed. This will propagate error in the signal at each point of rounding. If the number of bits internal to a system is sufficiently high, then the error introduced by simple rounding will not affect the significant bits available at the output of a system.

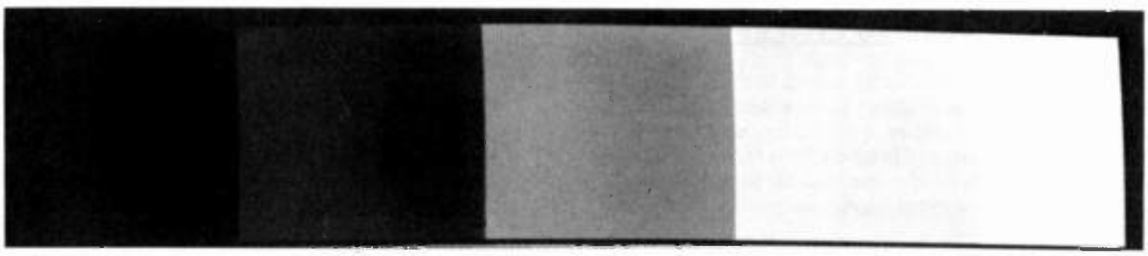

FIGURE 7 **BANDING ARTIFACTS** (severe truncation)

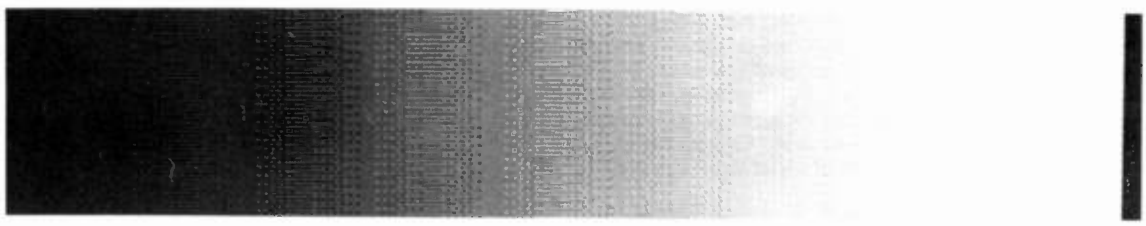

FIGURE 8 **DITHER APPLIED TO THE PREVIOUS SIGNAL AT TIME OF ROUNDING**

Adaptive rounding techniques make the effects of scaling and quantization less objectionable. Adaptive rounding implies that in a given system the rounding circuitry will introduce the appropriate amount of rounding and the appropriate spectrum of rounding based upon an analysis of signal spectral content. In limiting a signal to 8 bits, rounding of LSBs will manifest itself as an increase in noise. 1/2 LSB rounding produces low frequency noise, which is objectionable when mixed with an otherwise noise free digitally generated image. Adaptive rounding disperses the noise into higher frequencies by dithering the LSB of the resultant signal. It should not add noise on successive generations.

Dithering can also be applied to a system according to a psuedo random sequencer. The goal of dither in the rounding circuitry is to make the loss of SN ratio in a bit reduced system less offensive to the eye by changing the characteristic of the noise to higher frequencies. Ordered dither will accomplish this and the dither signal can be consistently and easily reproduced. If there is cooperation in establishing dithering algorithms then dither can be added and subtracted from a circuit depending upon the application. This will reduce visible error as well as prevent noise growth on successive generations through the digital format converter. If dither cannot be subtracted, then it should be applied only when necessary to mask distortions. The DST-300 implements this approach.

Increasing the number of bits of a given digital signal provides increased resolution and noise immunity. The digital signals discussed in this paper are currently specified at 8 bits for recording purposes. The interconnect allows for 10 bits to be sent between machines, however. The digital composite format takes advantage of that fact and uses the bits to increase the resolution of the blanking interval. Many devices operate internally with greater than 8 bits. The DST-300 is capable of operation in 8 or 10 bit environments.

THE AMPEX DST-300 CONCEPT

The DST-300 implements realistic solutions to the problems of digital signal format conversion. Design decisions were made that provide high-quality conversion without the burden of excessive circuitry. Flexibility has been included to allow the user to choose the method of encoding and decoding that best suits the job at hand.

The configuration of digitally interconnected equipment in a given studio may not necessarily remain static, but will have to adopt to changing needs as equipment is added in the future. The DST-300 has been designed with flexibility in this area also. The chassis is provided with space to insert circuitry to perform either one or two independent digital translations. Both can be in the same direction or one can be in each direction depending upon the needs of the user. In addition, a unique arrangement of input and output connectors on the rear panel allows alternate arrangements to be added in the future easily and at low cost. One possible example of this would be the inclusion of serial digital video connectors.

CONCLUSION

The conversion of digital video between the composite and component formats requires intensive digital signal processing. The processing is subject to some errors. Adaptive decoding and clean encoding minimize composite artifacts. Adjustable filters and blanking preserve the maximum possible active video content. Specialized rounding techniques make quantization errors less visible. The conversion can provide excellent results when properly implemented.

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S-VHS SIGNAL PROCESSING IN TIME BASE CORRECTORS

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Introduction

The arrival of the 1/2" S-VHS format was heralded by its developers as a significantly better alternative to the 3/4" U-Matic (Trademarked by Sony Corporation) format in numerous production applications. While it is not the purpose of this paper to delve into arguments that would attempt to prove or disprove the relative merits of each, there are major differences in the techniques required to process these signals in associated video equipment such as Time Base Correctors.

The purpose of this paper is to provide a background which traces the evolution of "color under" formats along with its impact on Time Base Corrector designs over the same time frame. Further, it will relate these evolutions to S-VHS, the newest member of this class of VTR signal processing.

The S-VHS Signals

The S-VHS signal in its most popular component form exists more precisely as two signals which represent the luminance and chrominance portions of a television picture. These signals are designated Y and C358, respectively. They are known together as Y/C358. The luminance part is a 1.0 Vp-p video signal with a defined bandwidth of 5 MHz. The chrominance signal is a bipolar signal on a 3.58 MHz carrier with a bandwidth of approximately 1 MHz. Its amplitude is approximately 800 mVp-p for 100% saturated color bars. It should be noted that these signals are neither "pure" component waveforms, such as those directly derived from RGB signals, nor composite video; they are in actuality, somewhere between the two.

The basic concept of S-VHS signals and their relative differences with respect to other formats on the same plane are best presented by first reviewing "color under" processing.

For a number of technical reasons, developers of the 3/4" U-Matic chose parameters that record the chrominance part of the signal spectrally at lower frequencies than luminance - hence the terminology "under". The spectral relationship is shown in Fig. 1. Typically, the color under subcarrier is either 688 kHz or 629 kHz. The luminance spectrum ranges from around 4 MHz at the bottom and 7 MHz at the top end, but these limits are dependent on the particular format. This technique requires that chrominance and luminance be separated from composite video before they can be recorded. Fig. 2 illustrates this point. On playback they become reunited after suitable and somewhat sophisticated color processing. More specifically, this color processing is a heterodyne recovery of the chrominance. This step cancels the phase and frequency deviations introduced by time base errors to provide a chrominance signal modulated onto a stable 3.58 MHz carrier. When added to luminance, which has not been effected by this process, the signal can be viewed on conventional monitoring equipment without chroma streaking and "rainbowing" that time base errors would otherwise create in the picture. Fig. 3 shows the spectral relationships for several of the most popular color under formats.

Referring back to Fig. 2, it was noted some years ago that separating the signal for the record process and then putting it back together in playback introduces additive distortion at each step. Further, multi-generation recording causes this to accumulate exponentially; at each pass driving the signal closer and closer to oblivian! When coupled with earlier conventional heterodyne TBC processing, which also had to separate Y and C to properly correct the errors and then put the signal together again, degradation was even further accelerated.

The first successful attempt to reduce the TBC's contribution to this escalating degradation was the concept of VTR subcarrier feedback shown in Fig. 4. This feedback signal to the VTR from the TBC eliminated the need to separate the signal into Y and C components within the TBC and it could treat the input composite signal as a full bandwidth representation of what had been originally recorded. The feedback signal is a 3.58 MHz subcarrier which is derived from the off-tape luminance and forces the chroma to follow its time base (and instantaneous frequency) errors. This signal replaces that of the 3.58 MHz oscillator shown in Fig. 2.

The next major step occurred in the VTR

where we were able to directly access the Y and C components in separated form for both the record and playback processes. This is shown in Fig. 5. The 3/4" U-Matic format was the first to do this and Y/C688 became popular as a VTR-to-VTR dub mode. It was frequently used as a input signal in more sophisticated TBC's to provide higher signal quality. The 688 kHz chrominance is "unprocessed" in this case; that is, the time base error effects which cause instantaneous frequency errors in these color signals are not cancelled as a result of the heterodyne processing described earlier. In the case of VHS, this carrier is 629 kHz as you will recall from Fig. 3.

Now, looking at S-VHS as shown in Fig. 6, we see that one other signal is produced. The chroma is modulated onto a 3.58 MHz subcarrier and designated as C358 to form one of the S-VHS signals I introduced at the start of this paper. The other, of course, is the Y luminance signal.

Aside from having reached another important milestone in the development of color under technology, with the introduction of S-VHS a significant signal system improvement in recording has extended the Y bandwidth to 5 MHz, thus providing higher picture resolution. Other advantages in maintaining separated Y and C processing are summarized in Fig. 7.

Cross-color and cross-luminance are spectrum related distortions, best controlled by keeping the Y and C spectra apart. The distortion shows up principally as transition dot crawl and edge fringing; highly saturated colors suffer the most from these effects. The next point has been previously described. Repeated decoding and encoding is detrimental to the signal, resulting in colorimetry losses, added ringing, loss of resolution, added artifacts and other effects which are produced as a by-product of accumulated multi-generation noise.

With respect to conversion to analog component signals, in the interest of maintaining the highest picture integrity, we believe it is important to be able to easily transcode from Y/C358 to three-wire analog component signals, such as Y/PB/PR. If the Y/C separation has already taken place, or better yet is not even necessary as in the case of Y/C generated camera signals, it is far better to remain in the Y/C format or transcode it to three-wire component for further processing. Certainly a Y/PB/PR format is a superior choice in maintaining lower introduced distortion compared to composite video for editing, effects, compositing of pictures or other such steps as would normally be encounted in most post-production operations today.

Hence, while Y/C358 falls into a space between three-wire parallel components and composite video, it can be straight-forwardly converted to either. As discussed, however, better overall picture performance is achieved if all post-production processing is done before conversion to composite video.

Processing In Time Base Correctors

Paralleling the evolution of video tape equipment in providing better pictures has been that of the TBC. Earlier TBC's, whose purpose was to correct 3/4" VTR errors, had to deal with heterodyne processed signals. This meant separation of Y and C and then special processing for chroma prior to time base correction. This processing is shown in Fig. 8. The heterodyne processor has to remove the chrominance from the stable 3.58 MHz carrier that had been introduced in the output stages of the VTR and remodulate it onto one that had been generated from the luminance (sync) signal. Clearly, we were undoing what had just been done in the VTR at the expense of the signal by adding more distortion. Such is the cost of advancing technology! As shown in the diagram, the Y and C signals were then added together and the resulting signal converted to digital form in the A/D converter. After digital processing, which removed the time base errors, the signal was converted back to analog form. Fortunately, as an extension of the heterodyne process, the 3.58 MHz feedback mechanism was discovered and used to help maintain higher picture quality. This scheme avoided the redundant processes in the VTR and TBC described above. This TBC block diagram is depicted in Fig. 9.

The S-VHS TBC is shown next in Fig. 10. The input and output interfaces support the Y/C358 signals that have been described above. Note, however, that a Y/C separator is included, followed by selector switches, so that archival 3/4" or 1/2" material can be mixed with S-VHS program material. Once converted to digital format, the time base correction process goes on as before and stabilized output signals are generated.

Fig. 11 shows more of the pertinent detail that constitutes the S-VHS TBC. One significant function in this diagram is the Y/C separator as noted above. Highest quality pictures are generated if the Y/C separation is accomplished using a comb filter design since this technique minimizes distortion and NTSC artifacts in the decoding step. Chrominance is processed as R-Y and B-Y color difference signals. This has been proven to be a wise choice over other component formats such as RGB or YIQ for a number of reasons. An important one relative to RGB relates to tolerance of timing and amplitude error differences. Y/R-Y/B-Y is more tolerant to channel path errors because luminance exists as a separate signal rather than being dependent on three signals and the influence of the differential path errors as is the case with RGB. An example is amplitude errors introduced into RGB channel paths. However small, they would tint the luminance gray scale and add distortion to the resulting picture. An argument in favor of Y/R-Y/B-Y over YIQ relates to the processing bandwidths as defined by each. Strictly speaking, the Q bandwidth must be restricted to 500 kHz to conform with the standards for YIQ signals. If this filtering occurs within the TBC or any post-production process, valuable resolution is lost and it is impossible to transcode to wider bandwidth Y/R-Y/B-Y or RGB signals again.

Another key function for the processing of S-VHS or color under signals in general is Y/C timing compensation control. By adjusting the delay of the luminance path relative to the chroma path, relative timing errors can be corrected.

Fig. 12 shows the output processing for the full S-VHS TBC. The wise choice of the Y/R-Y/B-Y signal structure is again apparent in that access to these signals as a transcoded output interface is direct. The use of these signals from an S-VHS TBC provides the highest degree of assurance that the signal quality going into the TBC will be preserved in further generation processing. This also allows other component signals to be mixed with S-VHS in component post-production systems. Note that this TBC system structure also provides composite video, which might be chosen for monitoring purposes or to interface to a 1" type C VTR; Y/C358, which may be chosen to dub to another S-VHS deck, as well as the Y/R-Y/B-Y signals mentioned above. This concept as a full S-VHS TBC exists as a product made by FOR-A Corporation of America, Model FA-300, Digital S-VHS TBC.

The Effects TBC

As a result of digital processing and full frame correction ranges in many current designs, the TBC offers the opportunity to incorporate picture effects capabilities. The most simple of these is freeze frame. Once in digital form and preferably component digital, other effects such as picture compression, mosaic and paint, pushes, cuts and dissolves can readily be achieved. Such an effects TBC is shown in Fig. 13. Here dual channel time base correction, each with a full frame of correction, can process two S-VHS or composite video signals from two VTR decks. Further, an effects memory with another frame of storage allows all of the effects mentioned above to be included. The concept is a product made by FOR-A Corporation of America, Model FA-740 Parallel Effects TBC.

Conclusion

The S-VHS TBC represents a parallel achievement in conjunction with the evolution of color under processing that has produced the S-VHS concept in analog recording. S-VHS TBC's must maintain the signal resolution as well as overall picture quality that this format can provide. Further, the TBC is in the best position in the system to create transcoded interfaces that allow the end-user to make the most flexible use of these signals. Whether simple S-VHS inputs and outputs or more sophisticated processing and picture effects, the TBC can do a great deal more than just time base correct the signal.

Steady improvements in 3/4" VTR technology have caused a dynamic growth in the use of this format in broadcast and industrial applications over the past several years. Whether we can expect as dramatic acceptance for S-VHS is not entirely clear at this point, however, I believe we can look forward to continued improvements and capabilities in both the S-VHS VTR and TBC in the future.

Acknowledgements

I wish to thank Mr. Paul Beck, Technical Facilities Manager, Emerson College, MA Communications Division, for his important contributions to this paper and to FOR-A Company Ltd., for their permission to present it.

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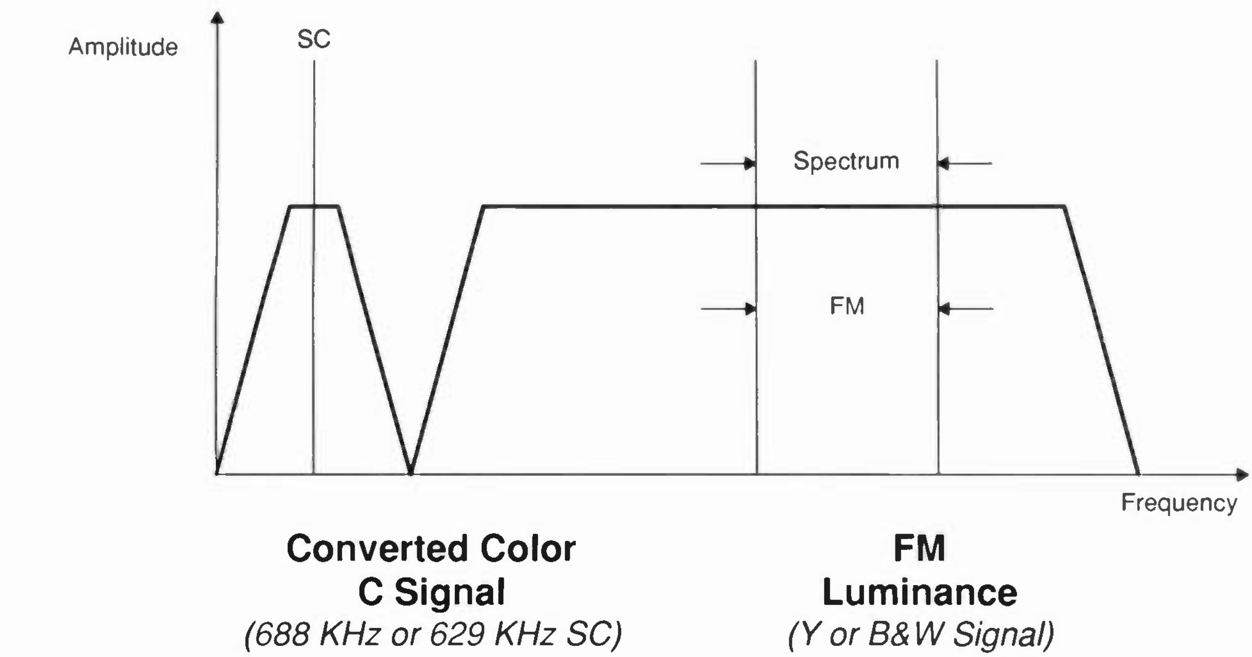


Fig. 1. The color under recording format records chrominance below the spectrum of FM luminance.

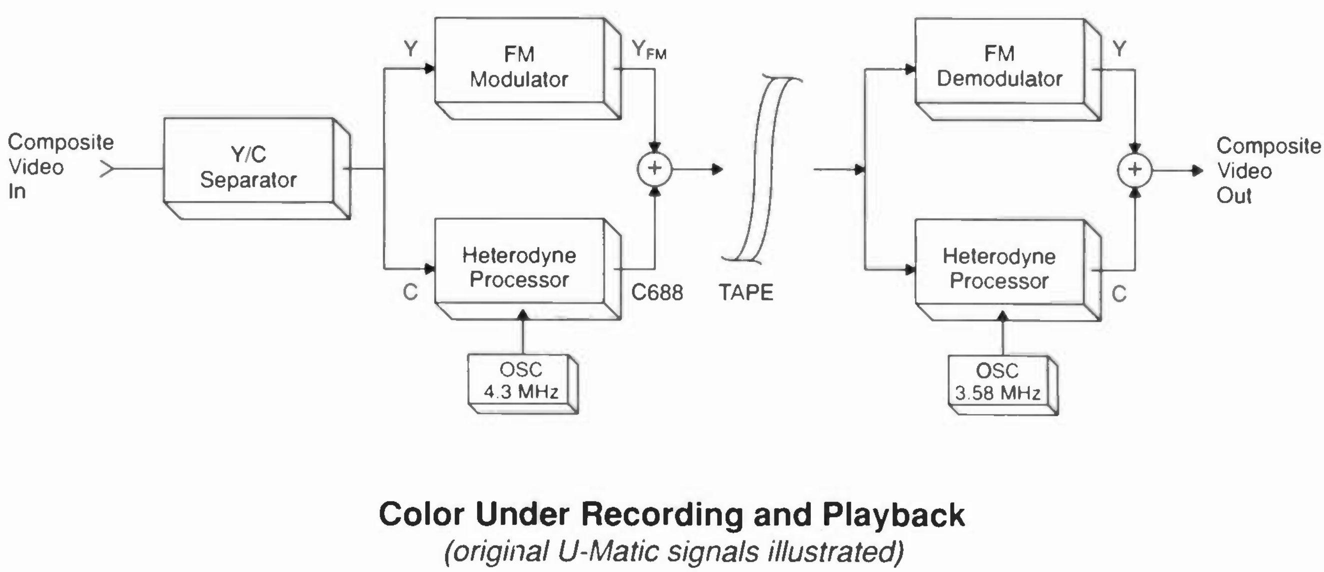
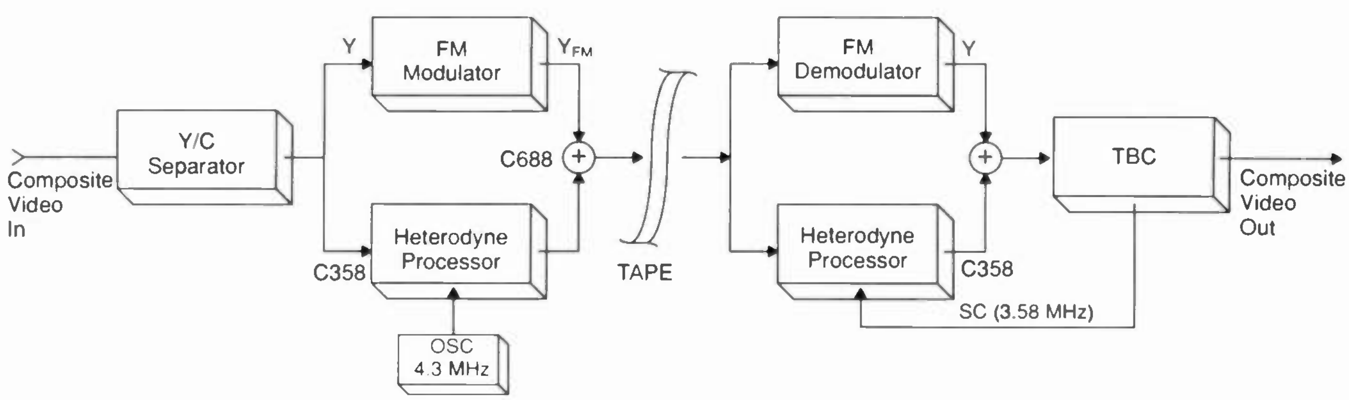


Fig. 2. Color under 3/4" VTR processing.

Format	Y		C	
	BW (MHz)	Spectrum (MHz)	BW (MHz)	Subcarrier (kHz)
3/4"	3.0	3.8 to 5.4	1.0	688
3/4" SP	4.2	5.0 to 6.6	1.0	688
VHS	3.0	3.4 to 4.4	<1.0	629
S-VHS	5.0	5.4 to 7.0	1.0	629

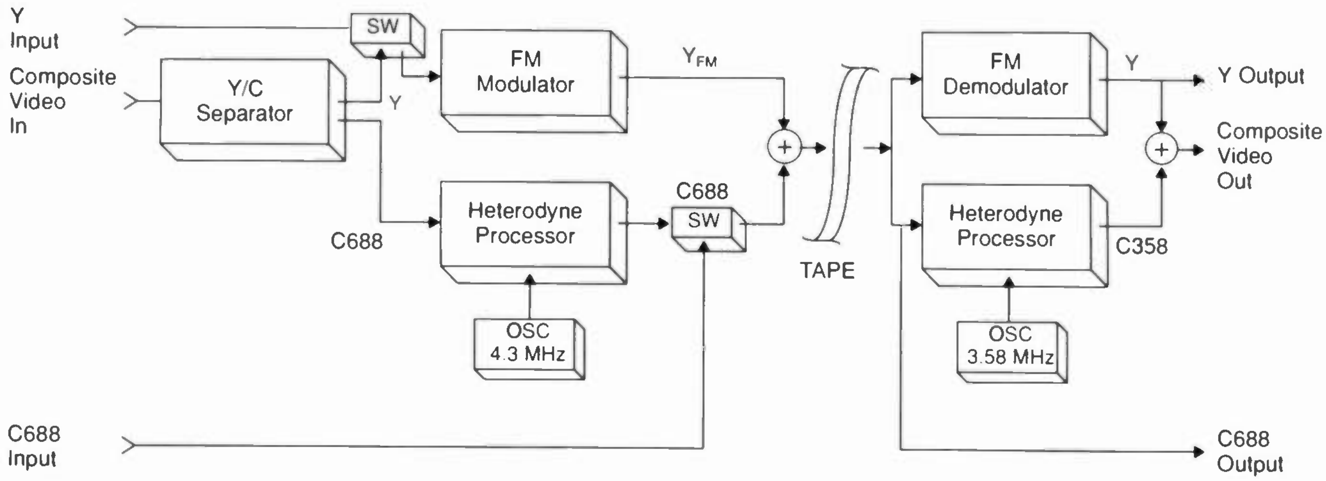
Comparison of Popular Color Under Formats

Fig. 3. Popular color under formats.



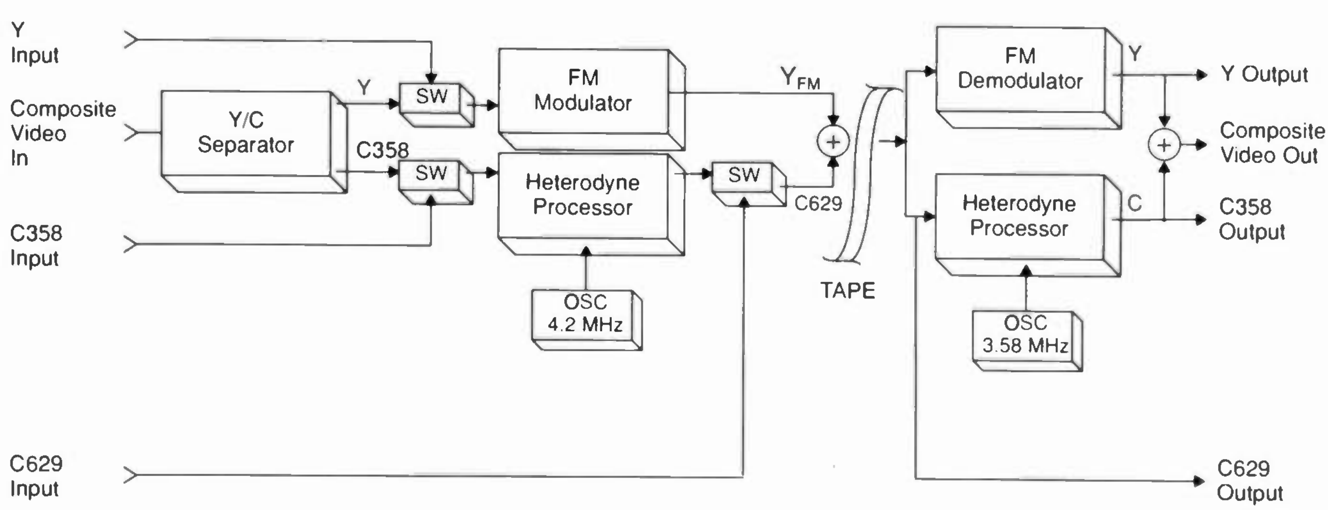
VTR Subcarrier Feedback System

Fig. 4. Color under processing
with VTR 3.58 MHz feedback.



Y/C Input and Output Interfaces

Fig. 5. Y/C688 input and output interfaces and processing in the VTR.



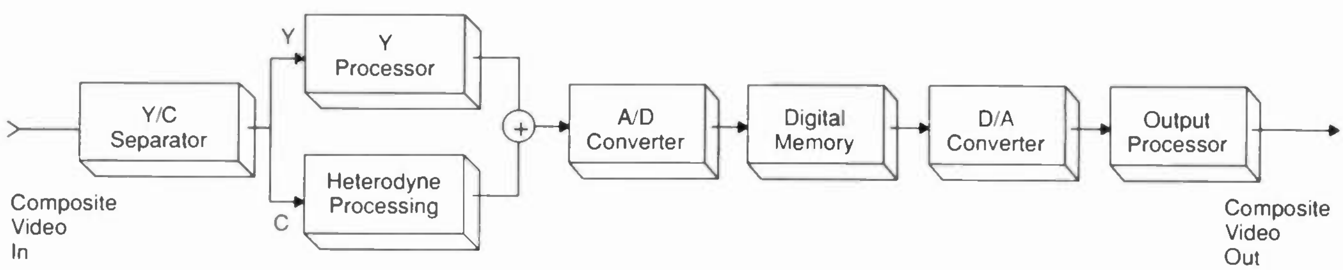
Y/C358 and Y/C688 Input and Output Interfaces

Fig. 6. Y/C358 and Y/C629 interfaces for S-VHS VTR's.

S-VHS Y/C Processing Advantages:

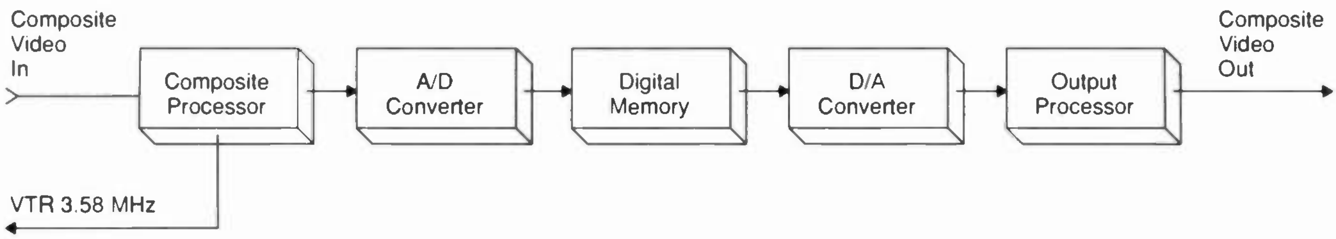
- Minimum cross-color effects
- Decode and encode degradation is eliminated
- Fewer NTSC processing artifacts
- Simple transcoding to 3-wire component signals

Fig. 7. Advantages of separated Y and C processing are shown. S-VHS is the most recent example.



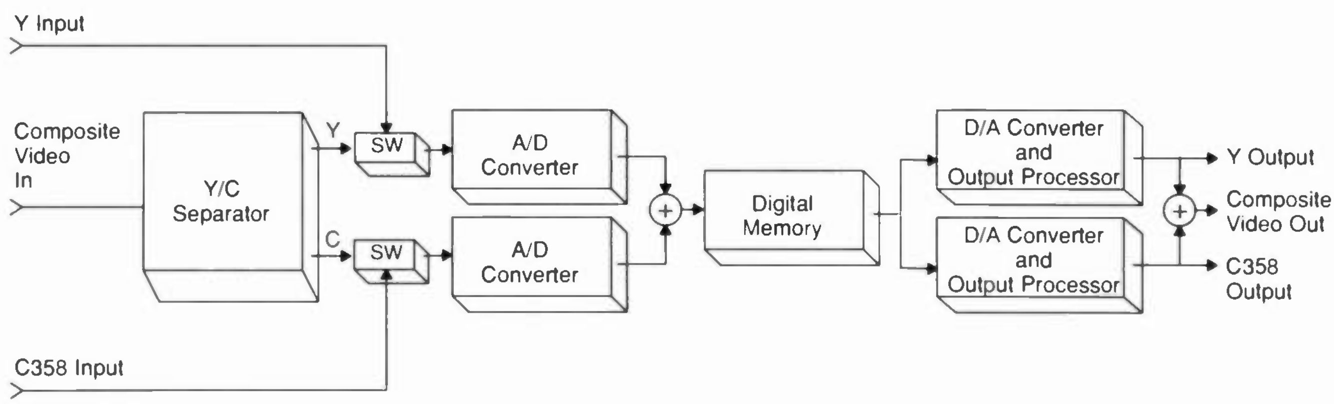
Heterodyne Process TBC

Fig. 8. The first heterodyne process TBC had to process chroma separately from luminance.



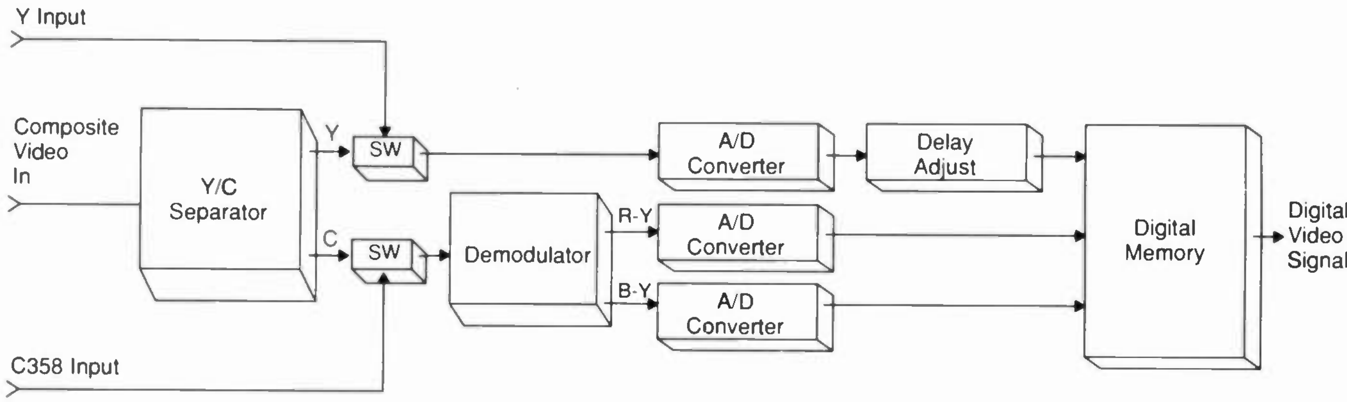
VTR 3.58 Feedback TBC

Fig. 9. The VTR 3.58 feedback TBC reduced added distortion.



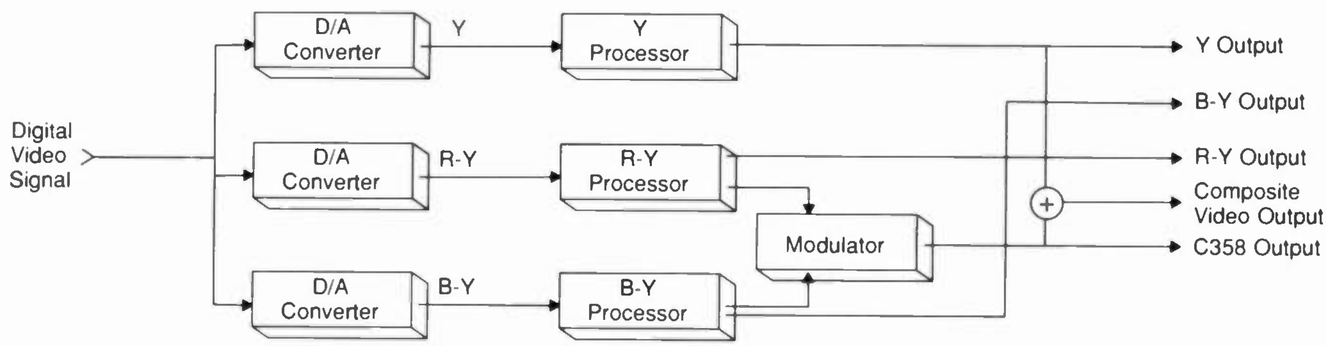
The Basic S-VHS TBC

Fig. 10. The basic S-VHS TBC interfaces Y and C358 signals.



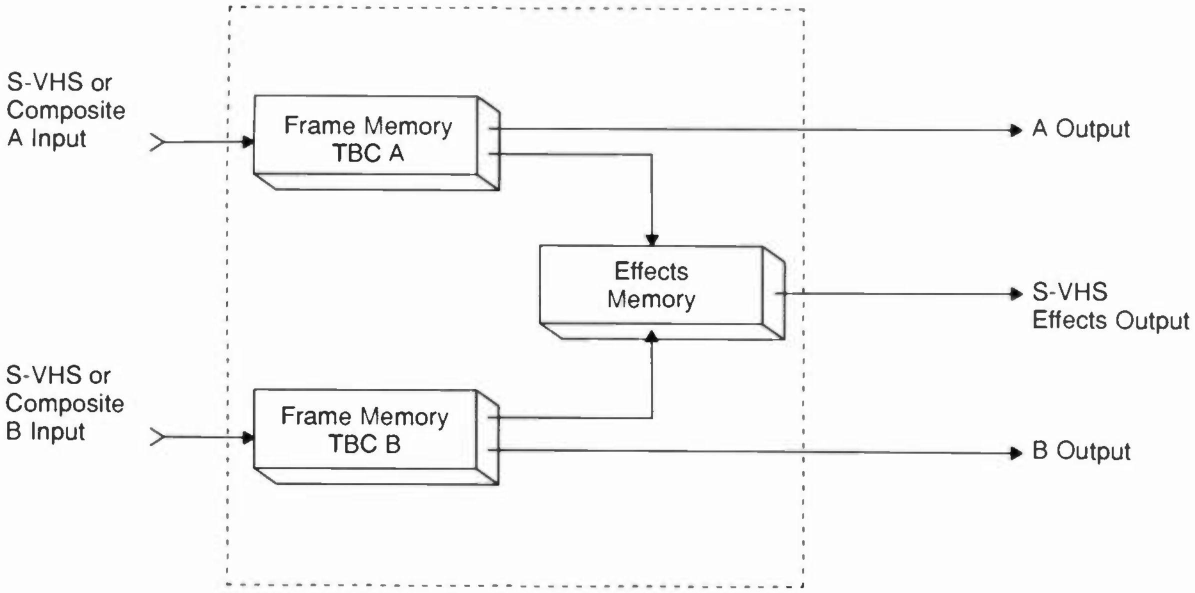
Full System S-VHS TBC Input Processing

Fig. 11. The full system S-VHS TBC input processing is shown.



Full System S-VHS TBC Output Processing

Fig. 12. The full system S-VHS TBC output processing is shown.



Dual Channel Effects S-VHS TBC

Fig. 13. Dual channel effects processing is shown with S-VHS interfaces.

INTERFACING THE TEKTRONIX VM700 VIDEO MEASUREMENT SET TO THE REAL WORLD

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Abstract

The Tektronix VM700 is destined to fundamentally change the way we make video measurements. It replaces the subjective interpretation of a video waveform or vector display with objective numerical results.

The VM700 employs a serial data format to connect to the outside world. Someday serial data inputs will be standard on broadcast remote controls; for now however, other provisions must be made for handling the vast amount of useful information the VM700 produces. Connecting the VM700 to a computer will satisfy even the most sophisticated need for processing the data gathered.

This paper deals with the specification and design of software and hardware to interact with the VM700. Flow charts and diagrams provide a graphic view of the interface to the VM700. Specific examples of the programming techniques used in the Modulation Sciences VMate will be provided. The VMate employs an embedded 80C88 processor to interface the VM700 to any conventional remote control. Samples of personal computer compatible code will be included.

Introduction

A quantum leap in video test and measurement, the Tektronix VM700 promises to fundamentally alter the way we characterize television signals. VM700 provides numerical values for many parameters of the video waveform and vector display that previously could only be estimated off a CRT. Such visual estimates are inherently subjective.

Each observer of the display often estimates a different value. And frequently, even the same technician viewing the waveform under different conditions will estimate a different value. With the VM700, the parameters of the video signal are available numerically, therefore unambiguously.

Once the parameters of a video signal are brought into the digital domain, many previously unimagined functions become easy to achieve. For example, placing high and low alarm and caution limits on all vital aspects of a video signal becomes simple. Measuring a video signal remotely WITHOUT having to transmit video over a "perfect" channel is no problem. Clear, bright, flicker-free waveform displays can now be created from the digitally processed video signal. Efectively, the VM700 has radically reduced the bandwidth need to remotely measure the parameters of a video signal.

The VM700 has extensive report generating and remote access facilities built into it. These functions are oriented around RS-232 serial data communications that connect the VM700 with a simple terminal and a variety of printers. These facilities demand that the user interact with VM700 on its terms - the reports are formatted by VM700, alarm and caution messages are created by VM700 and requests for information must be made by using VM700 mnemonics for video parameters. These restrictions pose no serious limitation to a maintenance technician or an engineering manager, but they do make difficult the direct use of VM700 data by a computer. Figure 4 is an example of a typical interaction with a VM700. The data in parenthesis are commands to the VM700. The plus sign is VM700's general acknowledgment of a command sent to it. Note that the report is formatted to make it easy for a person to read and understand it, but it's not so easy for another computer to interpret.

The data communications approach of the VM700 is useful and flexible if the broadcast plant is set up to deal with serial data communications -- and someday most plants will be. The reality today is that virtually all television transmitter facilities are built around conventional remote control systems, and they are based on contact closures and analog DC voltages. Thus, a method is needed to connect the VM700 to the transmitter plant. This paper will deal with interfacing a computer with the VM700.

A similar need for interfacing the VM700 exists in manufacturing applications. The VM700 is a nearly ideal tool for use in the production line testing and alignment of all types of video products. However, it lacks the facility for simple PASS/FAIL indication and for easy "alignment to a mark." Alignment to a mark is the process of "tweaking" some adjustment in a product until some specific value of some parameter of the video signal is achieved. Expecting someone working in a production line situation to read and interpret a video waveform monitor is asking a lot. Chances are that there will be a lot of mistakes.

The obvious solution to the needs of both broadcast monitoring and video equipment manufacturing is to connect a computer to the VM700. This external computer can then translate the VM700 data into contact closures and DC voltages for such applications as direct interfacing with a conventional remote control, or to have VM700 alarms operate warning lights for a video operator for broadcast applications. For manufacturing, PASS/FAIL indicators can be provided, as well as large analog panel meters with simple marks for the technician to adjust to.

Soon after VM700 was introduced, Modulation Sciences decided to design a product that would perform the interface functions described above. We named this new product "VMate." Vmate is a dedicated programmable controller that bridges the gap between the computer style output of the VM700 and the relays and analog voltages of a conventional remote control system. Figure 1 shows how the Vmate and a VM700 might be connected in a typical transmitter plant.

Hardware

The "brain" of the VMate is an 80C88 CPU based single board computer. This board includes most of the functionality needed to support the microprocessor. Built-in is a serial line interface, 16 bits of parallel interface, provision for onboard EPROM and memory that can be backed up by a battery.

The 80C88 CPU was chosen to provide software compatibility with the vast family of 8086 style personal computers. Being able to do software development on a personal computer whose CPU employed the same instruction set as the target machine made writing the program much easier and quicker.

In addition to the single board computer, an interface card was needed to connect the VMate with the real world. This includes the 16 relays that close for VM700 alarm conditions. A 17th relay indicates the status of the VMate. If for any reason the VMate becomes unavailable to provide data, this relay closes. Sixteen analog channels output DC voltages proportional to the numeric value of 16 parameters chosen from the VM700.

One consideration that became obvious during the early stages of design was that many stations do not have 16 unused metering (analog) channels on their remote control units. The solution was to provide a multiplexing function on VMate. Fifteen of the VM700 parameter measurements are combined so that they drive 5 remote control analog channels. The raise/lower function of the remote control selects one of three banks of five parameter measurements. With the raise/lower not operated, i.e. in the "neutral" position, parameters 1 thru 5 are available on VMate analog outputs 1 thru 5. With the remote control sending a "lower" command to VMate, those same 5 analog outputs now indicate parameter measurements 6 thru 10. With the remote control sending a "raise," parameters 11 thru 15 are available. The 16th parameter remains an orphan.

VMate also interfaces in the other direction - from the remote control to the VM700. This function allows the remote control operator to select which of the three video inputs (A, B or C) the VM700 is looking at. A DC output of VMate switches between 1, 2, and 3 volts to indicate the selection of video channel A, B, or C respectively. An external input allows a control voltage (a remote/local signal for example) to disable the VMate, thus preventing it from attempting to take control of the VM700.

Input and output connectors are industry standard "D" type. There are two 25 pin versions for RS-232 serial data communication and two 37 pin styles for connection to the remote control. One of the RS-232 channels connects the VMate with the VM700. For special applications a terminal or a personal computer emulating a terminal replaces the VM700. The second RS-232 channel allows direct access to the VM700's communication functions. The 37 pin connectors carry the relay closures and analog DC voltages to the remote control and the control signal inputs from the plant and remote control into the VM700.

Front panel controls place the VMate in program mode and reset it. Front panel indicators show the status of the 16 alarm relays, the VM700 video input selector, the analog signal multiplexer, the availability of the VMate and if it is in program mode.

SOFTWARE

Although hardware may be easier for many of us to relate to, software is the soul of a device like the VMate.

Several decisions needed to be made about software before any serious design of either the hardware or software could begin. The first was whether the VMate would include an operating system as part of its on-board software. An operating system would make the unit much easier to program, but greatly increase its cost. The greater cost would come from needing much more memory to contain and run an operating system and from the licensing fee that would have to be paid on each VMate sold. It was decided not to include an operating system in the VMate.

The next decision was what language to program the VMate in. There were many choices, including assembler, BASIC, FORTH, C, Pascal, Ada, Modula-2 and even FORTRAN. It quicky came down to C, Pascal, Ada or Modula-2 because we wanted a modern language that would lend itself to structured programming. The next requirement was for a language that had built-in concurrency. Concurrency is the ability of language to appear to have several activities going on at once without making the programmer do all the housekeeping. Concurrency is needed in working with the VM700 because several things can be taking place at the same time. For example, VMate could be updating information to the remote control at the same time a video signal parameter goes out of specification. Thus VMate would need to be doing two things at the same time.

Only Ada and Modula-2 provide concurrency as part of their language definitions. Conventional wisdom suggested that C be included in this list because it has become such a popular personal computer language. For example, the VM700 is programmed in C. However, C provides no concurrency in the definition of the language. Tektronix, in programming the VM700, got around that limitation by including an entire UNIX operating system in each VM700. The operating system supplied the needed concurrency. That was a reasonable choice for an instrument as complex as the VM700, but it would have been "overkill" in the VMate.

The choice between Ada and Modula-2 was a subjective one. Either language would have done the job well. Modula-2 was chosen because the programmers were experienced in Pascal. Modula-2, having been created by the same person who created Pascal, is very similar to Pascal. The compilers for Modula-2 were much less expensive than those for Ada. This is probably for two reasons; Ada is a more complex language than Modula-2, and most of the customers for Ada are aerospace-military contractors with big budgets. One special requirement for the compiler is that it produce code that could be put into EPROM and run without an operating system. The specific compiler chosen was Jensen Parners Inc. (JPI), Top Speed Modula-2. Although the support library for this compiler did not include stand-alone code and EPROM support by itself, source code was supplied for the library, making it possible for us to write the needed additions.

Talking to the VM700

Programming the VMate was challenging because of constraints created by the nature of the VM700. It must be pointed out that this discussion applies only to release 1.3 of the VM700 software. Later versions of the VM700 software may interface very differently.

Design goals:

* No programming of the VMate should be needed.
* The choices of what parameters of the video signal should be monitored and the value of the alarm trip points should be made by programming the VM700.
* Local operation of the VM700 should be possible with no thought of the VMate.
* All the remote control features of VM700 as provided by Tektronix should be available without interference by VMate.

All of the above goals of above were met in the production model of the VMate.

Interaction between VMate and VM700 is comprised of several major steps:

* Initialization
* Communications verification
* Control of VM700

* Data acquisition

The flowchart in Figure 3 shows the details of actual VMate/VM700 interaction.

Initialization takes place whenever:

* Vmate is powered up

* The RESET button is pressed

* Communication with the VM700 is restored after a break

The last item needs some additional discussion. Communications with the VM700 can break down because the connection between VMate and VM700 is broken, if the VM700 is turned off, or most commonly, because someone takes local control of the VM700. Taking local control of the VM700 requires no special action on the part of the user. It happens merely by pushing front panel function buttons. First CONFIGURE, then WAVEFORM, VECTOR or PICTURE as needed. As soon as any function button is pressed, VM700 will display the requested waveform or information. Vmate automatically disengages and the NOT READY indicator will light and the associated relay will close. After a user defined period, usually 5 minutes, VMate will attempt to take control of VM700 again. If control is regained, the VMate will initialize VM700 and return to normal monitoring. By the way, a hardware line on Vmate can be asserted to keep VMate in the NOT AVAILABLE mode and thus prevent it from attempting to regain control of VM700. This line might be operated by a master transmitter room remote/local switch.

In the initialization process, VMate first verifies the integrity of the communications channel, then attemps to read all VM700 files whose name begins with VMATE. The contents of these files tell VMate what data to request from the VM700. The VM700 files are read only once when the VMate initializes. After that, the VM700 mnemonics for the parameters or alarm status of interest are stored in VMate and used for actual data capture from the VM700.

Internally, the most important code module of the VMate is the parser. It extracts the measurement mnemonics from the formatted reports produced by the VM700. It is similar in structure to parsers found in programming language compilers. Once the parser has extracted the mnemonics of the video parameters selected by the user, those mnemonics are used by the VMate to request that parameter or alarm status from the VM700. The source code programming in Modula 2 for the one line parser is shown as Figure 4.

Diagnostics are an often ignored, but essential part of any software based system. Diagnostics are never truly appreciated until you try to get a system working that does not have good ones. The VMate's diagnostics divide up into two classes: those that run automatically each time the VMate is reset, and those that must be requested and used interactively by the operator.

The automatic diagnostic that executes whenever the VMate is reset causes all alarm relays to close for 2 seconds and all front panel LED's to light for 2 seconds. If, during the time that the automatic diagnostics are executing, the PROGRAM button is pressed, the VMate enters the user controlled diagnostic mode. In this mode the user can selectively assert any individual alarm channel relay as a steady closure or cause the relay to rapidly switch open and closed. In addition, any one analog channel can be individually forced to full scale. All of the function selections are made by repeatedly pushing the PROGRAM button. Pressing the RESET button causes VMate to leave this diagnostic mode. Figure 3, the program flow chart, shows exactly how the diagnostics fit into the operation of the system.

As an additional reliability feature, VMate includes a "watchdog" circuit. Included in the VMate software is a command to reset a hardware timer every 30 seconds. The timer is set to cycle out after 40 seconds, and so long as the reset commands are received every 30 seconds, the timer will never cycle through. However, should anything affect the proper operation of the program or the computer hardware, the timer will not be reset and the watchdog circuit will time out. This timeout will force the VMate into RESET mode.

Summary

The VM700 is a powerful and flexible tool for analyzing video signals. It is unique in its ability to reduce the parameters of a video signal to information that can be carried on a narrow band communications channel.

Often it is desirable to have an external computer further process the data gathered by VM700. By using the example of the Modulation Sciences VMate, we have shown the steps in interfacing the VM700 to a computer.

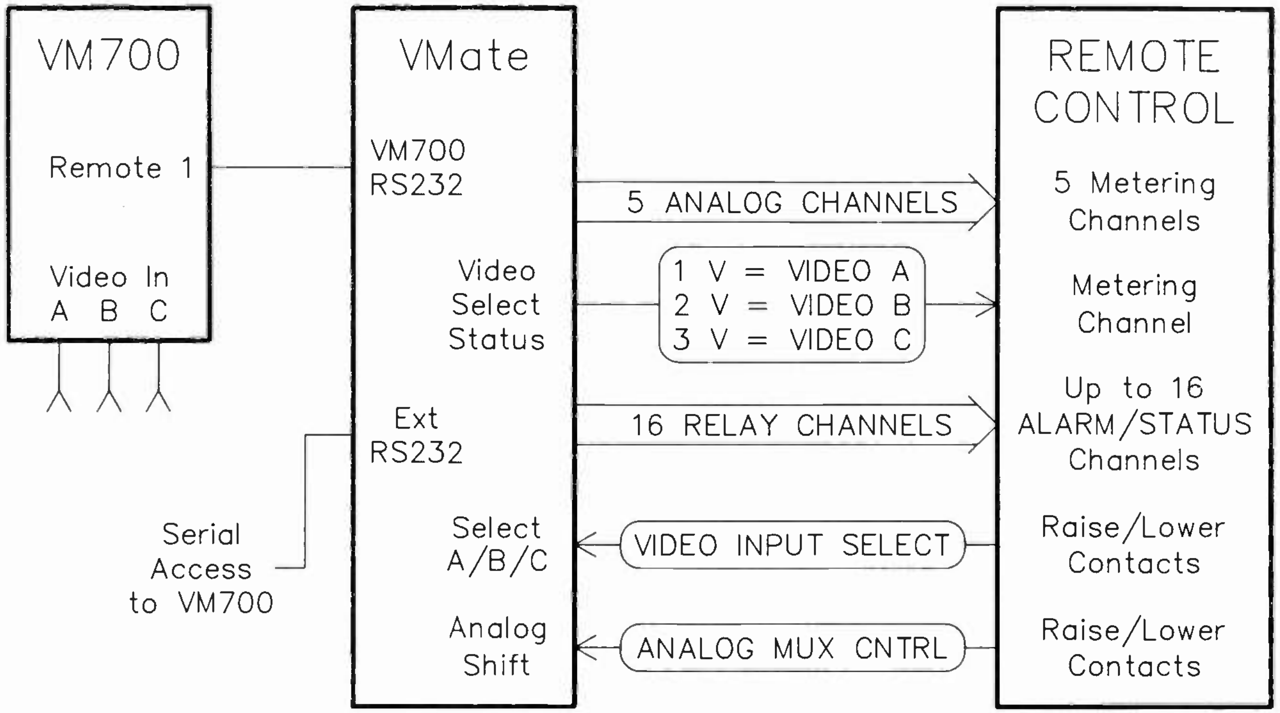

Figure 1 VM700 - VMate system interconnect diagram

```
PROCEDURE ParseLine( aLine : LineString ) : SHORTCARD;
(* Tries to match appropriate portions of the line with Descr and
   Units fields in the Parse Constants table.  If it finds a match,
   the corresponding Value and Status fields in the Parse table
   are updated, and the SHORTCARD returned is the parameter number.
   If no match is found, MAX(SHORTCARD) is returned.

   Color Bar parsing is a special case.  The color bar measurement
   produces 6 lines of data, each line with 3 data values.  ParseLine
   looks for the first of these lines, and sets BarCount to 0 if it
   finds it.  It then assumes that the current line and the 5
following lines contain Color Bar data.
*)
VAR   thisDescr : PCons.DescrString;
      thisUnits : PCons.UnitsString;
      value     : valueString;
      n : CARDINAL;
      match : BOOLEAN;

BEGIN
  ClearString( thisDescr );
  ClearString( thisUnits );
  Str.Slice( thisDescr, aLine, 0, PCons.DescrMaxI+1 );
  Str.Slice( thisUnits, aLine, 29, PCons.UnitsMaxI+1 );
  match := FALSE;
  IF BarCount > 17 THEN (* we're not doing Color Bars *)
    n := 18; (* don't look for Color Bars *)
    LOOP (* look for a match with any other measurement *)
      WITH PCons.Table[ n ] DO
        IF ( Str.Compare( thisDescr, Descr ) = 0 )
        AND ( Str.Compare( thisUnits, Units ) = 0 ) THEN
          match := TRUE;
          EXIT;
        END;
      END;
      INC( n );
      IF n > PCons.TableMaxI THEN EXIT END;
    END;
    IF match THEN
      Str.Slice( value, aLine, 21, valueMaxI + 1 );
      IF ValConv( value, PVars.Table[ n ].Value ) THEN
        (* valid measurement, check for alarm and caution *)
        IF Str.Pos( aLine, '**' ) < MAX(CARDINAL) THEN
          PVars.Table[ n ].Status := Alarm;
        ELSIF Str.Pos( aLine, '* ' ) < MAX(CARDINAL) THEN
          PVars.Table[ n ].Status := Caution;
        ELSE
          PVars.Table[ n ].Status := OK;
        END;
      ELSE (* not a valid measurement *)
        PVars.Table[ n ].Status := NG;
      END;
      RETURN SHORTCARD( n );
    ELSE (* see if it's Color Bars *)
      IF Str.Pos( aLine, 'Yellow' ) = 3 THEN (* it is Bars *)
        BarCount := 0;
      ELSE (* ignore it *)
        RETURN MAX(SHORTCARD);
      END;
    END;
  END; (* of non-Color-Bar parsing *)
```

Figure 2a   First half of ParseLine procedure from VMate program

```
(* The first line of Color Bar data will be detected by the
   preceding code.  If it found the first line of Bars, BarCount
   will be set to 0. Note that entries number 0, 1, and 2 in
   Pcons.Table (and therefore in PVars.Table) correspond to the
   first line (Yellow) of Color Bar data.  Entries number
   0 thru 17 in PCons.Table must correspond to the 18 pieces
   (6 lines * 3) of Color Bar data. BarCount is left set to 18 to
   signify that Color Bar processing is complete.
*)
IF BarCount < 18 THEN (* we are doing Color Bars *)
  Str.Slice( thisDescr, aLine, 3,  7 );
  TrimTrailingSpaces( thisDescr );
  IF Str.Pos( PCons.Table[ BarCount ].Descr, thisDescr )
    < MAX(CARDINAL) THEN
    (* we're in the right place, process this line *)
    n := BarCount;
    LOOP
      Str.Slice( value, aLine, 14, valueMaxI + 1 );
      IF ValConv( value, PVars.Table[ BarCount ].Value ) THEN
        (* valid measurement, check for alarm and caution *)
        IF Str.Pos( aLine, '**' ) < 31 THEN
          PVars.Table[ BarCount ].Status := Alarm;
        ELSIF Str.Pos( aLine, '* ' ) < 31 THEN
          PVars.Table[ BarCount ].Status := Caution;
        ELSE
          PVars.Table[ BarCount ].Status := OK;
        END;
      ELSE (* not a valid measurement *)
        PVars.Table[ BarCount ].Status := NG;
      END;
      INC( BarCount );
      IF ( BarCount - n ) > 2 THEN EXIT END;
      Str.Delete( aLine, 14, 20 ); (* delete processed data *)
    END;
    RETURN SHORTCARD( n );
    (* Note that the value returned marks the first of 3 entries
       in PVars.Table *)
  ELSE (* something's gone wrong, abort processing Bars *)
    BarCount := 18;
    RETURN MAX(SHORTCARD);
  END;
END;
END ParseLine;
```

Figure 2b   Second half of ParseLine procedure from VMate program

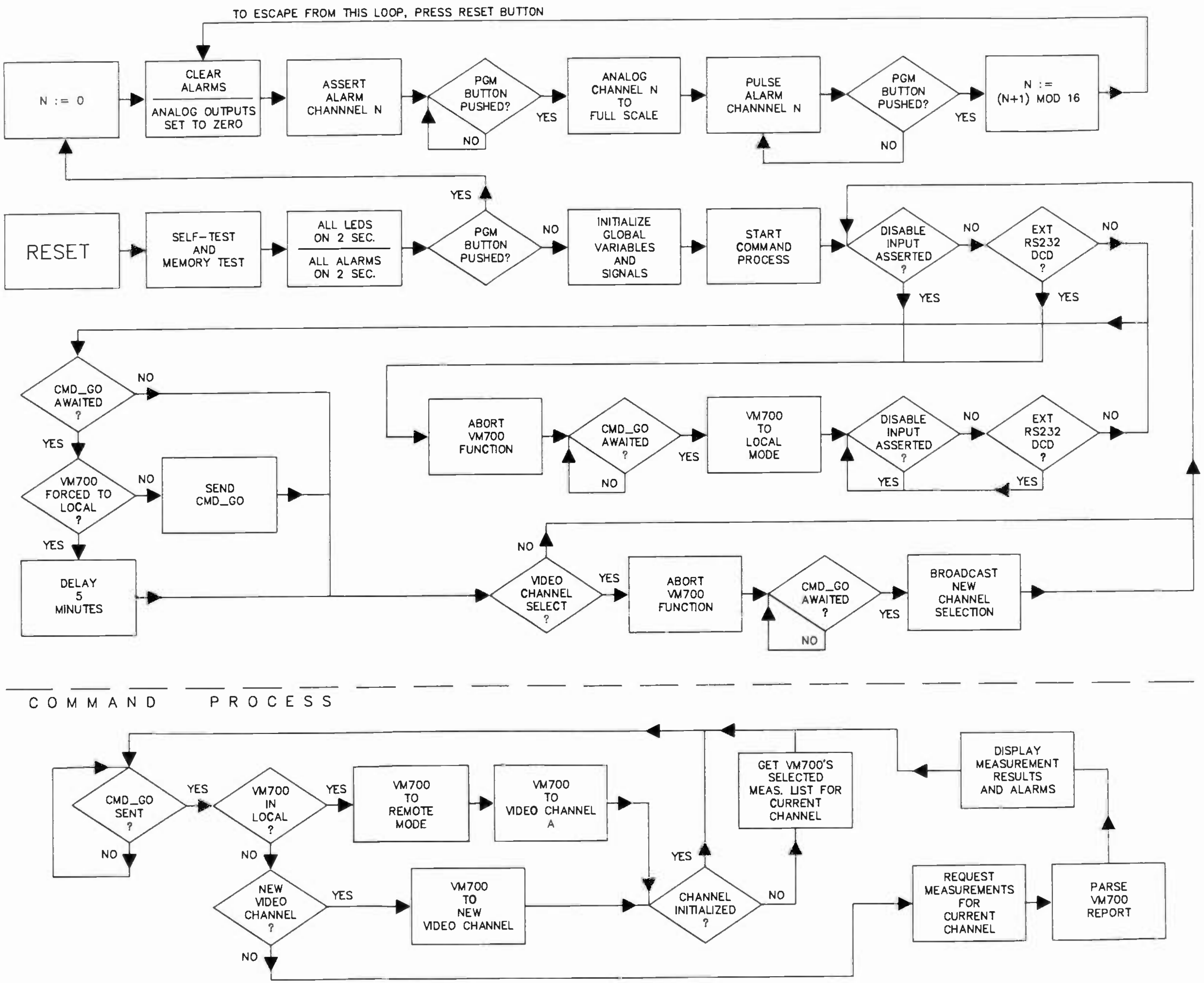

Figure 3 Simplified flow chart for VMate program

```
(rmt)
+
(gsm,A)
+Sparse^M
(ssm,A,System~Default)
+
(rep,A)
+
^L
Page 1

                         VM700 Video Measurement Set
      Channel A  Transmitter Input                      15-Sep-88    00:37:11

                                                  Violated Limits
                                                    Lower  Upper

Bar Top                ------  % Carr     **     10.0    15.0  Bar Not Found
Blanking Level         ------  % Carr     **     72.5    77.5  ZC Pulse
Unselected
Bar Amplitude          ------  IRE        **     96.0   104.0  Bar Not Found
Sync Amplitude           36.2  IRE        *      37.0    43.0  100 IRE = 714 mV
Blanking Variation     ------  % Carr                          ZC Pulse
Unselected
Blanking Variation        0.7  IRE                             100 IRE = 714 mV
Sync Variation         ------  % Carr     **      0.0     5.0  ZC Pulse
Unselected
Sync Variation            0.7  IRE                             100 IRE = 714 mV
Burst Amplitude         101.6  % Sync
Burst Amplitude          36.8  IRE        *      37.0    43.0  100 IRE = 714 mV

FCC H Blanking          11.02  us
FCC Sync Width           4.71  us
FCC Sync-Setup           9.43  us
FCC Front Porch          1.59  us
Sync to Burst End        7.57  us
Breezeway Width          0.79  us
FCC Burst Width          7.4   Cycles     **      8.0    11.0
Sync Risetime          120     ns
Sync Falltime          120     ns
RS-170A H Blanking      17.53  us         **     10.65   11.15
RS-170A Sync Width       4.59  us         *       4.61    4.79
RS-170A Sync-Setup       9.36  us
RS-170A Front Porch      1.65  us         **      1.38    1.62
Sync to Burst Start      5.45  us         **      5.18    5.42
RS-170A Burst Width      7.5   Cycles

V Blank 4 IRE F1        21.0   Lines      **     18.0    21.0
V Blank 4 IRE F2        21.0   Lines      *      18.5    20.5
V Blank 20 IRE F1       21.2   Lines      **     19.9    21.1
V Blank 20 IRE F2       21.0   Lines      *      20.1    20.9
FCC Equalizer           48.1   % S.W.
FCC Serration            4.83  us
RS-170A Equalizer        2.15  us         **      2.18    2.42
RS-170A Serration        4.71  us

S/N Unweighted          71.1   dB                              RMS (Ref 714 mV)
S/N Lum-Weighted        73.1   dB                              RMS (Ref 714 mV)
S/N Periodic            50.9   dB         **     57.0  ------  RMS (Ref 714 mV)
```

Figure 4   Typical dialogue with VM700 to produce report

RECENT DEVELOPMENTS IN SOLID STATE TV TRANSMITTERS

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Larcan Communications Equipment Inc.
Toronto, Canada

INTRODUCTION

Low power solid-state TV transmitters up to 1 kW have been in worldwide use for many years and have ably demonstrated their improved reliability and superior performance over tube type transmitters.

This paper deals with high power solid-state TV transmitters up to 60 kW. It will discuss the evolution of solid-state designs, explain the design criteria for high reliability, discuss typical performance and show test results.

This paper will also discuss amplifier designs and methods of combining several amplifiers for a single high power output.

BACKGROUND

In 1980 LARCAN introduced its first solid-state broadband amplifiers. These were rated at 1500 W sync peak and were used as drivers for high power tube transmitters. These amplifiers employed bipolar transistors and consisted of eight modules combined for a single output.

Phase 2 of LARCAN's solid-state development concerned the design and manufacture of amplifiers with powers up to 3 kW. These amplifiers also employed bipolar devices.

In 1986 LARCAN initiated Phase 3, a program to design, develop and produce all solid-state TV transmitters with single ended powers of 30 kW and combined powers of 60 kW. These transmitters, designated Model M, were introduced at the 1988 NAB convention. At the time this paper is being written, late February 1989, 33 have been ordered, of which 18 are on-air. These are in addition to solid-state transmitters provided by other manufacturers such as NEC and the Harris Corporation. A LARCAN 22 kW transmitter is shown in Figure 1 above.

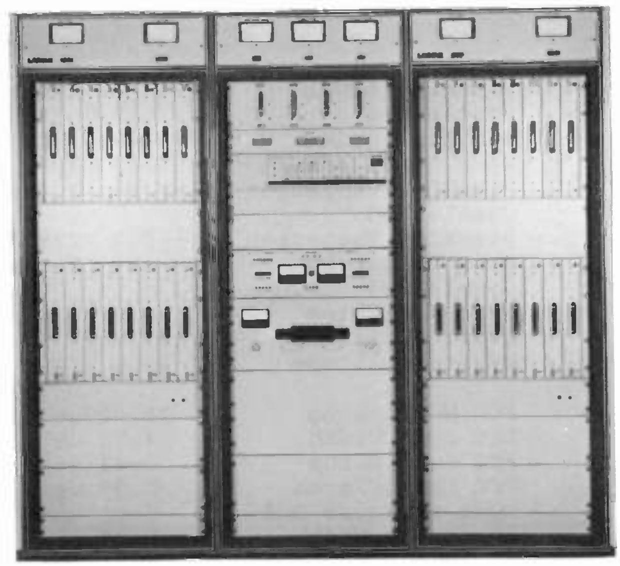


FIGURE 1

Initially when the decision was made to design and manufacture solid-state amplifiers LARCAN engineers investigated the use of FETS rather than bipolars. After several prototype FET amplifiers were built and tested we decided to stay with bipolars because we found the FETS to be more non-linear than bipolars and prone to oscillate.

When we initiated the Model M design program, we again looked at FETS and now found that FET technology had improved significantly. All Model M transmitters use FETS exclusively, as do the solid-state transmitters of other manufacturers.

The advantages of FETS over bipolars are:

1. They are more tolerant of poor VSWR conditions.
2. More power is available from each device thus fewer devices are required for a given output power.
3. They operate at 50 V rather than 28 V and thus draw less current.

4. Much simpler biasing circuitry requiring fewer components.
5. They are easier to broadband.
6. They have higher gain and require fewer stages of amplification.

TRANSMITTER BLOCK DIAGRAM

Figure 2 shows a block diagram of a 22 kW solid-state transmitter. The output stage of the transmitter shown comprises twenty-four 1 kW amplifiers combined for 22 kW. The isolation between the input ports on the combiner is such that the failure of one or more amplifiers affects neither the performance nor reliability of the other amplifiers.

DESIGN CONSIDERATIONS

There are five key design criteria to be considered in the design of solid-state transmitters and these, in order of importance, are:

1. Reliability
2. Performance
3. Serviceability or ease of maintenance
4. Features or operator interface
5. Appearance

Reliability

The reliability of a tube transmitter, with a single output tube and cavity and four separate power supplies, depends on the reliability of each and every one of these units. A solid-state transmitter, by design, has no such dependancy since it has redundant amplifiers, power supplies, blowers and control circuitry.

For optimum reliability, solid-state amplifiers operating class AB must be provided with VSWR protection. Some TV transmitter manufacturers use circulators at the output of each module for VSWR protection, while circulators do provide protection they have the disadvantage of being costly and bulky with a high insertion loss. In this transmitter each output amplifier is equipped with electronic VSWR protection. In the event of mismatch occurring in the output circuitry the resultant reflected power will be detected and processed to reduce the amplifier output to a safe operating level.

In addition to the VSWR protection provided for each amplifier there is also overall protection for the complete transmitter.

Device cooling is also a key consideration when designing solid-state transmitters for high reliability. Each of the two amplifier cabinets shown in Figure 1 is equipped with a blower rated at 2500 cfm. The formula for heat rise across the cabinet is

$$\Delta c = \frac{1760 \times Pd}{cfm}$$

where Pd = power dissipated
1760 is a constant
Δc = heat rise in degrees Celsius

The calculated heat rise, during the design stage, based on 2500 cfm and 11 kW dissipated at blanking level modulation

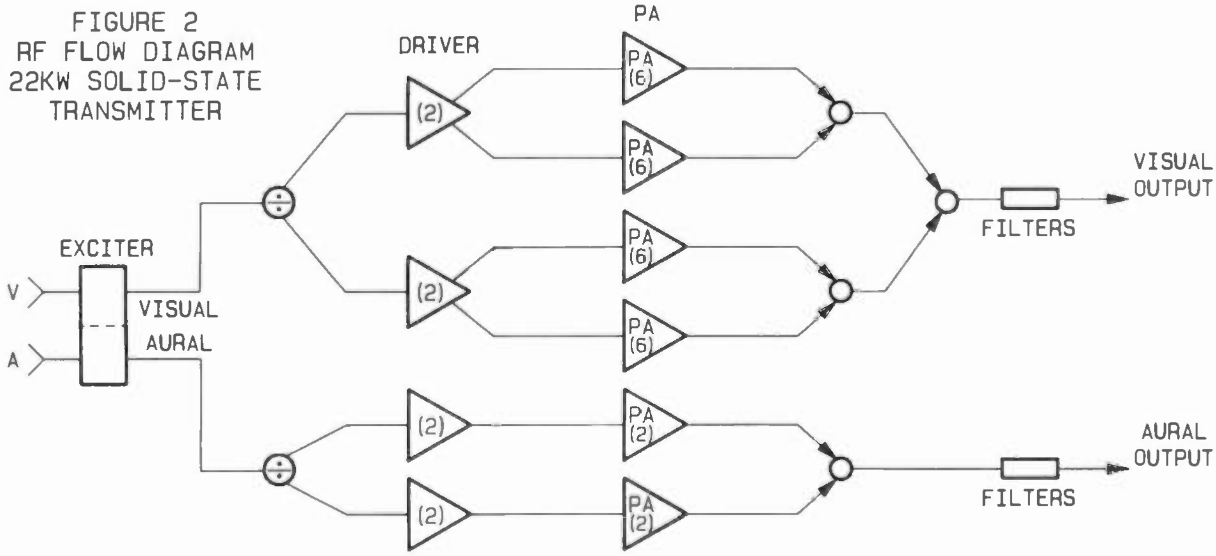


FIGURE 2
RF FLOW DIAGRAM
22KW SOLID-STATE TRANSMITTER

and 1200 W aural was 7.7°C, the actual measured heat rise was 8°C.

The flange temperature of the FETS in the top amplifiers when the ambient temperature is 30°C will be only 50°C. The maximum temperature for high reliability is 75°C.

Other areas where reliability must be designed into the product are:

a) Over-voltage protection
b) Over-current protection
c) Over-temperature protection

Performance

The performance of a solid-state transmitter is equal to a tube transmitter for virtually all parameters and superior for most. Figures 3A to 7A are oscilloscope photographs showing the performance to be expected from a solid-state transmitter. Figures 3B to 7B show the same parameters but with half of the transmitter shut off. No adjustments were made to the transmitter between taking the A and B photographs.

Solid-state transmitters maintain their performance over extended periods of time due primarily to a) their broadband design, and b) they have no tubes to lose emission or cavities to become mistuned.

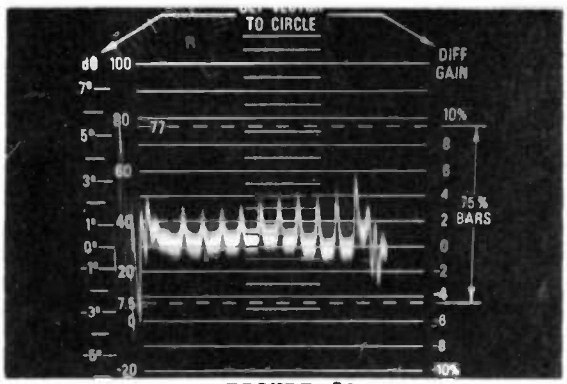


FIGURE 3A
DIFFERENTIAL PHASE

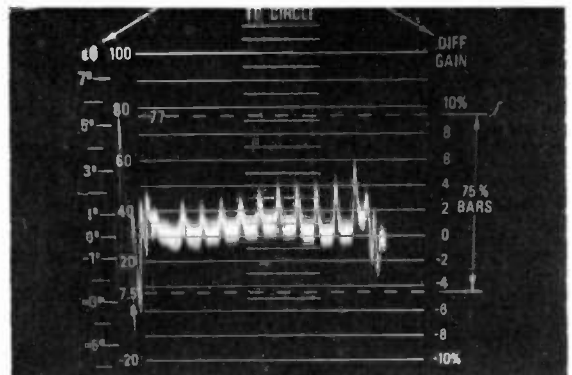


FIGURE 3B
DIFFERENTIAL PHASE

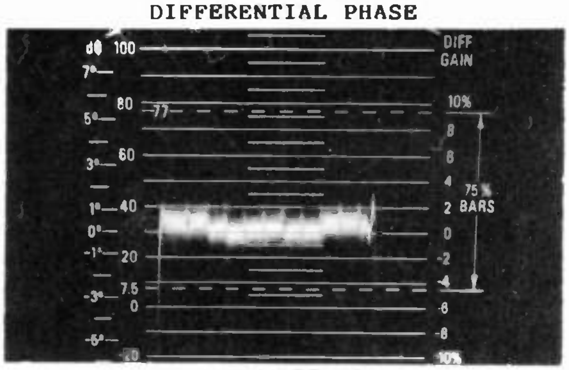


FIGURE 4A
DIFFERENTIAL GAIN

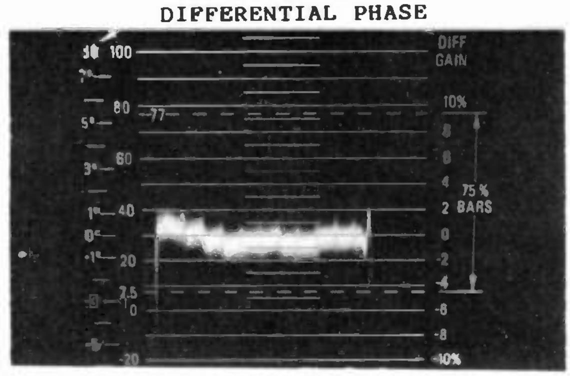


FIGURE 4B
DIFFERENTIAL GAIN

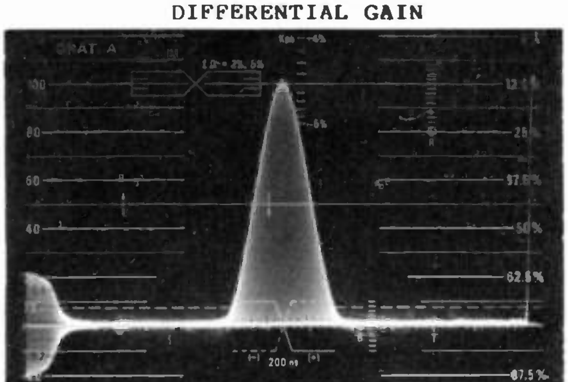


FIGURE 5A
12.5T PULSE

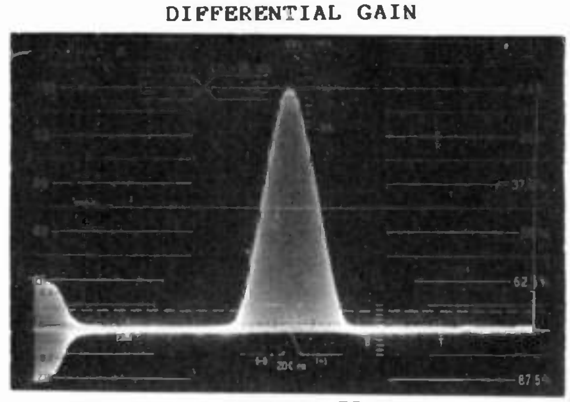


FIGURE 5B
12.5T PULSE

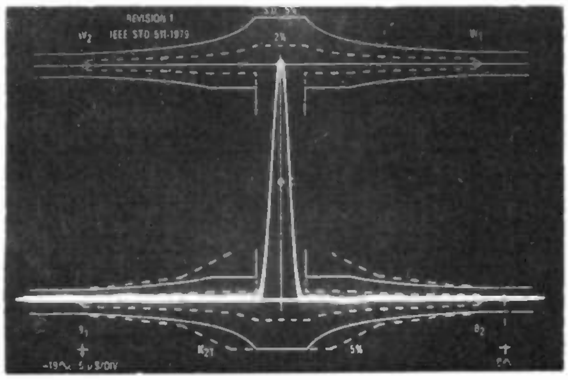


FIGURE 6A
2T PULSE

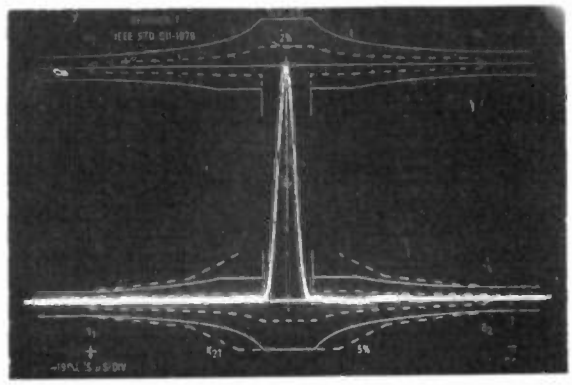


FIGURE 6B
2T PULSE

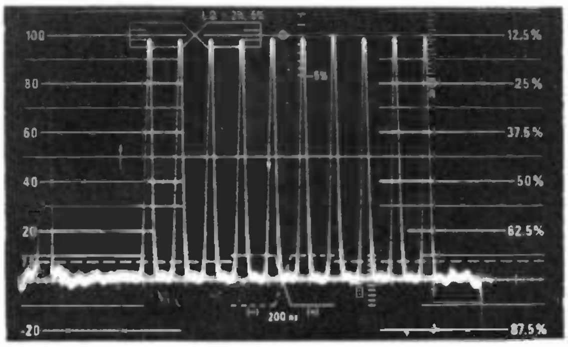


FIGURE 7A
LF LINEARITY

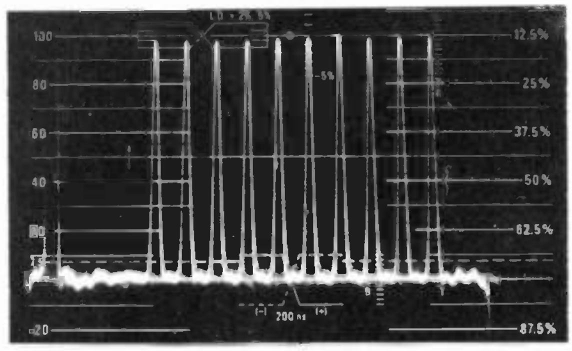


FIGURE 7B
LF LINEARITY

COMBINING CONSIDERATIONS

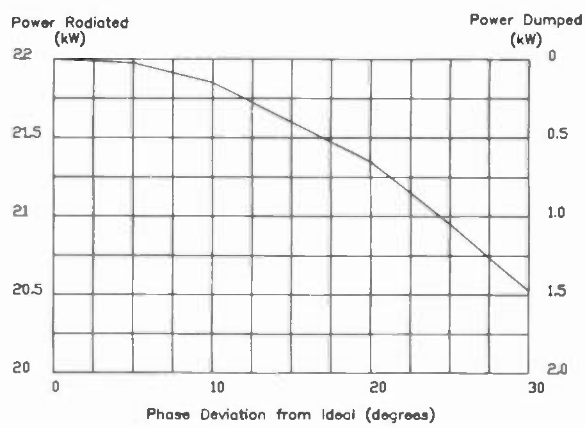
A question often asked regarding transmitters with parallel outputs is, what effect does an amplitude or phase imbalance have on output power?

Figure 8 shows the relationship between the power radiated and the power dumped in the combiner reject load when the two halves of a 22 kW transmitter are equal in power but not properly phased. Figure 9 shows the relationship when the two halves are properly phased but unequal in power. As these figures show there is relative insensitivity to the slight imbalance of power if one or two modules in one half of the transmitter were to fail. Because there is no tuning in the transmitter the phase relationship once set, never changes. The formula for deriving the graphs for Figures 8 and 9 is as follows:

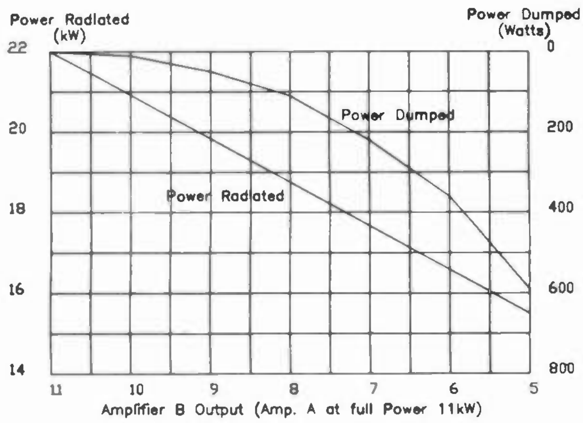
$$PY = \frac{P1 + P2}{2} + \sqrt{P1P2}\cos\theta$$

$$Pd = \frac{P1 + P2}{2} - \sqrt{P1P2}\cos\theta$$

where PY = power radiated
Pd = power dumped
P1 = power amplifier 1
P2 = power amplifier 2
θ = phase angle



POWER RADIATED AND POWER DUMPED
WHEN TWO HALVES OF A 22 kW TRANSMITTER
ARE EQUAL IN POWER BUT IMPROPERLY PHASED
FIGURE 8



POWER RADIATED AND POWER DUMPED
WHEN TWO HALVES OF A 22 kW TRANSMITTER
ARE PROPERLY PHASED BUT UNEQUAL IN POWER

FIGURE 9

There are several stages of combining as shown in Figure 2. In LARCAN transmitters all combining, except the final output, is done with stripline combiners.

The advantage of this method of combining is that it is non-adjustable, requires no maintenance, greatly reduces the need for coaxial connectors and ensures that the powers being combined are always properly phased. Figure 10 shows the method of combining two visual 6 kW blocks for a 12 kW output. Figure 11 shows the 6-way combiner used in the LARCAN 6, 12 and 22 kW transmitters.

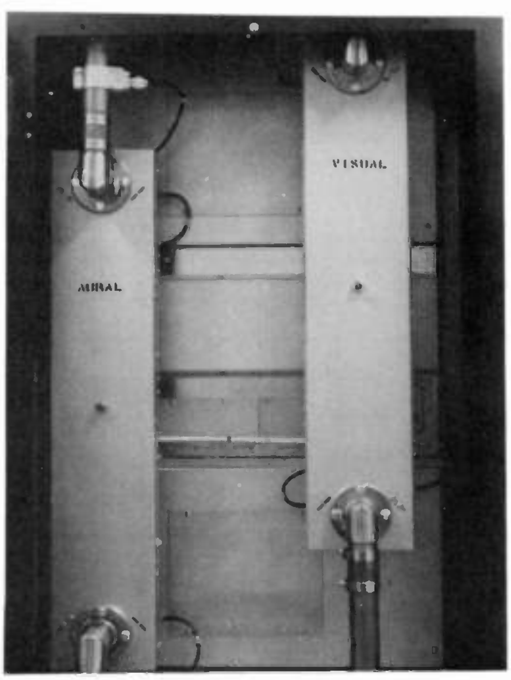


FIGURE 10
COMBINING TWO 6 kW BLOCKS
FOR 12 kW OUTPUT

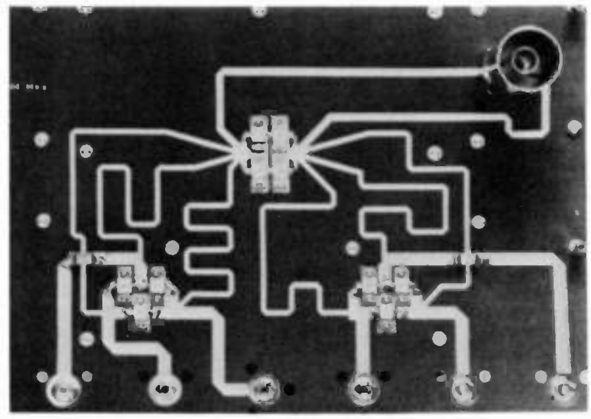


FIGURE 11
6-WAY POWER COMBINER

MODULAR CONSTRUCTION

Because of the limitation in the power handling capability of solid-state devices a large number are needed to achieve high power outputs.

This necessity becomes a virtue in that it results in redundancy and dictates modular construction. Figure 12 shows a 1 kW amplifier module. This module is used in all transmitters up to 22 kW. A similar module, but rated at 1500 W is used in the 30 kW transmitter.

The module shown employs four FET devices and makes extensive use of microstrip technology. The gain of this module is between 18 and 21 dB depending on the band. The bandwidth is 50 MHz on Band III.

The component count is extremely small resulting in improved reliability. The fuses are to protect the harness and amplifier boards in the event of a shorted FET. They also provide amplifier isolation when troubleshooting.

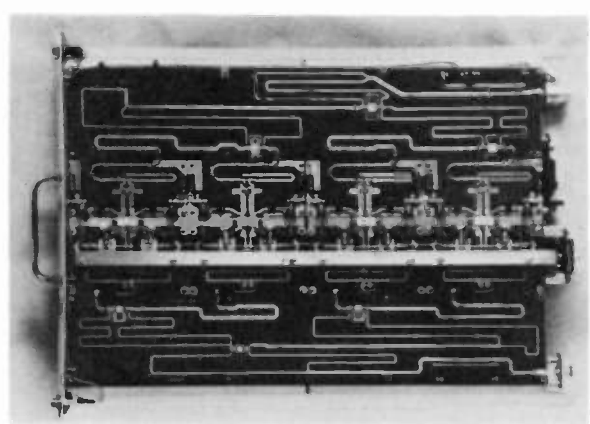


FIGURE 12
1 kW AMPLIFIER MODULE

The module shown in Figure 12 can be removed from its housing and reinserted while the transmitter is on-air without the need to turn off the transmitter or the module.

CONCLUSION

On-air experience has proven solid-state transmitters to be reliable, stable and to give excellent performance. Most major transmitter manufacturers are committed to solid-state technology. The advantages of solid-state are:

1. **IMPROVED RELIABILITY**

2. **LOWER OPERATING COSTS**
   There are some cost savings resulting from lower power consumption but the major savings are in tube replacement costs.

3. **LOWER MAINTENANCE COSTS**
   Maintenance costs are lower because most maintenance can be performed during daytime, but more importantly, can be performed by one technician.

4. **SIMPLIFIED CONTROL CIRCUITRY**

5. **NO EXTRA HEADROOM REQUIRED**
   Traditionally chief engineers specify transmitters with higher power ratings than actually needed. This is because tube transmitters are often mistuned and of course tubes lose emission. It is no longer necessary to pay for extra headroom because solid-state transmitters cannot be mistuned and the FETS do not lose emission.

THREE TUBE SWITCHLESS COMBINER

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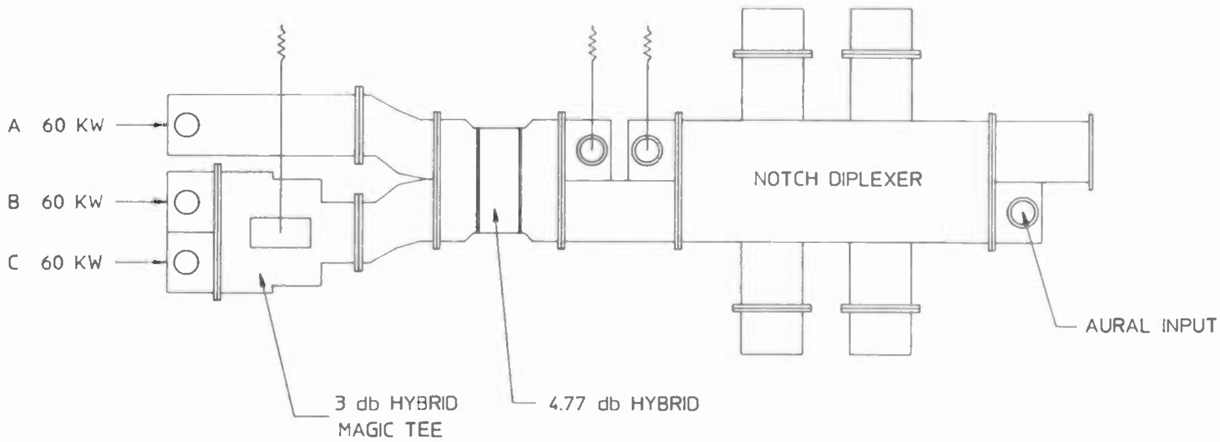
A three tube 180 KW magic tee diplexer combiner is not a completely new concept, but the switchless variety with total 3 tube selectivity is a brand new offering.

In the past, a 180 KW UHF system was a viable system, but it suffered from certain limitations upon failure of any tubes. Figure #1 depicts a system with no switching capabilities. If tube A fails, this is the best possible failure mode, tubes B & C go through the 3 dB Hybrid unaffected , but when the signal arrives at the 4.77 dB Hybrid only 2/3 of the 120 KW is directed to the diplexer. 2/3 of 120 KW is 80 KW. So with the failure of one 60 KW tube in this 180 system, the best possible result provides 80 KW on the air. The remaining 40 KW is dissipated in the load.

Failure of either B or C tubes creates a less favorable condition where all the power goes into the load until a second tube is extinguished. If the surviving tube is A then 2/3 of it's power goes into the load and 1/3 into the diplexer. This results in 20 KW on the air. If B or C is the surviving tube in this scenario, half of 60 KW goes into the first load and 1/3 of the remainder goes into the second load leaving 2/3 of 30 KW on the air. These types of failure modes discouraged broadcasters from using 180 KW Systems. No one likes to go from 180 KW to 80 KW or 20 KW upon the failure of one 60 KW tube.

There have been more attractive ways of obtaining 180 KW System, but they all involve switches that require shutdown before switching.

FIGURE #1: 180 KILOWATT DIPLEXER



MODES OF OPERATION		POWER	EFFICIENCY
A+B+C	TO DIPLEXER	180 KW	100%
A+B	TO DIPLEXER	0 KW	0%
A+C	TO DIPLEXER	0 KW	0%
B+C	TO DIPLEXER	80 KW	67%
A	TO DIPLEXER	20 KW	33%
B	TO DIPLEXER	20 KW	33%
C	TO DIPLEXER	20 KW	33%

Dielectric has recently built a 180 KW system for Silver King Broadcasting of New Jersey. It was installed last year in Tampa, Florida at Channel 50. The system has some very interesting features that overcome all of the disadvantages of the above system.

Figure #2 illustrates a simplified view of the system in Tampa. A standard Notch Diplexer appears on the right. The new switchless combiner consists of two standard switchless switching modules. Each module consists of a Hybrid, 2 Phase Shifters, a Magic Tee, and a load. The first module encountered when a signal is transmitted by tubes B & C works like any standard switchless combiner in a 120 KW or 240 KW system. It enables the output of either B or C tubes or both to be routed to the output of Magic Tee #1. The choice of modes is selectable by adjusting phase shifters #1 or #2.

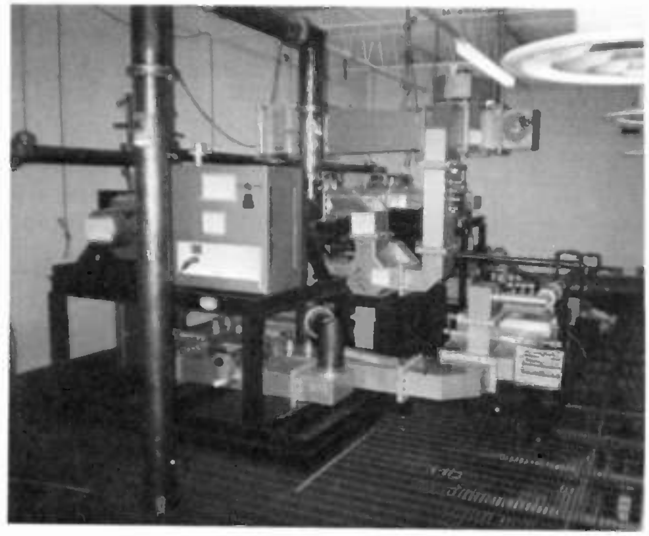
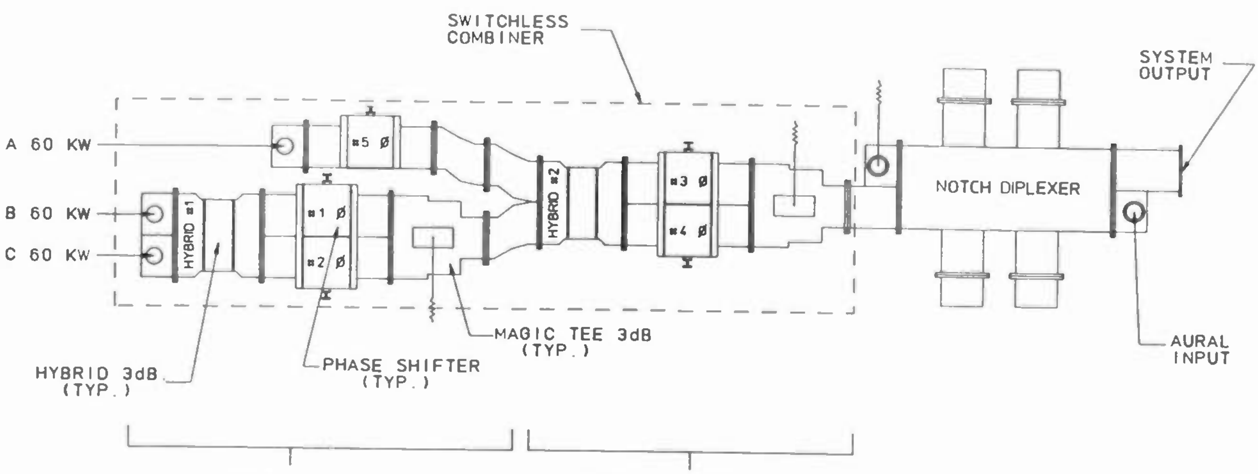


PHOTO #1

FIGURE #2: 180 KILOWATT SWITCHLESS DIPLEXER



ALLOWABLE MODES
B TO HYBRID #2
C TO HYBRID #2
B+C TO HYBRID #2

ALLOWABLE MODES	POWER	EFFICIENCY
A TO DIPLEXER	60 KW	100%
B TO DIPLEXER	60 KW	100%
C TO DIPLEXER	60 KW	100%
A+B TO DIPLEXER	120 KW	100%
A+C TO DIPLEXER	120 KW	100%
B+C TO DIPLEXER	120 KW	100%
A+B+C TO DIPLEXER	180 KW	100%

The novel feature of this 3 tube system becomes apparent when we observe the allowable modes of operation of the second switchless combiner. By suitable adjustment of each phase shifter each of the listed modes of operation can be obtained with similar efficiencies as two tube switchless combiners. See the table at bottom of figure 2 for the allowable modes.

Attaching a 3 tube magic tee system to a diplexer is no more difficult than attaching a two tube magic tee to a diplexer. Photo #1 shows a complete 180 KW 3 tube diplexer during installation. The magic tee of the 120 KW switchless is visible at the lower right. The coax elbow at bottom center is the input for the third tube. Phase shifter #5 from figure #2 is visible to the left of the elbow. Photo #2 shows the system from another view. In this photo, the 120 KW switch-

FIGURE #3: AVAILABLE NEW SYSTEMS

1. A FOUR TUBE 180 KW SYSTEM WHERE THE FOURTH TUBE IS A SPARE.

2. A FOUR TUBE 240 KW SYSTEM WHERE FAILURE OF A SINGLE TUBE ALLOWS 180 KW OPERATION.

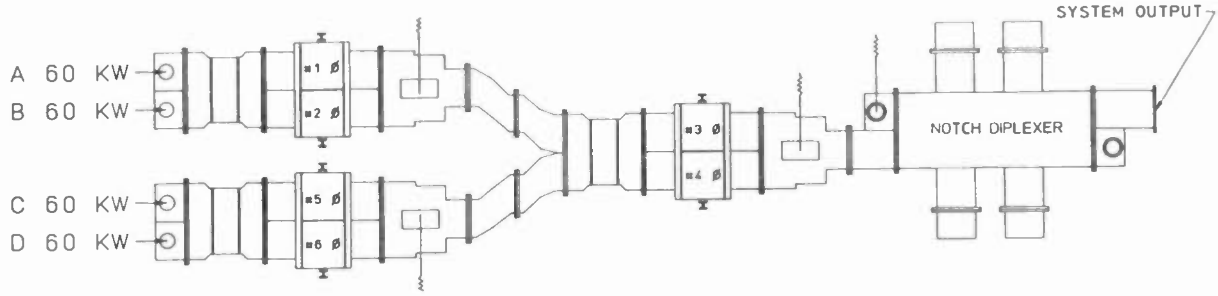
3. A FIVE TUBE 240 KW SYSTEM WHERE THE FIFTH TUBE IS A SPARE.

less is behind the railing and the phase shifter for the 180 system is visible below the round notch cavity and to the left of the load. The diplexer is the horizontal run of waveguide above the frame.

This system has performed so well that Dielectric is now offering several other variations on this switchless theme. See Figure #3.

In figure #4, we see the modes of operation for the 4 tube 180 KW Magic Tee switchless combiner system. They functionally appear the same and operate in the same modes. Using this configuration the broadcaster can operate on all four, any three, any two or any one tube and switch between any of these modes of operation at full transmitter power while never going off the air.

FIGURE #4 : FOUR TUBE 180 KILOWATT SWITCHLESS DIPLEXER
FOUR TUBE 240 KILOWATT SWITCHLESS DIPLEXER



MODES OF OPERATION	POWER	EFFICIENCY
A+B+C TO DIPLEXER	180 KW	100%
A+B+D TO DIPLEXER	180 KW	100%
B+C+D TO DIPLEXER	180 KW	100%
A+C+D TO DIPLEXER	180 KW	100%
A+B TO DIPLEXER	120 KW	100%
A+C TO DIPLEXER	120 KW	100%
A+D TO DIPLEXER	120 KW	100%
B+C TO DIPLEXER	120 KW	100%
B+D TO DIPLEXER	120 KW	100%
C+D TO DIPLEXER	120 KW	100%
ANY 1 TO DIPLEXER	60 KW	100%
ALL 4 AT REDUCED POWER	180 KW	100%
ALL 4 AT FULL POWER	240 KW	100%

Figure #5 demonstrates a five tube 240 KW switchless magic tee system. There may not be much demand for this configuration, but it illustrates that there are very few limitations to this concept. This enables a broadcaster to operate on any four tubes, rotating the fifth tube for maintenance and emergency use.

Existing 240 KW switchless systems are limited in that failure of a single tube requires the user to drop back to 120 KW. The systems illustrated here all allow the remaining tubes to be fully utilized upon failure of a single tube. In addition, they add to the available options for those who seek economical forms of redundancy for high power systems.

One final note for VHF broadcasters, all of the modes of operation illustrated here in waveguide are also available in coax at VHF frequencies. The two and three tube combinations might be attractive at VHF.

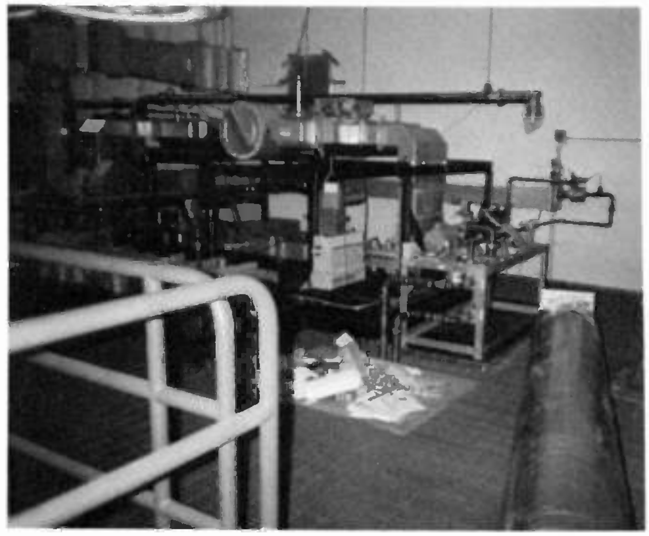
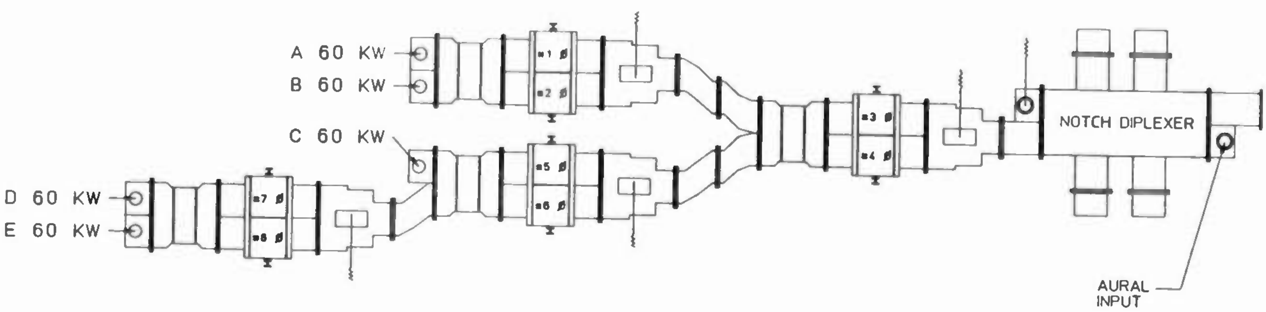


PHOTO #2

FIGURE 5 5 TUBE 240 KW SWITCHLESS DIPLEXER



MODES OF OPERATION	POWER	EFFICIENCY
ALL FIVE AT REDUCED POWER	240 KW	100%
ANY FOUR	240 KW	100%
ANY THREE	180 KW	100%
ANY TWO	120 KW	100%
ANY ONE	60 KW	100%

KLYSTRODE EQUIPPED UHF-TV TRANSMITTERS—REPORT ON THE INITIAL FULL SERVICE STATION INSTALLATIONS

N.S. Ostroff, R.C. Kiesel, A.H. Whiteside, A.B. See
Comark Communications, Inc.
Colmar, Pennsylvania

The Klystrode* tube has been incorporated into actual production UHF-TV transmitters by Comark Communications, Inc. Several transmitters have been installed in the field. This paper details the field experiences at these sites. It includes operational data as well as maintenance procedures and site problems. A discussion of a common amplification klystrode equipped transmitter, also installed in the field, is presented. Power consumption data and a cost analysis at each site has also been included.

* A registered trademark of Varian Associates, Inc.

I. INTRODUCTION

On June 5, 1988, the world's first Klystrode powered UHF-TV transmitter was commissioned into full time broadcast service at WCES-TV, Channel 20, in Wrens, Georgia. This 120kW installation was followed in October by a second 120kW transmitter at WABW-TV, Channel 14, in Pelham, Georgia. Additional installations also followed, including a 10kW common amplification air-cooled configuration for the Central Virginia ETV network and an 80kW common amplification parallel configuration serving Bloomington, Indiana, WIIB-TV, Channel 63.

This paper will detail these historic first uses of klystrode tubes. Operating maintenance and actual proof data will be provided to illustrate the advantages of this new technology.

Finally, a discussion pertaining to the direction of present and future klystrode equipment configurations and a discussion on the use of common amplification techniques, air cooling and RF system configuration will also be presented.

II. KLYSTRODE TRANSMITTER INSTALLATIONS

A. Wrens, Georgia - WCES-TV, Channel 20

The world's first klystrode equipped transmitter was placed into full time (19 hours per day) broadcast service at Wrens, Georgia, WCES-TV. WCES-TV is owned and operated by the Georgia Public Telecommunications Commission and is a PBS affiliate.

The 120kW klystrode transmitter replaced an aging 30kW G.E. unit. WCES-TV went off the air in early May 1988 to permit the removal of the old G.E. transmitter and the installation of the Comark 120kW unit. The new transmitter with its increased output power required replacement of all of the old transmitter's support systems including the RF, AC power controller and cooling systems. In addition, ten tons of building air conditioning was added.

The 120kW klystrode transmitter is shown installed at Wrens in Figure 1. The equipment utilizes three Eimac 2KDW60LA kystrode water cooled tubes, two independent high voltage power supplies and two separate heat exchangers with back up pumps and manual switchover.

The magic Tee RF system shown in Figure 2 permits the operation of the 60kW visual klystrode tubes in parallel or separately. It also permits operation of the visual 2 tube as an aural amplifier and multiplex operation of either visual tube both separately or combined. This highly redundant configuration provides multiple modes of emergency operation and is typical of other 120kW Comark installations.

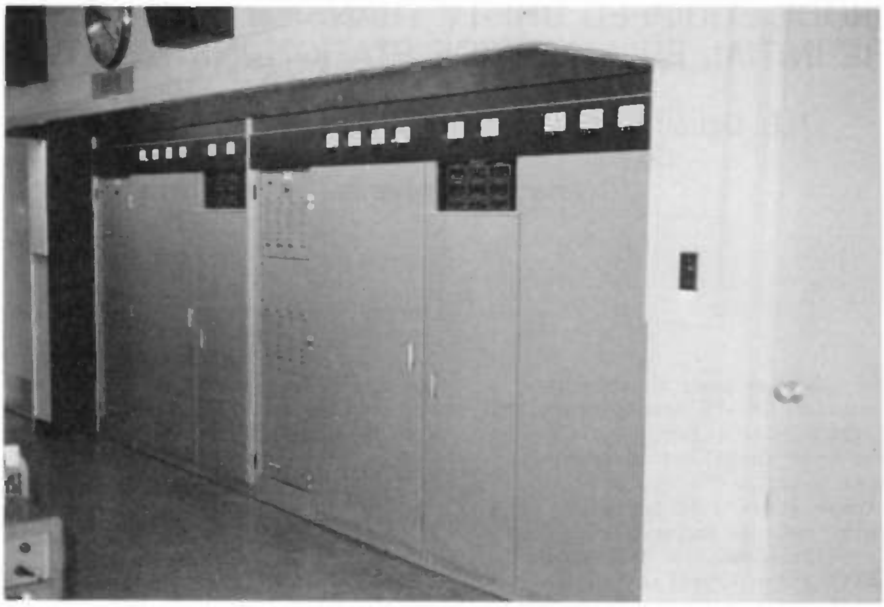


Figure 1 - 120kW Klystrode Amplifier Cabinets as installed at WCES-TV, Wrens, Georgia

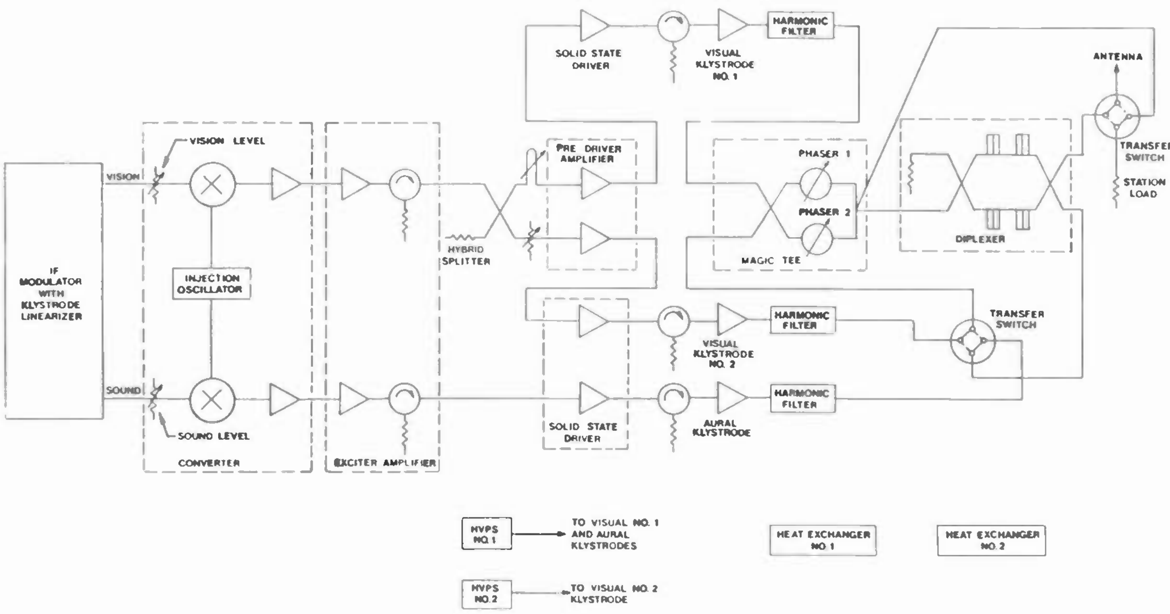


Figure 2 - 120kW Klystrode Transmitter System Block Diagram (Wrens, Georgia)

Figure 3 - Three Klystrode Tubes plus Spare ready for installation at Wrens, Georgia

Figure 4 - Modulator Linearity Correction System under adjustment (Wrens, Georgia)

Figure 3 shows three klystrodes plus a spare in their circuit assemblies ready to install into the Comark 120kW transmitter at WCES-TV.

Figure 4 shows the modulator's linearity correction system under adjustment during installation. The linearity corrector for the Comark klystrode transmitters is designed by Thomson-LGT. The superior capability of this corrector is demonstrated by the quality of the proof results.

The corrector is required to deal with non-linearities at both white and black levels. It independently corrects:

a. Low frequency linearity
b. Differential gain
c. Differential phase
d. ICPM
e. In band response flatness

The linearity corrector is part of the modulator unit. This modulator includes capability for sync reinsertion as well as overall system output power AGC. The AGC system uses a sample of the transmitter's output signal to compare to an operator preset power reference. Thus, after initial adjustment, the transmitter output power is fixed by the AGC loop against changes in drive level, input line voltage changes and other perturbations. The loop has an overall positive drive range of +1db to avoid overdrive conditions. The AGC is also used to provide a soft RF drive start which gradually increases output power over a 1 to 2 second period. This greatly reduces transient conditions that can damage both RF and DC systems.

The proof of the Wrens, Georgia transmitter revealed an extremely clean, text book, output reflecting the overall capability of the system as shown in Figures 6 through 11.

Figure 5 is a schematic diagram of the proof test equipment set up.

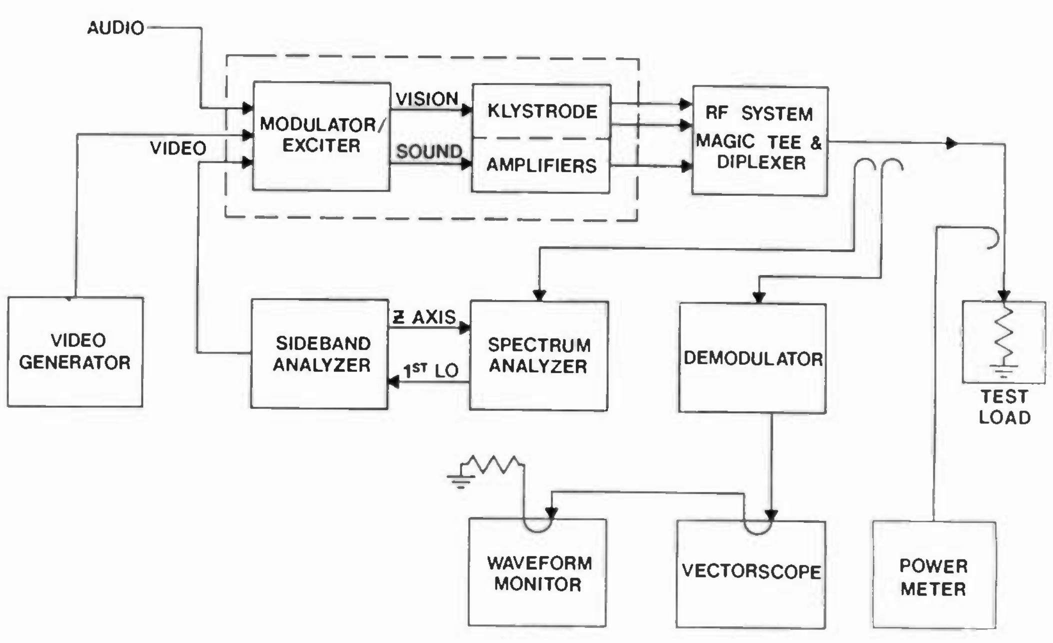


Figure 5 - Proof of Performance Test Equipment Block Diagram

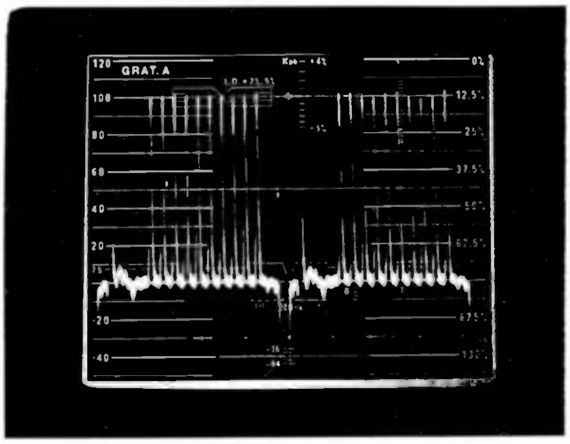


Figure 6 - Low Frequency Linearity - WCES-TV

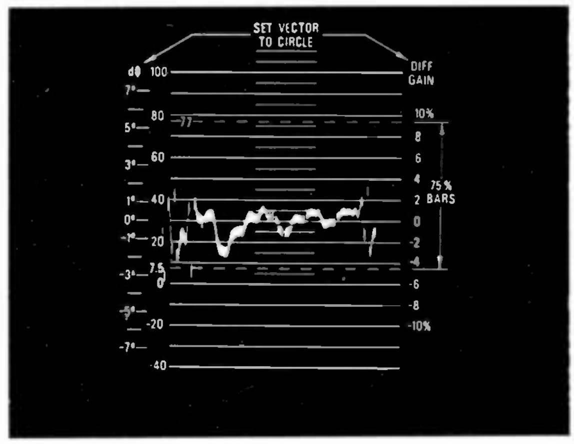


Figure 9 - Differential Phase - WCES-TV

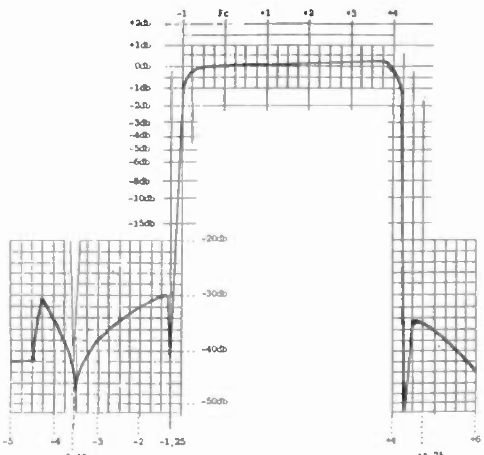


Figure 7 - Sweep Response, WCES-TV

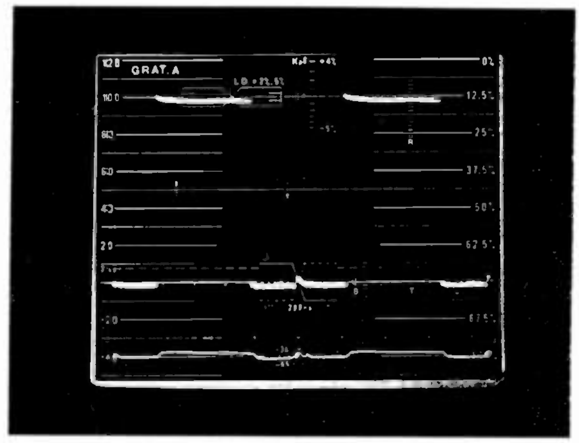


Figure 10 - Field Square Wave - WCES-TV

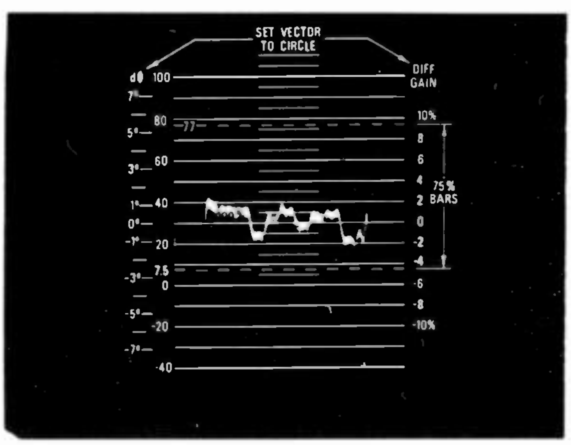


Figure 8 - Differential Gain - WCES-TV

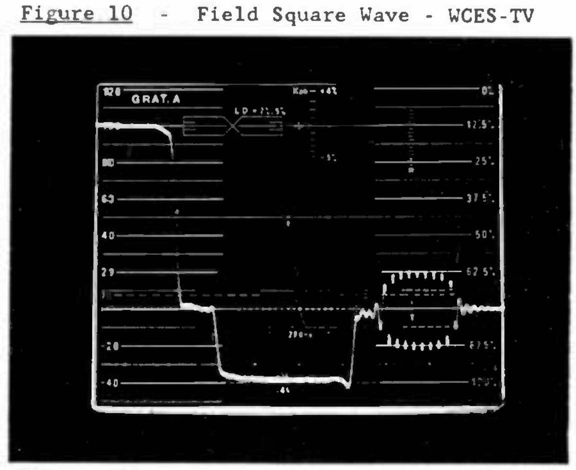


Figure 11 - Horizontal Rate - Sync Expanded WCES-TV

Figure 7 is the output spectrum showing both upper and lower sideband response. The superior lower sideband levels, at or below -30db, is proof of the system linearity.

Figure 10, full field square wave, is interesting in that it shows the effect of the power supply regulation. The variation of RF output is less than 2%. The AGC system helps to maintain this level.

Figure 11 shows the line rate horizontal sync. Note the relative lack of ICPM spikes and power supply effects.

The proof of the Wrens, Georgia klystrode equipped transmitter demonstrated the full visual performance capability of the klystrode tube for the first time in actual broadcast service.

The aural performance of the transmitter was also outstanding. Distortion levels ran less than 1% throughout the audio spectrum with AM noise down 63db and FM noise down 62db.

The real proof of the transmitter's performance was found on the klystrode's beam current meters. The average beam current at 50% APL and 104% peak RF output was 1.65 Amperes per tube. At Black level the beam current was 2.31 Amperes per tube. Combined with a beam voltage of 31.5kV the measured figure of merit was 120% and the total beam power was 104kW. If a fully pulsed advanced design klystron transmitter was installed instead of the klystrode equipped transmitter, the power consumption of the output visual klystrons at 50% APL would have been at least 192kW. This is based on an assumed 46% efficient klystron using pulsing to raise the operating efficiency to 65%. Higher efficiencies can be achieved but they are usually not stable over long term operation.

Assuming all other power consumption elements are equal, we can derive the operational cost savings of the Comark klystrode equipped transmitter. At 19 hours per day of transmitter operation and an average of $.075 per kW hour, the savings to WCES-TV, Channel 20, over one year is $45,800 using the klystrode equipped Comark transmitter. If 24 hour per day operation was undertaken by WCES-TV, the power cost savings would approach $60,000 per year. Finally, the cost savings of the Comark klystrode equipped transmitter as compared to a typical klystron transmitter currently in service in many stations today would be over $95,000 per year. This is based on a klystron beam efficiency of 50% and 24 hours per day operation.

The installation at Wrens, Georgia proved the cost savings of klystrode tube technology when used in the Comark transmitting system.

B. Pelham, Georgia - WABW-TV, Channel 14

The second 120kW klystrode equipped transmitter was commissioned on October 10, 1988. This installation proceeded along the same lines and satisfied the same technical requirements as the installation at Wrens, Georgia. Figure 12 shows members of the WABW-TV staff examining the installed Comark 120kW klystrode equipped transmitter. The visual #2 installed klystrode tube can be seen on the left side of the picture. The proof of the Pelham installation produced data that was very similar to Wrens. Figure 14 shows the measured lower sideband responses at -28db. This again established the fundamental linearity of the klystrode equipped system.

Figure 12 - WABW Staff Members Examining Comark 120kW Klystrode Equipped Transmitter - Pelham, Georgia

Left to Right:
Mrs. Irmgard Jones
Mr. Don Mitchell, C.E.
Mr. Bobby Poitevint

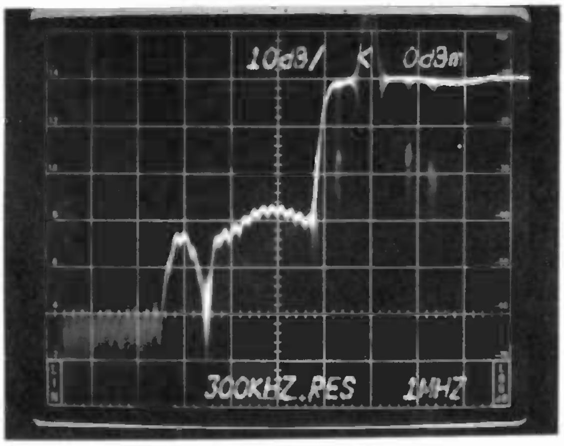


Figure 13 - Lower sideband responses - WABW-TV

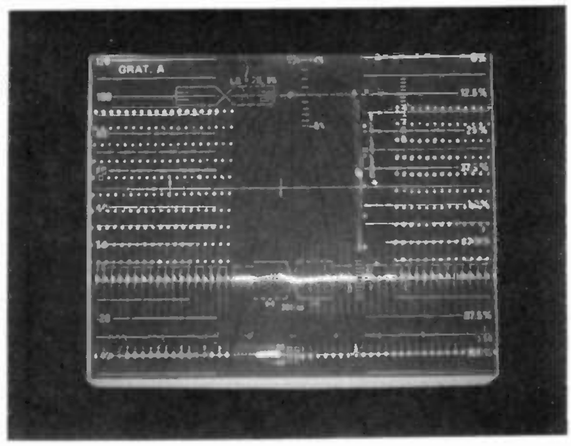


Figure 14 - Expanded Vertical Interval - WABW-TV

Chart 1 is a comparison of some of the normally tested visual parameters for both the Pelham and Wrens installations. The performance of both transmitters was very similar illustrating the reproducibility of this technology.

CHART I

Comparison of Wrens and Pelham Results

Parameter	Measured value Wrens	Pelham
Lower Sideband Rejection	-30db	-28db
Upper Sideband Rejection	-35	-42
In Band Flatness	0.5db	0.5db
-750KHZ Response	-0.4db	-0.7db
-1.25 MHZ Response	-40db	-30db
Low Frequency Linearity	2%	3%
ICPM	2°	2°
Variation of Output	2%	1.5%
Differential Gain	4%	4%
Differential Phase	$\pm 1^\circ$	$\pm 1^\circ$

Figure 14 shows the Pelham transmitter output vertical interval and Figure 15 shows the 2T and 12.5T bar responses. Both of these parameters are similar to the Wrens transmitter results.

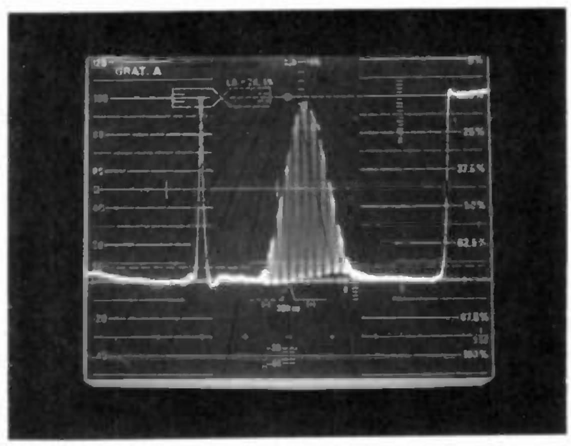


Figure 15 - 2T, 12.5T Response WABW-TV

The average beam current of the Channel 14 transmitter at Pelham, at 50% APL was 1.45A per tube. The beam voltage was 31.6kv, resulting in a total average beam power consumption at 50% APL for both tubes of 91.64kW at 120kW peak sync output. This represents a figure of merit of 131%.

The cost savings to WABW-TV by using the Comark klystrode equipped transmitter over the most advanced pulsed klystron transmitter is about $48,000 per year based on $.075 per kW hour and 19 hours per day. When compared to an average transmitter in service today, the cost savings is at least $95,000 per year based on 24 hour per day operation.

Note that the actual annual cost savings in dollars is not materially different ($48,000 vs. $45,800) at the Pelham Figure of Merit of 131% compared to the 120% achieved at Wrens. Figure of Merits above 120% do not result in significant additional cost savings as a result of the law of diminishing returns. It is, therefore, difficult to justify incurring major costs or technical risks to achieve Figure of Merit improvements beyond 125%. If further meaningful efficiency improvements are to be achieved, it is necessary to make changes in the way a high power T.V. transmitter is configured. The next Comark installation is representative of such a change in design philosophy.

C. Bloomington, IN - WIIB-TV, Channel 63

On December 27, 1988 an 80kW klystrode equipped transmitter was commissioned at WIIB-TV, Channel 63 serving the Bloomington-Indianapolis market. This transmitter represents an historic departure from traditional transmitter design philosophies and it is the first time that klystrode tubes have been used in common amplification service.

The superior linearity of the Eimac 2KDW60LF klystrode tube combined with the powerful linearity correction incorporated in the Thomson-LGT designed corrector, permit outstanding signal performance from the Comark Model CTT-U-80SKMR klystrode equipped transmitter. While multiplex operation of klystrons is normally associated with emergency service, or significantly reduced output ratings, the common amplification technique incorporated in the CTT-U-80SKMR permits full broadcast quality signals at high output power and excellent efficiencies.

The equipment configuration at WIIB-TV takes full advantage of the options offered by common amplification.

Two Eimac 2KDW60LF klystrode tubes are used in the Comark CTT-U-80SKRM, each operating at 40kW and combined in a magic Tee for 80kW. No diplexer or diplexer switching is required. Further, the parallel nature of the independent solid state drive chains, as well as the independent high voltage power supplies and heat exchangers, reduce the possibility of signal failure. In the unlikely event of a signal failure in one half of the transmitter, the magic Tee can be switched to provide 50% of full transmitter rated power (40kW) without removing the RF output from the remaining operating tube.

Figure 16 is a block diagram of the WIIB-TV installation. It illustrates the redundant nature of the transmitter system and the simplicity of the RF output system. If a second modulator and upconverter were added with auto switching, the configuration could be considered to be "bullet proof"!

The proof data for the WIIB-TV transmitter was taken at both a 10db and a 16db aural visual ratio. Final operation was set at 80kW and 16db A/V. The 10db A/V operation was proofed at 60kW output power.

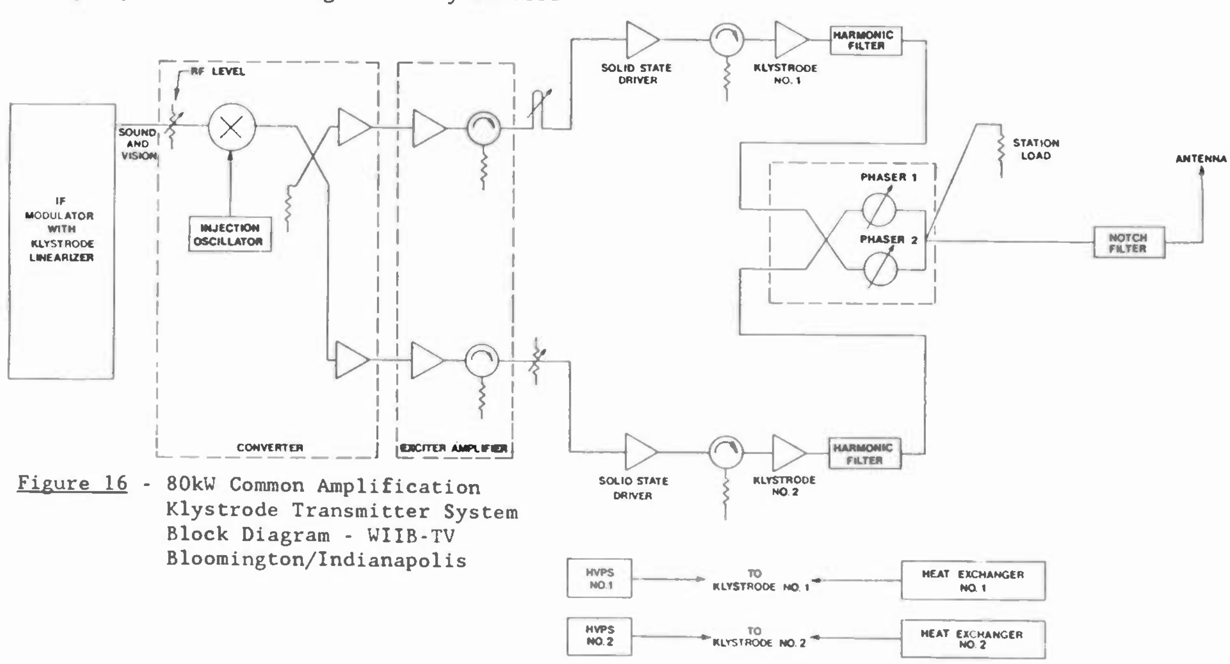


Figure 16 - 80kW Common Amplification Klystrode Transmitter System Block Diagram - WIIB-TV Bloomington/Indianapolis

Figure 17 - 80kW Transmitter installed at WIIB-TV - Bloomington/Indianapolis

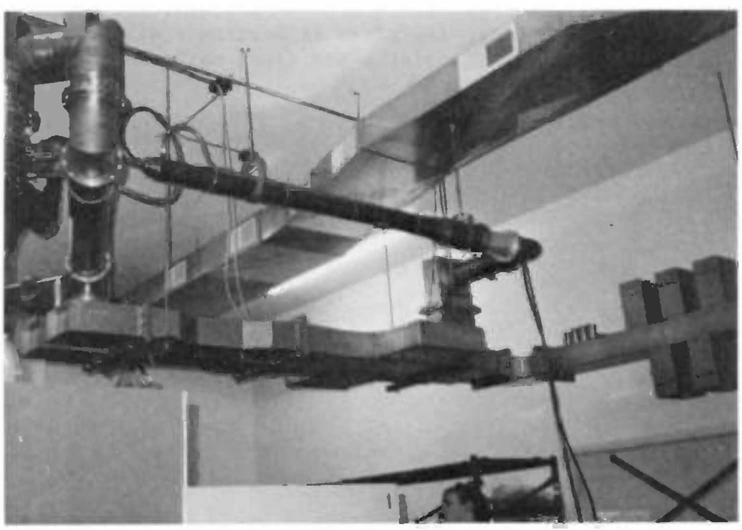


Figure 18 - Output RF "System" consists of only the output harmonic filters, Magic Tee Combiner, System Dummy Load (Water Column) and the Waveguide Output Notch Filters - WIIB-TV

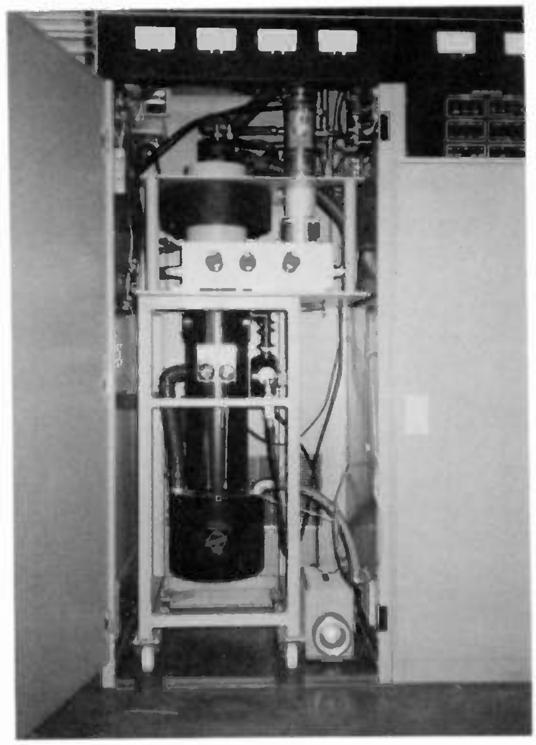


Figure 19 - Klystrode Tube installed in the Amplifier #1 Postion - WIIB-TV
This tube is amplifying both the vision and sound carriers simultaneously.

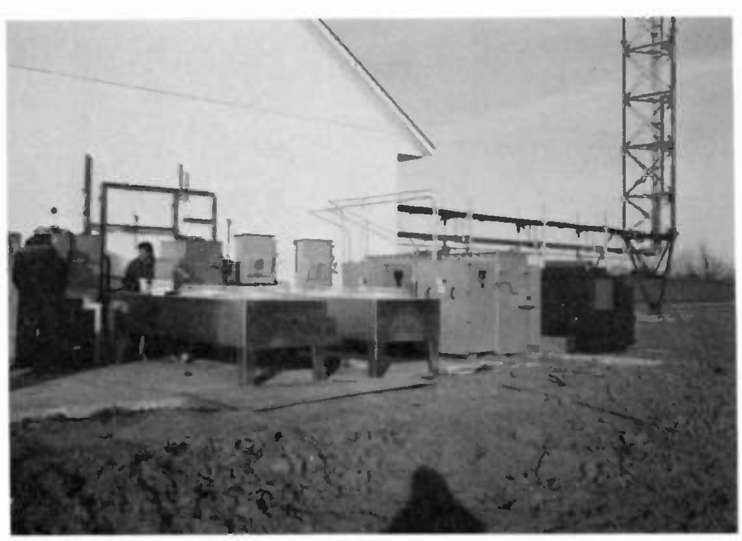


Figure 20 - Dual Heat Exchangers and Dual Power Supplies which add redundancy to the transmitter system at WIIB-TV

The electrical performance of the 80kW transmitter was outstanding. All visual and sound parameters were well within rigid broadcast standards.

Chart II summarizes the measured results.

CHART II

Measured Results WIIB-TV, Channel 63

Parameter	Measured Value
Lower Sideband Rejection	-30db
Upper Sideband Rejection	-40db
In Band Flatness	0.5db
Low Frequency Linearity	4%
ICPM	1°
Variation of Output	2%
Differential Gain	1.5%
Differential Phase	1°
In Band Intermodulation	
10db A/V (60kW)	-53db
16db A/V (80kW)	-56db
Out of Band Spurious	
Visual Carrier -4.5MHz	-62db
Visual Carrier +9.0MHz	-63db

Figure 21 shows the output spectrum at 80kW. (The in band IM product at Visual Carrier +920kHz can be seen to be better than -56db.

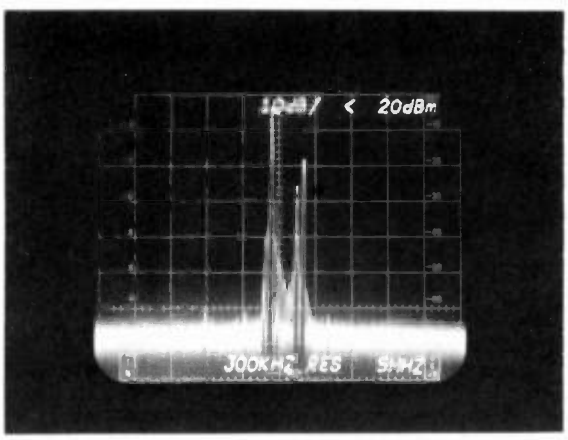


Figure 21 - Output Spectrum at 80kW-WIIB-TV

Power consumption of the 80kW transmitter was extremely low considering that both vision and sound carrier were being amplified simultaneously in each tube.

At 80kW peak visual output, the total beam power drain of the pair of klystrode tubes was 55.2kW at 50% APL. Black picture power drain was 83kW. White picture power drain was 42.5kW. The beam voltage was 28.6kV.

The figure of merit for this operation was 145%. This is calculated without considering the aural output power from the tube. Of course, the A/V ratio was 16db. At 10db A/V ratio and 60kW peak visual output power, the measured figure of merit approached 120%.

The surprising outcome of this installation was that the figure of merit for common amplification service closely follows that for visual only service! Considering the small economic impact of higher Figure of Merits, makes this finding truly significant for future transmitter designs.

The implications of the WIIB-TV installation are far reaching. They suggest a whole new generation of high power, super-efficient, simple and low cost transmitters based on klystrode tubes in common amplification. These implications will be discussed in the conclusion of this paper.

III. OPERATIONAL EXPERIENCE

Chart III shows the number of sockets at each site and the total hours of klystrode tube operation as of February 10, 1989.

CHART III

Sockets and Time in Service - 2/10/89

Site	No. of Sockets	Hours Per Day	Combined Hours in Service
Wrens	3	19	13,750
Pelham	3	19	5,300
Bloomington/ Indianapolis	2	24	2,200
Charlottesville* (10kW Air Cooled)	1	0	0

*Shipped to site but not installed as of 2/10/89

There have been no in service operational tube failures at any site since the equipment was commissioned. There have been three tubes replaced. Two replacements took place during installation and initial testing procedures. In one case at the Wrens site, efforts were undertaken by Eimac and Comark personnel to narrow the bandwidth of the aural klystrode tube to attempt to obtain more efficiency. The process was taken too far and the output ceramic was cracked. The extremely high voltage developed in the, intentionally, narrow banded (less than 2MHz) output cavity exceeded the strength of the ceramic.

The second failure occurred after 50 hours of operation of the replaced aural tube. The problem was traced to shipping damage. After these initial problems, the final aural tube has proven to be stable and reliable.

The third and final event was a failure of a feed-through capacitor in the RF circuit assembly for the visual tube at Wrens at approximately 3,700 hours in service. Upon disassembly of the input cavity, it was discovered that the V1 klystrode tube had suffered a burn through one of its filament contacts. This was not related to the feed-through failure. The vacuum was not lost. Further investigation revealed that several filament contact fingers in the socket were crushed. The tube was removed from service and replaced. The socket was repaired. It should be noted that the tube was removed while still operational. Inspection of the other tubes at Wrens showed no other filament contact damage.

There have been no tube failures at the Pelham or Indianapolis sites.

The tube removed from Wrens with the burned filament contact was returned to Eimac for evaluation. The tube was operated at the Eimac factory and then disassembled. Initial results indicated that the grid had not changed its physical dimensions and that there was no evidence of any high energy arc damage to the grid. Further, the focus electrode did show arc tracks indicating that the design was performing properly by forcing any tube arcing to the robust focus electrode.

IV. MAINTENANCE

The routine maintenance of a klystrode equipped transmitter is generally the same as that required for today's standard water cooled klystron transmitters except for the lack of pulser maintenance. Unlike the new MSDC klystron device, the klystrode is cooled with a glycol and water mixture in freezing climates and straight distilled water in warm climates. No de-ionization system with expendable filters is used and the entire water system is virtually at ground potential. There is no beam voltage on the klystrode's cooling water or multiple hose connections inside of the collector structure like that found on the MSDC klystron device.

The only maintenance item that differs from standard klystron equipment is the need to clean the klystrode's grid. This process involves using a built in semi-automatic system in the transmitter to place a positive voltage on the grid (60V) with respect to the cathode. With beam voltage off, the grid becomes a miniature anode of a diode and dissipates the conducted cathode current. This heating effect causes any cathode material that has migrated to the grid to boil off and be collected by the tube's vac-ion pump. The pyrolytic graphite grid structure is unaffected by this process. The entire process takes about fifteen minutes. Grid cleaning intervals have been experienced from every week in new tubes to once every two months. Monitoring of the tubes' grid current provides the criteria for planning a grid cleaning during the next routine maintenance activity of the station. The process of cleaning the grid is being reduced to a one push button operation in the latest versions of Comark's klystrode equipped transmitters.

Before and after cleaning, the parameters of the klystrode tubes appear to be unchanged. The major effect is the reduction of negative grid current.

There is some indication that grid cleaning intervals can be extended as the cathode of the new tube ages and less material is available to be boiled off.

V. THE KLYSTRODE & HIGH DEFINITION TELEVISION

The Klystrode tube has previously been demonstrated to be capable of wideband full power operation.[1] At all of the sites described in this paper, the wideband performance was confirmed. The combination of the wideband klystrode tube stage and the system's broadband driver stages make Comark klystrode equipped transmitters ideal candidates for future upgrade to HDTV applications, if extended channel bandwidth is required.

VI. CONCLUSIONS

During 1988, three klystrode equipped transmitters were placed into full time broadcast service. A fourth was delivered to a site for installation. The operating results at these installations have confirmed the economic claims for system efficiency. Further, these installations have demonstrated that simple transmitter systems can be built that will permit excellent electrical performance to be obtained from

One of these installations, WIIB-TV, used common amplification techniques. This transmitter eliminated the visual/aural diplexer and its associated switching systems. As a result, a very simple output R.F. system was possible using just a magic TEE and output filters. This configuration is reliable and inexpensive and leads to consideration of future possibilities. A second common amplification site, at the 10kW level, was also delivered. This transmitter is air cooled.

Common amplification has already been demonstrated at 80kW, 40kW per tube. At transmitter power levels of 120kW to 160kW, the loss of any single signal path would only reduce output by one-third or one-quarter. The output R.F. system would be simple and the transmitter inexpensive.

The high efficiency documented in this paper leads to the possibility of air-cooling at 30kW common amplification or 40kW visual only service. No other device, including Tetrodes, can offer this possibility. Air cooled 60kW and 90kW transmitters are clearly possible. In vision only operation, 80kW to 120kW air cooled transmitters are possible. Air cooling would eliminate heat exchangers, plumbing and pumps. Installation would be considerably easier and transmitter site requirements would be reduced. These advantages will be demonstrated with the installation of Comark's 10kW air cooled klystrode at Charlottesville, Virginia in early 1989.

The inherent wideband performance of the klystrode tube and the Comark broadband driver system make these transmitters ideal candidates for upgrading to HDTV if channel extension techniques are adopted by the industry. All of the klystrode equipped transmitters demonstrated a wide bandwidth capability (i.e. over 2 channels). This wide power bandwidth is available, without modification to the klystrode amplifier, for HDTV applications in the future.

The operating results of the first klystrode equipped transmitters and the subsequent field experiences, indicate a bright future for this technology and exciting possibilities for the UHF broadcaster. The klystrode tube in the Comark transmitters permits an efficient, simple, reliable and inexpensive transmitter to be built for high power UHF broadcasting applications.

Bibliography:

[1]"Using Klystrode Technology to Create a New Generation of High Efficiency UHF-TV Transmitters", N.Ostroff/A.Whiteside

Authors' Note:

The authors wish to thank the management and staff at WCES-TV, WABW-TV and WIIB-TV for their cooperation and support during the installation and initial operation of the equipment at their respective sites.

GIVING A RENEWED LIFE TO AN OLD UHF TRANSMITTER

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There seems to be little doubt that we must learn to get along with a new hard economic reality in the world of television broadcasting. Major capital expenditure for replacement equipment happens only if financial justification overwhelmingly points towards complete replacement as an only option. The impending replacement of an UHF television transmitter is not immune from this new requirement to conserve financial resources. Alternatively, it is often not economically viable to retain an unreliable, older technology transmitter. Loss of precious revenue generating air time, coupled with increasing maintenance costs and a shortage of critical spares can create a good deal of pressure to replace an older transmitter. Luckily, when faced with this dilemma, there is in many cases a workable alternative to total replacement of a transmitter.

In the abstract, a workable alternative requires making a definitive evaluation of the condition of your present transmitter, determine what areas, if replaced, would gain the most benefit in terms of air quality, reliability, maintenance costs and efficiency, then devise a remodeling plan that provides the greatest cost-benefits ratio and meets all the desired goals.

With all these lofty goals in mind, the staff at Monitor Television Inc., WQTV-68 Boston was faced with just such a problem. They had inherited from the station's previous owners a RCA Model TTU-60 transmitter which had been originally designed over 20 years ago and had been poorly installed and maintained over the past 10 years of it's life. This installation also had all of the predictable problems that go along with trying to operate a television transmitter reliably and efficiently at almost 800 Mhz. Totally unreliable operation, coupled with a shortage of of most major component spares, pointed towards replacing the transmitter. Unfortunately, total replacement of the transmitter would deflect capital resources away from other rebuilding projects that were important to the overall growth of the station.

As a result, it was decided to apply a little imagination to the project of improving the reliability and output quality of the present transmitter. The goal was to meet or exceed the air "look" and reliability of a new transmitter. A set of priorities and a plan of action was developed. The planning was not made any simpler with the added restriction of a 24 hour, seven day a week program schedule.

There was little doubt that the problem that needed to be addressed immediately was the output end of the transmitter. The power amplifiers are the source of the most catastrophic and costly failures and generally require the most immediate attention.

Most UHF transmitters provide power amplifier high voltage regulation by means of some type of variable transformer or transformers at the main power line source voltage. This is not the case in all UHF transmitters though, many designs have no power amplifier mains regulation at all. Although most high voltage power supplies are designed to work over a wide range of input mains voltages, the effect of reduced or excessive input supply voltage to high power Klystrons in visual service can be very dramatic. Compressed or stretched sync, high incidental phase, coupled with poor or unstable differential phase and gain performance is common. If more than one tube is used in parallel, poor power tracking coupled with lowered efficiency is a guaranteed result of poor or nonexistent regulation.

WQTV's particular installation included one General Electric Inductrol type regulator on each of three input power line phases feeding the power amplifiers power supplies. Under the best of circumstances this installation provided regulation of poorer than 5%. The regulators also often broke down because of the weakly designed fiber gear assemblies that have been become infamous.

Although regulation of the individual phases is in theory most desirable, the reality of most installations is that individual phase regulation is just not necessary. Whether the source voltage is delta or wye, a decrease in one phase caused by a temporary load imbalance is almost always accompanied by a corresponding rise in another phase. This is particularly true if there are no power factor complications. Most transmitters, fortunately, sum the outputs of the various phases in parallel or series so that the net result of source imbalances is minimized. The real need for regulation comes as a result of the rise or fall of all three of the input phases in the incoming mains. In WQTV's individual case the regulators were replaced with a Peschel type, simultaneous phase regulator made by Hipotronics Inc. of Millerton, New York. This unit has provided reliable regulation of better than 1% over an incredible 10% plus range of voltages. This new regulator totally solved the poor power output stability of the this transmitter.

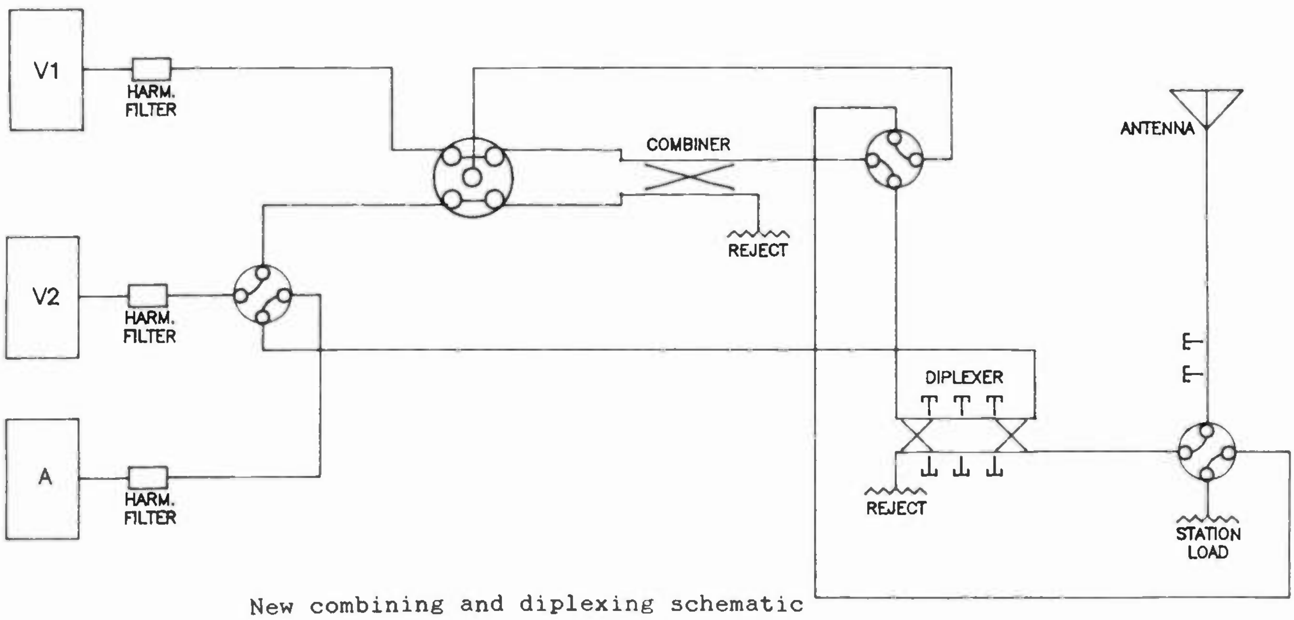
The next area that required an examination was the power amplifier's high voltage power supplies. Fortunately, the installation at WQTV had air cooled high voltage transformers with separate high voltage diode stacks. Many transmitters are fitted with oil immersed high voltage transformers. Often these make up a self contained power supply with built in diode stacks and filtering. If these transformers were built prior to 1978 there is a chance that they may contain PCB's as a fire retardant. Although nothing needs to be done with these installations under present law if they are not leaking other than having EPA approved labeling, one should consider the replacement of these units while they are in good condition and do not present a disposal hazard.

The RCA TTU-60 at WQTV is fitted with high voltage diode stack assemblies that are easily rebuilt. A little research led the station staff to a alternate source of the diodes. Fortunately, many of the failed stack assemblies had been saved over the years. Normally, only a few of the series diodes fail in these units so that they can be inexpensively rebuilt. The cost for new complete units are extremely high. After inspecting and repairing the high voltage diode stack assemblies, the rest of the high voltage power supplies were thoroughly cleaned and all the old suspect high voltage wiring was replaced.

The next area of the transmitter to gain attention involved overall transmitter reliability and came as a result of a major mistake in original installation design. This transmitter is made up of two combined 30 Kw visual amplifiers with a separate amplifier devoted to aural service. All of the transmitter combining and switching in this installation was accomplished with standard coaxial transmission line and not waveguide. As with any high band UHF television transmitter, the low loss and reliable method of combining and switching is with use of waveguide combining and switching. WQTV's original installation was complicated by the interchangeable use of 75 ohm 6-1/8" and 50 ohm 3-1/8" transmission line throughout the original design. This was originally done because only a small amount of room available for the overall installation and a poor choice of non-matched switching, combining and load components was made at the time. There were no less than five 50 ohm to 75 ohm transmission line transformers used throughout this design. This resulted in the inevitable heating, poor efficiency , I^2R and return losses. These in turn produced premature failures.

The very able aid of Tom Vaughan and the good people of Micro-Communications Inc. of Manchester New Hampshire was enlisted to design a new switching and combining system for this transmitter. The design criteria required that this combining, switching and diplexing system provide as much failure mode redundancy as possible because of the station's 24 hour schedule and the lack of a redundant transmitter. The design also had to keep in mind the anticipated stereo operation the station was considering. The final design is shown in the enclosed figure. The clever use of a five port waveguide switch allowed for the efficient switching of any one of the individual transmitters to the station load or to the antenna with or without the diplexer. Also provided in this design was the flexible capability for use of either of the visual transmitters for aural service or for the use of any of the three amplifiers in multiplexed or combined visual and aural service. This "belts and suspenders" approach provided a degree of redundancy that is a real luxury.

The next problem to be resolved was how to remove the present jungle of transmission line combining and switching and then install the new waveguide system and remain on the air with as near a normal schedule as possible. This not



New combining and diplexing schematic

unlike changing an automobile's engine oil while the engine remains running. The solution turned out to be easy and it solved more than one problem at a time. One of the least desirable areas of this particular transmitter was it's exciter. Although it is all solid state, it acted as if it where full of tubes. It required constant repair and adjusting with a norm of poor picture quality and even poorer audio performance. Also, the thought of later converting the aural portion of this unit to wideband MTS operation seemed impossible with its poor stability and even poorer signal to noise performance. The people at ITS Inc. of McMurray, Pa. make a retrofit exciter for the TTU-60 that exceeds the old exciter's performance and stability. Also, under present FCC rules this change in the transmitter's exciter does not require relicensing because it does not effect the transmitter's basic frequency or power output and it is already FCC type accepted. In addition, this unit also works particularly well in the multiplex or combined visual and aural mode. Much of ITS's business is in low power television where combined visual and aural service is the norm. As a result, their exciter designs are optimized for just that type of need. This turned out to be one of this exciter's greatest assets and gave the station the method of operation needed to accomplish the transmission line replacement.

The performance of the ITS Model 20 exciter seen in actual standard service has exceeded expectations. The transmitter typically has less than 1° of differential phase and 1% of differential gain. Incidental phase remains below 1°. This exciter can attain and keep these impressive numbers with the use of a series of multipoint video predistortion networks that can be switched in and out as needed. This affords the range needed for for correcting almost any type of power amplifier distortion that might be expected. The aural portion of this exciter is equally impressive. The station had opted over the years to run over fifteen percent aural power because of the poor signal to noise performance of the old exciter. Experimentation has shown that the station can now lower the aural levels to less than 5% of the visual power output with negligible aural quality degradation. The station is presently, as a matter of convention and safety margin, running a 10% aural level. Testing in actual stereo service remains, but no doubt equally impressive performance will be demonstrated. This exciter's aural and visual drivers can also put out over three watts each, which could in many installations remove the requirement of intermediate power amplification.

With the purchase of the new exciter the installation of the new waveguide combining system was easily accomplished. During a brief four hour outage on a Sunday night, the new exciter was installed and placed into multiplexed service into a single visual power amplifier. This amplifier was rerouted with two short sections of transmission directly to the antenna's feed transmission line. The visual power amplifier tube could easily be run at 35 to 40% power in this mode without any exotic tuning. The station ran in this mode without most of the audience noticing much of a signal quality change for the two weeks of the new combining and diplexer installation. Although there was a significant power output difference, it was more than made up for by a corresponding improvement in aural and visual quality afforded by the installation of the new exciter. The extensive planning by the people at MCI and DSI Communications, who actually installed the new system paid off. The whole system was installed and tested within the two week deadline and the transmitter was placed back into normal combined service during a five hour planned off period. It should be mentioned that this change in the combining system did not require relicensing under present FCC rules because all of these changes where made prior to the point at which the power output of the transmitter is measured and calibrated, namely the station's dummy load. Normally, the engineering affidavits that accompany a station's construction permit application indicate the losses incurred after transmitter power output measurement. Only if those losses are changed is relicensing required.

The transmitter remodeling completed at this point made dramatic and noticeable differences to WQTV's viewing audience. The losses in the old combining system were greater than 2db. The visual tubes rarely ran lower VSWRs of less that 1.2. This made making the station's licensed power virtually impossible. The new combining system had less than .3 db total insertion loss with nearly immeasurable return loss because of the custom tuning and matching afforded by MCI. The use of the six cavity diplexer also made future full MTS stereo possible. The new exciter was doing its job with it's much improved visual quality and aural signal to noise. To complete this installation a Datatek D-701 Group Delay Corrector was added to help the transmitter meet its group delay requirement. A home receiver envelope predistortion network is included in this unit and can be easily switched in and out at will for testing. The overall corrector provides many more points and methods of correction than the originally supplied RCA unit and is easily adjusted with the use of a synchronous envelope delay measuring set or a simple waveform monitor.

It was also decided during this period to also change the transmitter's present Intermediate Power Amplifiers. In this particular transmitter there are two 20 watt intermediate power amplifiers fed from the exciter's driver in quadrature by a hybrid splitter. The output of these IPAs were in turn fed to the two visual Klystron's input cavities. These IPAs needed to be linear and matched as to not complicate the predistortion necessary to drive the two power amplifiers. These solid state units proved not to be reliable in this transmitter in practice and were replaced with a single 30 watt ITS-256 solid state amplifier. This unit, as it is installed, is hybrid split in quadrature at its output. The Klystrons in this particular transmitter in visual service require less than 10 watts of actual drive to make full output power, so there is plenty of margin using this new scheme. Redundancy is provided by means of a bypass relay with the IPA itself. When the relay is activated the 3 watts available from the exciter's driver is split and sent to both power amplifiers. This provides a much more reliable and safe method of backup than the loss of a complete power amplifier. The transmitter can probably make over 30% combined visual output power in this mode.

A particular source of trouble in this and many other transmitters is the high voltage contactor. This contactor normally carries three phase 480 volt power to the main high voltage power supply. In WQTV's particular installation, this contactor is of the open contact type and is asked to carry nearly 300 amps per leg. This contactor was self latching and had no anti-pumping circuitry. Temporary overloads would send this 200 pound contactor into floor shaking oscillations. The self latching contacts on this contactor are infamous for their self induced misadjustment. This can often be a source of irritation when transmitter testing requires many on and off cycles in a short period. To improve overall reliability this contactor was replaced by a 400 amp vacuum style contactor made by Townsend Broadcast of Westfield Ma. The unit was easy to install because it is smaller than the original design. The use of vacuum contacts no only make the unit smaller, it greatly improves the reliability of the contact surfaces themselves. The new contactor also contains the anti-pump circuitry necessary to prevent dangerous contact oscillations. This unit is available in a number of different sizes and can retrofit most transmitter installations.

Much of the plumbing involved in the fresh water cooling system of this transmitter also needed to be rebuilt. All of the gate type valves provided in the original design are being replaced with chromed ball type valves. The rubber cooling hoses connecting the power amplifier Klystrons to the cooling system had become dry and inflexible. These were completely replaced with premanufactured armored cables made of space-age materials that will withstand the 24 hour a day, 180 degree water with little long term degradation. Remote sensing of the status of the transmitter's cooling system was also added to the transmitter's remote control system so that problems in this vital area could be headed off before severe and expensive damage could be done.

A simple cost analysis is helpful in making the complete case for rebuilding this transmitter. The list below includes the items described in this paper and are for reference only. There are no two transmitter installations that are the same.

Regulator w/install	$ 25,000
Combiner/Diplexer w/ loads, install & tune	95,000
Exciter	25,000
30W IPA	7,000
Group Delay Corrector	6,000
Misc. R.F. Components	1,000
Vacuum Contactor	10,000
Rebuild of Tube Cool.	5,000
	$174,000

This compares favorably with the current installed cost of a 60 Kw UHF transmitter with combining of approximately $750,000.

It is not presumed nor is it implied that rebuilding an old or unreliable transmitter is the only answer. The purpose of this paper was only to show what one station did when it was faced with a unreliable transmitter that looked as if it needed replacement. The reliability and air quality of this present rebuilt transmitter can be compared with any modern design. It would seem that the harsh economic realities of present day broadcasting environment demand the most creative approaches to problems that in the past were only solved with liberal doses of capital.

HIGH POWER ISOLATOR FOR UHF TELEVISION

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INTRODUCTION

Ferrite devices have been used for some time. Until recently they have not been used in power applications because of the high forward power loss.

This paper discusses the design, performance and application of new ferrite devices that are capable of handling 400 killowatts of forward power with less than 0.1 dB of forward power loss. These devices are now available to the Broadcast Industry.

Some of the new High Power RF Components that have been designed using these new ferrites are:

- o Impedance Stabilizer
- o Ghost Eliminator
- o Hot Switches
- o Harmonic Attenuators

NEW REQUIREMENTS FOR RF SYSTEMS

The changes occuring in the Broadcast Industry in the past few years have resulted in new requirements and new components.

Reliability -100% reliability is desired since stations are now on the air 24 hours. The system must permit maintenance and calibration during a very small window. Multiple RF paths have to be provided, in case of transmitter failure. Hand over to a new mode has to be done with no loss of air time.

Computer Control - RF components such as switches and switchless combiners must be microprocessor controlled and interfaced with computer controlled transmitter.

Handle Higher Power - Combining of two (120kW), three (180kW) and four (240kW) klystrons are now required. A much high power safety factor and more efficient components are required for these applications.

Frequency combining e.g. two 120kW TV channel into one antenna system is also becoming popular.

New Klystrons - The new all band high efficient klystrods and MSDC klystrons require a more stable impedance for harmonics as well as the fundamental.

SATISTY ATV & HDTV REQUIREMENTS - These requirements demand that the RF system be almost non-dispersive, be loss less, reflectionless and produce no cross modulation products.

GHOST LESS - Reflections must be well below preceptable levels.

NEW COMPONENTS

These requirements have generated a need for many new components:

- o Multiport switches
- o Switchless Combiners
- o Phase shifters
- o Better harmonic filters
- o Line stretchers
- o Impedance stabilizers
- o Ghost absorption devices
- o Hot switches
- o Microprocessor control

SYSTEM REQUIREMENTS

The condition for optimum performance of the RF system requires:

- o The output impedance of the generator to be exactly equal to the input impedance of the RF system.
- o The RF system to be completely transparent e.g. there should be no degradation of the video signal.
- o Reflections to be less than visible levels.

IMPEDANCE

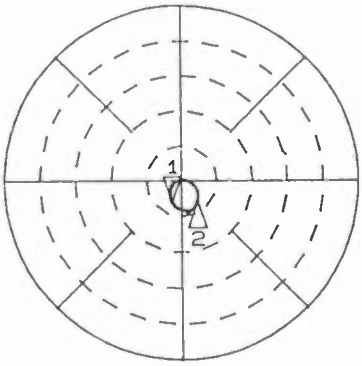
For maximum transfer of energy, the input impedance to the RF system must be equal to the output impedance of the generation. The input impedance of the RF system is dependent on the system loads.

In a TV system there are two different loads that are presented to the klystron.

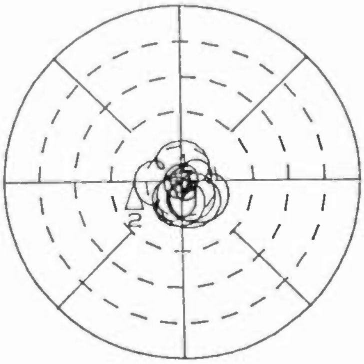
(A) The RF system with the station load on the output where the impedance is static and relative wide band.

(B) The RF system with a long transmission line and antenna on the output where the impedance is dynamic and relatively narrow band.

An example of the impedance measured at the input to the RF system is shown in the two Smith Charts, one with the station load on the RF system and the other with the antenna as the load.



INTO
STATION LOAD

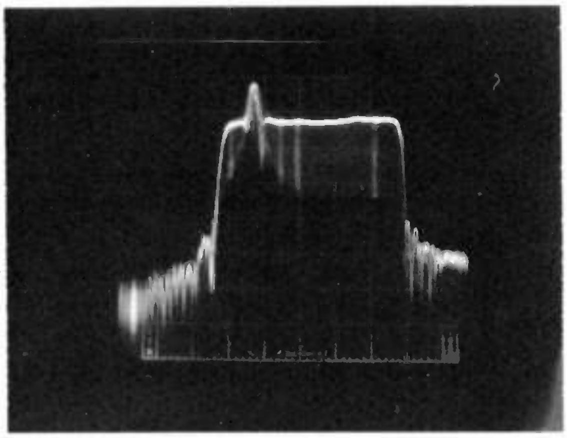


INTO
LINE & ANTENNA

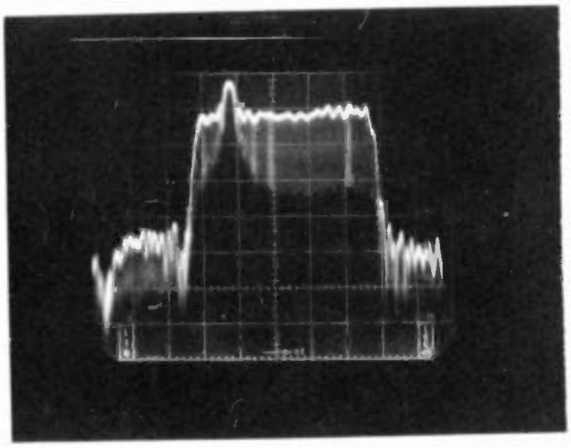
PASS BAND RESPONSE - The difference in pass band response is very dependent on the impedance change.

This is best illustrated by the enclosed pass band response which shows a relative flat response ± 1/2 dB into the station load and response of ± 2 dB with the antenna as a load.

The ripple amplitude and the frequency of the ripples are a function of the distance to the antenna and the antenna VSWR.
The power output variation of a klystron for a 1.15 VSWR varied through 360^0 is shown in Appendix A.



RESPONSE INTO
STATION LOAD



RESPONSE INTO
LINE AND ANTENNA

OPTIMIZATION - An RF system can contain over 200 components.

Because of the finite band width and large distance involved it is possible for reflection to add in phase at some frequencies. In that case, the worst case reflection (Γ) could be the sum of all the reflections (N) in the system.

$$\Gamma \text{ total} = \Sigma \ (\Gamma)$$

$$\text{VSWR} = \frac{1 + |\Gamma|}{1 - |\Gamma|}$$

It is pretty obvious that the components can not be just connected together since they could combine to a worst case VSWR of 3:1.

A schematic of a 120kW system and associated bill of material is shown in Appendix B, also tabulated is the best VSWR for each component.

An optimization process must be used. The optimization process consists of introducing conjugate matching or complementary resonant circuit. Components must be assembled as sub-assemblies, optimized, then connected as major sub-assemblies and re-optimized, until the complete system is optimized.

The process is tedious, but ultimately the complete RF system will result in system VSWR over the band width of interest equal to or less than the individual component VSWR.

As seen in the Smith Chart plot with the antenna as the load, the VSWR of the antenna will cause the impedance to change rapidly.

The frequency (Δ f) is one complete revolution of impedance.

$$\Delta f = \frac{498}{L} \times (\lambda 0 / \lambda g)$$

where (L) is the distance to the antenna. For a 2,000 ft. installation there would be 31 complete revolutions in a 6 MHz band.

The impedance will also change due to daily thermal changes. Daily variation of temperature can result in the line on a 2,000 ft. tower expanding one wavelength or 360°. The impedance of the complete system will rotate as a cluster about the center of the Smith Chart. This will result on power output changes that are much greater than the transmission loss due to VSWR variation.

SYSTEM TRANSPARENT

Ideally, the RF system should be transparent e.g., there should be no transmission loss or group delay.

Transmission Loss - The transmission loss in an RF system is that power which is lost due to reflections.

$$\text{Transmission Loss} = 10 \log (1 - |\Gamma|^2)$$

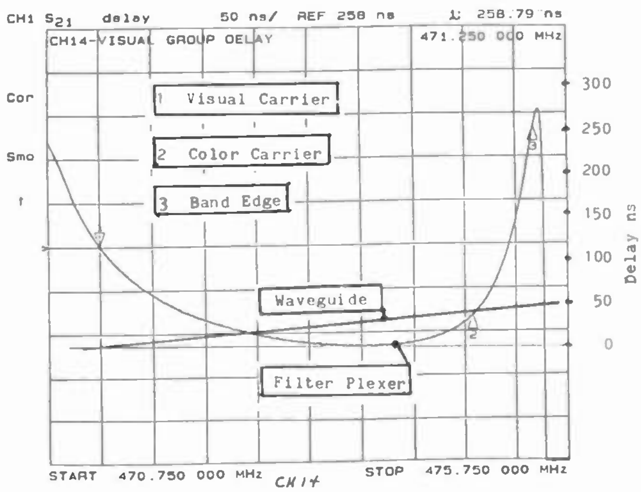
VSWR	Transmission Loss dB
1.05	.002
1.10	.010
1.20	.036
1.50	.177
2.00	.511

Group Delay - Group delay is the rate of change of phase with frequency.

For most passive RF coaxial components, this will be zero. This is not so with waveguide, where the phase velocity is greater than the velocity of light.

The resultant group delay is from two sources. The transmission line and high Q resonant circuits in the RF system.

The high Q circuits are the diplexer and filterplexer.



The variation due to the filterplexer is 120 ns, the diplexer 270 ns and the waveguide 30 ns across the visual band of 4.18 MHz.

ATV and HDTV systems will be very sensitive to group delay variation due to the additional carriers and side bands that will be added.

REFLECTIONS

Reflection from anywhere in the system will not only combine with other reflection to modify the impedance, but each individual reflection can cause a ghost. The ghost by definition is an exact reproduction of the original signal.

Reflections less than 250 nanosec are not normally a problem although they can cause edging effects.

Reflections greater than 250 nanosec can result in visible distortion to the recovered picture. This can be seen at edges of either a white or black trailing images of each sharp transition in the picture. Reflections greater than 500 nanosec will appear as a separate image (ghost).

When the number of lines are doubled as is proposed for HDTV, reflections greater than 250 nanosec will appear as a separate image.

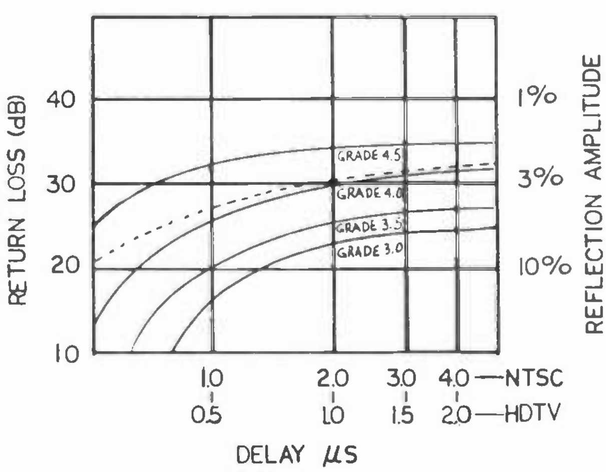
The relationship between reflection delay in nanosec, position and displacement of ghost on a TV monitor is shown below.

Distance Ft.	Delay ns	Displacement 25" monitor NTSC	HDTV
125	250	0.09"	0.19"
250	500	0.18"	0.38"
500	1000	0.36"	0.76"
1000	2000	0.72"	1.53"
2000	4000	1.44"	3.06"

Most of the reflections within the transmitter room will be within 125 ft. from the klystron and will not result in a ghost.

It is generally conceded by Lessman, Daderian & Wright that reflections greater than 2% (VSWR 1.04) are visible by a critical observer. Reference 1,2,3.

The dotted curve is based on recent studies of reflection in a mutitower environment [Vaughan & McClure].



GHOST DELAY IN AMPLITUDE FOR GIVEN PICTURE GRADE TYPICAL VIEWER SAMPLES

GRADE	IMPAREMENT
5	Excellent
4	Good
3	Fair
2	Poor
1	Bad

APPLICATION OF HIGH POWER ISOLATOR

The high power isolator will sclve some of the problems in RF transmission.

- o Stabilize the impedance
- o Reduce intercarrier distortion
- o Eliminate ghosts

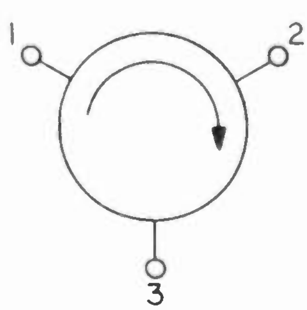
As discussed in the previous section, small impedance changes can result in large changes in power output of the UHF klystron. A high power isolator will provide a constant impedance to the generator and with isolation levels of 30dB or greater. It will also protect the klystron from even the largest of impedance changes, including catastrophic change due to short circuits and lightning hits. The isolation is sufficiently high to stabilize varying impedance and reduce inter-carrier phase distortion. Reflection from the antenna (ghosts) will be totaly absorbed in the third port of the isolator.

Another application of the isolator will be as a hot switch. The output switch on all transmitters is either a patch panel or a motorized switch, in either case the transmitter must be shut down before switching can take place. This requires the use of remote controls and various answer back commands to insure the switch has been fully engaged before transmitter power can be reapplied.

With the addition of a remotely controlled power tuner on the output of the isolator, the RF power can be switched from the antenna to the station load without shutting down the transmitter.

DESCRIPTION

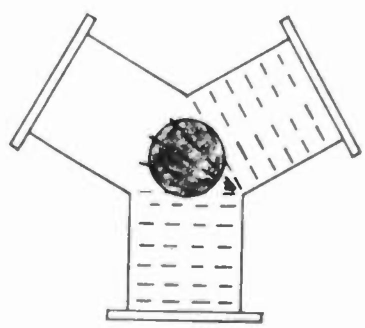
The high power isolator is a 3 port device. Reference (4)



A wave entering at port 1 comes out at port 2. A wave entering at port 2 comes out at port 3 etc. This unique property of circulating the waves around as indicated by the arrow makes it an ideal device for matching, isolating and switching.

If a load is used to terminate port 3 a wave from the generator enters at port 1 and comes out at port 2 going to the antenna. Reflections from the antenna enter the device at port 2 and are absorbed by the matched load at port 3. The generator always sees a perfect match at port 1 - independent of the impedance of the antenna at port 2. Also the antenna sees a perfectly matched generator at port 2 - even if the internal impedance of the generator is not matched; since the generator and antenna are isolated.

The basic principle of operation is the deflection of the plane wave by a premagnetized ferrite cylinder.



The ferrite cylinder is located in the center of a Y-junction of three waveguides. The ferrit cylinder is magnetized perpendicular to the plane of the waveguide. The wavefront of an incoming wave at port 1 is deflected and goes out at port 3.

In the past these devices were available for low and medium power applications up to a few kilowatts. The main problem in high power applications is the removal of the heat dissipated inside the ferrite cylinder. This required the development of a new ferrite material that had very low loss. The low loss permitted the use of more efficient and smaller cooling techniques. The new low loss ferrites have an insertion loss of 0.03 dB. To remove the heat, the ferrite cylinder is divided into several thin ferrite discs which are bonded in pairs to water cooled metal carrier plates.

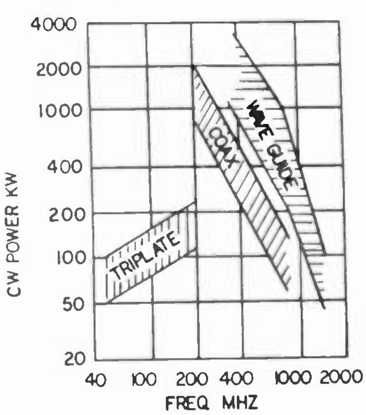
The power handling capabilities depends on the transmission line as well as the ferrite material and cooling techniques.

A typical waveguide device has a ferrite loss of 0.03 dB. For a 120kW TV transmitter ferrite losses of 400W would have to be removed. This is done by circulating water through the ferrite carrier.

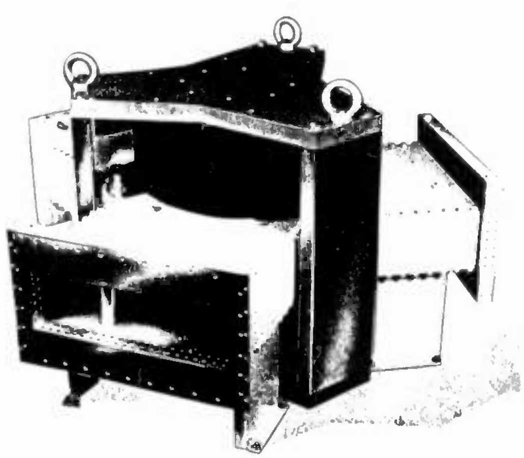
The mircowave behavior depends on the saturation magnetization of the ferrite material and the magnitude of the premagnetizing DC-field. The saturation magnetization of microwave ferrites is temperature dependent and is influenced by:

(A) Cooling water temperature
(B) Heating by RF losses. i.e., it is dependent on RF power level.
(c) Ambient temperature

The upper and lower limits of CW power handling show that devices are available that can handle up to 4 million watts.

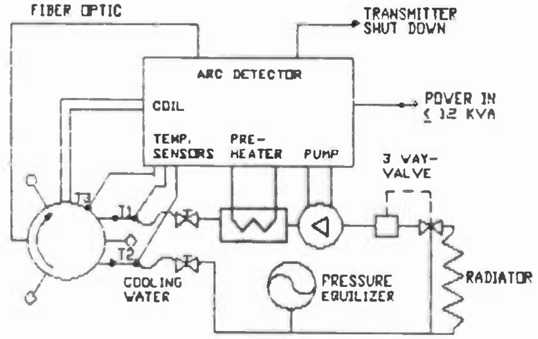


C.W. POWER HANDLING CAPABILITIES



SUPPORT SYSTEM

The cooling system is a closed system. The water is circulated by a pump and takes the heat from the ferrites and radiates it to the ambient air. A temperature controlled three way valve bypasses the radiator and stabilizes the water temperature for the circulator. The operating temperature must be above ambient and is between 40 and 55 deg. C. Temperature sensors, sense input, output and ambiant water temperature. From this information a correction current is fed to the coil. The unit automatically adjusts itself to all temperatures. An optical arc detector is included in the support system.

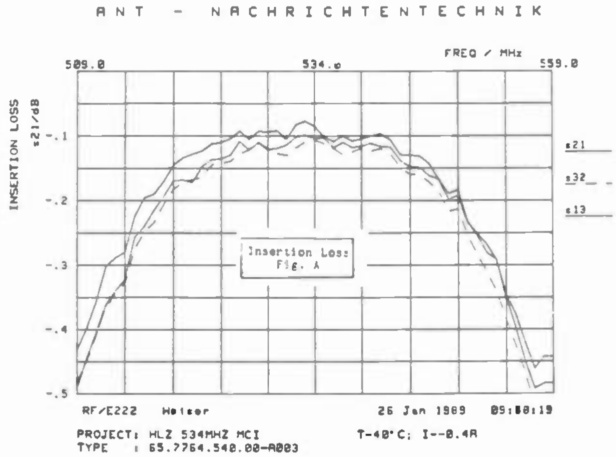


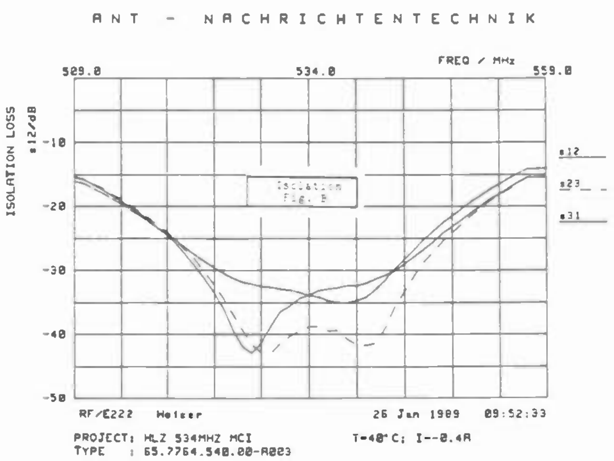
SUPPORT SYSTEM

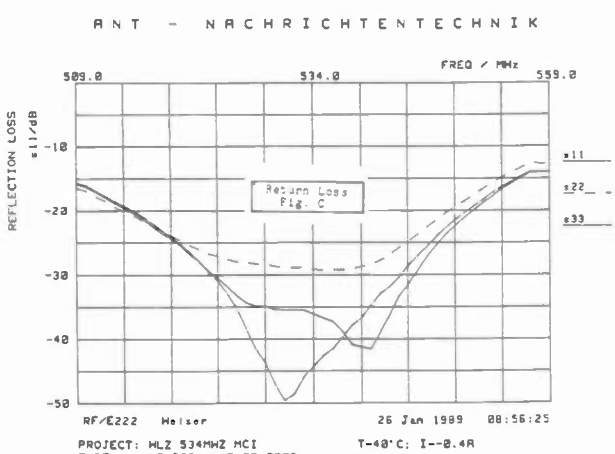
PERFORMANCE

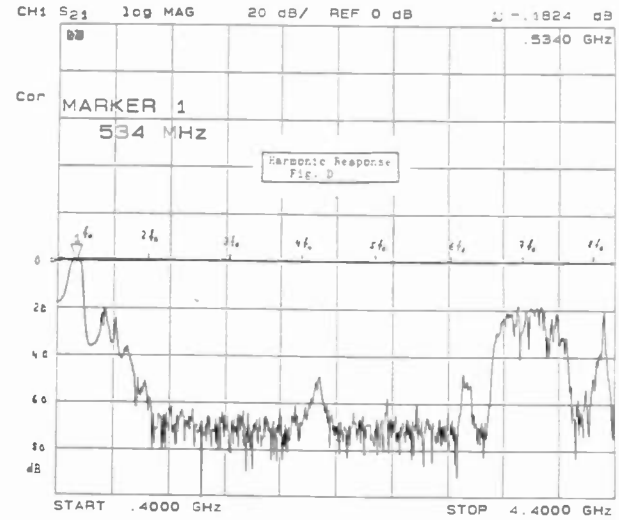
The enclosed data show the performance of the isolator used in the Channel 24 tests.

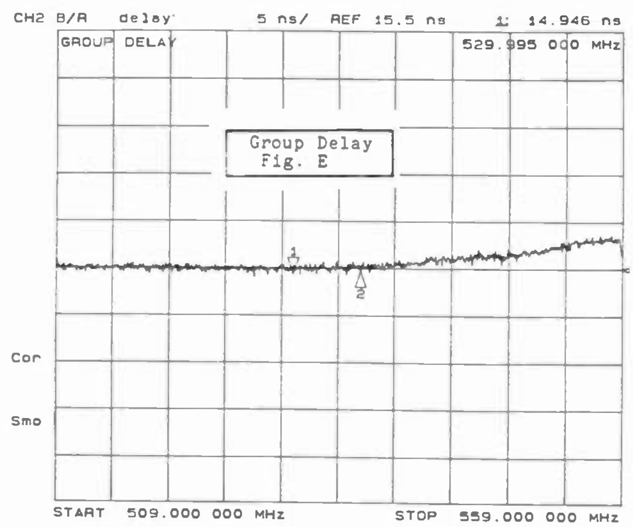
Insertion Loss Fig. A	0.1 dB	± 6 MHz
Isolation Fig. B	34 dB	± 6 MHz
Return Loss Fig. C	34 dB	± 6 MHz
Harmonic Atten. Fig. D	60 dB	2nd Harmonic
	65 dB	3rd Harmonic
	60 dB	4th Harmonic
	65 dB	5th Harmonic
Group Delay Fig. E	2 ns	











The first high power device was installed and tested at WEDH-24, Hartford, CT. The test parameters were:

Location	Input to Antenna Line
Power	60kW
Freq. MHz	530 - 536
VSWR Output	1.10, 1.20, 1.50

ACKNOWLEDGEMENT

Micro Communications, Inc. and ANT Telecommunications are jointly developing new products using the ferrites developed by ANT Telecommunications.

We would like to thank Jack Kean, Consultant and Fran Abramowicz, Director of Engineering of the Conn Broadcast System, for their permission and help in conducting the high power test.

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(2) Daderian & Wright "Predicting TV Ghost Interference & Picture Quality".
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(3) Vaughan & McClure "Reflection & Ghosts in a Multitower Environment".
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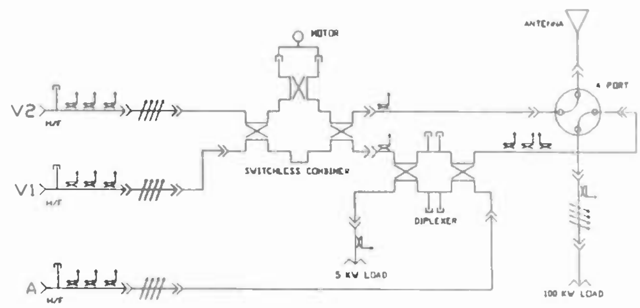
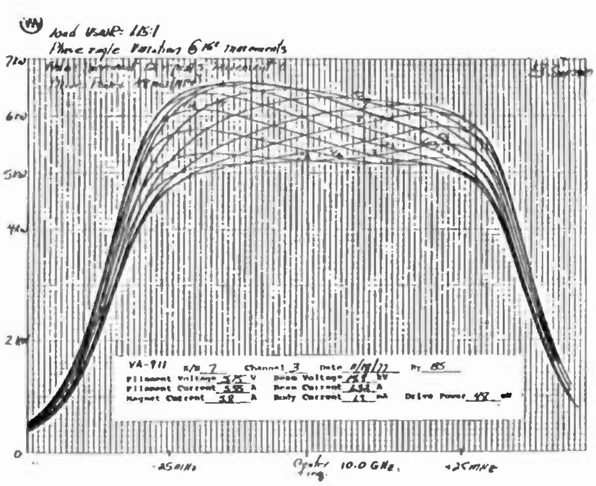
ANT Telecommunications



RF SCHEMATIC OF 120kW
RF SYSTEM



KLYSTRON POWER OUTPUT RESPONSE FOR 1.15 VSWR WITH PHASE CHANGE EVERY 15^{0}

APPENDIX A

Line between Klystron & Combiner

4	Elbows	1.03
1	Expansion Section	1.05
1	Harmonic Filter	1.05
1	Directional Coupler Section	1.03
4	Lines	1.01

Combiner, Diplexers and Switches

1	Transition	1.05
5	Hybrids	1.03
2	Tuners	1.01
2	Cavity Sections	1.05
8	Elbows	1.02
1	Switch	1.05
3	Directional Coupler	1.03
1	Test Load	1.08

Transmission Line

180	Line Sections	1.001
5	Elbows	1.02
1	Gas Barrier	1.05

Antenna	1.04

BILL OF MATERIAL

NUMBER, INDENTIFICATION AND TYPICAL VSWR OF A 120 kW RF SYSTEM

APPENDIX B

A 60 KW UHF-TV MSDC KLYSTRON TRANSMITTER

Earl McCune
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THE VARIAN KLYSTRON AMPLIFIER – INTRODUCTION

The Varian VKP-7990 60 kW UHF-TV klystron amplifier incorporates multistage depressed collector (MSDC) technology to provide extraordinarily high efficiency operation. This klystron represents the culmination of a development effort initiated in 1984 and supported by NASA, NAB, PBS, transmitter manufacturers, and Varian. Progress on this program has been reported annually to the NAB[1,2,4,6] along with technical reports to the IEEE[3,5]. This report describes the development efforts of the past year as the tube was evaluated to verify its suitability for entering commercial service. During the same time, a power supply design compatible with the unique requirements of the multiple collector elements was developed. The final result is a complete transmitter, which has been evaluated in TV service and has demonstrated efficient operation.

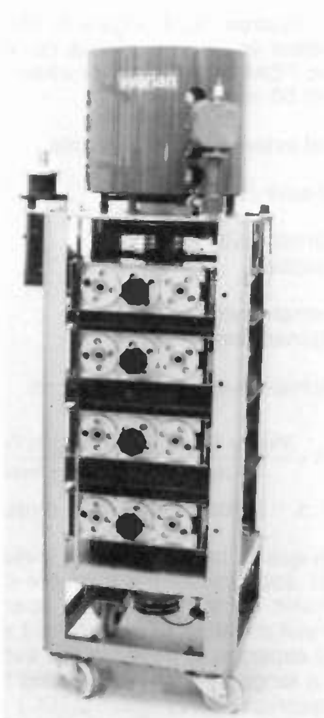


FIGURE 1. VKP-7990 MDC KLYSTRON

Klystron Development

A klystron with four external cavities based on the TEV PT-5090 was selected for application of the depressed collector technology. The resulting klystron is designated the VKP-7990, which is shown in the photo of Figure 1. The performance characteristics are listed in Figure 2. The enhanced efficiency performance is achieved entirely by the power recovery feature of the new collector design. All other tube elements are unchanged; electron gun, cavities, focusing magnet, etc. Consequently, the life and reliability inherent in the standard klystron will be maintained as long as the new collector design meets these standards. Therefore, considerable attention was directed toward ensuring the new collector design will meet the stringent life and reliability standards expected for the klystron.

- Description
  - Four External Cavities
  - Tunable, 470 to 810 MHz
  - Water and Air Cooling
  - Includes: MSDC Collector, Modulating Anode, Beam Control Electrode

- Performance

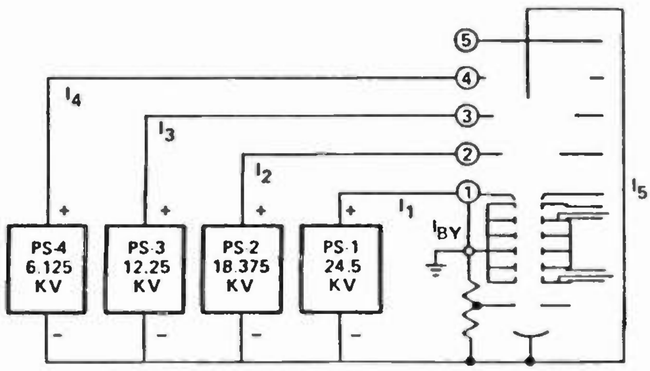
Output Power	64 kW Saturated
Drive Power	20 W
Bandwidth, 1 dB	6 MHz
Figure of Merit	Exceeds 130 in Typical TV Service

FIGURE 2. VKP-7990 PERFORMANCE CHARACTERISTICS–MULTISTAGE-DEPRESSED COLLECTOR KLYSTRON

To date, five tubes of this type have been constructed and evaluated. A test facility was established that uses the parallel power supply configuration shown schematically in Figure 3. Typical television operation includes pulsing the beam control electrode to achieve optimum performance. The test data of tube S/N L8-088 is shown in Figure 4 for operation in this manner. The Figure of Merit for this operating condition is 1.36.

Potential Problem Areas

The primary potential problem that we have been concerned about has been regeneration due to electrons turned around in the collector and returned to the beam in the opposite direction. Any depressed collector design has this potential problem and could lead to nonlinear performance. We ad-



PARALLEL CONFIGURATION

FIGURE 3. POWER SUPPLY SCHEMATIC

VKP-7990 S/N L8-088

SUPPLY	VOLTAGE	AVERAGE OUTPUT POWER	32.9 BLACK	18.9 AVERAGE	12.1 kW WHITE
1	25.4	CURRENT	0.922	0.440	0.355 A
		POWER	23.413	11.175	9.014 kW
2	19.05	CURRENT	0.915	0.596	0.347 A
		POWER	17.438	11.350	6.612 kW
3	12.7	CURRENT	1.171	1.107	0.435 A
		POWER	14.876	14.060	5.527 kW
4	6.35	CURRENT	0.230	1.200	2.355 A
		POWER	1.461	7.621	14.956 kW
TOTAL POWER			57.188	44.206	36.109 kW

FIGURE OF MERIT = 60/44.2 = 1.36

FIGURE 4. PARALLEL CONNECTED PULSED POWER SUPPLY REQUIREMENTS

dressed this concern in two ways: first by careful design of the collector electrode geometry to ensure no regions of reflecting electric field; and second, by coating the electrodes with a material with superior electron absorption. Analyses during the design phase indicated these methods should avoid the problem and allow satisfactory operation. Test data validates that conclusion. Regeneration is detectable but is at least 20 dB below the signal level and consequently does not disturb the klystron linearity.

A problem did arise due to outgassing of the collector coating material. While this material absorbs electrons well, it also absorbs gas atoms and consequently is difficult to evacuate to a high vacuum. To solve this problem, we improved the vacuum pumping capability during the processing of the klystron and are now able to achieve a satisfactory vacuum level.

The collector elements are water-cooled, but each element is at a different voltage, leading to concern about electrolysis at the coolant fittings. We have addressed this problem in three ways. First, the external coolant connections are at ground potential to avoid all problems outside of the klystron. Coolant hoses are enclosed in the collector shroud of adequate length to ensure a gradient of not more than 2 kV/foot. Second, each positive electrode is fitted with a replaceable sacrificial element to protect against tube damage. Finally, the water coolant purity is to be maintained with high resistivity. Operation to date has shown no deleterious effect from this problem.

Since the current distribution to the collector electrodes changes as a function of power level, there are video currents on these elements. Consequently, there is concern about radio frequency interference (RFI) problems due to the collector. To address this concern we have completely enclosed the collector with a shield. In addition, the collector element leads are filtered. No RFI problems have been noted to date.

Each potential problem has been addressed and satisfactory solutions have been achieved. Therefore, we feel confident that the VKP-7990 MSDC klystron will provide the same reliable service that the previous standard klystrons have provided for years. At this point, our laboratory tests support this expectation.

The MSDC Klystron – Conclusion

It has been over eight years since the concept of applying depressed collector technology to UHF-TV transmitters was recognized. Since that time the television community has worked together to make the concept a reality. A million dollars of funding was secured to enable the development program to begin in 1984. That program produced a demonstration klystron that met all expectations; the concept was validated. Since the completion of that program, Varian has continued the development effort to achieve an MSDC klystron that meets all of the exacting standards expected by the television community. The VKP-7990 klystron has successfully been put to the test and is now ready to provide commercial service.

THE MSDC TRANSMITTER – INTRODUCTION

Klystrons have been used as high-power amplifiers from the time UHF television was introduced to the present day. Successive generations of integral and external cavity designs have achieved improved gain, bandwidth, power, tuning range, and efficiency. As station transmitter powers increase, efficiency remains the parameter offering the best payback for improvement.

The evolution of klystron FOM (Figure of Merit) is shown in Figure 5. The latest version of external-cavity klystrons has achieved a basic FOM of 50%, which when pulsed may be raised to between 60 and 75%.

Early integral and external-cavity klystrons	30-40%
Modern external-cavity klystrons	40-50%
ACE pulsed external-cavity klystron – first generation	50-60%
ACE pulsed external-cavity klystrons – present generation	60-75%
ACE pulsed external-cavity MSDC klystrons	140%

$$\text{*Figure of Merit} = \frac{\text{Peak of Sync Power at Klystron Flange}}{\text{DC power into the klystron beam}} \times 100\%$$

FIGURE 5. FIGURE OF MERIT* EVOLUTION

To achieve high-quality performance at these levels of efficiency demands sophisticated exciter pre-correction, with particular emphasis on the stability of operation. This is achieved by careful attention to the inherent stability of each function. All the expertise acquired in this evolution has now been applied in a range of transmitters using the Varian four-stage multiple depressed collector (MSDC) klystron, which has enabled the FOM to be raised to 140% with average picture modulation.

Implementation of the MSDC Klystron

The decision to develop an MSDC klystron transmitter was based on the following considerations:

- The MSDC klystron is identical to a well-proven modern external cavity tube up to the last interactive gap. The collector is changed to the MSDC design, which collects the used beam more efficiently with less waste heat.
- Klystrons are inherently rugged and are well-known for long life.
- The MSDC klystron achieves high gain.
- Tuning techniques are simple and well understood.
- One version of klystron and cavity covers the entire UHF band.
- Normal klystron beam voltages are used permitting air-cooled supplies.
- The ability to use highly developed pre-correction and pulsing techniques improves performance.

The choice of an MSDC klystron as a means of improving transmitter efficiency has allowed a new design to be introduced relatively quickly and with a minimum of engineering risk. Essentially, only the HV power supply and cooling are changed. The ability for tube and transmitter designers to work closely on the first demonstration of the klystron design has brought the benefits of MSDC operation to the market in record time for such a significant improvement.

Transmitter Efficiency

The power consumption of the MSDC Transmitter compared with previous generations of equipment is shown in Figure 6. Corresponding savings in electricity costs are shown in Figure 7.

	Consumption kW	Saving Ref 1 kW	%	Saving Ref 2 kW	%
1. Older generation transmitters using "A" series integral-cavity tubes 1969–1972	250	–	–	–	–
2. Best modern transmitters using PT5093 ACE external-cavity tubes. Visual pulsed. 1988	118	132		–	–
3. VISTA MSDC 1989	75	175	70	43	36

NOTES:

a. For comparison with 2 and 3, the consumption for 1 has been calculated for a 60 kW output although the transmitters of this period were rated at 55 kW.

b. Tx 1 uses "A" series integral-cavity tubes (basic FOM = 30). No pulsing. Tx 2 uses PT5093 ACE external-cavity tubes (basic FOM = 52). 30% pulsing. Tx 3 uses VKP-7990 in visual and PT5080 ACE in aural.

c. Aural ratio in all cases is 10%.

FIGURE 6. COMPARISON OF POWER CONSUMPTION USING VISTA MSDC FOR A 60 kW TRANSMITTER

	Annual Saving Ref Older Generation $	Annual Saving Ref Best Modern $
1. 60 kW installation	92,000	22,600
2. 120 kW installation	191,300	52,700
3. 240 kW installation	381,700	104,500

NOTES:

a. For comparison with 2 and 3, the cost for 1 has been calculated for a 60 kW output, although the transmitters of this period were rated at 55 kW.

b. Annual cost based on 18 hours/day and 8 c/kWH.

FIGURE 7. COMPARISON OF ELECTRICITY COST SAVINGS USING VISTA MSDC

These savings are for a pulsed MSDC klystron with an FOM of 140, which has been achieved using normal klystron operating procedures. It is worth noting that at these levels of efficiency the "background" consumption of other parts of the transmitter is increasing in relative size. Further FOM improvement would therefore be subject to the law of diminishing returns.

The Range of Transmitters

We have followed our usual practice and designed two basic transmitter modules which can be assembled in many different combinations to provide all the required power and reserve modes.

The two configurations discussed in Figure 8 are

- The type 1890, which is a single VKP-7990 MSDC klystron amplifier complete with its own power supply and, if required, exciter.
- The type 1891, which contains two klystrons and can act as a complete 60 kW transmitter with a power supply and exciter. A VKP-7990 MSDC klystron is used for the visual amplifier and a normal PT5080 external-cavity klystron for the aural.

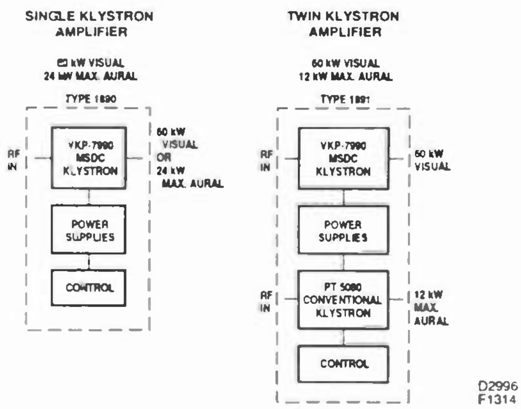


FIGURE 8. VISTA MSDC KLYSTRON TRANSMITTERS

To achieve 60 kW, a single type 1891 may be used. The alternative approach would utilize two type 1890s (one for the visual and one for the aural), both fitted with MSDC klystrons.

At 120 kW, one type 1891 and one type 1890 or three of the type 1890s, each separately controllable, are possible choices.

At 240 kW, either five or six klystron versions may be used. Our preferred combinations are one type 1891 and three type 1890s, or five of the type 1890s.

All the usual reserve modes are possible using waveguide switches or switchless combiners for hot switching after an amplifier failure.

Transmitter Layout (Figure 9)

A design objective was to retain the small physical size of the existing klystron transmitters. This is attractive to both the user and the manufacturer, as well-proven designs can be employed. As usual, the air-cooled power supply is integrated. Underfloor access is not required, and no rear access is needed for normal operation.

FIGURE 9. 60 kW TRANSMITTER WITH MSDC VISUAL KLYSTRON

All low-level circuits are in modules with plug and socket connections. Extension boards are not required anywhere in the signal path. Remote control and monitoring connections are all marshalled together. All connections to the klystron and its circuit assembly are ergonomic and carefully labelled. The same magnetic circuit and cavities are used as for the single-collector external-cavity klystrons.

The use of locked doors eliminates the need for additional covers for high-voltage components and thus reduces the danger of arcing to ground.

Operation

MSDC klystron tuning is exactly the same as for a conventional external-cavity klystron, and is entirely predictable and straightforward. The cavities are carefully designed for temperature stability, and the magnetic circuit is massive and rugged. All tuning controls are equipped with counters and are accessible from the front (Figure 10).

All klystron parameters are metered with a separate indication for each collector. Power normal indicators are fitted on the output of all exciter and IPA modules to give immediate indication of faults. The integrated control and monitoring unit provides separate latched indication of all normal and fault states; all local metering, monitoring, and controls are repeated remotely.

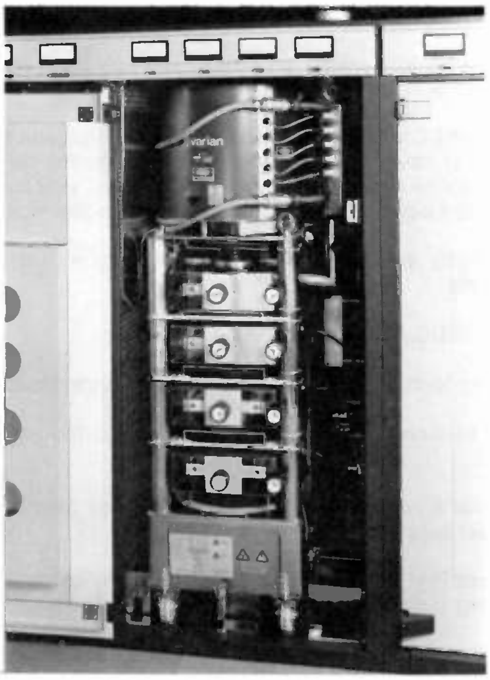


FIGURE 10. VKP-7990 KLYSTRON MOUNTED IN THE VISUAL SOCKET

Beam Pulsing

A third-generation beam pulser is used that provides the full 1200 V permitted for the tube with 10 mA capacity to allow for a possible increase in ACE current as the tube ages. Separate controls for sync pulse amplitude and shape, together with advanced pre-correction techniques, allow good sync pulse shapes to be achieved.

As well as correcting the violent phase changes in the sync pulse region, the exciter allows adjustment to the sync drive level. This enables the klystron output coupling to be safely optimized and prevents possible overdriving.

The limit to correction is now the stability of all the circuit elements. Careful control of the signal level in the pre-correction stages by alc loops permits operation at powers up to 0.5 dB below saturation in the picture with good performance parameters.

Performance

The extra stages added to the collector to change the klystron are all in the region after the signal amplifying interactions of the beam. Klystron performance is, therefore, only slightly affected. The influence of electrons produced by secondary emission is primarily seen as a redistribution of current between the collectors. Therefore, secondary emission has been minimized to ensure that the figure of merit is not degraded. It is just possible to detect the presence of secondary emitted electrons in the drift tube, but their minimal effect has only a slight influence on linearity.

The klystron is set up to produce low-frequency nonlinearity of 60-70%, with corresponding differential gain and phase. These values are readily correctable and are held stable by the exciter circuits (Figure 11).

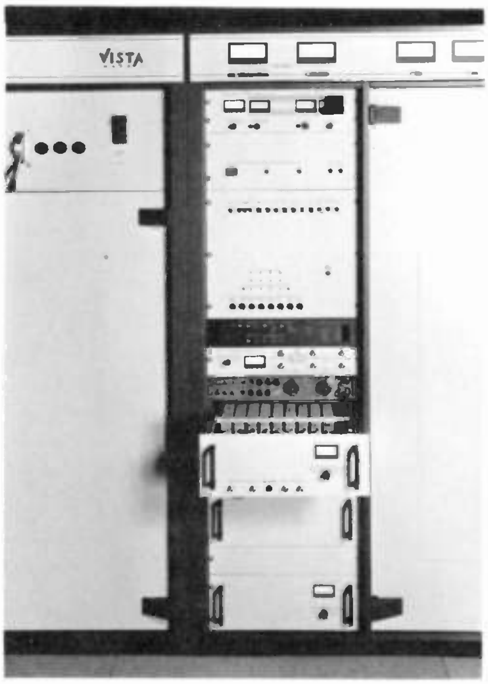


FIGURE 11. EXCITER AND CONTROL CIRCUITS IN THE MSDC TRANSMITTER

Klystron Beam Supplies

The novel requirement to supply four collectors may be addressed either by a series or parallel arrangement of supplies. Both methods have been investigated and used. A series supply has been chosen for technical and economic reasons.

Although the current to each collector varies widely with instantaneous output level, the total current stays within narrow limits. Using a single transformer for all supplies thus minimizes the size of iron core needed. The usual 12-pulse rectification technique provides inherently low ripple and reduces the need for filtering. The increased size and complexity of the rectifier stack is minimal beyond that of a normal beam supply because the total potential of the four supplies is the same as for a normal klystron, i.e., 24.5-27.5 kV. This maximum voltage allows the use of an environmentally acceptable air-cooled transformer and eliminates the hazards of oil-cooled supplies.

A simple 6 A current-regulated supply feeds the refocusing coil at the collector end of the klystron. Its adjustment, which is made for best distribution of current between collector elements, is not critical.

Cooling

The transmitter is cooled by air passing in and out of the top of the cabinet. Like all electronic equipment it benefits from constant temperature but is designed to handle inputs up to +45°C. An internal high-pressure blower cools the IPAs, exciter, and klystron circuit cavities.

The MSDC klystron collector is water-cooled, but all other parts are air-cooled as are all modern external-cavity tubes.

The collector stages are water-cooled by a single circuit that loops through each one. The high voltage on the stages increases the importance of water purity. To ensure this purity, a two-stage system is used with a water-to-water-plate heat exchanger separating the primary and secondary circuits.

In the primary circuit, 10% of the coolant constantly flows via a bypass "polishing" circuit. This contains long-life resin deoxygenation and deionizing elements to keep the water purity high. A purity meter with warning alarm is provided in the main primary flow to the collector. Copper in the circuit is minimized by the use of stainless steel and plastic.

Advantage is taken of the secondary circuit to provide a water/glycol mixture to the external heat exchanger, permitting operation in low-temperature environments.

Safety and Protection

Operator protection is a prime consideration. All hazardous potentials are behind locked doors and accessible by release of trapped keys only after all dangerous voltages are visibly grounded. The many potential hazards of the klystrons are safely contained with all controls brought to the front panel and accessible via inner doors. Comprehensive monitoring of the presence of all vital supplies protects the valuable klystrons from any circuit or cooling failure. Means are provided to check all detection and overload circuits routinely.

The MSDC Transmitter – Conclusion

The new range of MSDC transmitters embodies the well-known virtues of the traditional klystron – long life, stability, ruggedness, and ease of operation. At the same time, the MSDC klystron has the best figure of merit of any tube: this results in significant lowering of power consumption for the transmitter.

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LIGHTING FOR HDTV

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Abstract

On February 3, 1989, Sony introduced the 300 Series HDTV camera to SMPTE in San Francisco. The 300 camera had a number of modifications and improvements over the aging 100 Series camera. The one point that caught the interest of many was the increase in sensitivity on the new camera. There they were, side by side, six year old technology and the newest and latest in electronic imaging. The 300 Series was one and a third stops faster than the old series. The 300 Series creates new applications for advanced television usage, which will challenge the use of 35mm film in field production.

The lighting for the SMPTE presentation was created by HMI. The intentions was to match the monitors with the light source to allow the viewers of both the live set and the monitor a consistent color temperature which was daylight.

The Look

The advanced television concept, in addition to its other merits, celebrates two intentions. The first is increased resolution and second is an advance in contrast ratio over existing NTSE standards. ADTV is directly pursuing the image qualities found in 35mm film. For us to begin to draw nearer to the look of film, we must begin to address the merits of the filmaker's concern with lighting as an essential element in the production process, without which no dimensional definition of image can be achieved. What this means is that the videographer can no longer think of lighting video by using "flat lighting" or a two to one lighting ratio.

The Light

During the past two decades there has been a revolution taking place in the world of lighting. This upheaval in lighting primarily focused on location motion picture production. This transformation was brought about by the advent of the A.C. enclosed arc discharge lamp, more commonly known as an HMI.

Cinematographers have been applauding the efficiency of HMI's and have been using these instruments almost exclusively for interior and exterior location film work. Many video crews are simply not yet acquainted with the HMI and its flexibility.

What exactly is an HMI and how does it work? An HMI is an A.C. enclosed arc discharge lamp. An arc on the order of one half to one inch long is enclosed in a quartz double ended arc tube where a mercury arc heats metal halide and rare earth additives until they radiate. The mix of these additives is essentially a continuous spectrum with an initial color temperature of about 5600 degrees Kelvin.[1]

The color temperature is usually described in terms of degrees Kelvin. Tungsten halogen, also known as quartz iodine, has a color temperature of 3200 degrees Kelvin. HMI's have a relative temperature of 5600 degrees Kelvin.

Correlated Color Temperature of Typical Light Sources

Artificial Light			
Source			**Mireds**
Match flame	1700K		588
Candle flame	1850K		541
Tungsten-gas filled lamps:		Camera filter	
40-100W	2650-2900K	82B (100W)	317-345
200-500W	2980K	82A	336
1000W	2990K	82A	334
Daylight			
Sunlight:			
Sunrise or sunset	2000K		500
One hour after sunrise	3500K		286
Early Morning, late Afternoon	4300K		233
Average noon, (Wash. D.C.)	5400K		185
Midsummer	5800K		172
Overcast sky	6000K		167
Avesage Summer Daylight	6500K		154
Light Summer Shade	7100K		141
Average Summer Shade	8000K		125
Partly cloudy sky	8000 -10000K		125-100
Summer skylight	9500 -30000K		105-33

Sunlight should not be confused with Daylight. Sunlight is the light of the sun only. Daylight is a combination of sunlight plus skylight. These values are approximate since many factors affect the Correlated Color Temperature. For consistency in the text, we shall use 5500K as Nominal Photographic Daylight. The difference between 5000K and 6000K is only MIREDS (the same photographic or visual difference as the between household tungsten lights and 3200K photo lamps (the approximate equivalent of ¼ Blue or 1/8 Orange lighting filters).

Color temperature describes the actual temperature of a "black body radiator" and thereby completely defines the spectral energy distribution (SED) of the object. When the object becomes luminous and radiates energy in the visible portion of the spectrum, it is said to be incandescent. Simply stated, this means that when an object is heated to an appropriate temperature some of its radiated energy is visible.[2]

Comparison of Photographic Light Sources

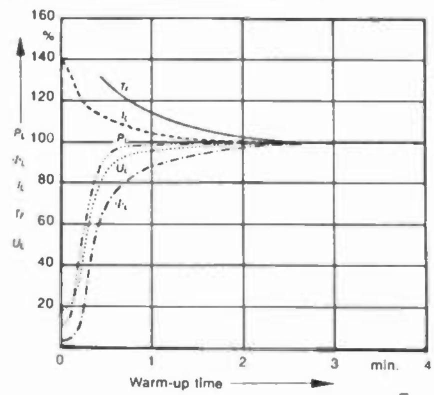
Description	Correlated Color Temperature (at rated voltage)	Mired Value	Efficacy Lumens/watt
Incandescent			
Standard and tungsten/ halogen	3200K	313	26
CP gas filled	3350K	299	32
Photoflood	3400K	294	34
Daylight blue photoflood	4800K	208	
Carbon arc (225A Brute)			
White flame, Y-1 filter	5100K	196	24
" " no filter	5800K	172	
Yellow flame YF 101 filter	3350K	299	
*Xenon, high pressure DC short arc	6000K	167	35-50
*Metal halide additive AC arc			
HMI	5600K	179	80-102
CID	5600K	179	80
CSI	4200K	238	85

Indicated on the above charts is the MIRED value. When dealing with sunlight and incandescent sources (both standard and tungsten halogen types), the MIRED system offers a convenient means for dealing with problems of measurement when adjusting from one color temperature to another. It must be kept clearly in mind that this system is only for sources than can be truly described as having a color temperature. The term MIRED is an acronym for MIcro REciprocal Degrees. The MIRED number for a given color temperature is determined by using the following relationship:

$$\text{MIRED Value} = \frac{1{,}000{,}000}{\text{Color Temperature (degrees Kelvin)}} = \left[\frac{10^6}{^{\circ}\text{K}}\right]$$ [3]

MIRED values are not commonly sought in field production. The standard indicator is color temperature.

As mentioned, these lamps operate on alternating current only. Initial start up as well as hot restrike require high voltage ignition. There is a ballasting device which is supplied to limit current. As a general characteristic, all of these lamps tend to have a light output which is modulated in relation to time. This is due to the fact that the light output follows the current, and these lamps are operated on alternating current. As the current rises through zero and up to a maximum and back down to zero to the opposite polarity peak, the light output tends to modulate between a minimum and maximum value.[4] This point is of value in understanding that a "flicker" problem could result with a framing rate of a film camera and its shutter angle.



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HMI 2500-W power consumption P_L, luminous flux O_L, current intensity, I_L, nearest color temperature T_R, and operating voltage U_L (relative values), as a function of time after starting the cold lamp.

This rhythmical beat of light output poses no threat to video application. The continuance of the video system is such that it will bridge the pulsations and there is no possibility of "flicker".

HMI lamps require some time to "come up" to color temperature, this may vary slightly by the wattage of the lamp but usually from one to two minutes to begin shooting. The reason for this waiting period is to allow the five or so rare earth metals to heat up enough to vaporize. This vaporization produces the color temperature which we equate to daylight (5600 degrees Kelvin). After the lamp has been "burning" for at least five minutes, the fixture should be capable of being turned off and then on again to properly called "hot restrike". This again

It should be stated that there are in use, ballasts which typically have a varies from the instrument manufacturer as to whether or not their ignitor will hot "restrike".

square wave and are "flicker free".

Color temperature of the HMI lamp is affected by its use. Usually one hour of burning time reduces the color temperature by one degree Kelvin. This very gradual drop in color temperature forges the HMI as a very stable medium in comparison to incandescent and halogen sources.

Why has the HMI become the "light of choice" on locations? The answer is threefold: efficiency, efficiency, efficiency. The lumen output per wattage draw on the HMI is about 80 to 106 lumens per watt. Tungsten halogen are about 25 to 30 lumens per watt. Lumens are the total light output of a source. HMI has a three to four time lumen output as compared to tungsten sources.

HMI LAMPS—SUMMARY OF ELECTRICAL AND PHYSICAL CHARACTERISTICS

Lamp Power Rating (Watts)	200	575	1200	2500	4000	6000	8000	12000
Minimum Open Circuit A.C. Voltage to the lamp for ignition (Volts)	198	198	198	209	360	220	380	380
Lamp Operating Voltage (Volts)	80	95	100	115	200	135	220	225
Lamp Operating Current (Amperes)	3.1	7.0	13.8	25.6	24.0	55	—	65.0
Luminous Flux (Light output in Lumens)	16,000	49,000	110,000	240,000	410,000	630,000	800,000	1,008,000
Luminous Efficacy (Lumens/Watt)	80	85	92	96	102	105	100	84
Average Life (Hours)	300	750	750	500	500	350	500	—
Burning Position	Horizontal ± 15°	Any	Any	Horizontal ± 15°	Horizontal ± 15°	Horizontal ± 15°	Horizontal ± 15°	Horizontal ± 15°

6

This lumen/watt efficiency travels to the location and in terms of power needed, cable and distribution requirements are all cut back. Many locations can be lit by existing circuits thereby abolishing "tying in" for power. The lumen/watt factor also maintains a "cooler" set. By "cooler" we look again at lumen/watt efficiency. It is understood that 3.14 BTU = 1 watt. Let us suppose that our net production area (NPA) is 500 square feet, and our lighting level required is 150 foot candles. With a "quartz" package we are going to need about 20KW of power and this condition will produce 62,800 BTU's. The equal NPA with HMI as our source and our power drops to 7KW and BTU's are 22,000.

Service Power in Kilowatts (KW) For Net Production Area (NPA) And Required Footcandle (fc) Level

NPA (ft^2)	50 fc	100 fc	200 fc	300 fc	400 fc	500 fc
500	7	14	27	40	54	67
1000	14	27	53	80	106	134
1500	20	40	80	120	160	200
2000	27	54	107	160	213	267
2500	34	67	134	200	266	334
3000	40	80	160	240	320	400
3500	47	94	187	280	372	467
4000	54	106	213	320	426	534
4500	60	120	240	360	480	600
5000	67	134	267	400	532	667
6000	80	160	320	480	640	800
7000	94	187	374	560	745	934

The average illumination requirements for the various types of entertainment installations are generally as follows:

1. Typical average illumination requirements for proscenium type stages is 300 footcandles. For high school and amateur theater, a value of 100 footcandles can be used.
2. Typical average illumination requirements for color television production facilities should be 150 footcandles.
3. Typical average illumination requirements for major film studios is 250 footcandles.

The above table is based on the use of tungsten halogen lamps; if standard incandescent lamps are used, multiply the kilowatt value by 1.2.

The above table is based on the use of tungsten halogen lamps; if standard incandescent lamps are used, multiply the kilowatt value by 1.2.

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Conclusion

There are then four main areas that summarize the technical advantage brought to Advanced Definition Television production by the introduction and widespread use of the HMI:

- access to 5600 degrees Kelvin daylight color temperature light source
- cooler ambient temperatures
- greater lumen output per power draw
- less set up time with fewer fixtures needed

All these elements give the videographer a higher degree of control over the subject matter being recorded and thus offer more creative choices.

HARP—HDTV HIGH SENSITIVITY, HAND-HELD CAMERA

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ABSTRACT

HDTV is now in the stage of practicalization after holding out expectations for many years as a future television system. Practical HDTV equipment are being developed in earnest and many HDTV programs are being produced. Of the HDTV equipment, it was at a relatively early stage that HDTV cameras reached a practical level. Because its sensitivity was inferior to that of the existing NTSC camera, however, shooting has been subject to more restrictions. A high-sensitive HDTV camera, developed recently by NHK (Japan Broadcasting Corporation), provides the solution to this problem. This camera, due to its high sensitivity, about ten times greater than that of HDTV cameras currently used, allows shooting of darker scenes and covers a larger depth of field. Shooting from various angles and at various places is also facilitated due to the hand-held type. The high sensitivity is realized by employing HARP tubes (High-gain Avalanche Rushing Photoconductor).[1,2] Avalanche multiplication, occuring in each HARP tube's target layer, brings about a high gain in sensitivity. Satisfactory signal-to-noise ratio (SNR) is also attained by employing GaAs FETs for the pre-amplifiers. In addition, on-line registration correction system is employed to simplify the camera adjustment.

This camera was used in experimental HDTV broadcasting at the Seoul Olympic Games with considerable success.

This paper reports the basic characteristics of the camera, including sensitivity, SNR and resolution. Below are some of the major performance and specifications:

Performance and Specifications

· Type	hand-held (Self-contain)
· Camera tube	2/3-inch HARP
· SNR	45 dB
· Sensitivity	200 lux F2.8
· Resolution	30% (at 800 TVL)
· Registration	on-line correction system
· View Finder	1.5-inch CRT
· Size	378 (D) × 333 (H) × 108 (W)
· Weight	camera head unit 6.5kg
· Power Consumption	camera head unit 33W base-station 230W

NECESSITY OF HIGH SENSITIVITY

Sensitivity of HDTV cameras may be defined as either physical sensitivity or effective sensitivity. Physical sensitivity is defined as the amount of incident light required to generate a reference output signal level when a white object (reflection: 89.9%) is shot. This sensitivity depends almost entirely on the photoelectric transfer characteristic of the camera tube. Therefore, there appears to be no substantial difference in physical sensitivity between HDTV and NTSC cameras if the same photoelectric transfer layer is used. To be exact, however, the HDTV camera suffers some deterioration of physical sensitivity and requires more incident light (about 1.5 times), or a smaller F number (about half). This is because the reference signal current of the camera tube is set higher than in NTSC cameras (in order to improve SNR) and the total amount of light received by the scanning area of the camera tube is smaller due to the wide aspect ratio of 16:9.

On the other hand, effective sensitivity is defined by the amount of light required to obtain satisfactory HDTV pictures. The depth of field is a major factor which can restrict the effective sensitivity. This term represents the distance area a camera can keep in focus when the camera lens is focused on an object point. As is well known, brighter lenses of smaller F numbers provide a smaller depth of field. Even if a very bright (small F number) lens is developed, therefore, many portions of an object cannot be in focus with such a lens. Upshooting a man's face is an example. Although the end of the nose is in focus, the eyes may be out of focus, deteriorating the advantage of HDTV high resolution. This indicates that actually available F numbers are lower limited. Fig. 1 shows the difference in depth of field between HDTV and NTSC cameras. A, B and C are objects while A', B' and C' are their respective images. The continuous lines show the respective courses of light reflected from A, B and C when the lens is focused on A. Although the light from A converges on the surface of the camera tube, the other two are respectively spreading to PP' and QQ' on the tube surface, that is, out of focus. These spreads are no problem if they are not larger than the one picture element of the camera because the resolution of the camera is determined by the size of the picture element. If they are larger than the picture element, however, these spreads determine the resolution.

Here, let us define the depth of field as a distance which causes the same size spread as the picture element of the camera tube. The picture element in HDTV is about half the size of that in NTSC, and the depth of field is more or less halved. To obtain the same depth of field, reversely, the light flux from C must be restricted to within FF', that is, the lens must be halved in diameter. This corresponds to increasing the lens's F number by 2 and, therefore, reducing the amount of incident light to a fourth. In other words, to obtain the same amount of incident light as in NTSC cameras, object illumination must be raised four times, and six times when including the deterioration of physical sensitivity discussed earlier. Developing a high sensitivity HDTV camera has been needed to compensate these deteriorations in effective sensitivity, or to solve the substantial problem on field of depth.

CHARACTERISTICS OF HARP TUBE

Principle of HARP Tube

In the HARP tube, signal charges are multiplied in the photoelectric transfer layer. Fig. 2 illustrates this mechanism. A light beam goes through the lens and reaches the photoelectric transfer layer, generating electron-hole pairs on the surface. There, holes move with high velocity due to a strong electric field applied to the target and collide against atoms in the layer, generating other electron-hole pairs. These newly generated holes also move to generate other pairs. Such an action is called avalanche multiplication. This process is repeated to generate many holes, which eventually reach the back side of the layer and are stored there as signal charges. These charges are read out by scanning beam and output as signal current.

That is, the HARP target layer can generate more signal charges than those generated directly by the incident light (primary signal charges, equal to those generated by non-avalanche photoelectric transfer layer). In addition, the avalanche multiplication process is almost free from generating excess noise. The HARP tube can be considered a high-

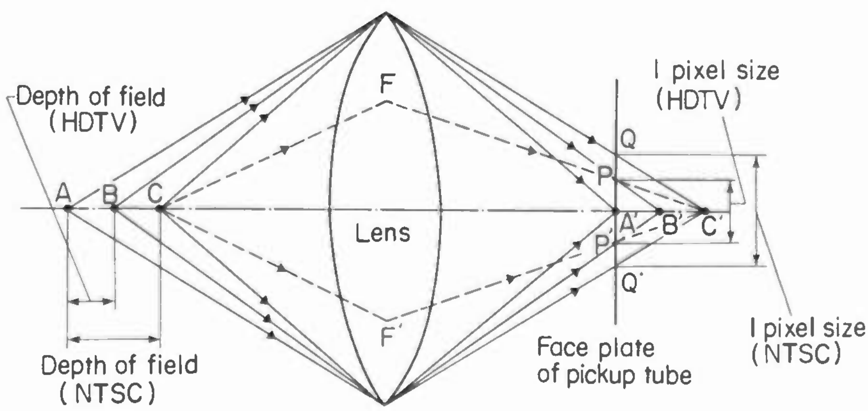

Fig. 1 Depth of Field

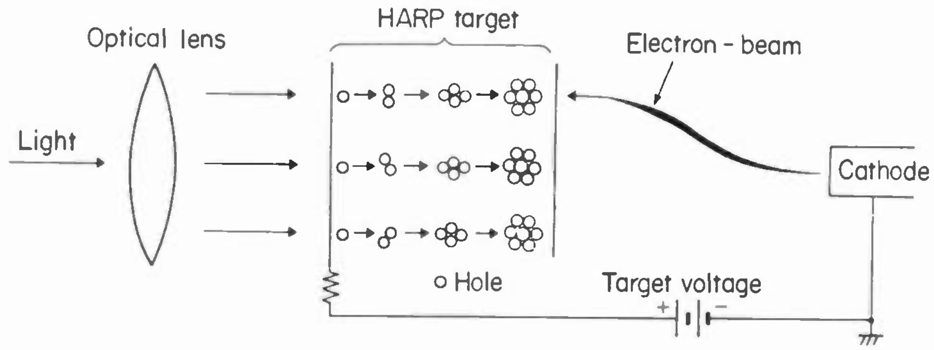

Fig. 2 Mechanism of HARP Target

sensitivity tube that can output sufficient signal current with a good SNR for a very small amount of incident light.

The avalanche multiplication of the HARP tube depends much on the voltage applied to the target. Fig. 3 shows the dependence of the signal current on the target voltage. At around 50 V, the sensitivity is equal to that of the Saticon because no avalanche multiplication occurs. The signal current begins to increase at around 200 V and shows an increase of about 10 times at between 240 and 250 V. Beyond this range, however, the dark current begins to rapidly increase (because thermally excited holes cause avalanche multiplication even when no incident light) although the current signal increases further, making it impossible to use the tube for image pickup.

Gamma Characteristic

Signal amplitude compression results from the characteristic that the sensitivity depends on the target voltage. This is because the target voltage, accelerating holes in the target layer, is modulated by the signal charges themselves. This phenomenon may be explained as follows:

Let us consider a picture element on the HARP target layer. When the electron beam is positioned onto this picture element, the HARP target's back side area of this picture element is set to the same voltage (0 V) as the cathode because the back side area is connected to the cathode through the electron beam. Then, when the electron beam moves to the next picture element, the back side area starts floating (insulated from the cathode).

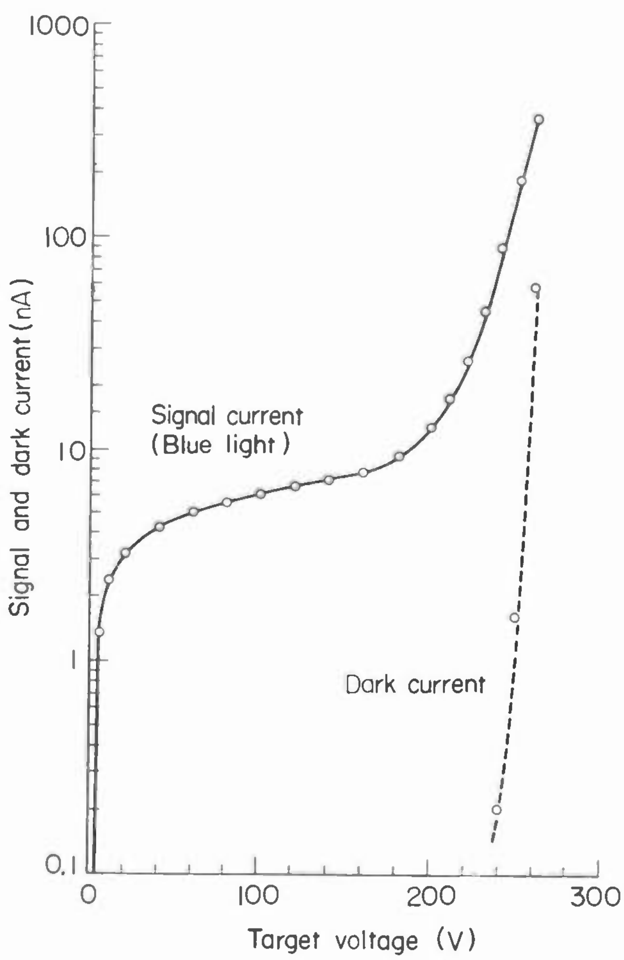

Fig. 3 Voltage-Current Characteristics of HARP-TUBE

During floating, holes excited by incident light are multiplied and accumulated on the back side because a voltage is applied to the surface of the HARP target. This accumulation changes the potential of the back side area to the positive direction and, therefore, effectively reduces the target voltage, resulting in suppressed avalanche multiplication. That is, the output signal is not proportional to the amount of incident light. This effect depends on the output signal level. Below the reference signal level, the gamma value is between 0.9 and 0.95. Beyond the reference level, the value becomes so small that for eight times the amount of incident light, the four times output signal is obtained. This characteristic alleviates the requirement for maximum beam current. As the result, good performances of the electron beam system have been achieved, including improved resolution and reduced beam bending likely to occur at the highlights.

SIGNAL-TO-NOISE RATIO (SNR)

Thermal Noise of FET

The major noise source of an HDTV camera is the thermal noise of the FETs used for first stage amplification. In general, the input signal attenuates in proportion to the frequency as far as a voltage amplifier device with capacitive input impedance is used for amplification. To compensate this attenuation, these pre-amplifiers are designed to increase their gains in proportion to the frequency (6 dB/oct). The noise spectrum of an amplifier becomes so called triangular noise of which power increases in proportion to the 3/2 power of the required signal bandwidth. Due to the wide signal bandwidth, the HDTV camera is inferior to the NTSC camera in terms of SNR.

In general, the noise characteristics of a pre-amplifier can be evaluated by the figure of merit, or g_m/C of the first stage FET (g_m: mutual conductance, C: FET's input capacitance). GaAs FETs show larger g_m/C than Si FETs and are more suitable for the pre-amplifier[3,4]. However, they have the so-called 1/f noise which, in inverse proportion to the frequency, increases (-3dB/oct) from a certain frequency (corner frequency, f_c). Conventional GaAs FET's f_c is around 100 MHz. Using these FETs in place of Si FETs has no merit because the HDTV's signal bandwidth is 30 MHz. Recently, however, low corner frequency (lower than 10 MHz) GaAs FETs have been developed and manufactured.

Beyond the corner frequency, the output noise of a GaAs-used pre-amplifier increases its amplitude in proportion to the frequency (6 dB/oct), like a triangular noise. Below the corner frequency, the noise has a total increase rate of 3 dB/oct because the 1/f noise characteristic of -3 dB/oct is added. That is, below the corner frequency, the noise amplitude is in proportion to the square root of frequency. If the corner frequency is low enough, therefore, the 1/f noise is not so large a problem for the TV camera and its high g_m/C provides a merit.

In our camera, GaAs FETs are also employed to improve the SNR. For a signal bandwidth of 30 MHz, a high SNR of about 45 dB is obtained according to actual measurement.

Shot Noise

In addition to the FET noise, the shot noise caused by photons is inherent. Future performance improvement of FETs will make the existance of this noise clear although it is not being considered seriously at present. Here, let us explain in what manner sensitivity is restricted by this shot noise. Assuming a noise free FET, as is well known, the SNR of the camera can be represented as:

$$SNR = i_s / \sqrt{2 \cdot q \cdot i_s \cdot B} \qquad \ldots\ldots. (1)$$

q: electron charge
B: signal bandwidth
i_s: signal current

$$i_s = \eta \cdot i_0 \qquad \ldots\ldots. (2)$$

Where, quantum efficiency η is the ratio that incident photons are converted to primary signal charges. i_0 is an output current obtained when $\eta = 1$.

For a signal current i_s of 0.3 µA, an SNR of 45 dB is obtained from equation (1). The SNR with respect to the shot noise is in proportion to the square root of the amount of incident light because the amplitude of the shot noise is in proportion to the square root of the amount of incident light, while the signal current is proportional to the amount of incident light. This means that a certain amount of incident light is required at least to attain sufficient SNR for HDTV.

Although no measurements have as yet been taken regarding the limit of impairment perception for shot noise, analogical inference is possible using transmission channel noise. The shot noise depends on the signal level, that is, it becomes large in bright areas in a picture but small in dark areas. However, the gamma processing circuit has a large gain for a low signal level (dark area) but a small gain for a high signal level (bright area). Therefore, the output of this circuit tends independent on the signal level,*1 allowing the shot noise to be treated as a level-independent transmission channel noise. To be correct, however, the shot noise is equivalent to a smaller transmission noise of about 6 dB because shot noise amplitudes are compressed by the gamma processing circuit. For example, the above-mentioned 45 dB SNR shot noise is equivalent to a 51 dB SNR noise introduced through the transmission channel.

The limit of impairment perception for transmission channel noise is said to be about 45 dB in terms of SNR. This is equivalent to about 39 dB of shot noise, corresponding to 80 nA of signal current. This means that it will be possible to reduce the reference signal current of the camera tube down to 80 nA if noise-free FETs are developed. In other words, if the reference signal current is set lower than 80 nA, enough high picture quality cannot be attained for HDTV. That is, this reference signal current of 80 nA determines the theoretical limit of sensitivity.

For HDTV camera tubes currently used, the reference signal current is set at between 0.3 and 0.4 μA, indicating that the sensitivity can be improved about 4 or 5 times. In addition, because the quantum efficiency of conventional photoelectric transfer layer such as the Saticon are 0.3 or 0.4 (for G light), the sensitivity may be raised about 3 times further if the quantum efficiency is improved to 1. In total, therefore, sensitivity can theoretically be improved by 12 - 15 times compared to HDTV cameras currently used. This theoretical limit of sensitivity corresponds to the light level of 2000 lux at F11. Our camera's 10 times improved high sensitivity is considerably close to this theoretical limit.

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*1 Using signal level x, shot noise can be represented as $nx^{0.5}$ (n: constant). Gamma processing circuit output y is represented as $y = (x + nx^{0.5})^{r}$. If noise components are small, this can be approximated by $y = x^{r} + nrx^{r-0.5}$. For r = 0.5, this equation results in $y = x^{r} + 0.5n$, not depending on the signal level, and the noise amplitude is halved.

RESOLUTION

Fig. 4 shows the resolution characteristics of the HARP camera. Regardless of the small size, 2/3-inch, it provides over 30 percent amplitude response at 800 TVL, thanks to the amorphous target which is similar to that of the Saticon. In addition, a high performance MS (magnetic focus static deflection) type electron optics system and DIS[5] type electron gun are employed for high resolution and low lag characteristics.

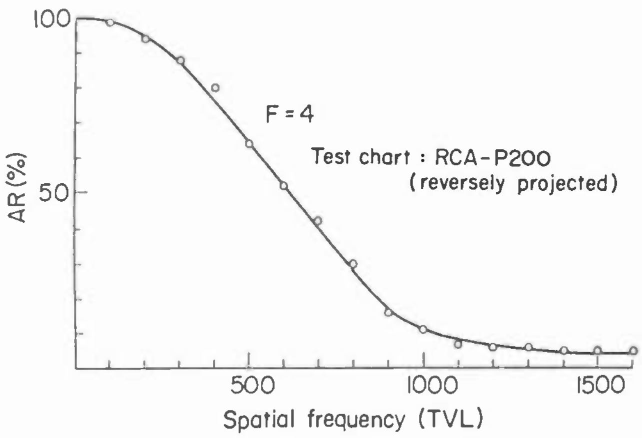

Fig. 4 Resolution Characteristics of HARP Camera

The camera's resolution depends on not only camera tube but much on the optical lens. Firstly, consider an ideal lens free from any aberration. In this case, the resolution is limited by diffraction. Fig. 5 shows the resolution of the ideal lens in terms of the MTF (moduration transfer function). The resolution becomes higher according as the lens aperture is opened, or, the F number is reduced. For an actual lens, however, the resolution has a peak at between F2.8 and F5.6. This is because the resolution is deteriorated by aberration below F2 and limited by diffraction beyond F8. The latter limit by diffraction is also theoretical and, therefore, cannot be exceeded by any actual lens.

Thanks to the about 10 times improved sensitivity, the HARP camera can set lens aperture at around F5.6 to shoot a dark object which the currently used HDTV camera cannot cope with unless the lens aperture is fully opened (to around F2). This provides advantages of the high

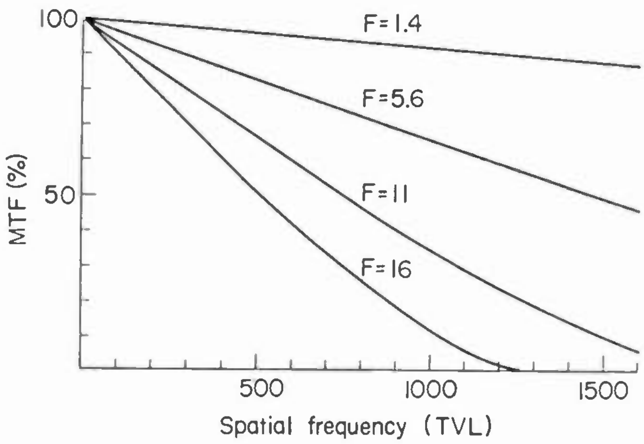

Fig. 5 Resolution Characteristics of Ideal Lens.

sensitivity, such as less deteriorated resolution of lens and larger depth of field. On the other hand, if the lens aperture is set larger than F11 for a bright object, diffraction limit the resolution. It is therefore desirable to use a neutral density filter (ND filter) to restrict the amount of incident light and set the lens aperture appropriately when shooting a bright object. This consideration is unique to such a high sensitivity camera.

OTHER HIGHLIGHTS

On-line Registration Correction System

Registration errors not only cause color shift errors, but also deteriorate the resolution of the luminance signal synthesized from the R, G and B signals[6]. In the HDTV system characterized by high resolution, particularly high accuracy registration is required[7]. In our camera, a new registration correction system is employed which can detect and compensate a registration error even during a camera operation.

The R, G and B signals are written into the memory respectively. The registration error is detected as displacement between the two signals, i.e., R and G or B and G, at which the maximum value of cross-correlation function is obtained. The detection error is within a fourth of the scanning line spacing, ensuring high enough accuracy. This registration correction system can result in improved operability because no test chart is necessary and it can quickly cope with registration errors caused during operation by the fluctuations of the temperature, terrestrial magnetism and so on.

Assistance Device for Focusing

In the HDTV system characterized by high resolution, more accurate camera focusing is required than for the NTSC TV system. A camera is usually equipped with a small CRT as the view finder. But it is difficult for camera operator to obtain accurate focusing by such a small CRT. In order to solve this problem, we have developed a new method[7] that makes a contour signal flicker and superimposes it onto the original video signal. When the camera is out of focus, the contour signal is small resulting in a small flicker on the view finder. When the camera is in focus, the contour signal at its largest causes the largest flickering on the view finder. This flicker can be cleary recognized even on the small view finder, ensuring satisfactory focusing.

CONCLUSION

This HARP camera has achieved a remarkable sensitivity which satisfies a requirement in HDTV program production. Such technology as GaAs FET pre-amplifier for SNR improvement, on-line registration correction system, have been incorporated to develop a small size, light weight hand-held camera. As the result, the camera can be used to shoot pictures from various angles and at various places, in addition to enabling to cope with dark scenes which has been difficult with currently used HDTV cameras.

The camera was used to broadcast the Seoul Olympic Games in HDTV which was done in an experimental basis. Since then, it has been operated satisfactorily. It is expected that the HARP camera will create a new field in HDTV program production.

ACKNOWLEDGEMENT

With deep respect, we thank every person who participated in the development of the HARP tube, the key device of this camera. And we thank every person who cooperated with us in operating this camera in experimental HDTV broadcasting at the Seoul Olympic Games and elsewhere.

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HDTV TRANSCODING—A VERSATILE STANDARDS CONVERTOR

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ABSTRACT

Over the last few years a number of HDTV system specifications have emerged. These all, to varying degrees, present compatibility problems with the broadcast standards in use today, which are themselves largely incompatible.

This paper discusses the down-conversion aspects of the documented HDTV systems to the current broadcast standards.

INTRODUCTION

The HDTV standards being reviewed are all conveniently described as 'production' formats. They do not offer a solution to the problems of transmission and reception/display. Obviously not all HDTV production is intended for broadcast, though much of it will be aimed at as wide an audience as possible. To achieve this, in the short term requires downconversion to an existing transmission format, with minimal degradation to the picture quality.

Downconversion equipment will, in addition find an important role in programme production. Often when shooting film a video camera is coupled to the film camera to enable a quick review of shot composure, etc. In the HDTV production centre this could obviously be achieved by replaying the HDTV recorder, but in many cases this will be in a remote site and so may take a significant time. A better approach would be to use a local portable downconverter coupled to a 1/2" recorder and monitor.

For outside broadcast use it may be inconvenient to have HDTV monitors on set, especially if more than one is required, as cabling G,B,R signals to each, (with DA's) will take time and effort. Again a portable downconverter connected to NTSC or PAL monitors may improve efficiency, considering that in many cases it is not necessary to have full resolution pictures on set.

A further application of a downconverter is in post production, where for convenience it is usual to generate edit lists off-line using existing 1/2" or 3/4" suites.

The remainder of this paper will consider the HDTV formats and typical approaches to downconversion.

HDTV FORMATS

The production standards considered are:-

1.	1050/59.94	Proposal to SMPTE Aspect ratio 16:9 Interlace ratio 2:1
2.	1125/60	Sony sync format Aspect ratio 5:3 Interlace ratio 2:1
3.	1125/60	SMPTE 240M Aspect ratio 16:9 Interlace ratio 2:1
4.	1250/50	Eureka proposal Aspect ratio 16:9 Interlace ratio 2:1

The Sony 1125 system is being phased out and replaced by SMPTE 240M. Sony provide a 'zone converter' to interface between the 5:3 aspect ratio system and 240M which has an aspect ratio of 16:9. However in a discussion of downconversion it is important to keep the 5:3 and 16:9 systems separate as aspect ratio conversion to 4:3 will use a different expansion ratios for each.

Both the 1050 and 1250 specifications define a family of related standards with non-interlaced formats at the top. Although equipment is unlikely to be available at the higher standards for a while yet, these formats may be of significant use in the standards conversion process.

One further system for which hardware exists is the 1049/59:94 system used by Hitachi in their high resolution disk recorder system. This is not intended to be a programme production system though downconversion is desirable for reviewing archives off-line.

INTERCONNECTIONS

Likely interconnections between the HDTV production formats and existing standards are shown on Figure 1. The numbers alongside the connections give a crude estimate of the relative complexity of the conversion process, with one being the simplest. It is interesting to note that all the downconversion processes shown require less complex processing for good picture quality than does NTSC to PAL and vice versa conversion. The simple reason for this is that with a HDTV source there is more information to start with. We can therefore expect HDTV downconversion to give better results than the PAL/NTSC conversions we see today.

The dashed connections show paths of lower significance; these are conversions across the national field frequency boundaries. It is unlikely that a production facility using, say, 1050 line equipment will need to off-line process in PAL. The demand for downconversion here will be at the top quality end, typically for programme distribution. Such conversions will, if possible make use of the non-interlaced formats proposed in their specifications, hence downconversion from 1050 line to PAL would first convert to a non-interlaced 1250 format.

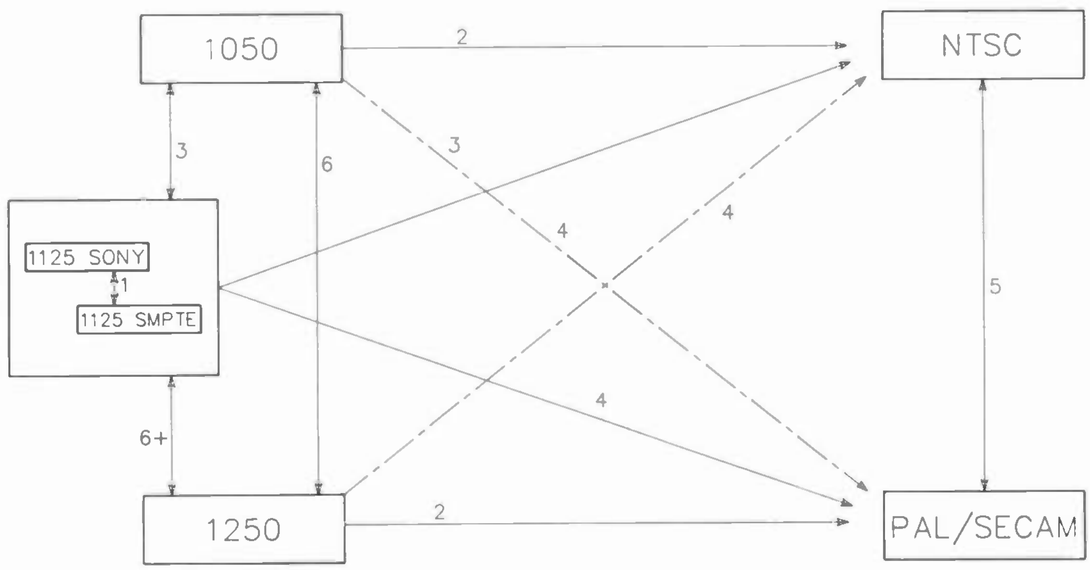
Film may be used as a means of downconverting HDTV as systems for direct recording of HDTV source material on film have been demonstrated. However, this is an unsatisfactory approach in respect of both cost and motion performance. The only acceptable route would be from 1050 or 1125 to film at 30 fps, and then for transmission as NTSC. To use 30 fps film for a PAL/SECAM output or 24/25 fps film for any system would give unacceptable motion performance.

Interconnection format

All the HDTV systems so far defined use the analogue components G,B,R for interconnection which are of equal bandwidth. However, allowance is made for processing using luminance and colour difference (Y,Pr,Pb) signals where the colour difference may have half the bandwidth of the luminance. Bandwidths specified are between 20 and 30 MHz for G,B,R,Y signals and 10-15 MHz for Pr, Pb (if used).

Equivalent bandwiths after line expansion, in the 525/625 domain will be about half the source bandwidth, which is still approximately twice the available bandwith of PAL/NTSC etc. So the first prob-

Figure 1. Interconnections



lem encountered in the downconverter is what to do with the surplus bandwidth. The actual low pass filter cut point will depend on the aspect ratio of the input and the mode of display on the output together with parameters such as active line length. Typically the bandwidth of the input luminance will be cut by half. To use a simple anti-alias filter will give, often severe ringing due to the energy in the source above the cut-off frequency. A gaussian or transitional filter will be necessary, though as a consequence, this will introduce some roll-off in band. For the chrominance process the filtering is more severe owing to the approximately 4:1 ratio between luminance and chrominance bandwidths of NTSC, PAL etc. Some alias will be inevitable - to what extent this is seen obviously depends on picture content.

Aspect ratio

The aspect ratio of the source is either 16:9 or 5:3, the aspect ratio of PAL, NTSC etc., is 4:3. There are a number of modes of display possible; which to use will depend on the application, whether in production, post-production or transmission. Any down-converter will have to offer a range of display modes; the four most useful modes are shown in figures 2-5.

Figure 2 is the 'edge cut' mode where full picture height is maintained. A pan facility is required to move the viewed area to the most appropriate portion of the input picture. For sequences where it is necessary to show the whole source picture a squeeze mode is used, as shown in figure 3. Here the source is anamorphically distorted to fit in the 4:3 window. The edge cut mode of display is used by the U.K. broadcasters for 'cinemascope' or other widescreen format films. Title sequences often occupy the full screen width and so are shown in squeeze mode, though the remainder of the film can satisfactorily be shown 'edge cut' with careful use of the pan-scan control.

More common in European countries is transmission in the letterbox mode shown in figure 4. Here the full width of the source image is maintained, with blank lines above and below to fill the 4:3 window. This is obviously the simpler mode of display as there is no pan-scan control and so no risk of losing a vital part of the picture. The blank lines need not just be left black, they may be coloured or used for subtitles. For 16:9 down-conversion the blank lines occupy 25% of the active picture area, so there is a real loss of vertical resolution owing to the reduced number of output lines and an apparent loss of horizontal resolution due to the 'smaller' picture size.

Letterbox display is probably the most useful in production, although a larger monitor may be required.

Figure 5 shows a compromise position, suitable for viewing and transmission, whereby the output picture is slightly reduced in height, and to maintain accurate geometry, a compensating amount of

Figure 2.
Edge Cut Mode

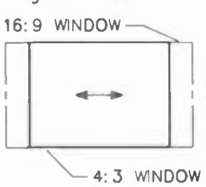


Figure 4.
Letterbox Mode

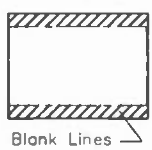


Figure 3
Squeeze Mode

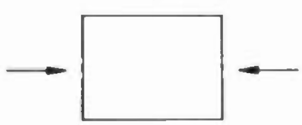
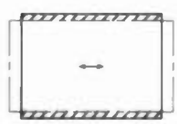


Figure 5.
Reduce Edge Cut Mode



edge cut is added. It is unlikely that a pan-scan control would be necessary in this mode as the small loss of picture content could be allowed for by instigating a 'safe area' at the time of production.

The squeeze mode may be useful in one other instance; if the new European satellite broadcasters were to begin transmission in a widescreen 625 line format, as is permitted in the MAC specification, then HDTV production systems will be used to provide programme material. A downconverter will then be required with a component interface and significantly wider bandwidth than necessary for PAL.

CONVERSION PROCESSING

There are many ways in which to approach downconversion. Choices must be made as to the number of formats allowed on the input and output and on the desired picture quality. As illustrated in figure 1 there are significant differences in the complexity of the various conversion processes.

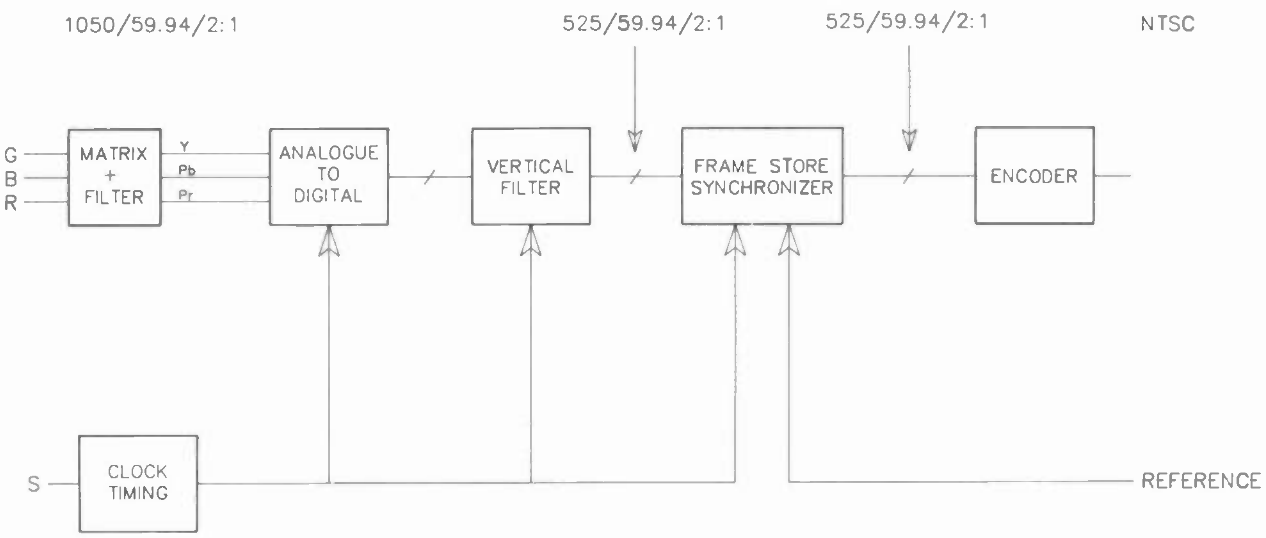
The simplest downconversion process is from 1050 to NTSC or 1250 to PAL/SECAM, this is shown in figure 6.

Incoming G,B,R is matrixed (for convenience) to Y,Pb,Pr and passed to a vertical low pass filter. Ideally this cuts the vertical resolution by half, necessary before dropping alternate lines. In practice a filter of limited length, say 5 lines, cannot fully remove alias frequencies whilst maintaining a reasonably flat in-band response. This is not really a problem as vertical alias is of secondary significance to interline flicker. However, if it is envisaged that the PAL/NTSC may be upconverted in the display to a non-interlace scan format then control of the vertical alias becomes important. In this case it would be better to perform the vertical filtering and down sampling on a non-interlaced HDTV source e.g. 1250/50/1:1.

To create an accurate interlace in the output image the vertical filter must be offset by half an input line interval on alternate fields, or to give a constant aperture function one field is offset by + 1/4 line and the alternate field by - 1/4 line. Following the vertical filter is the line expansion process whereby the HDTV line of approximately 30 uS is expanded to the 63.5 or 64 uS of the NTSC or PAL line. Expansion is achieved by the ratio of input to output clock frequency in a simple line store; the exact ratio depends on the input and output active line lengths, aspect ratio and display mode. A frame store sychronizer is used to lock the down-converter output to an external pulse generator.

If we wish to include 1125 line sources the simple vertical filter described will need replacing with a more complex interpolator. For a 525 line output the downsampling ratio is 15/7. The coefficients of the filter must be sequenced in a 15 input line cycle, with an appropriate offset field to field to restore interlace.

Figure 6.
Basic Downconverter



Synchronisation between the 60Hz field rate of the 1125 line systems and NTSC at 59.94 will in most cases be achieved by dropping one field every sixteen seconds or so. If tests show this to be unacceptable in a high end downconverter then relatively simple processing may be included to search for fields with minimal motion. A larger buffer store is required in this case.

Field rate conversion

So far only line rate conversions have been considered. Referring back to figure 1., there is one significant downconversion route that requires field rate conversion, that is from 1125/60 to PAL/SECAM. This may be approached in two ways; most easily the source could be converted to an intermediate standard of 525/60/2:1 with the line rate converter already described, then passed to a conventional 525 to 625 line standards converter. There are two drawbacks to this method; first the output vertical resolution has been limited to that of a 525 line system and secondly the artefacts of the standards converter are introduced. The conventional standards converter must preserve as much vertical resolution of the source as possible, even though this information is aliased in the temporal domain due to the interlace. The aperture function of these converters is a careful compromise between vertical and temporal response. With the extra lines of the HDTV source it is possible to split the vertical and temporal filters so each may be optimised independently.

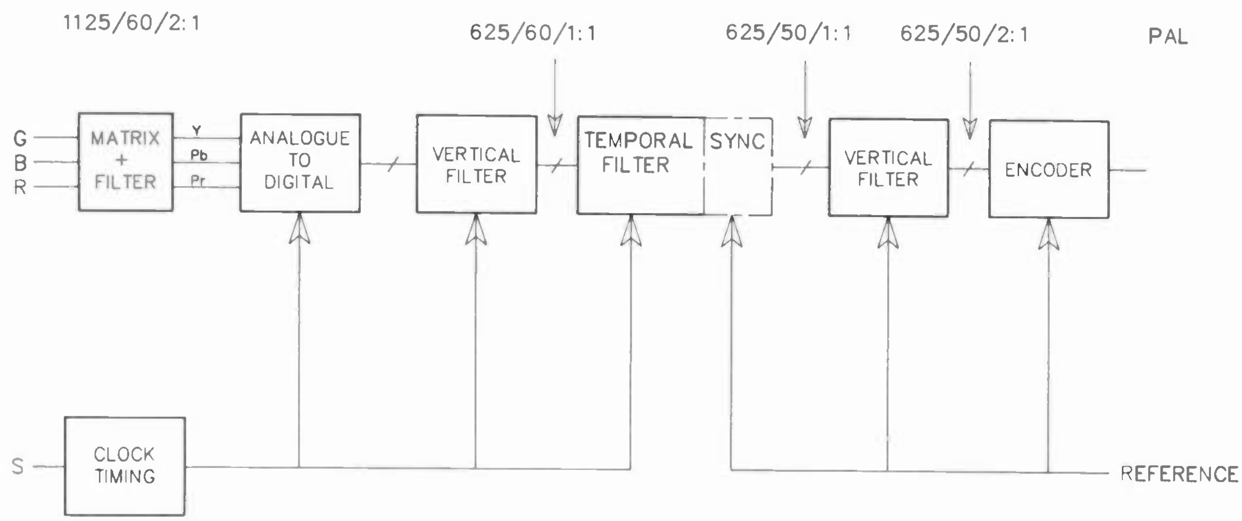
Such a downconverter is shown in figure 7. An non-interlaced intermediate standard of 625/60/1:1 is created by line interpolation only, which is then passed to the temporal filter. The output of the temporal filter is a standard 625/50/1:1 which a second vertical filter with appropriate field to field offsets, converts (by downsampling) to 625/50/2:1.

A similar process is equally valid applied to conversions 1050/59.94 to PAL and 1250/50 to NTSC. For a 1050 line source the intermediate standards would be the same as for 1125, only the first vertical filter would change. With 1250 line sources the intermediate standards would be 525/50/1:1 after the first vertical filter and 525/59.94/1:1 into the second vertical filter.

All temporal conversions described above rely on a linear temporal filter, which should be adequate for HDTV downconversions that split the vertical and temporal processes. However, with a realistic temporal filter limited in length to say, four fields there is obviously going to be a compromise between temporal bandwidth and rejection of the primary alias frequencies.

Motion compensation techniques theoretically offer a solution to the main artefacts of linear filtering, namely motion blur and judder, but problems associated with accurately generating motion vectors in real time are still to be solved. It will be the case that any motion adaptive eqiupment will be significantly more costly than that using simple linear or adaptively controlled linear filters.

Figure 7.
Downconvertor with field rate conversion



A VERSATILE STANDARDS CONVERTER

Snell & Wilcox Ltd., introduced what we believe to be the worlds first universal HDTV downconverter at IBC, Brighton '88. At that time it would except 1125 or 1250 line sources for conversion to PAL, NTSC, NTSC4.4, SECAM, PAL-M, or PAL-N. Control facilities included a dynamic squeeze feature to enable edge cut or anamorphic display modes; a pan control to enable full movement of the 4:3 window, together with the usual video controls including colour balance.

The equipment has now been extended to include the 1050/59.94/2:1 standard submitted to the SMPTE by the American broadcasters. A letterbox display mode is now included, this may be in the form shown in figure 4 or 5. In addition it is possible to shift the letterbox display to the upper part of the 4:3 window giving more room for subtitles or time-code.

A key feature of the equipment is portability making it particularly suited to the various off-line applications.

HDTV CAMERA LENS REQUIREMENTS-FROM 525 LINES FIRST CCD GENERATION TO HDTV: THE EVOLUTION OF OPTICAL REQUIREMENTS FOR TV CAMERAS

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From the first CCD cameras of only a few years ago to the presently available HDTV cameras, there has been a tremendous increase in image quality. The camera lens has been and still is a crucial factor. Camera optics have been required to deliver better and better performance.

To the broadcaster the best lens is the one which optimizes the level of optical performance he needs with operating characteristics such as angle of view, aperture, focusing distance as well as reliability, size and cost.

This paper explores the limiting factors of optical performance, today's State-of-the-Art, and actual or future possibilities of improvement.

1.- THE SENSORS

The first CCDs for Production Broadcast cameras were able to resolve 400 points per TV line. This has now gone up to 700, or even 850 pixels. Tubes are presently able to resolve 1250 TV lines. High Definition horizontal resolution will be of the order of 2000 points on a horizontal line.

For reference, 35 mm motion pictures is able to read 1700 TV lines or more.

Figure 1 shows Modulation Transfer Function (MTF) of various sensors as a function of image spatial frequency.

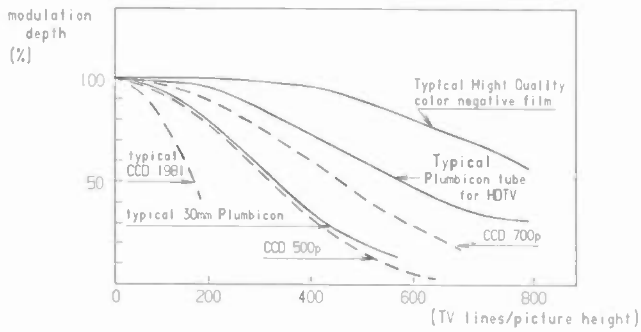

FIGURE I: MODULATION DEPTH OF VARIOUS SENSORS

2.-CONSEQUENCE: REQUIREMENTS FOR THE LENS.

The improvement in the camera sensor would be useless unless the lens is itself able to separate elementary images not larger than what the sensor can resolve.

Resolution of lenses is limited by:

- diffraction
- geometric aberrations
- chromatic aberrations

3.- DIFFRACTION

Diffraction is caused by the vibrating nature of light. The edge of lens elements, or of the iris, acts as secondary sources of light. The unfortunate result is that an optical instrument cannot be absolutely stigmatic, that is, it cannot give an infinitely small image of an infinitely small object. In other words the image is always a spot, even for a point. The radius of that spot is given by the well known formula:

$$R = \frac{1{,}22\,\lambda}{2\ n\ \sin U}$$

λ = light wave length, n is the refractive index, U is the semi aperture angle.

Diffraction thus limits the Modulation Transfer Function. A totally perfect lens still cannot have a MTF higher than as shown on Figure 2.

back focus) make the task even more difficult.

It would be too involved to analyze here in detail the effect of each of these aberrations (spherical, coma, field curvature). Of course, they all add together and the designer aims at minimizing the spot image. He is totally successful with a CCD sensor when the spot is essentially smaller than a pixel. This is typically the case for present ENG lenses for CCD's such as ANGENIEUX 14 x families for 2/3" and for 1/2".

It is worth mentioning the significance of strict manufacturing tolerance for optical surfaces, and for their exact tilt alignment, a particularly difficult problem in zoom optics with many mobile elements. Again, practical cost considerations are a limit and make the job of the designer even more taxing.

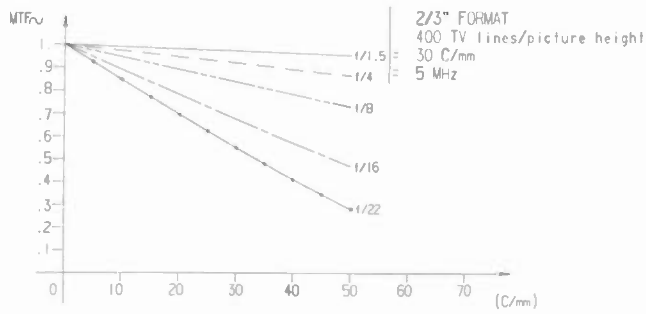

FIGURE 2: DIFFRACTION LIMITED MTF FOR λ = 540nm

As an illustration of this, it is interesting to note that the ANGENIEUX 15 x 12 for High Definition is limited by diffraction only for any aperture equal to or smaller than f/4 as shown on Figure 3. For these apertures the lens has reached the maximum resolution possible dictated by the laws of optics. Practically between f/4 and f/8 the lens performance is substantially better than what can be perceived by the sensors.

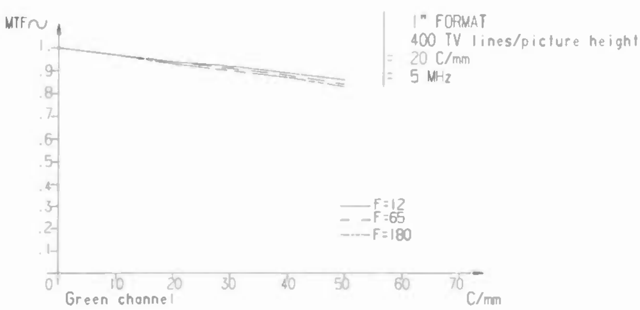

FIGURE 3: ZOOM 15x12 HDTV at f/4

4.- GEOMETRIC ABERRATIONS

Rays of light are deviated by each optical surface according to Newton Law (n 1 sin I 1= n 2 sin I 2), i.e. according to the index of refraction of the optical material and according to the angle of incidence of the ray. It is the science, and the art of the optical designer to compute radii of curvature and thickness of lens elements in order that all rays of light emitted by one object point do converge to one image point.

It is always possible to achieve a satisfactory result with enough spherical elements. However, in practice the number of parameters is limited. Further constraints such as the clearance for the color splitting prism (the so called

5.- DISTORTION

Distortion does not affect resolution, but geometry. It may be of particular significance with CCDs which have none whatsoever; Incidentally, just like film.

To illustrate values currently found in present State-of-the-Art lenses, Figure 4 shows the distortion of an ANGENIEUX 40 x 9.5 and of the ANGENIEUX 20x8.5 as a function of focal length.

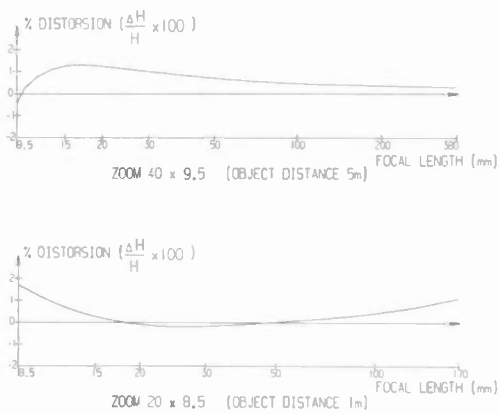

FIGURE 4: DISTORTION

6.- CHROMATIC ABERRATION

Chromatic aberrations are due to the fact that the index of refraction of optical materials varies with the wave length of light. They can show themselves in two perverse ways:

- longitudinal chromatic aberration
- lateral chromatic aberration

Longitudinal chromatic aberration means the green, red and blue images do not focus in the same plane. While moderately inconvenient with tubes which can be individually positioned, it may be a problem if not properly corrected in CCDs which are cemented in a fixed position. Obviously out of focus red or blue picture have

have a degraded MTF.

There has been recently a proposal to position the CCDs according to a standardized amount of longitudinal chromatic aberrations. This interesting approach, not without merits, is however still raising questions under discussions at the present time.

Lateral chromatic aberrations are in effect distortion or geometry defects varying with the color of light. The designer must reduce it to the point where it is no longer objectionable. For CCD cameras it is the case when the blue and red pictures are imaged on the same pixel as the green image.

To supply a numerical example, the ANGENIEUX 14 x lenses for 2/3" CCD cameras show a maximum distance of red to green, and blue to green, of no more than 10 microns, at all focal lengths on an object at the usual working distance of 8 feet. This can be favorably compared to the distance of 17 microns, or even 13 microns to be found between the centers of two pixels in the most recent CCDs. The same can be said for the 1/2" format. The lateral chromatism in the ANGENIEUX 14 x 7 is less than 6 microns, for a pixel to pixel step of 13 microns.

7.- HOW TO FURTHER IMPROVE?

As good as they are, television lenses will require improvements to satisfy the future challenge of improved sensors, and sooner of later, of Enhanced TV and HDTV. Which directions within research are open to lens manufacturers? Fortunately there is a number of ways:

- The use of new optical materials as they are developped or made available with larger dimensions by glass manufacturers.

- Improve the machining of optical surfaces. This will allow an increase in the number of elements, therefore of design parameters. Presently, more elements mean a higher correction but it may be destroyed by the accumulated manufacturing errors.

- Use aspherical surfaces to correct aberrations connected with large apertures. This is somewhat futuristic as the technology is not really there for the larger diameters required by the aperture and focal lengths of television lenses.

- Control the positioning of optical groups as a function of more than one variable. Micro processor control, as used in the ANGENIEUX 20X and 40Xm allows the optimum displacement of focusing groups. This is extremely helpful, in more ways than one, at the longer focal lengths.

- Change the glass path of the camera prism in real time. One of the most difficult problem of zoom lens designs is the change of aberrations with zoom configuration (i.e. focal length, focusing distance, aperture). ANGENIEUX has developped this unique patented device (Figure 5). An "intelligent" lens with an "intelligent" prism can deliver an image virtually free of any chromatic aberration.

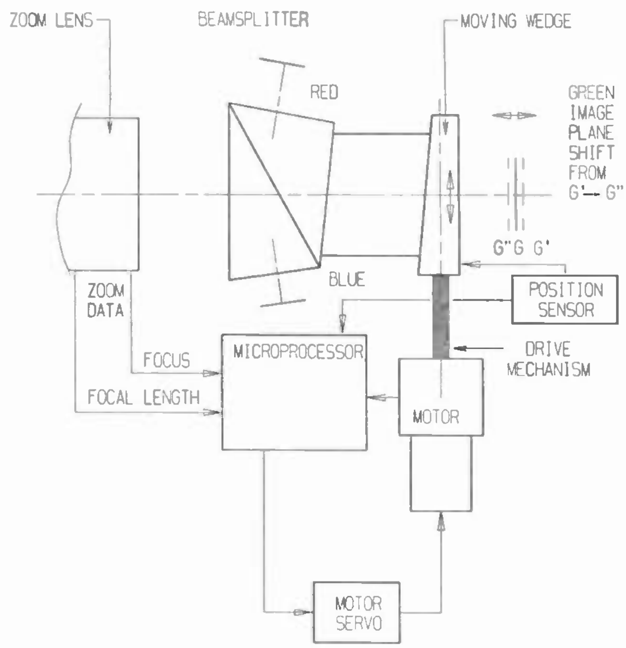

FIGURE 5: DYNAMIC PRISM FOR LONGITUDINAL CHROMATIC ABERRATION CORRECTION

MULTI-STANDARD HDTV SIGNAL GENERATION

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Abstract

Whatever standard or standards are actually adopted for HDTV in addition to the existing SMPTE 240M, there will need to be sources of test signals and synchronizing information. This paper describes the essential parts of a generator which can provide three channels of synchronizing signals as well as four different GBR test signals from non-volatile memory. Signals are available in any of the currently suggested HDTV formats. With an interconnected personal computer, the test signals available may be changed by downloading different signal instruction sets from existing libraries, or signals may be modified and new ones created using user-friendly software. Synchronizing information in either "normal" or tri-level formats and locking between synchronizing signals and to both the test signals and the outside world are provided.

Background

Continuing arguments between all the interested parties in Advanced and High Definition TV leave only one thing absolutely clear -- that there will be no total consensus in either the standards for production or emission. It is not the purpose of this paper to argue in favor of any particular standard(s) but to show how one manufacturer has gone about providing a tool that will allow the proponents of the various systems to test out their theories rather more accurately than with mere computer projections.

Like any television system, all the proposed HDTV systems require synchronizing signals and test signals to provide system timing and to verify performance. It would be totally uneconomic, at the present time, for a manufacturer to provide dedicated generators for each of the proposed standards, or to respond effectively to the needs of those wanting custom modifications each time another proposed standard raised its head.

Against this background, Magni took their 2015/2021 generator line and created a new product to satisfy the needs of the HDTV user and researcher. The generator, the Magni 2030, can provide for the synchronizing needs of the HDTV studio operation whilst also producing test signals for that studio. The HDTV standard at which the unit is shipped can be changed very easily on-site, so that an end-user who might be working in more than one HDTV standard has a multi-standard sync and test source available.

General Arrangements

The general mechanical arrangements at the rear of the 2030 generator are shown in Fig. 1. The upper part of the system provides the Test Signal Generator (TSG) and the lower part the Sync Pulse Generator (SPG) functions. The two units can be interlinked by a timing cable if synchronous timing between the SPG and TSG outputs is needed; otherwise, it is possible to split the timing between the units and to have the TSG locked to a station reference of some kind while the SPG is timed ahead, or behind, in another station function. The two units could also be interconnected through a delay system so as to have the TSG signals at a permanent timing difference for, for example, a switcher input.

Each of the two halves of the system has three signal generation channels; for the TSG these are three 1V peak-to-peak channels for G, B and R. For the SPG the three channels are capable of 2V peak-to-peak outputs, and are normally set with H and V drive at 2V pk-to-pk on channels 2 and 3 but with mixed sync on channel 1 at 0.3V pk-to-pk (for normal syncs) or 0.6V pk-to-pk (for tri-level syncs). Each channel (the three from the TSG and the three from the SPG) actually has two outputs.

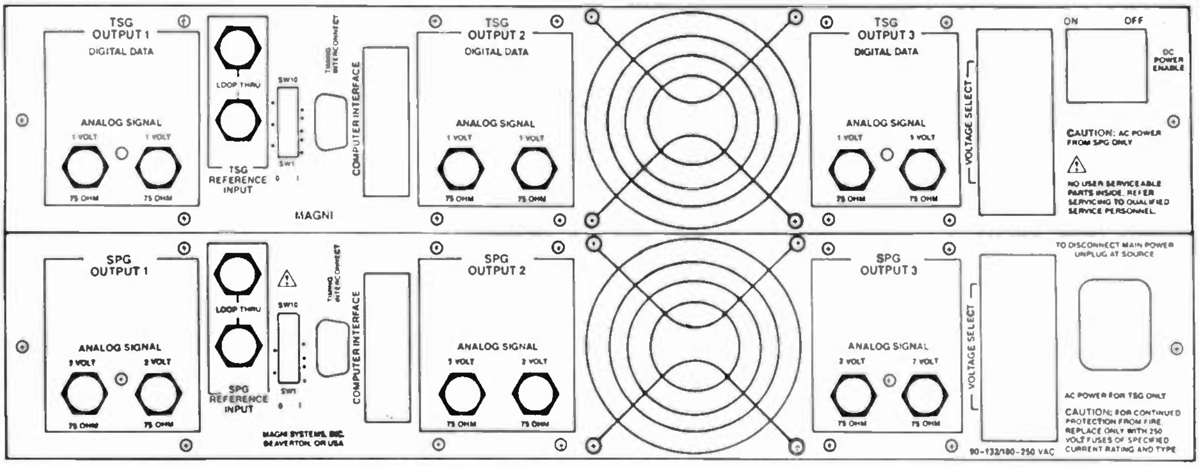


FIGURE 1 - 2030 REAR PANEL

Because of the operational environment in which the 2030 may be used, there is no power switch to the SPG half although there is to the TSG half; power for the TSG is looped from the SPG, but both parts are separately fused.

Each half of the generator has its signal instruction sets in non-volatile memory, so that any interruption to power will not require the reloading of any signals.

Each half of the system also has its own computer interface port; in normal use, where an operation is running in only one HDTV standard, it would be unlikely that the SPG interface would need to be accessed by the facility: if a change was to be made there, it would probably occur when the unit was being commissioned and the SPG signals that had been loaded were not those needed for the operation -- there is still some confusion, for example, about drives/ blanking uses in timing various boxes in the overall HDTV studio.

The TSG part of the generator is the one most likely to be accessed by the end-user. When a PC is connected to the interface port from the computer's parallel port, the four factory-loaded signal selections can be changed to any other four from the appropriate library file for that HDTV standard. That library file could be the original one supplied by Magni -- containing about a dozen signals -- or, more likely, would also contain signals that the end-user had modified or generated for his particular purpose. It is expected that, often as not, the availability of four signals from the front panel switch would be sufficient for operational use, so that a PC would not be required on a daily basis.

Principle of Operation

It was decided very early in the design of the 2000 Series generators that the standards which could be produced by the generators should not be limited by a controlled memory size; so, for example, the use of a frame of memory would have limited the unit to whatever "frame" meant in that respect (525/59.94, 625/50, or whatever). The signal RAM inside the units therefore bears no direct relationship to either line or field rates.

The Signal Master Software that is used with the Series allows the design of signal instruction sets on a PC; those are then digitized and downloaded into the 2030 from the Centronics parallel port for simplicity. These instructions produce the signals from the generator in real time.

To make a new HDTV standard requires the heart of the signal to be defined by what is known as the frame descriptor. This system description defines the repeatable, non-active video of the signal: how many lines, line length, sync shaping, what happens in the field interval, interlaced or not, color framing, and so on.

Once the frame descriptor for a particular standard is produced, the actual addition of the active video information -- the production of field splits and the like -- is comparatively simple using the Signal Master Software.

Fig. 2 shows the basic operational schematic of the 2030 generator. The two parts of the generator are very similar, with the essential differences being in the amount of signal memory that is provided, the signal output levels from the channels, and the normal connection of the synclock.

For each half of the generator, a standard 8-bit parallel computer interface port is used to pass the data and instructions to the generator's CPU (Z80). The CPU itself controls the order in which data is read from the Control Memory and the frequency at which the precision crystal oscillator is running. For 1125/60 HDTV signals, for example, sampling is at 108.00 MHz, while for 1125/59.94 HDTV signals sampling is at 107.892096 MHz, the ratio being chosen so that the signals themselves for the two files are identical.

The synclock module can be set to different configurations to cover the different HDTV standards and the different natures of normal and tri-level syncs. Normally, the TSG part of the system would be synclocked to the mixed syncs on the channel 1 output of the SPG using a supplied interconnect cable; however, there might be studio circumstances where the TSG was required to be locked to a separate reference, and the facilities are available to do that. Similarly, the SPG part of the system can be allowed to free-run and can be, if required, the master reference for the operation, or it can be synclocked to an external reference.

The combination of the Timing Frame Descriptor, the Timing Generator and the Address MUX determines and specifies the nature and addresses of the signal data and the start and stop points. The facilities provided also allow for the construction of Multi-modules, which are selected signals in a sequencing order (manually or automatically selected) such that a whole sequence of tests can be run through very simply.

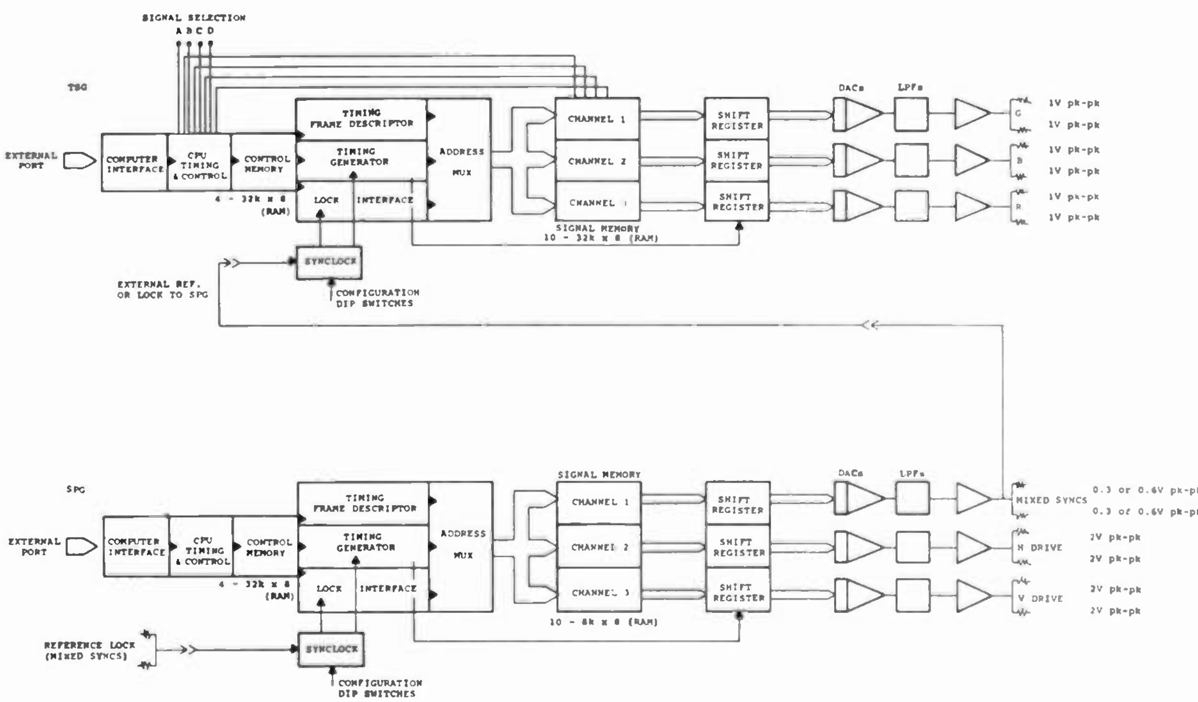


FIGURE 2 - BLOCK SCHEMATIC,
2030 CONTROL AND SIGNAL GENERATION

Selection of the GBR output signal is made from a front panel switch which, through the CPU, controls which area of signal memory is output.

The signal memory itself is identical for each of the three channels on the TSG and each of the three channels on the SPG. Data is read out as 80 parallel bits, and the shift registers perform a parallel-to-serial transformation at a timing sequence determined by the Timing Generator. The outputs from the shift registers are then 10-bit parallel signals. Non-volatility in the memory RAM is provided by the use of "smart" sockets.

Precision Digital-to-Analog Converters (DACs) convert the 10-bit signals into wideband analog outputs. Because of the high sampling frequencies used, analog signals up to about 40 MHz are possible, but the spectrum is limited by a 5-pole Low-pass filter (LPF) in each channel with a -3dB point at about 40 MHz. The published frequency response specification at 30 MHz is +2%/-3% compared to 50 kHz.

The LPFs are followed by output amplifiers which provide two 75 Ohm outputs from each channel. To provide a "standard," the 2030 uses 2 V outputs for drives on the SPG and 1 V outputs for test signals. The mixed sync channel (if one is loaded that way on the SPG) would still be capable of a 2 V output signal, but the standard output would be 0.3 V pk-pk for normal syncs and 0.6 V pk-pk for tri-level syncs.

Test Signal Files

The Test Signal Files for the 2030 are available in a variety of HDTV formats:

1125/60.0 2:1 "Normal" Sync
1125/60.0 2:1 Tri-level Sync
1125/59.94 2:1 Tri-level Sync
1050/59.94 2:1 Tri-level Sync
1050/59.94 1:1 Tri-level Sync
525/59.94 1:1 Tri-level Sync
1250/50.0 2:1 Tri-level Sync

This list does not represent the limits of the standards which could be made with the 2030, and others have been asked for as both standard files and as specials.

On the Test Signal Files there are the synchronizing signals for that particular HDTV standard, plus test signals which normally take the form of:

100% Color Bars
75% Color Bars
100% White - Full Field
Multiburst - Packets at 1 MHz increments, 1-12 MHz
Multiburst - Packets at 2 MHz increments, 10-32 MHz
Multipulse - Modulated at 1 MHz increments, 1-12 MHz
Multipulse - Modulated at 2 MHz increments, 10-32 MHz
10-step Staircase
Linear Ramp
60 Hz Square Wave
Colorimetry Chart

Any of the signals that are loaded in the switch positions at delivery from Magni (75% Color Bars, High-Frequency Multiburst, Linearity Ramp, Colorimetry Chart) can be modified to any of the Test Signal File Signals, or any of the signals may be modified by the user to suit their environment.

```
                 M A I N   M E N U

Build      new signal from existing signal
Setup      user disk.  Generates signal
Transfer a signal to the generator
dOs        command
Create     first signal of a new system
Edit       current signal
Display    current signal
eXit       end of program
```

FIGURE 3 - MAIN MENU - SIGNAL MASTER SOFTWARE

Signal Master Software

It is also very straightforward, if the user requires, to write new signals in the GBR format or even in Y, B-Y, R-Y. This process is undertaken using the Signal Master Software supplied with the system. The main selections for new signals in the main menu of that software (see Fig. 3) are for either "Build" or "Create." The other functions provided in this menu allow the actual Setup of the user disk once a new signal has been put together in terms of its criteria; Transfer of the signal to the 2030; execution of DOS commands; and Editing or Display of the current signal. The Display function yields a digitization of the waveform, with graticules and a cursor provided so the data that has been input can be checked prior to the Setup of the signal; this gives the user some confidence that what was done was what was intended and can save considerable time if a mistake has been made.

"Build" is the simplest file to work in. This program is for the modification of an existing signal; that signal is chosen from the directory provided, and a new name may be given to the new signal. A window is then provided (Fig. 4) that displays the signal characteristics of the original signal and details the features that may be added: luminance, pedestals, sine-squared pulses, sine wave packets, modulated frequency pulses and user functions. The latter may be any mathematically-based description of a signal feature: line sweeps might be defined in this way, for example.

The Edit window at the bottom of this screen provides a listing of all the features of the signal in terms of what the feature is, when it starts, when it finishes, rise times, magnitudes and so on. It is possible to display time in microseconds, cycles or clocks; and amplitude in terms of IRE, mV or Bits. The use of clocks for time indication is particularly useful for HDTV systems where the basic synchronizing and line features are defined in clocks: in the signal shown in Fig. 4 (which is an 1125/60 Multiburst), for instance, the reference clock frequency defined is 74.25 MHz, allowing all the fixed features of the signal to be defined in clocks. This is for convenience only in that there is no operating hardware in the system at that 74.25 MHz.

The general arrangement of the Multiburst signal (Fig. 4) is shown together with the full detail of its construction in Fig. 5, which is taken from a Utility File known as DECOM (Decompose). DECOM allows a permanent paper record of a signal to be made. This version shows the advantage of working in clocks.

"Create," the other primary main menu selection, is used for construction of a New system: the menus which follow prompt the user through a set of needed specifications of the system, including its name, sampling frequency, samples per line, reference clock, display offset and reference time. Following the same "Create" process, macros for the frame descriptor are written for the new system.

When it is needed to change signals in the 2030, the **TRANSFER**

ADD SIGNAL EVENT, EDIT, or DISPLAY. Esc to EXIT

Luminance
Pedestal
Sine^2 pulse
sine Wave packet
modulated Frequency pulse
User defined signal
Display signal
Edit signal

T)ime_toggle [μSEC,CYCLES,CLOCKS] M)agnitude_toggle [IRE,mV,BITS]

EDIT WINDOW

FILE TRMBH.SIG	Samp Freq 108.0000 MHz	Clock Freq 74.2500 MHz		Subcarrier
SINE PACKET	Sine^2	from 18.59 μSEC	to 19.93 μSEC	IREpp = 98
SINE PACKET	Sine^2	from 20.44 μSEC	to 21.78 μSEC	IREpp = 98
SINE PACKET	Sine^2	from 22.29 μSEC	to 23.63 μSEC	IREpp = 98
SINE PACKET	Sine^2	from 24.14 μSEC	to 25.48 μSEC	IREpp = 98
SINE PACKET	Sine^2	from 25.99 μSEC	to 27.33 μSEC	IREpp = 98
LUMINANCE	Sine^2 Pulse	Center at 3.70 μSEC	IREp = 98.000	HAD o
LUMINANCE	Sine^2 Pulse	Center at 4.71 μSEC	IREp = 0.000	HAD of

E)dit_field Del)ete_line U)ndel R)epeat_line I)gnore_line O)ffset S)ave

FIGURE 4 - "ADD" FEATURES AND EDIT WINDOW

function is called up and the screen shown in Fig. 6 appears. When the on-screen cursor is put over a particular signal and "ENTER" is keyed, that signal will be transferred to the 2030. A "Link" module also exists so that four signals can be transferred at one time to all four positions on the 2030 signal selector. Various other Utilities are provided to the user under the Transfer function, including the setup for Multimodules and automatic sequencing.

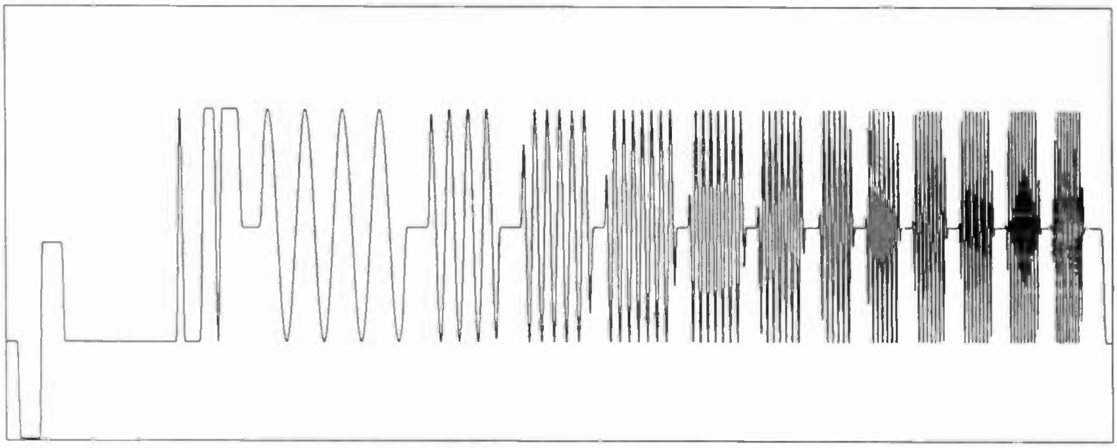
Summary

The Magni 2030 provides the necessary multi-standard capability to allow it to be used for all the HDTV standards that have been proposed to date. It is simple to use in its

* *

File: **TRMBL.SIG** Printed: FEB 17, 1989 Created: FEB 10, 1989

* *



Sampling Frequency 108.00000 MHz   Clock Frequency 74.25000 MHz
Reference time 69.71250 CLOCKS   Starting Level 240.0000 BITS   3200 Samples per Line

LUMINANCE Sine^2 at -44.00 CLOCKS Ending Mag of 4.800 BITS 10-90% time of 4.00 CLOCKS
LUMINANCE Sine^2 at 0.00 CLOCKS Ending Mag of 475.200 BITS 10-90% time of 4.00 CLOCKS
LUMINANCE Sine^2 at 44.00 CLOCKS Ending Mag of 240.000 BITS 10-90% time of 4.00 CLOCKS
LUMINANCE Sine^2 at 318.75 CLOCKS Ending Mag of 788.800 BITS 10-90% time of 5.94 CLOCKS
LUMINANCE Sine^2 at 393.00 CLOCKS Ending Mag of 514.400 BITS 10-90% time of 5.94 CLOCKS
LUMINANCE Sine^2 at 2112.00 CLOCKS Ending Mag of 240.000 BITS 10-90% time of 5.94 CLOCKS
SINE PACKET Sine^2 from 436.07 CLOCKS to 721.19 CLOCKS BITSpp = 548.800 10-90% time of 5.94 CLOCKS freq = 1.0000 MHz
SINE PACKET Sine^2 from 770.19 CLOCKS to 906.81 CLOCKS BITSpp = 548.800 10-90% time of 5.94 CLOCKS freq = 2.0000 MHz
SINE PACKET Sine^2 from 955.82 CLOCKS to 1092.44 CLOCKS BITSpp = 548.800 10-90% time of 5.94 CLOCKS freq = 3.0000 MHz
SINE PACKET Sine^2 from 1122.88 CLOCKS to 1259.50 CLOCKS BITSpp = 548.800 10-90% time of 5.94 CLOCKS freq = 4.0000 MHz
SINE PACKET Sine^2 from 1289.94 CLOCKS to 1396.86 CLOCKS BITSpp = 548.800 10-90% time of 5.94 CLOCKS freq = 5.0000 MHz
SINE PACKET Sine^2 from 1427.30 CLOCKS to 1514.40 CLOCKS BITSpp = 548.800 10-90% time of 5.94 CLOCKS freq = 6.0000 MHz
SINE PACKET Sine^2 from 1549.82 CLOCKS to 1612.19 CLOCKS BITSpp = 548.800 10-90% time of 5.94 CLOCKS freq = 7.0000 MHz
SINE PACKET Sine^2 from 1642.63 CLOCKS to 1705.00 CLOCKS BITSpp = 548.800 10-90% time of 5.94 CLOCKS freq = 8.0000 MHz
SINE PACKET Sine^2 from 1735.44 CLOCKS to 1797.81 CLOCKS BITSpp = 548.800 10-90% time of 5.94 CLOCKS freq = 9.0000 MHz
SINE PACKET Sine^2 from 1828.25 CLOCKS to 1890.62 CLOCKS BITSpp = 548.800 10-90% time of 5.94 CLOCKS freq = 10.0000 MHz
SINE PACKET Sine^2 from 1921.07 CLOCKS to 1983.44 CLOCKS BITSpp = 548.800 10-90% time of 5.94 CLOCKS freq = 11.0000 MHz
SINE PACKET Sine^2 from 2013.88 CLOCKS to 2070.08 CLOCKS BITSpp = 548.800 10-90% time of 5.94 CLOCKS freq = 12.0000 MHz
LUMINANCE Sine^2 Pulse Center at 275.00 CLOCKS BITSp = 788.800 HAD of 8.91 CLOCKS
LUMINANCE Sine^2 Pulse Center at 349.94 CLOCKS BITSp = 240.000 HAD of 8.91 CLOCKS

FIGURE 5 - "DECOM" PRINT OF TEST SIGNAL AND FEATURES

```
HDTV-RGB (1125/60.0 2:1) TRI-LEVEL TEST SIGNALS.  TSF-1502
                     DIRECTORY: E:\MAGNI\TSF1502
    use FUNCTION keys to select signal or CURSOR keys and Enter to send
        Press the  ?  or  H  key for help with additional functions

        CB75              CB100              WHITE              BLACK
        COLORCHART        CONSTAL            VARLUM5%           VARCHROM10%
        VARPHASE10%       SPLIT5&10%         VARLUM10%          VARCHROM20%
        VARPHASE20%       SPLIT10&20%     F1 SET1            F2 FREQRESP
        MBRST_1-12        MBRST_10-32        MPLS_1-12          MPLS_10-32
        LINRAMP           TENSTEP            FLDSQRWV           CBSMPTE

                    2030 SERIES TRANSFER  VERSION 1.0
                 Copyright (C) 1989  by MAGNI SYSTEMS, Inc.
```

FIGURE 6 - TRANSFER SCREEN - 2030 TEST SIGNAL FILES

delivered form, and the software that is provided allows very simple changes to signals within one standard, or between standards themselves. In these respects, the unique use of signal memory has allowed for a totally flexible design which still retains user friendliness. It is to be hoped that the system allows the studio user to have the standard SPG/TSG in a rack, much as he has always been accustomed to, so that all the concentration may be made on the programming itself -- whatever the HDTV standard.

For the researcher into HDTV, the generator allows for the very speedy verification of a newly-proposed or a modified HDTV system without the manufacture of hardware for each system or expensive modifications because of small changes to a system standard.

TESTING OF ATV SYSTEMS FOR TERRESTRIAL BROADCASTING BY THE ADVANCED TELEVISION TEST CENTER
A PROGRESS REPORT

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Advanced Television Test Center
Alexandria, Virginia

INTRODUCTION

The Advanced Television Test Center (ATTC) was established in 1988 by CapCities/ABC, CBS, INTV, MST, NAB, NBC, and PBS to test proposed advanced television transmission systems for terrestrial broadcasting. In general, the tests will measure the quality, ruggedness against interference, ability to co-exist with NTSC signals, and operating characteristics of the new systems in the nation's electromagnetic environment.

The process will include objective and subjective testing, and, eventually, over-the-air testing. Objective measurements will include elements such as resolution for both static and moving pictures; subjective (psychophysical) testing will measure picture and sound quality and the effects of interference on them. The Test Center is working in cooperation with the FCC Advanced Television Service Advisory Committee to organize this process. ATTC is also keeping all announced proponents advised and is seeking their comments on development of both facilities and test plans.

In addition, the Test Center will explore the suitability of various spectrum bands for transmission of advanced television (VHF, UHF, and bands above 1 GHz) for single channel and wideband systems, and where appropriate, for such purposes as studio-to-transmitter links, network interconnection, and the like.

The objective is to provide the FCC and its Advisory Committee with the technical data to be considered in the selection of a broadcasting advanced television (ATV) standard. This data will help the government, industry, and public assess the ATV options for the U.S.

The Test Center's facilities are located in Alexandria, Virginia, near Washington National Airport, Metrorail, and highways US-1 and I-95. Additional facilities will become operational in the fourth quarter of 1989, keyed to when the FCC and/or its ATS Advisory Committee make various determinations which will affect the test design and when ATV system proponents themselves are ready with system hardware for testing.

PROPAGATION TESTING

The first tests, now being conducted by the Test Center, are those described in the Propagation Test Plan created by the Advanced Television Systems Committee (ATSC) and recommended by the FCC Advisory Committee. This effort is intended to provide information concerning how advanced television (ATV) systems, whose RF spectrum requirement exceeds 6 MHz, might perform when broadcast terrestrially--either in existing broadcast bands or in spectrum above 1 GHz. The work above 1 GHz is being undertaken in keeping with the recommendation through the Advisory Committee that research be done in this area, and in order to assess possible ATV system performance in existing broadcast microwave facilities (e.g. studio-to-transmitter links).

Several ATV proposals employ an NTSC channel and an augmentation signal which are combined in the ATV receiver to provide ATV features such as wide screen, increased horizontal and vertical resolution, and extremely high quality audio. Such approaches may be regarded as frequency diversity systems. In terms of the existing TV broadcast bands, for example, a VHF broadcaster might employ a UHF channel for the augmentation signal, while a UHF broadcaster--who uses a channel in the lower half of the band--might require use of an unused, higher UHF channel.

The propagation tests are intended to determine the differences in propagation between VHF and UHF, across the UHF band, and in spectrum above 1 GHz in terms of propagation loss, fading characteristics, multipath differences over the same path, and propagation delay differences.

The Test Center has been granted an experimental license for channels 58/59 in Washington, D.C., and experimental transmissions were begun early in November 1988 from the tower of WUSA-TV (Channel 9) in Washington, D.C. Channel 58 is used at times to simulcast WUSA programs to compare propagation characteristics between VHF and UHF bands over the same path to a number of sites within the service range of both transmitters.

Test Signal: In the other mode, the visual carrier (there is no aural carrier) frequency of the UHF transmitter is shifted to 740 MHz, the upper edge of channel 58, for double sideband transmission of a special broadband test signal. This test signal is the sinx/x pulse specially devised for this purpose so that its spectrum extends uniformly from DC to 6 MHz, or DC to 12 MHz, as a double sideband transmission. The special test signal is provided by a Tektronix signal generator.

Demodulator: A double sideband TV demodulator capable of receiving both sidebands of our 740 MHz carrier over the range 734-746 MHz was supplied by the Communications Research Centre (CRC) in Ottawa, Canada. This demodulator has two synchronous video detectors operating in phase quadrature so that both the in-phase and quadrature phase components of the received signal (and echoes of it) are recovered. Both the in-phase and quadrature phase video outputs from the demodulator are recorded on a digital oscilloscope. This device acts both as a waveform recorder and as a noise reducer enhancing the signal-to-noise ratio to permit measurements of the sinx/x pulse.

Measuring Device: The measurements are made under software control using custom software developed by CRC. The result is a plot of frequency response and group envelope delay across the 734-746 MHz band which the sin x/x pulse occupied as double sidebands. The effects of echoes on these measurements show up as frequency selectivity. This concept was pioneered by CRC and applied to testing of wideband cable television channels in Canada in 1988.

The Test Center will compare UHF propagation characteristics of channels at the upper and lower ends of the UHF television band. The field testing, using several hundred sites in the metropolitan Washington area, is being carried out with a field truck supplied by CBS and manned by ATTC and CBS personnel.

Tests Above 1 GHz: The double sideband sin x/x tests will also be carried out at 2.5 GHz and 12.5 GHz with transmitters which have been supplied by ITS Corp. and by CBS respectively. ATTC is conducting tests at two widely different frequencies in the microwave band over the same path to characterize wideband TV transmissions above 1 GHz.

Report on Findings: Propagation tests are expected to be completed in June. The data resulting from these tests will be reduced to a written report which will require at least several months. What ATTC hopes to determine is whether TV signals propagated over such bandwidths, terrestrially, are practical and whether the characteristics of radio paths at VHF, UHF and above 1 GHz are such that augmentation signal approaches to ATV may be technically feasible. In addition, the Test Center hopes to determine if spectrum above 1 GHz can be used for terrestrial television broadcasting.

These tests in Washington, D.C., alone cannot determine whether such possibilities will work everywhere; but, they should tell the industry and the government whether they might work at all. Further tests may be required over a longer period of time and conducted in more than one community by the government itself or other parties. This, then, is a first step--results of which should be able to be reported later this year.

LABORATORY TESTING

Laboratory tests of ATV systems will be conducted by the Test Center in its own facilities. Objective performance measurements such as horizontal, diagonal and vertical resolution, color gamut, etc., have been identified by the FCC Advisory Committee Planning Subcommittee Working Party 2 (PS/WP-2). Subjective picture and sound will be measured both for quality and system sensitivity to impairments. Guidance on this test plan came from the Planning Subcommittee Working Party 6 (PS/WP-6). The Test Center will also measure the interference caused to NSTC reception by these ATV signals on co-channels, adjacent channels and UHF taboo channels.

Testing guidelines for ATV systems were recommended by the Advisory Committee to the FCC last summer. ATTC technical staff is in the process of drafting detailed test procedures to determine if they can be implemented. In so doing, the Test Center must take into account, in addition to the costs to ATTC, the equipment actually available and the characteristics of ATV systems, both of which were not fully known in June 1988. The draft test procedures will be given to all known proponents for their comment.

All testing will be over an RF link simulating the limited RF bandwidth available to terrestrial broadcasting and simulating the "perils of the vehicular radio path": noise, interference and multipath, which can be expected to affect different ATV systems differently.

RF Test Bed

The Test Center has designed a terrestrial RF test bed. This design was reviewed by the Systems Subcommittee Working Party 2 (SS/WP-2) last fall and is now being constructed in the laboratory. When completed, it will be fully documented and plans will be available to proponents to permit testing and refinement of systems before delivery to ATTC.

In the case of ATV systems which employ an augmentation channel, the RF Test Bed can introduce different noise power into each channel, and different echoes can be introduced to each. The effect of radio transmitters using the same channel (for either channel) can also be simulated.

Present plans call for computer control of the operation of the RF test bed to avoid operator errors and to provide an accurate, detailed record of test conditions and results.

Subjective Testing

Preliminary thinking had called for objective (technical) and subjective (psychophysical) testing to be done sequentially in the laboratory, with a proposed system on-line. It became apparent that this would require considerable time and resources. For example, ATTC assumed the following basics. The number of viewers' opinions required to have good statistical validity was taken as 32. A larger number would only slightly increase the confidence in the data; however, this rapidly escalates costs--a limit of four viewers can sit 56 inches (3 X picture height) from a 38" HDTV Monitor. This would require using eight monitors or, with fewer monitors, repeating the tests for different groups of viewers. The use of different groups of viewers brings into question the uniformity of groups, particularly since the number of people in a group is small. Other cost factors include the cost of building, equipping and maintaining multiple viewing rooms.

Video-Taped Test Results

Given the above, ATTC considered video tape recording the tests so that all viewers could view tapes, perhaps always on the same monitor, and so that the Test Center could view the recorded signals displayed on the same monitor. Recording the output subjective test material (stills and motion sequences) would also permit comparing systems by specific impairments. The performance of all systems, for example, under co-channel interference conditions could be evaluated by the same set of viewers.

A number of ways of doing this has been investigated, and ATTC believes it has found a way to record the video signals for all ATV systems disclosed to date.

While 1125 line, 2:1 (interlaced) signals used by MUSE, VISTA, and Del Rey can be recorded on an 1125, 2:1 high definition digital video tape recorder (HD DVTR), it is, of course, the 525 and 787.5 line progressive scan systems, and the 1050 line interlaced systems, which provide the challenge. ATTC believes these formats can be recorded using a special data multiplexer which, in effect, "tricks" a high definition DVTR into thinking it is recording 1125 line video. This data multiplexer deals only with the digital video signals and it does not affect the spatial or temporal resolution of the recorded images. The Test Center has developed a design for such a multiplexer and is contracting for the full implementation of a prototype to demonstrate the feasibility of this concept.

Two-DVTR Testing: The Test Center plans to employ two digital VTR machines--one to play the motion sequence video tapes with which to test systems, the other to record the test results with the help of the special multiplexer.

Motion considerations: These are critical to the evaluation of ATV systems since these systems, in general, trade temporal resolution (ability to portray motion) for increased spatial resolution (definition). This tradeoff is justified by conclusions drawn from experiments which suggest that the human eye-brain system cannot perceive fine detail of moving objects. Different ATV systems have exploited this concept in different ways. Therefore, one may infer that while most, if not all, ATV proposals will produce excellent still pictures, significant performance differences may be found in the way they deal with image movement: camera movement or subject movement.

ATTC has chosen to use a digital HD VTR for its motion sequence testing. The motion sequences for this testing will be arranged for and provided through the FCC Advisory Committee (PS/WP-6). Should the sequences selected be available on analog tape, they will be copied to a digital recorder. Using the second digital recorder, the master tapes can be digitally duplicated so that an adequate number of effectively identical copies are available. Proponents could also obtain copies. In this way, all tests will be conducted on master-quality tape and the condition and adjustment of the DVTR should not be a factor. Such elaborate measures appear warranted considering the significance which may be attached to these tests.

It should also be noted that copies of digital recordings of system test results could be made available after completion of testing for evaluation by the FCC Advisory Committee, the FCC, proponents and others.

Still Pictures: Important as motion sequences are, still pictures will also play a key role in testing. Among others, resolution test patterns are still pictures.

ATTC is purchasing a special still-store system, the PIXAR, to generate test patterns and to process digitized photographs. A software specialist firm is developing custom software to control the PIXAR. This still store is entirely digital and can read out digital image data at real time HDTV clock rates, even though it is loaded from a data tape at a much slower rate.

A number of test pictures have already been produced and digitized for the Test Center by Kodak. These test pictures come in the form of digital data recorded on data tapes. The data is copied onto a data disc which, in turn, can be rapidly copied to the PIXAR high speed random access memory (RAM).

The original data are in the form of RGB linear camera signals digitized in 16 bits and color corrected to SMPTE "C" assumed phosphor colorimetry. The data come as a super-high resolution image of some 2000 lines by 4000 pixels. The PIXAR system includes a Sun Computer which performs digital filtering and resampling functions (off-line) and processes the data to generate a test pattern in a given system format. Therefore different files will therefore exist containing the exact same picture in 525/59.94, 1:1, 1125/60, 2:1, and so forth, without having to shoot such pictures with a TV camera. ATTC does not plan to employ any camera in its testing work.

Interference Issues and Measurement: Both simulcast systems and augmented NTSC systems would require the use of more channels than are used today. These systems must be carefully tested to determine whether they can be transmitted without causing significant interference with existing broadcasts and whether these signals could survive the interference caused to them by NTSC signals.

Evaluating interference susceptibility is an especially important part of ATTC's testing program. Such tests are subjective by nature. Some ATV systems transmit a signal claimed to be directly receivable on existing receivers without significant performance impairments to either sound or picture. At least one simulcast system claims that simple converters can be made so that NTSC receivers can be used, to view the picture. The quality of picture and sound reception will be examined using a bank of consumer receivers and video cassette recorders, selected by and supplied through the Electronic Industries Association (EIA), which represent a cross-section of such equipment.

SUMMARY

The foregoing information outlines the Test Center's present plans for propagation tests in Washington, and for laboratory tests, as they now stand. Because ATTC is working on the frontier of TV technology, some unanticipated difficulties are to be expected. Nevertheless, the Test Center believes it is proceeding in the right direction. Some of these advanced techniques developed by ATTC for testing ATV systems may be useful to the broadcasting industry once the standard for ATV has been chosen.

Before its work is complete, laboratory testing should be confirmed by over-the-air tests which the Test Center will help organize. On the basis of the laboratory tests results, only a small number of systems may need testing over-the-air. Already ATTC has received several offers by broadcasters to make their facilities available for air tests.

The ultimate goal of the Advanced Television Test Center is to provide conclusive data which will be useful in determining the future of Advanced Television in North America.

The author wishes to thank Mr. Robert Niles, Chairman of the Test Center's Technical Committee for his encouragement to write and deliver this paper.

CABLE TESTING FOR ADVANCED TELEVISION SYSTEMS

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Introduction

The cable industry is a diverse collection of companies with different approaches, goals, and opinions. There really can't be just one cable industry position on Advanced Television, ATV, Systems. I've put forth my thoughts on the subject based on talking with others in the cable industry, other related industries, and potential competitors. But after all is said and done, this is just one opinion.

Rational Expectations

Realistically, when can we expect ATV to arrive? When will proponents have hardware to test? When can testing begin? And how long will it take? As long as we're exploring our crystal ball, when can we expect ATV to be commercially significant?

The Federal Communications Commission, FCC, has created an Advisory Committee of industry experts to recommend a course of action. Ultimately, the FCC is the only entity in the US which is empowered to decide the outcome for terrestrial broadcast. The FCC Advisory Committee has three subcommittees and a passel of Working Parties. The Systems Subcommittee Working Party 1, SSWP1, is charged with the initial analysis of proponents. This is the point at which proposals are submitted to the Advisory Committee for consideration. SSWP1 conducted a "Hell Week" during which all proponents presented their systems and answered questions. One of the important questions concerned the availability of hardware for testing. Many promised fourth quarter 1989 while some said mid 1990 would be the earliest possible delivery date. Anyone experienced in engineering knows Murphy's laws and how they apply to delivery schedules. The well intentioned promises made by proponent *management* and the reality of producing the hardware by proponent *engineers* will likely have some discrepancy. Hardware availability forms an important limiting condition on testing. Testing cannot begin without hardware. Testing *should not* begin until a steady flow of proponent hardware is available. Otherwise costly periods of inactivity will inefficiently consume limited testing budgets.

Cable Labs is scheduling its own mini "Hell Week" to serve as the mechanism for proponents to indicate their interest in being considered for cable application. These sessions are likely to last one or two days and concentrate on the cable aspects of proponents' systems.

At the 1989 Electronics Industries Association, EIA, Winter Consumer Electronics Show in Las Vegas, there were three panel sessions on High Definition Television, HDTV. Alex Felker, FCC Mass Media Bureau Chief, probably made the most important statement about HDTV standards: "...we're already about eighteen months into this process ... and in many ways we haven't come all that far ... we're looking at somewhere around another eighteen months, perhaps another two years before testing is complete and the Commission would be in a position to select a transmission standard." After testing is complete, a few months will be required by the FCC Advisory Committee to digest the results and make its recommendation to the Commission. The Commission can only then make its decision. A year or so longer and the first products will be introduced.

The Advisory Committee's System Subcommittee Working Party 2, SSWP2, is charged with the actual testing. Of course, SSWP2 has no budget and is staffed with volunteers. It must be dependant on others to implement the tests. Specifically, the Advanced Television Test Center, ATTC, and Cable Labs are expected to be helpful. SSWP2 produced an interim report in February of this year. It was reported that ATTC expects to be ready for testing in October of 1989 and that testing will take from one and a half to two years. This is in agreement with the assessment by Alex Felker.

The growth in penetration of consumer electronics products is a well known phenomena. Experience with Radio, Black and White TV, Color TV, and VCR's provides guidance in making predictions on how HDTV will grow. Typically, when first introduced a new consumer electronics product is very expensive and growth is very slow. When the first Black and White television receivers were introduced, they cost about as much as a compact car. Likewise, when the first color TV receivers were introduced, they cost as much as a compact car of that day. It would be reasonable to expect the first HDTV receivers to cost about as much as a Hyundai. The growth curve for these kinds of products involves a long shallow rise over ten to twelve years reaching less than ten percent of TV house holds. Then a certain price point is reached and the penetration curve turns almost straight up. Seventy to eighty percent penetration is achieved in just three to four more years. It is rational to expect that it will be between twelve and fifteen years before commercially significant penetration of HDTV receivers is achieved. Any more aggressive projection is out of line with past experience and begs for an explanation of the deviation.

Cable Labs

The cable industry has formed an R&D consortium in 1988. The consortium is called Cable Television Laboratories, or Cable Labs. It is funded by members at the rate of two cents per cable subscriber per month. This generates slightly in excess of eight million dollars per year. This funding supports many activities. Just one of the activities will be ATV testing. It is important to realize that only a minority fraction of the budget is available for ATV. Some have assumed that ATV testing will be the principal activity of the Cable Labs. This is incorrect. It is expected that the Cable Labs ATV work will at least partially support the efforts of the FCC Advisory Committee.

At this point in time, Cable Labs has very limited staff and has not yet found a permanent location. Much of the work must be done by volunteers while Cable Labs is establishing itself. Time is required to hire staff, plan tests, and implement them. If the rational expectations discussed above are correct, there will be ample time to do the testing in a professional and reasonable manner. The fact that test facilities, staff, and plans are not yet firmly established is not a problem if placed in the proper time perspective. Adequate progress is being made.

Cable Labs has appointed Walter S. Ciciora of American Television & Communications, ATC, as its Director of Advanced Television Projects. He is on part time loan from ATC for this effort. In this capacity, he has responsibility for formulation and conduct of ATV programs and efforts including HDTV testing. In addition, Nick Hamilton-Piercy of Rogers Communications serves as chairman of the ATV Subcommittee of the Cable Labs Technical Advisory Committee, TAC. The ATV Subcommittee advises and oversees Cable Labs efforts in ATV.

Cable Labs is most anxious to cooperate with Broadcasters in general and on ATV issues in particular. Cable Labs has a cooperative attitude towards ATTC. As just one example of how Cable Labs and ATTC are working together toward common goals, consulting work on subjective test plans are jointly being sponsored. Cable Labs and ATTC have freely shared information and opinions on testing procedures and plans. There is every reason to expect that this cooperation will continue. We have many of the same goals and objectives and wish our viewers to be satisfied.

Limited Resources

Because of limited resources, Cable Labs is most interested in finding suitable existing facilities to hire for handling the tests. Either a single facility to conduct the entire project or a collection of facilities to address various aspects of the tests are being considered. These facilities must be competent, respected, affordable, available, and free from conflict of interest. A search is underway for such facilities. A number of research, industrial, academic, and government labs are being considered as well as specialized efforts such as ATTC. Only as a last resort will Cable Labs seek to build its own facilities and hire its own staff. This latter course of action has the serious drawback of what to do with it after the HDTV work is complete. This is a particularly troublesome issue for the staff which would have to be temporarily hired.

Another consequence of limited resources is that the testing objectives must be limited also. There are two ways of applying limited resources. A limited number of proponents can be thoroughly tested or all proponents can be tested in a limited way. The former approach is to be preferred. Testing all proponents poorly may not help in making a choice, or worse yet, may yield an erroneous result. Prudence requires limiting the field so careful testing of a limited number of proponents will result in meaningful answers.

Cable Goals

With the above discussion as background, what are cable's goals in ATV? The two most important goals are that cable must preserve its ability to compete and that cable must be able to deliver the signal chosen by the broadcasters. The cable subscriber must see no apparent loss of quality when terrestrial broadcast signals are received via cable. It is important that this be done cost effectively. Being able to compete boils down to the need to insure that no artificial ceilings are placed on the quality cable can deliver. Cable's most important competitors in the ATV arena are: 1) Pre-recorded media, 2) Telco, 3) Direct Broadcast Satellite, DBS, and 4) Broadcast. The first three of these are in a position to deliver truly excellent video. Cable must not be second rate.

A third goal is that of compatibility. The 140 million television receivers built to the current technical standard, called the National Television Systems Committee, NTSC, standard, will be around for the foreseeable future. They must be served. But must all ATV signals be viewable on NTSC receivers? For those signals which are viewable in both formats, what is the best way to achieve this?

A fourth goal is a need to insure that ATV is capable of those things that are somewhat unique to cable service. Included here are the need for addressability, truly secure scrambling, and delivery via satellite to cable head ends. These must be accomplished cost effectively. They must not be band-aids or add-ons. They must be built in to the ATV system itself. Addressability is the capability to individually control each subscriber's access to programing for which fees are expected. The addressing signal must be secure, fast enough, and capable of a large enough audience. Addressing of cable subscribers must be under the direct control of the cable operator. The scrambling must both hide the video from young eyes so parents don't object and be undefeatable to those who would wish to steal the programing. If the video can be occasionally made unrecordable on consumer VCR's, perhaps early pay-per-view "windows" can be enjoyed. But this capability must be switchable to a recordable mode for more normal use.

The last goal is cost. Cost includes the matters of spectrum space, level of signal quality, and the most difficult question of all: "How good is good enough?". Cost also has a time horizon. Quality which is adequate today may be embarrassing tomorrow.

Critical Issues

A number of critical issues remain unresolved. These problems must be solved if meaningful tests are to be conducted.

The various proponents have built their systems on one of four different sets of scanning parameters. Several of the proponents claim that if the test video is not generated in their particular format, they will be so disadvantaged that they might just as well not participate in the tests. There is another group of individuals which claim that *only* the 1125/60 production standard should be used as original source material. Some elements of that group are trying to abuse this issue to build a case that 1125/60 should be the *only* production standard. These positions are incompatible and must be resolved. A practical aspect in all this is the very limited availability of equipment in any of these scanning configurations. If equipment is not available, testing cannot be done. If important proponents decline to participate because of inappropriate input signals, testing likewise cannot commence. This will not be easily resolved.

All systems proposed for terrestrial broadcast and cable involve the compression of bandwidth from studio grade 30 MHz signals to 6 MHz or some similar much smaller value. All systems promise near perfect still pictures. This is not expected to be a major difficulty. The trouble arises when motion sequences are considered. Here's where noticeable differences can be expected. The design of the input video and the test procedures to ferret out the differences between the proponents is vitally important. The issue is made complex by the above discussed apparent need for four different scanning parameter sets. In particular, proponents with progressive scanning rather than interlaced scans claim that a progressively scanned source is necessary if their motion methods are to be adequately tested. Conversion from interlace to progressive scan cannot adequately make up for the motion information lost in the taking process. A further complication is the problem of camera lag and camera light sensitivity. Current ATV cameras are tube based, not employing Charge Coupled Devices, CCD's. These tubes have artifacts generating smears on moving objects which mask the motion artifacts created by the proponents' systems. If motion performance is to be adequately tested, a source of lag free video must be found. This will be difficult.

Another issue revolves around the display device used for testing. Anyone experienced in video system testing knows that differences between display devices can overwhelm differences between systems under test. The best approach is to observe all proponents on the *same* display device. However, the fact that there appears to be four different scanning parameters makes the problem exceedingly difficult. There is an interaction between the scanning raster and the dot structure of the shadow mask tube resulting in a Moire pattern which looks like a large finger print on the screen. The design of display devices is an art which seeks balance between conflicting requirements. One of the requirements is to minimize the Moire pattern. This is difficult enough when only one scanning raster is involved, but to accommodate four parameters may be asking too much. One possible solution is to fund one manufacturer's effort to make several display devices with carefully matched electron guns and identical phosphor formulations, preferably from the same batch. The drive circuitry would have to be the same and the adjustment of the displays carefully supervised with specialized test equipment. This approach will be expensive, time consuming, and controversial. Ignoring the display problem will only result in a less creditable result subject to later challenge. The likelihood is that the tests would eventually be repeated more carefully after legal action.

Another display related issue is the impact of the viewing conditions on the subjective observers' results. A common problem with currently available ATV displays is that they are not very bright. Every ATV demonstration to date has taken place in darkened rooms or with a bezel to exclude ambient light. The subjective viewing tests must be designed to minimize the negative impact of these display deficiencies on viewer response.

A further deficiency of current displays is their size. This makes testing difficult when it is realized that viewers need to be no further than 1.5 times the picture height if careful testing of motion artifacts is to be accomplished. As discussed above, motion artifacts are the likely area of critical difference between various proponents. An experience at the 1989 EIA Winter Consumer Electronic Show in Las Vegas put this matter into perspective. Panasonic displayed a prototype Improved Definition Television, IDTV, receiver with three by four aspect ratio and seventy inches of diagonal measure. The receiver displayed pictures from a video disk source. Most observers positioned themselves at about 1.5 times picture height. At that distance, moving objects in the picture were tracked by turning the head and moving the eyes. Under these conditions, the usual ATV bandwidth reduction technique of allowing moving objects to have less resolution than stationary objects results in unpleasant effects. Pre-recorded ATV media may be free of such problems resulting in unfavorable comparisons between tape and disk versus cable and broadcast. Only by viewing the test at 1.5 times picture height can these problems be properly included in the test results. Greater viewing distances will mask the effect and provide incorrectly optimistic results. These results will be invalid when viewers have large screens.

Another critical issue involves whether testing will be done on a proponent by proponent basis or on an impairment by impairment basis. The first approach involves setting up a proponent and running through all subjective tests. The remaining proponents are taken in sequence, one at a time. Of practical necessity, different sets of viewers would be involved from proponent to proponent. The second approach requires recording all test results for later play back. Then all proponent results of a single impairment test, say multipath, are presented to one group of observers, and the results tabulated. This approach is likely to be the more valid since all proponents are observed by the same group of subjects for a particular impairment. Testing by proponent will result in tests being done by different observers.

Much discussion has taken place over whether incomplete systems should be tested. "Incomplete" has usually meant systems without the sound transmission method in place. Proponents have argued that they do not have the time to prepare sound equipment and since sound will be digital, it doesn't matter anyhow. Others have pointed out that allocation of the spectral resource between sound and picture involves trade offs that must be included in the testing. Left out of this discussion has been the fact that the addressability and encryption technique signals also involve trade offs that need to be tested. A strategy for dealing with this ignored issue needs to be evolved.

There are many other unresolved issues of lesser importance. Those discussed above are of a "show stopping" nature. They need fair, affordable answers to which proponents will agree. Without this, testing cannot fairly begin. If testing is forced without adequate resolution of these issues, the lawyers will consume more time and money than is saved by taking short cuts.

Likely Form of Cable Testing

The FCC Advisory Committee Planning Subcommittee Working Party 4, PSWP4, is charged with considering alternate media, including cable. They have created a test plan. This plan is not a test procedure, but a statement of what is to be tested, and broadly speaking, how it is to be tested. The detailed procedure is left to those who will be doing the testing. Presumably this means Cable Labs and/or Cable Labs contractors.

While details have not yet been set, a few general principles can be described. Even these are subject to review and revision by Cable Labs staff.

Cable has two main categories of interests: 1) those proposals which would be suitable for use to deliver cable signals to the home, and 2) those proposals which are likely to be used by broadcasters for terrestrial delivery. Since cable will want its subscribers to have access to broadcast signals, cable wishes to contribute to efforts aimed at insuring quality delivery of broadcast signals to cable homes. In the best of all worlds, these two sets of proposals could be merged. But we must be prepared to deal with the possibility that the surviving cable and broadcast systems will be distinct. Most likely, a review of the proposals will reveal a number which fail to serve either of these purposes and can be eliminated without test.

There will likely be a small number of objective tests done by experts to determine if any of the proponents are technically incompatible with cable technology likely to be practical in the time frame when ATV is commercially significant. The surviving proposals will then be tested for the range of impairments over which they provide some degree of reasonable performance. A common set of ranges for levels of impairments to be used in all tests will be determined by the panel of experts.

It is most desirable that all surviving proponents be tested over identical ranges and the results recorded, preferably in digital format. Then the tapes are edited so that all surviving proponents' responses to a given set of impairment values can be displayed in random order to the same group of subjective observers who will scale their observations. This "impairment by impairment" approach insures most equal treatment of all tested proponents. The test results will then be weighted by Cable Labs staff and compiled into system evaluations to determine the system or systems to be supported.

Cable Labs will likely use a test bed to simulate a cable system for the tests. Impairments to be tested include but are not limited to random noise, impulse noise, power supply noise, non-linear effects such as composite triple beat and cross modulation, micro reflections from cable impedance miss-matches, and phase noise introduced by converters and modulators. Satellite links, microwave links, FM super trunk cable links, and fiber links will likely be included or simulated in the tests.

In the interest of being cost effective, a set of screening tests will be sought will can be used to limit the number of surviving proponents which will be subject to the more complete testing. The first step in the process will be the Cable Labs "mini Hell Week" during which proponents will indicate their interest in cooperation with Cable Labs and commit to joint efforts. A search will be made for "leading indicator" or "surrogate" tests which can comprehensively represent performance to a group of related impairments. This strategy not only saves time and money, but lessens the burden on the subjective observers.

Of course, the final surviving proponents will likely be tested on a selection of actual cable systems, probably with simultaneous delivery via satellite. It is likely that the broadcasters' chosen system will be finally tested via off-air pick up and fiber studio link to cable head ends. These ultimate tests will insure nothing was over looked in creating the cable test bed.

Other Issues

There are a number of other issues which must be understood if valid testing is to be undertaken. They will be discussed in the spirit of shedding light on cable's objectives and needs in testing ATV proponents.

Large Screen TV

ATV and really large TV screen displays are so synergistic that I believe one can't happen without the other.

All of the early consumer research on ATV has indicated that viewers see little difference between NTSC and ATV if they are more than three or four picture heights away from the display. In normal living rooms, people sit six to eight feet from the screen. For ATV to be noticeably better than NTSC, the screen must be two to three feet high. Since TV's are normally described in terms of diagonal measure, the size screen needs to be fifty to seventy five inches for the wider aspect ratio. Less than that and most viewers won't see the difference between NTSC and ATV.

Conversely, anyone who has seen a large NTSC projection set is dismayed at how poor the picture can be. This is due to the shortcomings of NTSC which was created at a time when technology had to strain to provide a twelve inch picture. ATV is critical to the sale of large screen TV's.

Research on cost effective, bright, large screen displays is well underway in Japan and elsewhere. However, consumer products are still about ten years away. This is very much in concert with the time frames described earlier. Large screens and the growth in penetration of ATV will go hand in hand. Incidentally, the need for a large screen to enjoy ATV will keep ATV as an upper price point product. It is unlikely that inexpensive receivers in the $200 to $300 range will ever be anything other than good old NTSC.

Pre-recorded media

The most immediate cable competitive concern is over pre-recorded media. Magnetic and optical recording technologies are the video arts which have made the most progress in the last ten years. They are also the areas most likely to make dramatic progress in the future. A startling demonstration was provided to the National Cable Television Association, NCTA, HDTV Blue Ribbon Panel members who participated in the March 1988 visit to Japan. Mitsubishi showed a 20 MHz baseband VCR prototype which recorded and played back near-studio quality, wide screen HDTV. Since the recording was at baseband, no video compromises were required. There were none of the motion artifacts we've come to expect from bandwidth-reduced HDTV. The mechanism of the prototype machine was nearly identical to standard VHS design. What we saw was an eminently practical approach. Mitsubishi subsequently demonstrated the device at the 1988 NCTA convention in Los Angeles.

The 1988 Institute of Electrical and Electronic Engineers, International Conference on Consumer Electronics, IEEE-ICCE, in Chicago included several papers on digital VCR's for consumer electronics. The IEEE-ICCE technical papers generally appear two to five years before products are introduced. The message is clear, consumers will have digital quality VCR's in the near future. First they will record ordinary NTSC, then they will evolve to ATV.

The optical disk is slowly building momentum. A huge and growing list of titles are available. Rental stores are popping up. The disk is a low noise media providing cleaner video. The best pre-recorded video I've ever seen comes from a Sony professional HDTV video disc. The absence of noise adds tremendously to the realism.

From cable's perspective the concern is this: five to ten years from now, a subscriber rents or buys a disk or tape to view at home on his large screen display. Afterwards he continues viewing a cable channel such as HBO. The tape or disk was recorded at baseband using 20 MHZ to 30 MHz with no compromises. The cable signal is bandwidth compressed video with motion and other artifacts. On the large screen, the direct comparison may prove disturbing.

Broadcast

Broadcasters are in an extremely difficult position. Spectrum is scarce. Most importantly, the spectrum that is available may not be of sufficient quality to provide ATV in all locations. Broadcasters face the problem of multipath. Reflections from hills, mountains, towers, airplanes, buildings, etc. cause multiple signals to be received. These appear on the screen as ghosts or as blurring of the main signal. The vestigial sideband nature of the signal makes the ghosts particularly bothersome. ATV's doubled number of scan lines means that the scanning speed is doubled. This doubles the displacement of a ghost on the screen. Since the objectionability of a ghost is an exponential function of its displacement, ATV ghosts are nastier than NTSC ghosts of the same severity. Additionally, many ATV systems use time compression to separate the luminance information from the chrominance signals. When these signals are uncompressed in the receiver, a single ghost is converted into two ghosts with different locations and sizes.

The consumer electronics industry has worked on "ghost cancellers" for a couple of decades. They still remain impractically complex systems with marginal performance for even the less demanding case of NTSC. It is likely that ghost cancelling will remain out of reach for quite some time to come.

The importance of direct broadcast studio links to cable head ends will increase with ATV. This technique best serves cable subscribers and broadcast viewers alike. This especially makes sense when we consider that ATV is a large screen, relatively expensive phenomena, to be enjoyed in the primary viewing location of the home. In the time frame when ATV becomes commercially significant, cable penetration will be at least eighty percent of house holds. Most anyone with an interest in video will be a cable subscriber. From this we conclude that at least ninety to ninety five percent of initial purchasers of ATV receivers will be cable subscribers.

Compatibility

"Compatibility" is a rubber word often stretched to meet the needs of the moment. One respondent to the FCC's Notice Of Inquiry, NOI, has created an elaborate six level hierarchy of compatibilities. From the cable operators' perspective this is nonsense. NTSC compatibility can only mean that an ATV signal is also viewable on essentially all existing NTSC receivers with acceptable quality and without adaptor boxes or modification of the subscriber's NTSC receiver. Anything else is simply not "compatibility".

The last thing we need is to be put in the position of having to provide adapter boxes so existing subscribers can continue to view current programing.

The Myth of the Single Universal Standard

In the best of all worlds, a single universal ATV standard would reduce consumer confusion and increase efficiencies. Lower prices and faster adaptation of ATV would result. It would provide a "level playing field" for all video delivery competitors. Price, service, and quality of programing would be the instruments of competition. The example of the free market approach to AM-stereo clearly displays the problems of multiple standards.

Unfortunately we do not live in the best of all worlds. In our world there are real, physical differences between video delivery methods. Video has changed a lot since the simple days when black and white TV was introduced. Then the consumers' only choice was off-air reception. Compatibility then meant only that all broadcasters used the same technical standards so the consumer could get by with only one receiver. When color was introduced, cable was an insignificant part of the video scene. Compatibility again meant that all broadcasters used the same technical standards. But "compatibility" gained an additional meaning. The old black and white receivers had to be served by the same signal that put color pictures on new sets.

Today the situation is much more complex. The consumer not only receives signals off-air, he also subscribes to cable, rents tapes and discs, and maybe has a DBS dish receiving frequency modulated, FM, video. Furthermore, he hears of the phone company wanting to provide digital video signals over fiber. Since these delivery means are diverse in their fundamental technology, a single universal standard is impossible. Vestigial Side Band Amplitude Modulation, VSB-AM, FM satellite, digital fiber, and tape and disc recording are just too different in their basic technologies to come under a mandatory, comprehensive, single universal standard. This would be like coming up with a single universal standard for home heating which applies to gas, oil, coal, wood, cow dung, and electricity.

The regulatory situation is also much more complex. While the FCC has rigid control over some of the video transmission means, it has virtually no control over others. For example, no one asked the FCC for permission to introduce Super VHS. Likewise, no one needs to ask for permission to introduce a prerecorded ATV format. Unless this is changed in ways that seem highly unlikely at present, a single universal standard is simply impossible! Cable must retain its ability to compete in this complex new world.

Signal Robustness

The broadcasters' signal is important to cables' subscribers. They expect it. When broadcasters begin transmitting ATV signals, cable subscribers will insist on receiving those signals over their cable connection.

Cable's concern over the broadcaster's signal is primarily over robustness. Cable processes the signal many times before it reaches the subscriber. In doing so, the signal may become bruised. If the broadcasters achieve ATV in the present 6MHz, they will have to squeeze even more information into the existing bandwidth. This will most likely reduce the robustness of the signal and make it more subject to damage. The cable industry must work with the proponents of systems meant for use in terrestrial broadcast to minimize this cable hazard.

Conclusion

ATV will be a source of concern, effort, and excitement for at least a decade. We have our work cut out for us. When considered carefully, the problem of ATV testing is very complex indeed.

Biography

Dr. Ciciora is Vice President of Technology at American Television & Communications, ATC, in Stamford Connecticut. Walt joined ATC in December of 1982 as Vice President of Research and Development. Prior to that he was with Zenith Electronics Corporation since 1965. He was Director of Sales and Marketing, Cable Products, from 1981 to 1982.

Earlier at Zenith he was Manager, Electronic System Research and Development specializing in Teletext, Videotext and Video Signal Processing with emphasis on digital television technology and ghost canceling for television systems.

He has nine patents issued. He has presented over seventy papers and published about thirty, two of which have received awards from the Institute of Electrical and Electronic Engineers. Walt writes a monthly column titled "Ciciora's Forum" for Communications Technology magazine.

He is currently chairman of the National Cable Television Association, NCTA, Engineering Committee, Chairman of the Technical Advisory Committee of Cable Labs, and Director of Advanced Television Projects for Cable Labs. He is a past chairman of the IEEE International Conference on Consumer Electronics. Walt is a Fellow of the IEEE and a Senior Member of the Society of Cable Television Engineers. Other memberships include the Society of Motion Picture and Television Engineers, Tau Beta Pi, Eta Kappa Nu, and Beta Gamma Sigma. He has served on several industry standard-setting committees. Current interests center on competitive technology, the consumer electronics interface with cable, and High Definition Television.

Walt received the 1987 NCTA Vanguard Award for Science and Technology.

Walt has a Ph.D. in Electrical Engineering from Illinois Institute of Technology dated 1969. The BSEE and MSEE are also from IIT. He received an MBA from the University of Chicago in 1979. He has taught Electrical Engineering in the evening division of IIT for seven years.

Hobbies include reading, wood working, photography, skiing, and a hope to someday become more active in amateur radio (WB9FPW).

PROPAGATION TESTING FOR ADVANCED TELEVISION BROADCASTING SYSTEMS

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ABSTRACT

The development of an advanced television system for broadcast facilities in the United States requires that there be some understanding of the characteristics of the radio frequencies over which the new system will be transmitted. A program to conduct propagation tests in the present broadcast and SHF bands has been developed and field tests will be conducted during the first half of 1989. The results of the tests will be used by advanced television system proponents in the development of their systems for broadcast applications.

BACKGROUND

One of the tasks of the Advanced Television Systems Committee ("ATSC")[1/] in 1987 was to establish a task force to design and conduct propagation tests in various frequency bands to determine the ability of each band to support a broadcast advanced television service ("ATS"). The task force was composed of representatives of the three television networks, industry associations, equipment manufacturers and the Canadian Government.[2/] Washington, D.C. was chosen as a test site because of its proximity to most of the participants and its apparent suitability for the tests to be conducted. Because of widespread interest in advanced television, The Federal Communications Commission ("FCC"), in November, 1987, formed an industry advisory committee for an advanced television service to examine all aspects of the issue and report its findings to the Commission.[3/]

In 1988 the Advanced Television Test Center ("ATTC")[4/] was formed to conduct objective and subjective tests of various advanced television ("ATV") systems offered by a wide range of proponents. The propagation testing project was turned over to the ATTC in order to coordinate the testing effort and provide the substantial funding needed to operate the ambitious test program. The Test Center is working with Working Party 2 (Test & Evaluation) of the Planning Subcommittee of the FCC's ATS industry advisory committee and other groups and committees which have an interest in the results of the propagation tests.

Purpose of The Tests

The rationale for conducting the tests, in view of the fact that considerable literature and knowledge exists on propagation of the UHF and SHF frequency bands, is that advanced television systems incorporate complicated modulation and transmission schemes which may be affected by characteristics of over-the-air transmission not presently known or quantified. Some ATV systems employ an augmentation channel in addition to the normal NTSC channel which must be combined at the receiver in order to display an ATV image. Because it is unlikely that the two channels can be contiguous and may be contained in separate bands, the propagation differential between the two bands must be determined. There is little in the literature that describes this differential.

Some ATV systems employ sophisticated modulation schemes which may be adversely affected by changing characteristics within the transmission channel. Again, there is little available information on this subject except for standard differential gain and phase measurements.

The need for more spectrum to support those ATV systems which employ either an augmentation or simulcast channel requires substantial investigation to determine if the existing bands have the space or if spectrum in bands other than traditional television bands can be used for ATV transmission. Working Party 3 (WP-3 Spectrum Analysis & Alternatives) of the ATS Planning Subcommittee determined that not only would the existing VHF and UHF bands be needed for ATV but that frequencies above 1 GHz should be investigated as well. To this end the WP recommended that propagation tests be conducted in the broadcast bands and bands to determine if frequencies above 1 GHz are suitable to support ATV if the exiting bands cannot.

One of the original tests to be conducted by the propagation task force was to transmit a signal wider than the normal 6 MHz channel in order to determine the characteristics of a 9 or 12 MHz channel for those ATV signals that use a wide band transmission format. However, the FCC, in its Tentative Decision and Further Notice of Inquiry,[5/] issued in November 1988, recommended that the industry narrow its options and tentatively decided that ATV transmission formats should be limited to systems that a) are compatible with the existing NTSC system, b) employ a maximum of 6 MHz for an augmentation channel or c) employ up to 6 MHz for a non-compatible simulcast channel. While the FCC does not advocate using a channel wider than 6 MHz for ATV, the propagation tests will use a special 12 MHz transmission format in order to more accurately characterize the existing 6 MHz channel and to explore wider channels above 1 GHz.

In order to avoid using any specific ATV format the committee agreed to use only test signals during the propagation tests.

Tests To Be Conducted

The propagation task force determined that two distinct test programs needed to be conducted to properly characterize propagation conditions for ATV. The first test program, referred to as the dual-band test, is a long-term test to evaluate a) the time of arrival differential, b) amplitude differentials, and c) group delay between identical signals in the VHF and UHF bands and between two signals widely separated in the UHF band. To conduct this test the same signal would be transmitted from the same site over the same path to a single receiving location. The first stage of this test would be to evaluate the differences between a signal in the VHF band and another in the UHF band with the same modulating signal on both. A second stage would be to evaluate the differences between two signals widely spaced in the UHF band.

The second test program, referred to as the wide-band test, and certainly the most ambitious of the two, is the field test project using a mobile measurement unit with antennas on a 30-foot mast and special wideband receivers for UHF, 2.5 GHz and 12 GHz bands and a special computer control system. These tests include a) measuring absolute and relative signal levels, b) determining phase and multipath characteristics of the channel (using a 6 MHz sinx/x test signal on an amplitude modulated double sideband carrier in the UHF, 2.5 GHz and 12 GHz bands) and c) measuring de-polarization effects in the UHF band.

The equipment taking the measurements is under computer control to increase efficiency and reduce errors.

Transmission Test Facility

In order to conduct propagation tests there must be a transmission site as well as reception facilities. For these tests the transmission facility must be capable of supporting the transmitters and test signal generators, have convenient access and provide a reasonable level of security. WUSA-TV (channel 9), owned by Gannett and a CBS affiliate, offered their building, transmitter room and tower to be used for the tests.

The channel 9 tower is located in the northwest corner of Washington, D.C. There are two towers of which the taller is used for their (and channel 7) main antenna and the shorter is used for the channel 7/9 standby antenna and auxiliary services. Channel 9 offered space on the shorter tower for the three antennas the project planned to erect - UHF, 2.5 GHz and 12 GHz. Channel 9 also offered space in their transmitter room for the project's equipment.

FCC Transmission Authorization

The selection of Washington, D.C. for the test site, while convenient to the task force, nevertheless created problems for obtaining frequencies for conducting the tests because of severe frequency congestion. However, through the use of engineering consultants, frequency coordinators, cooperative licensees of various existing facilities, the channels were found to permit the tests to be conducted at the frequencies and in the geographical area of interest.

UHF Facility: Channels 58 and 59 were chosen in the UHF band for the wide-band signal.[6] A search of the UHF spectrum revealed that transmission on channel 58 and 59 would be possible if the energy on channel 59 would not cause sound image interference to reception of WBFF-TV channel 45 in Baltimore, Maryland located only 35 miles to the north of Washington, D.C. With the agreement of channel 45 and the provision to cease transmission if sound IF image interference occurred, the FCC granted an experimental radio license for a facility in northwest Washington, D.C. with a power level of about 1,700 watts ERP and a directional antenna oriented away from Baltimore. The FCC granted the call sign WWHD-TV for the UHF facility.

Two modes of operation were established to permit the specially modified ITS 1 kW UHF transmitter to operate a) as a normal NTSC signal on channel 58 for the dual-band tests and b) as a double-sideband, wide-band signal on channels 58 and 59. For the wide-band test, the visual carrier is placed between channel 58 and 59 and there is no aural carrier.

Two transmitting antennas, specially designed by Micro Communications Inc. were installed at the 275 foot level of the channel 9 tower. One antenna is horizontally polarized and the other is circularly polarized. Both have the same gain and essentially the same beamwidth.

2.5 GHz Facility: Finding two 2.5 GHz channels was more difficult than obtaining the UHF channels. The combined mix of licensees (ITFS, MMDS, OFS) and uncertainty about the potential for interference in this heavily used band in Washington, D.C. and vicinity forced the project to look at other nearby bands.

An examination of the 2.5 GHz broadcast auxiliary band (2,450-2,500 MHz), which is shared with the operational fixed service (OFS) and industrial, scientific and medical (ISM) facilities, appeared to offer the best prospect for an experimental test facility. Again, because of the mix of licensees in this band, and the lack of formal frequency coordination procedures, substantial time was required to contact actual and potential users of the band. It was found that a) no fixed facilities were in operation, b) broadcasters using the band for ENG operations were willing to share a channel, and c) existing devices in the ISM band could cause some reception problems in some areas, which could be accommodated by the project.

The project applied to the FCC for an experimental radio station for the spectrum of 2,469-2,481 MHz with the center frequency at 2,475 MHz. This channel would be used for an AM double sideband signal with an occupied bandwidth of 12 MHz just as the UHF signal would employ.

Andrew Corporation provided the specially modified ITFS cardioid 16 dB antenna and transmission line.

12 GHz Facility: For the 12 GHz band, the project applied to the FCC for an experimental radio station on the frequency of 12.450 GHz in the newly allocated direct broadcast satellite (DBS) band that was formerly an OFS band. There were no licensed fixed users listed and frequency coordination was straightforward. The license was granted with the call letters of KA2XYE.

CBS provided the 12 GHz facility including the high power amplifier, antennas and receivers. Cablewave Systems provided the transmission line.

Location Of The Tests

The test site was chosen on the basis of convenience to most of the members of the task force, the availability of test sites, equipment, appropriate terrain and the willingness of various project personnel to conduct the tests in an area familiar to them. Appropriate terrain is an essential ingredient in the propagation test program. The terrain over which the signal pass must provide enough variation to demonstrate proper propagation characteristics. The terrain should not, however, include extremes of terrain such as flat plains, large bodies of water or mountains.

Access to the sites, on paved roads, within the appropriate terrain is also essential. The area chosen for the tests includes portions of Washington, D.C., northern Virginia and a portion of Maryland. Within these areas and within the beamwidth of the test transmitting antennas was found terrain representing rising, constant and declining elevation features which are considered to be adequate models for propagation testing.

Measurement Site Selection Criteria

The selection of sites involves several specific measurement schemes which have been used in the past for propagation tests and are considered standard procedure in the industry. These include arcs, radials and grids in which up to 410 specific sites will be measured.

Arcs: Two arcs have been identified along which measurements will be made. The arcs are at 15 and 30 km from the transmitter between the bearings of 180 degrees (South) and 330 degrees. Measurements will be made along the 15 km arc at 5 degree increments and at 2.5 degree increments along the 30 km arc.

The total number of sites on arcs to be measured is 90. Because an individual site, as determined on the map from the selection criteria may, in fact, be impossible to access when the vehicle is at the site, the adjustment conditions are to maintain the distance and to adjust the bearing.

Radials: Six radials have been selected for measurement along which sites will be chosen at one mile intervals. The terrain for radials should have certain characteristics. Two of the six should have a rising slope characteristics, two with a constant slope and two with a decreasing slope characteristics. Such characteristics can be found in terrain within the pattern of the transmitting antennas in the northern Virginia, and portions of Washington, D.C. and Maryland.

The radials selected for this test program include 190 and 195 degrees for decreasing slope, 210 and 225 for constant slope and 325 and 330 degrees for rising slope. Measurements will be made at 1 mile intervals from 1 to 31 miles along each radial. For six radials the total number of measurement sites is 192. Measurement conditions for adjustments at each site, if the vehicle cannot actually visit the site chosen on the map are to maintain the bearing and adjust the distance.

Grids: Two grids measuring 2.0 miles by 2.0 miles have been selected for on-site measurements. One is described as "high-density urban" and the other is "lower-density residential." Each grid will be measured at quarter mile intervals for a total of 64 measurements per grid and 128 for the two grids. Measurement conditions for adjustments at each site are at the discretion of the test personnel in the vehicle.

100 Foot Runs: These measurements are made by moving the vehicle slowly along a 100 ft. run in order to obtain an average signal level and ascertain the maximum and minimum levels in the area. A 100 ft. measurement run will be made at each measurement site in addition to a series of other measurements.

Pilot Tests and Calibration: A series of pilot tests were conducted to refine measurement and site procedures prior to the beginning of the measurement program.

A series of calibration measurements will be made each day of the field tests at close-in, specially selected and uncluttered sites with an unobstructed view of the transmitter for calibrating antennas and equipment prior to making measurements.

Reception Test Facilities

Two separate receiving facilities were constructed to conduct the required propagation tests. The dual-band test facility was designed to be placed at a fixed location for a period of time and record short and long-term changes between two signals in different bands. The field test, which includes the wide-band tests, was designed to be mobile to visit up to 410 test sites and measure a variety of propagation characteristics and parameters in up to four different bands.

Dual-Band Facility: This facility consists of two high quality television demodulators (for channel 9 and channel 58), a signal analyzer, waveform monitor, picture monitor, antennas, digital weather station and personal computer. The analyzer automatically measures time differential and group delay of the test signals. The demodulators were modified to provide a digital readout of signal level. The digital weather station provided weather information. The computer was programmed by PBS engineering staff to instruct the signal analyzer to take measurements and record the results along with signal levels and weather information.

The equipment was located 15.5 miles from the transmitter site at a bearing of 187 degrees in order to include enough average terrain on the path but not so far away as to receive a signal from channel 58 too weak to measure reliably. The site chosen is the office and laboratory of Carl T. Jones Engineering Consultants, located south of Springfield, Virginia where space in their offices for the equipment and on a tower for the antenna is available. The site is light-industrial with no high buildings near the antenna site. The terrain towards the transmitter is relatively clear with only scrub vegetation and rolling hills. An additional advantage of this location is that the equipment can be checked daily by competent personnel.

The test signal is the normal channel 9 (WUSA) programming and channel 58 configured for normal NTSC operation and using the channel 9 video programming but without program audio (instead, a station identification is transmitted each half-hour). Measurements are made each night from 6 pm to 8 am and continuously during weekends. This schedule permits the UHF transmitter to be configured for the wide-band (channels 58 and 59) operation during the day for the field tests.

The time differential is measured by comparing the leading edge of sync of the two stations. The video outputs of the two demodulators are compared by the signal analyzer and recorded in the computer. The signal analyzer also performs group delay measurements on a 4.2 MHz sinx/x signal inserted onto line 14 of the vertical interval of the two channels. RF signal levels are obtained from each demodulator by adding a circuit to obtain the digital signal level directly from the demodulator level indicator.

Weather information is recorded each hour in order to provide the ability to correlate weather with changes in the observed propagation conditions.

The initial term for this site is for three months. The time can be extended if necessary. Additional sites for the receiving system will be selected based on the results of the data obtained from the first site.

Wide-Band Facility: The wide-band receiving facility (field unit) was constructed in a small van, used in the past as an ENG vehicle. The van, provided by CBS, was recently refurbished and is equipped with a 30 ft pneumatic mast, 5 kW generator, racks for the equipment and a desk for the computer and digital oscilloscope. Test equipment in the van includes special wideband demodulator, with In-Phase and Quadrature (I & Q) channel output, digital oscilloscope, digital voltmeters, test signal generators, waveform and picture monitors, 2.5 and 12 GHz down-converters, frame counter and assorted antennas, RF preamps and filters. A personal computer was programmed to instruct the equipment to take the measurements, record the results, prompt the operator on procedures, and maintain the check list and site map information.

A general purpose instrumentation bus (GPIB) card was installed in the computer and connected to the UHF demodulator, digital oscilloscope, and digital voltmeters (used to measure signal levels from voltages provided by the demodulators). The digital oscilloscope was used to integrate, over several minutes, and record the waveform of the received I and Q channel demodulated double-sideband 6 MHz sinx/x test signal. Impulse response analysis of this waveform provides information on group delay and multipath. The output of the oscilloscope is fed to the computer for recording. A plotter will provide hard copy of the waveform for instant visual analysis.

The double sideband 6 MHz sinx/x signal is transmitted on the UHF (channel 58 and 59), 2.5 GHz and 12 GHz facilities. The sinx/x signal is gen-locked to the channel 9 video. Only a reference level is taken from the channel 9 signal.

Two antenna arrays were constructed for use on the 30 foot mast on the van. One array contains two UHF yagi with corner reflector antennas for the cross polarization tests. The other array contains the VHF, UHF, 2.5 and 12 GHz receiving antennas used for all other tests.

Control and monitoring of the three transmitters and UHF H and CP antenna switch is through a dial-up remote control system provided by Gentner. A cellular telephone is used in the field unit to call the transmitter for monitoring and control from each test site.

The software for the computer was provided by the Communication Research Centre of Canada and is an adaptation of a program the CRC used to conduct similar tests on cable television systems in Canada. The programming for the display of the recorded information is being created by PBS enaineering staff.

Test Parameters and Procedures

The task force, with the advice of the working parties and an ad hoc group that provided coordination for the test operations, developed a series of tests and test procedures to collect propagation data.

A. Dual-Band (Fixed Location Facility) Measurements, Resolution and Rate.

1. Delta-T: Measure the time-of-arrival difference between channel 9 and channel 58 to a resolution of 10 nanoseconds (ns). Take the measurement as often as the integration time to achieve the desired resolution will allow.

2. Relative Level: The RF level of the two signals will be observed once each second and recorded if a variation of more than 0.5 dB is noted.

3. Airplane Flutter: This is a subset of the combination of RF level and delta-T or selective fading and not measured as such as a specific parameter.

4. Group Delay: The group delay of a 4.2 MHz sinx/x signal on line 14 of channel 9 and channel 58 will be measured once each minute and recorded.

5. Weather Information: Temperature, humidity, barometric pressure and rain fall will be measured once each hour and recorded.

B. Dual-Band (mobile) Test Facility Measurements, Resolution and Rate

1. Absolute and Relative Signal Levels (field intensity): These measurements will be made over runs of at least 100 ft while continuously recording the signal level of the channels 9 and 58/59. The resolution of the signal level will be 0.5 dB and the rate will be 0.5 second for each channel. For this test a calibrated dipole will be used as a reference at a test location and the normal operating antennas (with gain) will be used for the field tests. A bandpass filter is used on the channel 58/59 signal to help protect the preamplifier from a nearby channel 56 signal. For the 2.5 and 12 GHz signals, separate runs, multiplexing the receiver (for 4 signals during the initial run) or spot checks will be employed to obtain signal levels.

2. Impulse Response: This measurement uses the digital oscilloscope to perform impulse response calculations on the 6 MHz sinx/x full field signal on channel 58/59, 2.5 GHz and 12 GHz. The signal will be observed for the length

of time needed to integrate out the noise (e.g. 4 minutes per channel is needed for 33 dB noise improvement). The measurement will be made at the beginning and end of the 100 foot runs.

3. Multipath: This test is conducted at the same time as the impulse response test and uses the same test signal. The digital oscilloscope and subsequently the computer) records the received waveform signal. By analyzing the waveform the delay and amplitude of multipath can be measured.

4. Signal Polarization: A separate antenna array using two antennas, one horizontally polarized and one vertically polarized, is employed to receive the UHF signal. Through a switching and phase network the antennas may be switched to provide an output representing horizontal, vertical or circularly polarized signals. At the transmitting end of the UHF circuit, a switch is used to select between the CP and H polarized antenna. During the field test the operator will use a cellular telephone to operate a dial-up remote control at the transmitter site to operate the switch.

Software Development

The use of computers to provide control and documentation for the project was a primary objective of the project. In addition to the ability to recall exact recorded information in any manner suitable for a desired display, the computer provides the means to automate the operation of the equipment. PBS engineering staff prepared the software for the dual-band tests and will prepare the software for subsequent display of the recorded results. The CRC provided much of the operating software for the wide-band field tests. Because of the additional complexity of using 4 bands and several pieces of computer compatible measuring equipment, the CRC software was adapted by propagation project personnel to conduct the measurement regimen as well as record the results.

Preparation of Results

Much of the value of a test project is the manner in which the test results are displayed and can be interpreted. Special consideration is given to recording the data in such a way as to allow relatively simple translation programs to be used to feed data into standard display software. The results of the propagation tests will be made available to the various advanced television committees and advisory groups and to the advanced television system proponents when the data have been verified and published in a manner useful to all parties. The test report is expected to be released in the second half of 1989.

SUMMARY

A propagation test project expected to provide useful data on the characteristics of various portions of the electromagnetic spectrum for evaluating the performance of advanced television broadcast transmission systems has been described. The project has taken months to prepare, conduct and will take additional months to evaluate and produce the final report. The broadcast industry cooperated fully to help design, equip, operate and analyze the results of the project which involved experimental wideband transmitters in the UHF, 2.5 GHz and 12 GHz bands.

ACKNOWLEDGMENTS

Special thanks goes to Will Haggerty, an engineer from Rubin-Bednarek & Associates, who operated the van during the field measurements and developed most of the field unit software.

On behalf of the propagation project, the author wishes to publicly thank the following organizations which have provided their expertise and equipment without which the project would not have been possible (listed alphabetically);

Andrew Corporation for the 2.5 GHz antenna and transmission line; **Cablewave** for the UHF and 12 GHz transmission line; **CapCities/ABC** for expertise in RF and propagation; **Carl T. Jones & Associates** for space for the dual-band test facility; CBS for the field unit, 12 GHz equipment and personnel support; **Communications Research Centre** (Canada) for equipment and software support; **Gentner** for the remote control equipment; **Information Transmission Systems** for the 2.5 GHz transmitter; **Jules Cohen & Associates** for frequency coordination; **Micro Communications, Inc.** for the UHF transmit and CP receive antennas; **MST** for their expertise in propagation and FCC liaison; **NAB** for support equipment; **NBC** for video support equipment; **PBS** for their support in development of software and laboratory space; **Rubin-Bednarek & Associates** for special arrangements for personnel support; **Tektronix, Inc.** for video equipment support; **WUSA-TV** for their cooperation and use of tower and transmitter space.

NOTES

1. The Advanced Television Systems Committee was established in 1983 to coordinate voluntary development of standards for advanced television systems in the United States. The Committee has several technology groups which in turn are composed of specialists groups. One such group (T3S4) established a task force to devise and conduct propagation tests for advanced television systems.

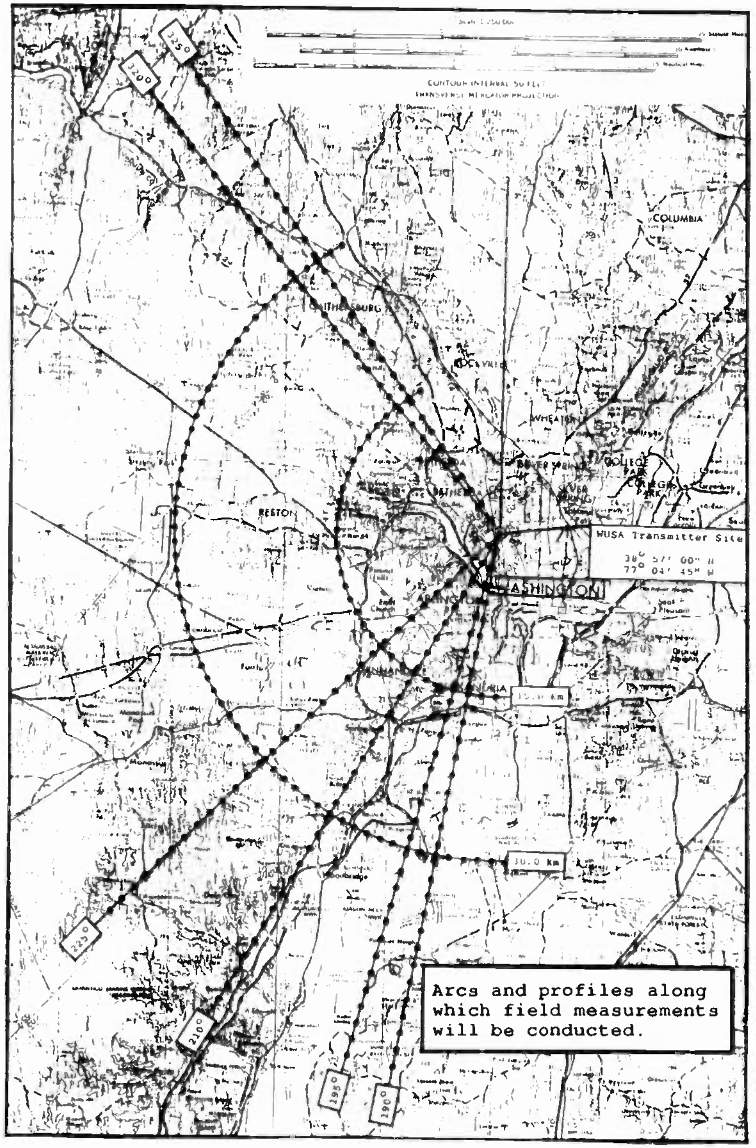
2. Representatives of the Communication Research Centre ("CRC"), a division of the Canadian Government Department of Communications participated in the task force as observers and offered their expertise in software and signal processing.

3. Docket 87-268 "Notice of Inquiry" released August 20, 1987, established the Advisory Committee. Three Subcommittees were formed for Planning (PS), Systems (SS) and Implementation (IS). Each subcommittee in turn established several working parties (WP) to examine various aspects of advanced television systems.

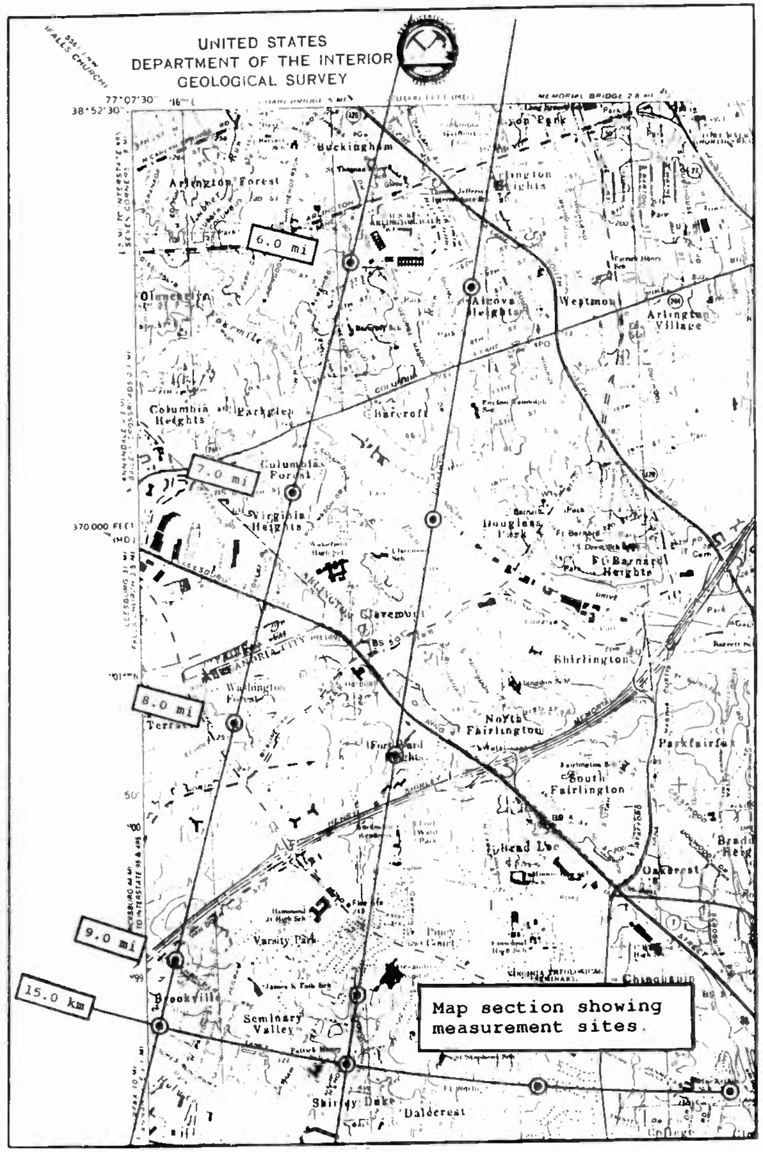
4. The Advanced Television Test Center is a non-profit corporation formed jointly by four television networks (ABC, CBS, NBC & PBS) and three industry associations (NAB, MST & INTV). It has offices in Alexandria, Virginia and expects to construct a laboratory for subjective and objective testing of advanced television systems for broadcasting.

5. Docket 87-268, "Tentative Decision and Further Notice of Inquiry".

6. These channels were used in 1986 and 1987 for a demonstration transmission of the MUSE HDTV signal that was developed by NHK for domestic direct satellite broadcasting in Japan. This test was conducted to demonstrate the feasibility of transmitting a 9 MHz wide MUSE signal on two TV channels by using AM vestigial sideband transmission and occupying about 11 MHz. This test used a modified 50 watt ITS transmitter and was received using a special wideband demodulator at three sites in Washington, D.C. including the FCC, U.S. House of Representatives and the National Association of Broadcasters. The call sign for the transmitter was WWHD-TV.



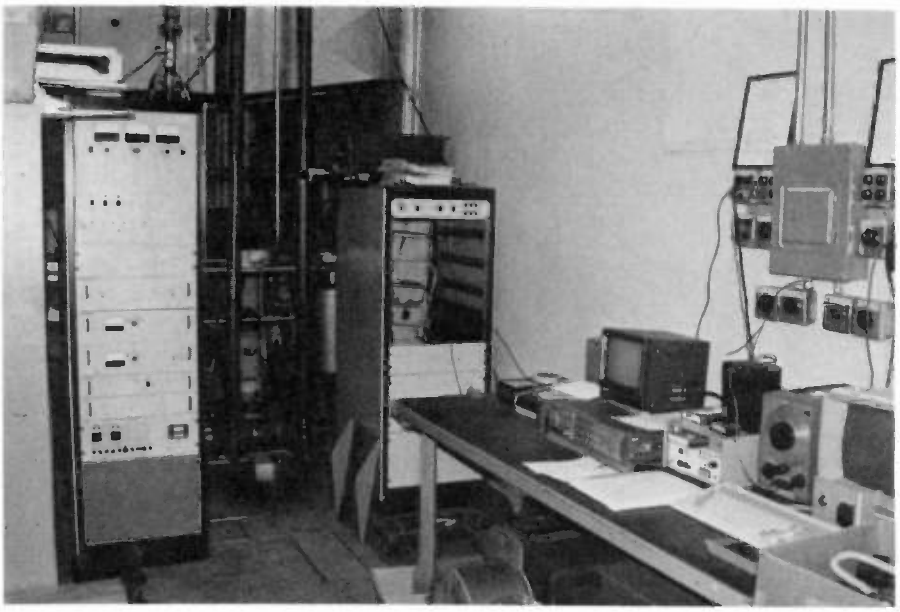
Arcs and profiles along which field measurements will be conducted.



Map section showing measurement sites.

Left: WUSA-TV (channel 9) towers. Shorter tower supports the propagation test antennas.

Above: Close-up of top of shorter tower showing the two UHF horn antennas. The 2.5 and 12 GHz antennas will be mounted in the same area.



Above: View of corner of WUSA-TV transmitter room showing UHF and support equipment. Open area is for 2.5 & 12 GHz equipment.

Left: Field unit with mast extended supporting the VHF, UHF, 2.5 and 12 GHz ant.

Below: Interior of field unit showing some of the measurement equipment.

THE COST OF CONVERTING A BROADCAST FACILITY TO HDTV

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Abstract

If from this paper you had hoped to find the answer to the question "How much will it cost to convert my station to Advanced Television (ATV)" I am afraid you will not find a complete answer. As of the end of February 1989 the advisory committee to the FCC still has 17 transmission systems under consideration whose complexity have a direct relationship to the total system cost. What this paper will give you is insight to some of the technical considerations of a conversion to ATV and associated costs.

Historical Equipment Costs

A manufacturer will introduce a new product for any number of reasons. It could be very cost effective or has advanced features and pushes the current technology. ATV products most certainly push the technology and that push always costs more than current technology. But how expensive is current HDTV equipment as compared to products introduced twenty years ago? In 1966 the RCA TK-42 cost $76,000, in 1969 the TK-44 cost $68,000 and in 1964 the Ampex VR-2000 cost $95,000. By using the Gross National Product implicit price deflator we can get a feel for the cost if we were to purchase the same equipment today. Shown in the first part of Chart #1 is the TK-42 & TK44, the LDK-6A and the newest Sony HDTV camera. The second part shows the Sony BVH-3000 one inch video tape recorder, DVR-1000 composite digital VCR, the Digital HDTV VCR and the 1989 cost of the VR-2000. The cost of the Sony HDTV camera is 13% more than the TK-42 and the Sony HDTV Digital VCR is 10% less than the VR-2000. The picture quality improvement requires no comment and the price/performance ratio between todays NTSC equipment and the 20-year-old equipment is spectacular.

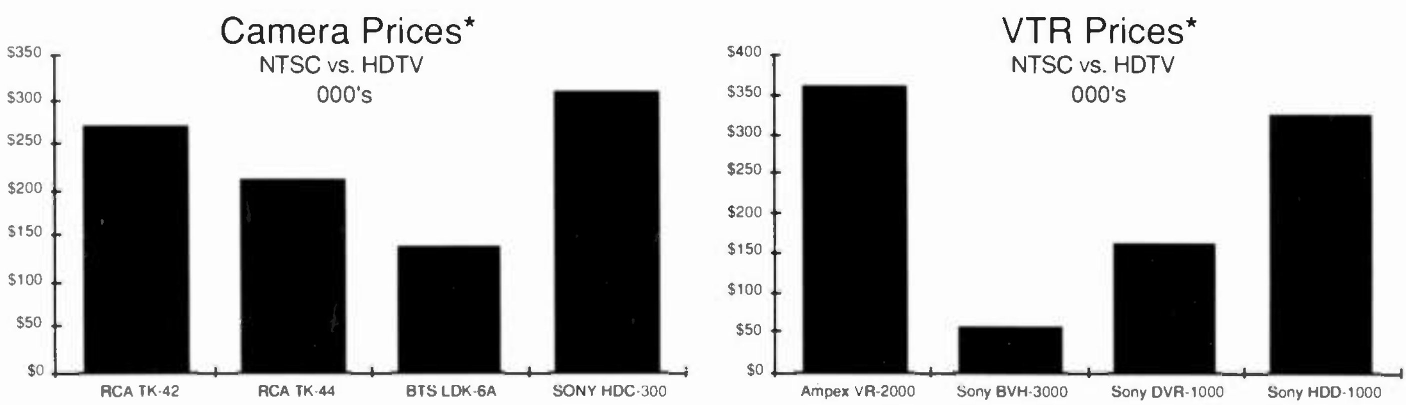


Chart 1

* The RCA TK-42 and TK-44, and the Ampex VR-2000 were converted to 1989 prices using Gross National Product implicit price deflator.

General Issues in Converting to ATV

Program Material Conversion

For many years after the introduction of any ATV system our viewers will continue to watch programming on their existing NTSC televisions. This presents two problems. First is the straight electrical conversion from NTSC to the selected ATV format and the second is the aspect ratio conversion. The electrical conversion is the simplest from a manpower standpoint. It requires some number of black boxes to pass the signal through but those black boxes will be quite complex. The process to convert from a 60.00 vertical rate to a 59.94 rate is not a simple mater. The aspect ratio problem is more complex as it will require a black box but also manpower to convert both to, and from, 16:9 and 4:3. There is no computer system that can automate the Pan and Scan process. In the beginning HDTV programming material could be shot for the 4:3 format and ignore the wide aspect ratio, or material can be prepared in post production for broadcast in the 4:3 format. Live programming may require a new job description "Pan and Scan Specialist" to get the most from both formats.

Signal Continuity Monitoring

If the cost of an ATV television set is $3,000, how many can we afford for signal continuity monitoring? There are over 75 color sets in offices and various other locations around my station. When we converted to color from B&W you could at least use your existing B&W monitor to view for signal continuity, and purchase a new color TV for the General Manager's office and the lobby. The conversion to an ATV system, however, will require a new monitor or television at every location that needs to view the wide aspect ratio.

Equipment Wiring

Currently, the only in-plant cabling system for implementing full HDTV is separate cables for the Red, Blue and Green signals. We are all experts in signal path timings because of NTSC. A 2 nanosecond error will cause a color shift between sources. For HDTV you can tolerate up to 5 nanosecond errors between the RG&B signals with no picture impairment. If you have experience in component signal routing then you have a feel for HDTV signal handling. HDTV RBG requires triple the number of wideband distribution amplifiers, three router levels, and three times the number of connections for each input and output termination. The last item to keep in mind is that your technical staff must make three times the number of cable terminations. All these add to the cost of introducing ATV.

Off-Air Monitoring

In most stations the in-house RF antenna system will not pass the new ATV signal. If your system modulates the cable from baseband then you will need new modulators for the approved standard.

Transmitters

This is the one real unknown for the future of ATV. The cost could be as simple as modification to the existing transmitter plant to a complete second transmitter system. This second transmitter may require a second tower and transmitter building and if that second transmitter is on a UHF channel the operating costs for it may come as a shock to a VHF broadcaster.

Studio Transmitter Link (STL)

Most broadcast facilities are not co-located and require a STL to get the NTSC signal to the transmitter plant. Many of the systems under consideration by the FCC Advisory Committee address only transmission over the air portion of a broadcaster's plant and not the problem of getting the signal to the transmitter. If you simulcast you will need a minimum of a second STL microwave system.

Core Equipment Area

The core equipment area contains the central signal router and its support equipment. If a local station originates in ATV and simulcasts NTSC, it may require the majority of the core equipment to be duplicated. Even a modest entry into ATV will require quite a few new racks to contain all the RGB distribution. Many plants will require major reworking of their central equipment areas. The electrical requirements for the increased amount of equipment must be provided and that electricity will generate more heat and require more air conditioning.

Monitor Size

A control room monitor wall is either made up of standard EIA 19-inch wide racks or wood construction. If a new HDTV monitor slides into the existing hole, the horizontal dimension of the monitor will be approximately the same as the NTSC monitor it replaces. If the production staff sits at same distance from the monitor wall, the viewed image on the new HDTV monitor will appear smaller because of the wider 16:9 aspect ratio. Producer/Directors want to see larger images not small ones. To install 30 or 40-inch wide racks to accommodate the larger monitors, you will need a larger control room or rebuild the interior to move the production staff closer to the central monitors and forego the increased viewing angle to the outer monitors.

Conversion Scenarios

To implement ATV at a local television station one must consider the cost of complete conversion to full HDTV. Costs could exceed 40-million dollars and station engineering staffs need to offer a way into ATV that will not put us out of business. One scenario for the introduction of ATV will follow how MTS sound was introduced to the United States. The majority of stations introduced MTS by receiving the stereo sound from their network, performing minimum processing and then transmitting it. This required the minimum amount of equipment and gave the viewer access to true stereo programming. The cost of passing MTS stereo through a station could be anywhere from $30,000 to $100,000 and local origination in stereo increased those costs several fold. The same is true for ATV. Costs to pass a network signal through the local station are inexpensive as compared to local origination. See Figure 1.

Pass the Network - Syndicated Programming Playback.

The equipment necessary to pass a network signal includes:

1. Signal delivery to the studio from the network.
   (The costs of signal delivery to the studio may or may not be included as part of your current network agreement.)
2. Signal processing at studio.
3. Delivery to transmitter site.
4. Transmission over the air.
5. Minimum monitoring.

Studio Operations and Playback

The equipment necessary to originate local programming in an ATV format requires the items from 'Pass the Network'. This equipment list will produce programming with the same production elements we use today.

1. Main equipment signal routing and distribution system.
2. Studio cameras.
3. Production switcher and digital video effects.
4. Still stores, character generation, and graphics.
5. Video tape recorders and commercial spot playback.

Field Operations (News & EFP)

The equipment necessary to operate field operations must include items from the two previous lists.

1. Field camcorders.
2. Edit rooms.
3. Expanded signal routing.

The Spread Sheets

Equipment costs for NTSC and HDTV products used in this paper have been developed for use by the FCC Advisory Committee on Advanced Television Service, Systems Subcommittee, Working Party Three on Economic Assessment. Many hours have been spent in collecting and estimating the numbers. A major contributor has been S. Merrill Weiss, Managing Director, Advanced Television for NBC. As shown they are only a first pass at costing until the FCC sets a transmission standard or further information becomes available from the ATV system proponents. It is a snapshot of where we are today. The summary of the three spread sheets is shown in Table 1.

	Total NTSC Price	Projected HDTV Price Today
Pass the Network	$5.957.3	$9,392.0
Full Studio Operations	$4,538.2	$14,810.5
Field Operations	$3,594.0	$14,314.6
Total	$14,089.5	$38,517.0

Table 1

A number of assumptions have been made to create these sheets; they include:

No installation has been provided for. Those costs will vary wildly because of location, unions, degree of system documentation etc.

All prices are list price for quantity of one.

A second RF system for augmentation is necessary and the needed spectrum will be available at whatever frequency or channel spacing is required. The second RF system needs to be full power UHF (worst case); it is assumed the space will not be available on the broadcaster's current tower.

A new tower, building and land to locate it are required.

The proponent encoded signal will pass through a standard UHF exciter with no modifications.

The antenna will not require special bandwidth, linearity, etc.

The proponent system will be encoded at the studio and the signal will pass through a standard microwave STL system and not require a second microwave.

Minimal audio/video monitoring has been provided for.

The current spread sheet shows a blue-sky cost of the proponent encoder and decoder.

The spreadsheets are organized with columns A through H. Their column descriptions are:

A = Quantity used.
B = Description of the item.
C = Current NTSC price on the equivalent HDTV product.
D = Quantity in A multiplied by the NTSC price.
E = Current unit HDTV price if a product exists today.
F = Quantity in A multiplied by the current HDTV price.
G = If there is no price available for the HDTV item themultiplier is used on the NTSC price to estimate the cost.
H = Either the true price of the HDTV item from F or the NTSC cost from D times the multiplier in column G.

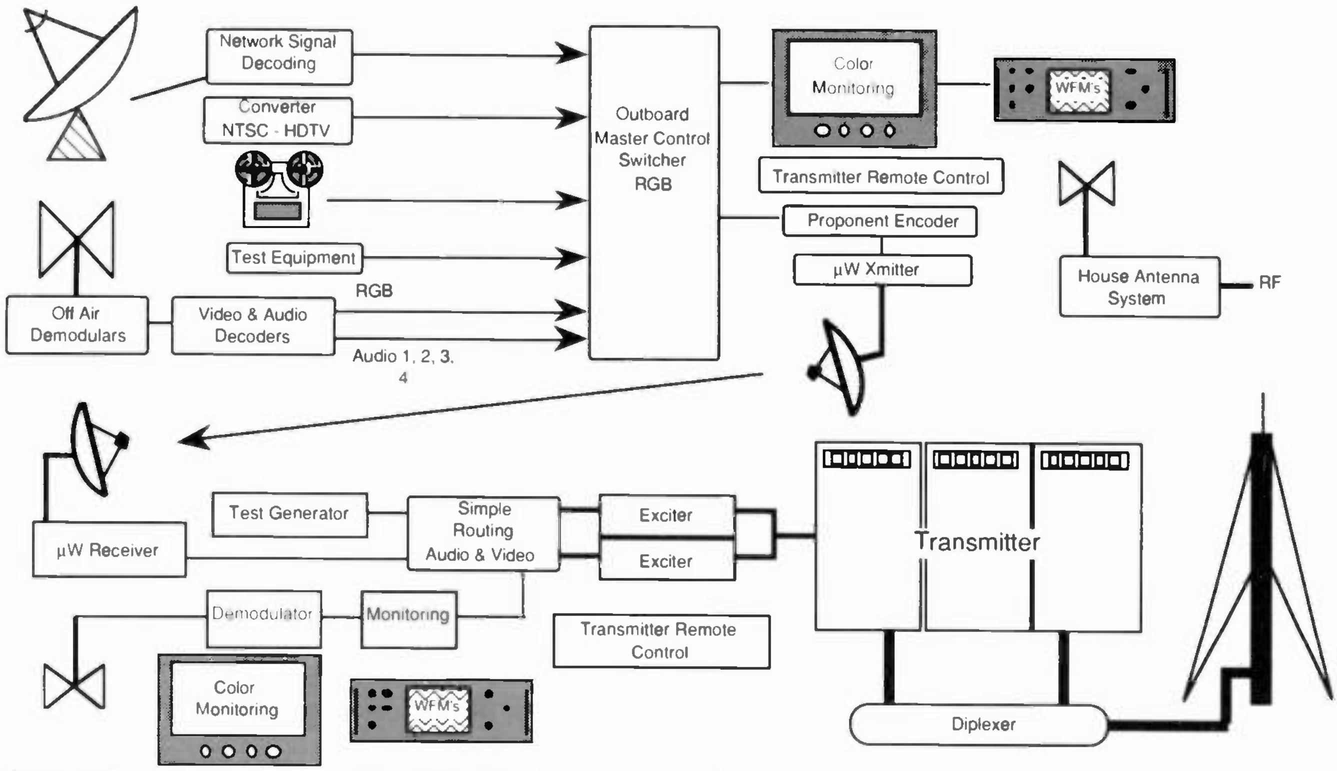
The 2.7 multiplier used in a number of places in column G was developed by taking the average price of a group of HDTV products that are available today and comparing their cost against the equivalent NTSC products. Some HDTV items did not fit that average and another value was used.

Conclusion

There is no inexpensive path to an ATV system. The first in the market will pay a premium. The necessity to triple-wire for RGB is very costly. Fibre or digital transmission must be developed to lower manpower installation costs. The possibilty of needing a complete new transmitter plant and very expensive format converters all add to the expense of conversion.

But, as you can see from Chart 1, the cost of broadcast equipment is always declining. The LDK-6 studio camera and the Sony BVH-3000 one-inch VTR provide levels of performance far greater than their twenty-year-old grandparents. Twenty years from now the cost of HDTV equipment will be far less costly than today. We must make purchase decisions in the next 5 to 10 years. The story is just unfolding; we are in chapter 3.with lots of pages remaining. One thing is for certain, we will have improved pictures in our future.

Passing a Network ATV Signal



Full Studio ATV Production

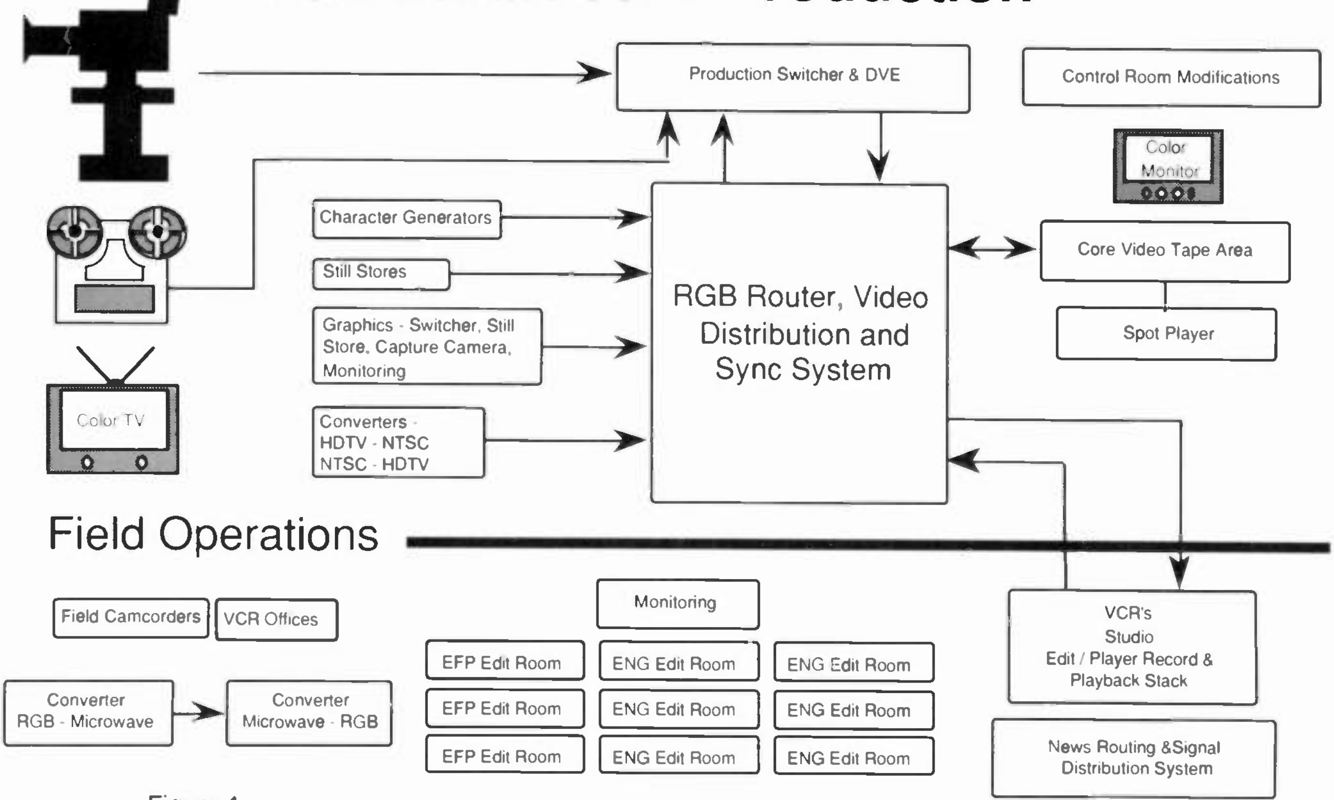


Figure 1.
The Cost of Converting a Broadcast Facility To HDTV

A Qty	B Item (All amounts in thousands)	C Unit NTSC Price	D Total NTSC Price A*C	E Unit HDTV Price	F Total HDTV Price A*E	G Multiplier for HDTV Today	H Projected HDTV Price Today D*G or F
	Transmitter Plant						
	5 Meg ERP - 180 kw Transmitter						
	Base Transmitter	$750.0					
	Tubes & Magnets	$210.0					
	Spare Tube	$60.0					
	Diplexer & Control	$160.0					
	Transmission Line	$150.0					
	Antenna	$200.0					
	Installation	$100.0					
1	Subtotal		$1,630.0			$1.5	$2,445.0
	New Transmitter Location						
1	Site Development	$100.0	$100.0			$1.0	$100.0
50	Acres - Land	$10.0	$500.0			$1.0	$500.0
5280	1.0 mile road	$0.2	$897.6			$1.0	$897.6
1500	sq. ft. building @ $75/ft.	$0.1	$112.5			$1.0	$112.5
1	Electric (substation & generator)	$400.0	$400.0			$1.0	$400.0
1	Tower & Installation	$1,500.0	$1,500.0			$1.0	$1,500.0
	Engineering Services		$350.0			$1.0	$350.0
	Antenna System Design						
	FAA & FCC						
	Tower						
	Legal						
	Soils & Survey						
	Transmitter Subtotal		$5,490.1				$6,305.1

A Qty.	B Item (All amounts in thousands)	C Unit NTSC Price	D Total NTSC Price A*C	E Unit HDTV Price	F Total HDTV Price A*E	G Multiplier for HDTV Today	H Projected HDTV Price Today D*G or F
	Transmitter Terminal Equipment						
1	Remote Transmitter Control	$85.0	$85.0	$0.0		$1.0	$85.0
1	Microwave System	$75.0	$75.0	$0.0		$5.0	$375.0
1	Routing Switcher, 16x16	$12.0	$12.0	$0.0		$3.0	$36.0
2	Waveform Monitor, Precision	$6.5	$13.0	$12.0	$24.0	$0.0	$24.0
2	B/W Monitor 14	$1.4	$2.8	$3.7	$7.4	$0.0	$7.4
2	Color Television	$0.3	$0.6	$0.0		$10.0	$6.0
1	Color Monitor 19 HD18	$8.0	$8.0	$18.0	$18.0	$0.0	$18.0
2	Proc Amp.	$4.5	$9.0	$0.0		$3.0	$27.0
2	VITS Inserter	$9.9	$19.8	$0.0		$2.7	$53.5
2	Modulation Monitors	$12.0	$24.0	$0.0		$5.0	$120.0
2	Demodulator	$5.0	$10.0	$0.0		$10.0	$100.0
1	Decoder -Proponent ATV > RGB-	$0.0	$0.0	$250.0	$250.0	$0.0	$250.0
1	Audio System	$50.0	$50.0	$0.0		$1.0	$50.0
1	Spectrum Analyzer	$40.0	$40.0	$0.0		$1.0	$40.0
1	Sweep Generator	$6.0	$6.0	$0.0		$1.0	$6.0
1	Frequency Counter	$2.0	$2.0	$0.0		$1.0	$2.0
1	Test Signal Generator	$7.0	$7.0	$26.0	$26.0	$0.0	$26.0
1	Portable Scope	$6.5	$6.5	$0.0		$1.0	$6.5
	Transmitter Terminal Equipment Subtotal		$5,860.8				$7,537.5
	Pass The Network - Studio Equipment						
1	Switcher, 10x1 -GVG Ten-X-	$1.5	$1.5	$0.0		$3.0	$4.5
1	Decoder -Proponent ATV > RGB-	$0.0	$0.0	$250.0	$250.0	$0.0	$250.0
1	Tape Machine 1 inch	$60.0	$60.0	$320.0	$320.0	$0.0	$320.0
1	House CATV System -2 Modulator	$25.0	$25.0	$0.0		$10.0	$250.0
2	Proponent System Encoder	$0.0	$0.0	$500.0	$1,000.0	$0.0	$1,000.0
2	Video DA's (8), Tray/PS	$5.0	$10.0	$15.0	$30.0	$0.0	$30.0
	Pass the Network Total		$5,957.3				$9,392.0

A Qty.	B Item (All amounts in thousands)	C Unit NTSC Price	D Total NTSC Price A*C	E Unit HDTV Price	F Total HDTV Price A*E	G Multiplier for HDTV Today	H Projected HDTV Price Today D*G or F
	Studio Operations -Full Local Production-						
4	Camera & Zoom, Studio	$136.0	$544.0	$375.0	$1,500.0	$0.0	$1,500.0
2	Character Generator, Color	$56.0	$112.0	$120.0	$240.0	$0.0	$240.0
1	Convert Format -NTSC>HDTV-	$0.0	$0.0	$463.6	$463.6	$0.0	$463.6
2	DVE, 1 Channel	$180.0	$360.0	$0.0		$2.7	$972.0
2	Graphics -Aurora 250-	$60.0	$120.0	$0.0		$2.7	$324.0
1	Camera, Overhead Graphic	$12.0	$12.0	$375.0	$375.0	$0.0	$375.0
14	B/W Monitor 14	$1.4	$19.6	$3.7	$51.8	$0.0	$51.8
35	Color Monitor 14 HD12	$4.0	$140.0	$10.9	$381.5	$0.0	$381.5
12	Color Monitor 19 HD18	$8.0	$96.0	$18.0	$216.0	$0.0	$216.0
1	Routing Switcher - Horizon	$734.0	$734.0	$1,543.0	$1,543.0	$0.0	$1,543.0
	128 x128 3 Video levels, 4 Audio, 100 ctl panels						
1	Commerical Spot Player	$375.0	$375.0	$0.0		$3.0	$1,125.0
2	Still Store -DLS 6030-	$80.5	$161.0	$0.0		$2.7	$434.7
1	Production Switcher, Large	$250.0	$250.0	$0.0		$3.0	$750.0
4	Sync Generator	$5.1	$20.4	$9.4	$37.6	$0.0	$37.6
13	Sync Distribution System -VDA's-	$5.0	$65.0	$15.0	$195.0	$0.0	$195.0
1	Test Signal Generator	$7.0	$7.0	$26.0	$26.0	$0.0	$26.0
10	Waveform Monitor, Precision	$6.5	$65.0	$12.0	$120.0	$0.0	$120.0
22	Waveform Monitor, Utility	$3.0	$66.0	$6.0	$132.0	$0.0	$132.0
50	Color Television	$0.3	$15.0	$0.0		$10.0	$150.0
17	Video DA's W/Equalization	$5.8	$98.6	$17.4	$295.8	$0.0	$295.8
6	Tape Machine 1 inch	$60.0	$360.0	$320.0	$1,920.0	$0.0	$1,920.0
4	C-Band Descramblers	$10.0	$40.0	$0.0		$6.0	$240.0
	Studio Operations Subtotal		$3,660.6				$11,493.0
	Post Production						
1	Production Switcher, Large	$250.0	$250.0	$0.0		$3.0	$750.0
1	DVE, 1 Channel	$180.0	$180.0	$0.0		$2.7	$486.0
3	Color Monitor 14 HD12	$4.0	$12.0	$10.9	$32.7	$0.0	$32.7
1	Character Generator, Color	$56.0	$56.0	$120.0	$120.0	$0.0	$120.0
1	Still Store -DLS 6030-	$80.5	$80.5	$0.0		$2.7	$217.4
1	Color Monitor 19 HD18	$8.0	$8.0	$18.0	$18.0	$0.0	$18.0
2	Color Monitor 9 inch	$2.0	$4.0	$0.0		$2.5	$10.0
2	Waveform Monitor, Precision	$6.5	$13.0	$12.0	$24.0	$0.0	$24.0
2	Video DA's W/Equalization	$5.8	$11.6	$17.4	$34.8	$0.0	$34.8
3	Tape Machine 1 inch	$60.0	$180.0	$320.0	$960.0	$0.0	$960.0
2	Tape Machine M-II - Beta Studio	$35.0	$70.0	$67.0	$134.0	$0.0	$134.0
1	Tape Machine 3/4 inch	$12.5	$12.5	$67.0	$67.0	$0.0	$67.0
1	Convert Format -NTSC>HDTV-	$0.0	$0.0	$463.6	$463.6	$0.0	$463.6
	Post Production Subtotal		$877.6				$3,317.5
	Full Studio Operations Total		$4,538.2				$14,810.5

A Qty.	B Item (All amounts in thousands)	C Unit NTSC Price	D Total NTSC Price A*C	E Unit HDTV Price	F Total HDTV Price A*E	G Multiplier for HDTV Today	H Projected HDTV Price Today D*G or F
	Field Equipment - News/EFP						
17	Camera, portable	$50.0	$850.0	$375.0	$6,375.0	$0.0	$6,375.0
7	Frame Sync	$14.0	$98.0	$0.0		$2.7	$264.6
15	Color Monitor 5 inch	$1.0	$15.0	$0.0		$2.5	$37.5
31	Color Monitor 9 inch	$2.0	$62.0	$0.0		$2.5	$155.0
5	Proc Amp.	$4.5	$22.5	$0.0		$3.0	$67.5
7	Switcher, 10x1 -GVG Ten-X-	$1.5	$10.5	$0.0		$3.0	$31.5
4	Waveform Monitor, Precision	$6.5	$26.0	$12.0	$48.0	$0.0	$48.0
10	Waveform Monitor, Utility	$3.0	$30.0	$6.0	$60.0	$0.0	$60.0
49	Tape Machine M-II - Beta Studio	$35.0	$1,715.0	$67.0	$3,283.0	$0.0	$3,283.0
7	Tape Machine 3/4 inch	$12.5	$87.5	$67.0	$469.0	$0.0	$469.0
25	Tape Machine -Field Beta	$9.5	$237.5	$0.0		$5.0	$1,187.5
8	Tape Machine -Viewing Beta	$5.0	$40.0	$67.0	$536.0	$0.0	$536.0
4	Microwave System	$75.0	$300.0	$0.0		$5.0	$1,500.0
1	Routing Switcher, 32x32	$100.0	$100.0	$0.0		$3.0	$300.0
	Field Operations Total		$3,594.0				$14,314.6

THE SPECTRUM-COMPATIBLE HDTV TRANSMISSION SYSTEM

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Introduction

The Spectrum-Compatible High-Definition Television (SC-HDTV) System is a simulcast system. The HD version of a program is transmitted in a second 6 MHz channel while the NTSC version is simultaneously transmitted in the current channel. See Figure 1. The SC-HDTV transmission signal is so designed that the potential for mutual interference between SC-HDTV and NTSC signals is significantly reduced. As a result, Taboo channel spectrum space in the existing TV bands is recovered for SC-HDTV channels. The continued NTSC broadcasts prevent the 160 million TV sets in current use from becoming obsolete. At the time when these receivers will have been replaced by high-definition or dual-standard receivers, in perhaps 15 or 20 years, much of the present spectrum will have become available. On the other hand, NTSC-based receiver-compatible and augmentation HDTV/EDTV Systems will lock up this spectrum in perpetuity.

Concise Description of the SC-HDTV System [1]

The high definition source signal is encoded for transmission by limiting transmitted power to picture power only and, additionally, by distributing it evenly over the 6 MHz channel. A combination of methods are used to achieve this and are illustrated in the block diagram of Figure 2. The first method includes elimination of conventional syncs, carriers and subcarriers. It is well-known that the picture carrier is the chief contributor to interference although sound and chroma carriers do their part. The power is more evenly distributed by transmitting the lowest 200 kHz of the video signal (which contain most of the power) in digital form. This portion of the video signal is converted to a 4-level 16-QAM signal and is transmitted during the vertical interval. The digital stereo audio is a part of the same vertical interval data signal.

The process of significant power reduction results in equally significant reduction of the interference into NTSC signals. It raises the concern that, conversely, the NTSC signal might increase interference into the HDTV signal. This is minimized by timing the scanning rates of the HDTV signal compatibly with NTSC signal scanning rates. Such timing makes possible the use of precision carrier offset. Temporal filtering (frame combing) of the HDTV video signal helps to reduce interference by (static) NTSC signals when combined with precision offset. Temporal filtering also contributes to power reduction since only a fraction of the current frame plus the difference from the previous frame is transmitted. To accomplish further peak signal reduction, which is not always guaranteed by average power reduction, signal compression and time dispersion are introduced.

The video source is a progressively scanned 787.5 lines/frame, 59.94 frames/sec RGB signal with a horizontal line rate exactly three times the NTSC line rate. The encoding process, in addition to the data signal, yields six video components as illustrated in Figures 3 and 4. The components of low frequency content are transmitted at full frame rate for good motion rendition. Higher frequency components, for good detail in static images, are transmitted at lower rates and color difference components also at a lower rate. After time expansion the bandwidth of all components is slightly less than 3 MHz. The six components are arranged into two groups, I and Q, which quadrature-modulate two carriers located in the center of the 6 MHz channel by DSB-SC-AM modulation. A small pilot signal is transmitted at the carrier frequency to aid in receiver demodulation. Frame and line synchronization are achieved by a clock signal transmitted during part of the vertical interval. The data signal also includes bits to help motion rendition interpolation.

Transmitter processing of temporal filtering, compression and time dispersion are complemented by receiver temporal post-filtering, expansion and inverse time dispersion. The receiver also includes frequency de-emphasis to minimize co-channel interference from an NTSC visual carrier, which is offset by 1.75 MHz. The de-emphasis (which also improves the noise performance) is complemented by transmitter pre-emphasis.

The I and Q signal components modulating the two quadrature carriers can be time-compressed and subsequently time-multiplexed into a single 6 MHz signal as illustrated in Figure 5. This is the preferred signal for FM modulation. It is useful for satellite communication, for VCR recording and play-back and for STL links. The inverse process is illustrated in Figure 6 and the time relationships in Figure 7.

The Reduced Taboos [2]

Broadcasters may be somewhat familiar with the taboos, the set of requirements for minimum distance separation between TV stations in order to avoid interference. The dominant mechanisms of interference are the entries in Table I. Comparing the NTSC and the SC-HDTV columns shows that the latter system is significantly less sensitive than the former.

When NTSC and SC-HDTV co-exist there are four sets of taboos to be considered. Data on two TV Receivers are listed in Table II. The HDTV column shows significantly higher possible levels for the undesired channel before perceptible interference starts. Measurements of interference of NTSC into SC-HDTV, and of SC-HDTV into SC-HDTV, though not yet available, are expected to yield satisfactory numbers.

The need for Taboos has been a source of confusion in the past and a better understanding between broadcasters and receiver manufacturers is still needed. Without implying a relative weight on the three causes, the necessity for Taboos can be listed as: (1) receiver limitations, (2) the F.C.C. allocation scheme and, (3) the nature of the NTSC signal. With regard to receiver limitations, experimental dual-conversion receivers have been built with reduction or elimination of taboos in mind. Such receivers have not been accepted in the market place for a variety of reasons such as degradation of VHF performance, inability to tune some CATV channels and increased cost without demonstrable benefit to the consumer. The dual-conversion receivers did eliminate I.F. Taboos. However, adjacent channel and intermodulation taboos remained. Even if a "perfect" receiver were available, the Taboos could not be eliminated or reduced as long as the current 160 milllion TV sets are in use, which will be many years.

With SC-HDTV the situation is different, all NTSC type receivers remain useful. New receivers will be introduced for reception of the SC-HDTV program and their construction will both influence and be influenced by the new reduced Taboos.

Broadcasters' Concerns

Simulcast transmission implies the operation of two separate transmitters, the current one of NTSC standards and one for SC-HDTV on another channel. The video source may be high-definition video for both transmitters. This requires a down-converter to transcode the high definition video to NTSC video. Amplifiers, switchers and, possibly, cabling will have to be acquired for the high-definition bandwidth. The STL link can benefit from the FM baseband version of the SC-HDTV signal described above. The SC-HDTV exciter at the transmitter site will be designed for in-phase and quadrature input as shown in Figure 8. No separate aural transmitter is required. A NTSC Transmitter block diagram is found for comparison in Figure 9.

The reduced average power of the SC-HDTV transmission system is of special importance to the UHF broadcaster who annually incurs a significant electrical power bill. An early estimate for a 5,000 kW transmitter is a reduction from in excess of $180,000 to less than $1,000 per year. In addition, water cooling or forced air cooling can be eliminated. Voltage breakdown requirements of the major transmitter components, however, have not changed.

The simulcast channel needs a transmission line to the antenna but no notch-diplexer. The antenna may be colocated on the mast for the NTSC channel antenna. If this is not possible, another mast, in another location in the community can be used.

Questions have been raised recently regarding the drop of ERP towards the channel band edge of antennas with narrow vertical beam. [3] The low power of the SC-HDTV transmitter allows simple equalization, if necessary.

Summary

A casual observer of the HDTV scene would probably decide that a receiver-compatible HDTV system is to be preferred, comparable to, for example, the transition from black-and-white TV to color TV. A more detailed study of the current proposals shows the disadvantages of retaining the NTSC system which is inefficient in power and timing and contains basic flaws in color treatment. Today's technology suggests that the venerable NTSC System, developed in the 1930's, is becoming obsolete. Basing a HDTV system, be it receiver-compatible or augmentation, on NTSC, means that NTSC will be with us in perpetuity. The SC-HDTV system allows a gradual phase-out of NTSC as the majority of receivers in the field have become HDTV or dual-standard. This frees extra spectrum space for which there certainly will be a need.

After authorization of HDTV broadcasting, the broadcaster must decide whether to expand to the new service or not. The current trend towards larger picture tube sizes and projection sets and competition from other non-broadcast consumer TV delivery means such as cable and VCR, make it desirable for a broadcaster to do so. It will require a capital outlay for new equipment. If SC-HDTV is the system of choice the outlay will be modest, probably not more than for other HDTV Systems (but excluding EDTV and IDTV proposals). Taking the long-term view, a simulcast system and, specifically, the SC-HDTV System is the best choice.

REFERENCES:

[1] "Spectrum Compatible HDTV System" Monograph published by Zenith Electronics Corporation, Glenview, IL., September 1, 1988.

[2] "UHF" Taboos, Protection Ratios and Current NTSC Television Receiver Performance." Report PS-WP3-21, dated February 22, 1988, by J. Rypkema to Working Party 3, "Advanced Television Systems Spectrum Utilization and Alternatives" of the Planning Subcommittee of the "Advisory Committee to the FCC on Advanced Television Service."

[3] "TV Signal Degradation: Is your Antenna the Culprit?" by Richard D. Bogner, BM/E Magazine Vol. 24, No. 3, March 1988, pp 164-169.

DOMINANT TABOO INTERFERENCE MECHANISMS

		NTSC	ZENITH HDTV
N - 1	A)	LINEAR FEEDTHROUGH OF SOUND CARRIER	N.A.
	B)	CROSSMODULATION OF RF ENVELOPE OF INTERFERING SIGNAL	LOWER PEAK RF ENVELOPE AND NO LOW FREQUENCIES IN ENVELOPE HENCE LESS VISIBLE
N + 1	A)	THIRD ORDER I.M. OF S_{N+1} WITH P_{N+1} RESULTS IN FM MODULATED 1.5 MHz BEAT FREQUENCY PROPORTIONAL TO THE SQUARE OF P_{N+1}	N.A.
	B)	CROSSMODULATION	AS N - 1
N ± 2		CROSSMODULATION	AS ABOVE
N ± 3		CROSSMODULATION	AS ABOVE
N - 4		CROSSMODULATION	AS ABOVE
N + 4		HALF IF BEAT P_{N+4} RESULTS IN 2.25 MHz (43.5 MHz IF) BEAT	ONLY LOWEST LOWER SIDEBAND FREQUENCIES COULD CREATE VISIBLE BEAT
N ± 7		IF BEAT SECOND ORDER BEAT OF P_N AND P_{N+7} RESULTS IN 42 MHz IF SIGNAL WHICH SHOWS AS CHROMA INTERFERENCE AT 3.75 MHz	NO SYNC AND LOWER PEAK RF ENVELOPE VASTLY REDUCE THIS TABOO
N ± 8		IF BEAT S_N AND P_{N+8} OR P_N AND S_{N-8} GIVES 43.5 MHz (2.25 MHz) BEAT	NO SOUND SUBCARRIER
N + 14		SOUND IMAGE S_{N+14} RESULTS IN 42.25 MHz (3.5 MHz) BEAT IN CHROMA	ONLY UPPER HALF OF UPPER SIDEBAND COULD CAUSE VISIBLE (H.F.) INTERFERENCE
N + 15		PICTURE IMAGE P_{N+15} RESULTS IN 1.5 MHz BEAT	ONLY LOWER SIDEBAND COULD CAUSE VISIBLE INTERFERENCE

TABLE I

MEASURED PERFORMANCE ON TABOO CHANNELS

	UNDESIRED TO DESIRED SIGNAL RATIOS AT RECEIVER FOR PERCEPTIBLE INTERFERENCE	
TABOO	HDTV INTO NTSC *	NTSC INTO NTSC **
COCHANNEL	-14 TO -6 dB	-22 dB
ADJACENT CHANNEL (n ±1)	> 30 dB	2 TO 20 dB
INTERMODULATION: (n ± 2, n ± 3, n - 4)	> 40 dB	13 TO 40 dB
HALF I.F. (n + 4)	> 30 dB	4 TO 19 dB
I.F.BEAT (n ± 7, n ±8)	> 45 dB	0 TO 40 dB
IMAGE (n + 14)	> 45 dB	3 TO 28 dB
IMAGE (n + 15)	> 45 dB	-17 TO 6 dB

DESIRED SIGNAL LEVEL: -45 dBm

* BASED ON MEASUREMENTS (WITHOUT USE OF THE TEMPORAL FILTER) ON SEVEN TV RECEIVERS REPRESENTATIVE OF EXISTING RECEIVER POPULATION

**FROM FCC/OET TM-1 REPORT BY HECTOR DAVIS

TABLE II

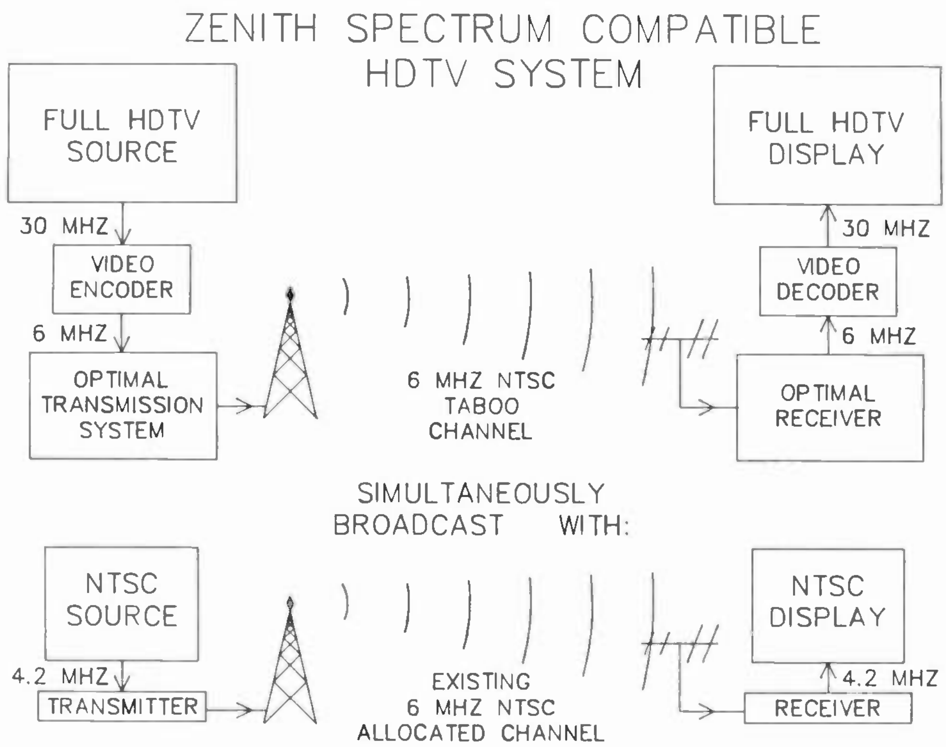


FIGURE 1

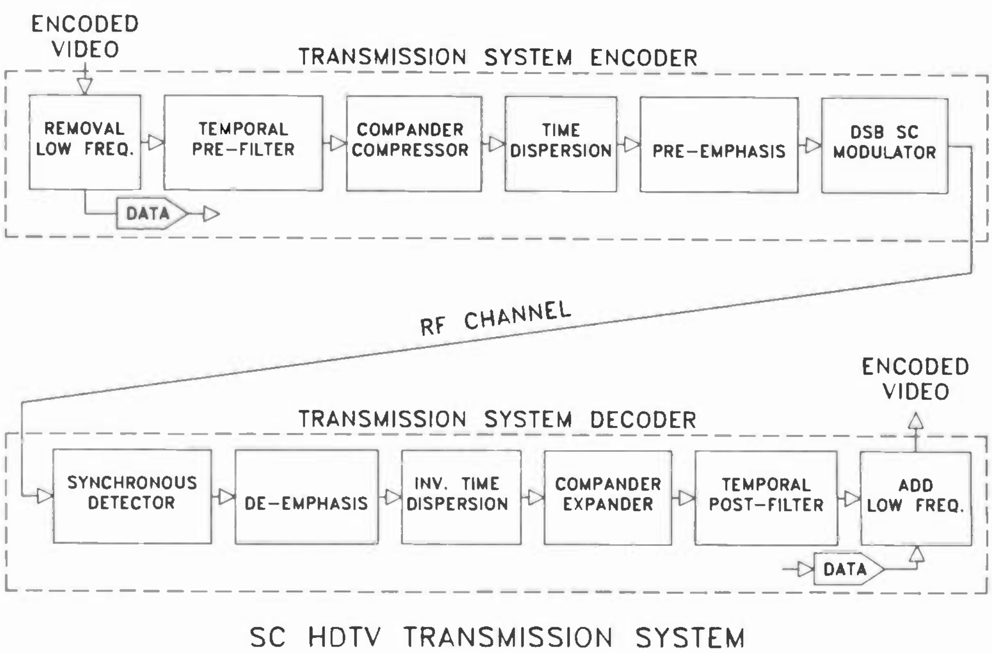


FIGURE 2

SPATIAL - TEMPORAL RESOLUTION COMPARISON
NTSC vs. SC-HDTV

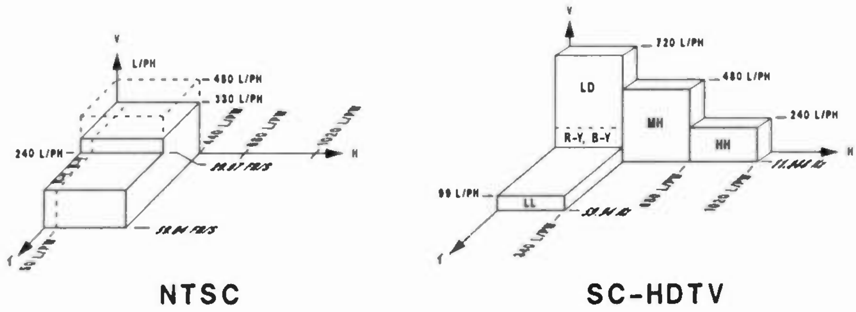


FIGURE 3

MODULATION ON IN-PHASE AND QUADRATURE CHANNELS

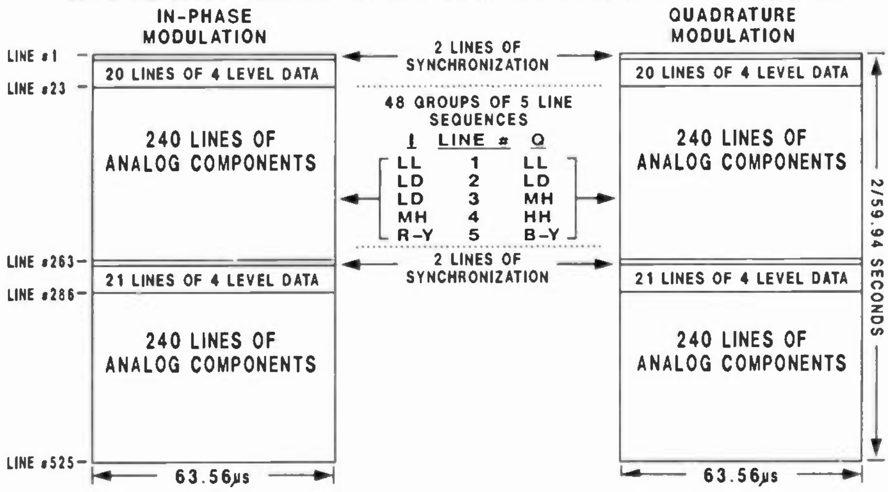


FIGURE 4

FM FORMAT OF HDTV SIGNAL

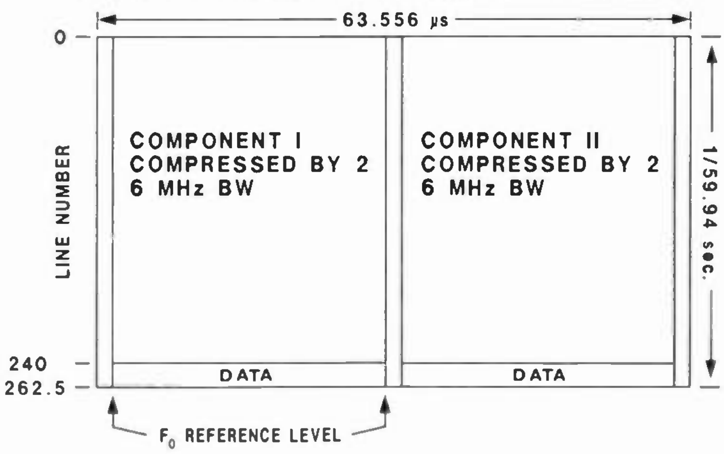
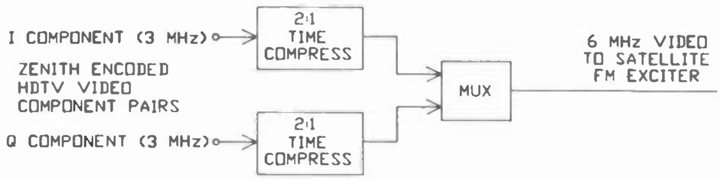
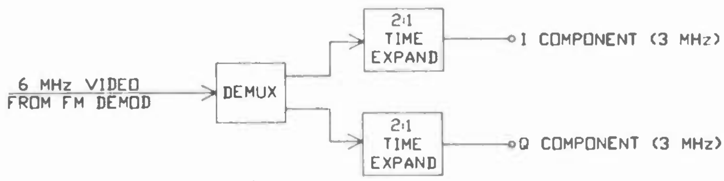


FIGURE 7



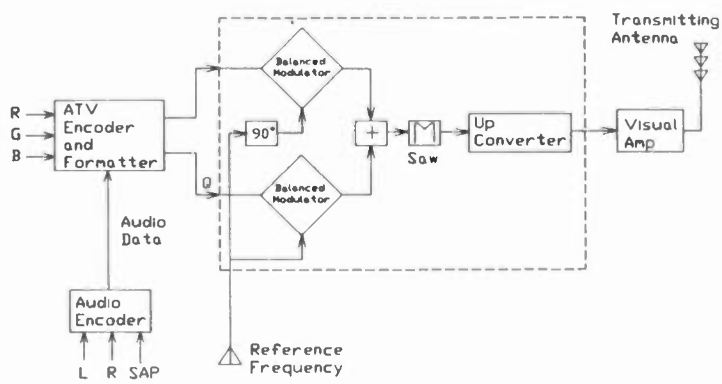
COMPONENT PAIR TO FM FORMAT CONVERSION

FIGURE 5



FM FORMAT TO COMPONENT PAIR CONVERSION

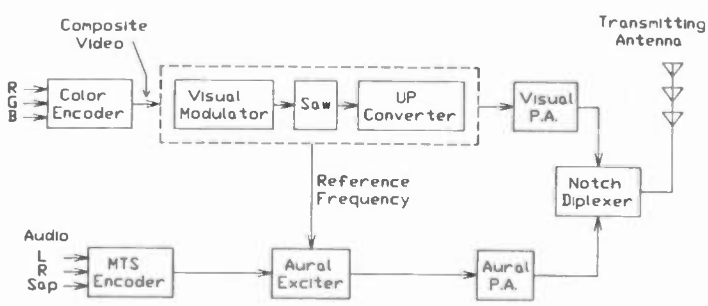
FIGURE 6



ATV Transmitter

Zenith Spectrum Compatible HDTV System

FIGURE 8



NTSC Transmitter

FIGURE 9

COMPATIBLE INTRODUCTION OF HDTV IN NORTH AMERICA

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Abstract

The arrival of HDTV (High Definition Television) in North America will present a myriad of technical problems for both the designers of the new television signal and the broadcast, cable (CATV) and satellite television industry. By the early 1990s, HDTV is expected to spark a revolution in mass communications and become the most significant development in the television industry since the advent of color. However, in the short time frame remaining, many crucial questions about the eventual implementation process remain unresolved. How to approach this huge television marketplace with a new HDTV system in a balanced and compatible manner is the subject of this paper.

Methods of HDTV Signal Coding

HDTV will be a higher quality television format than conventional NTSC and will require much greater bandwidth capacity than present television. This requirement poses the central challenge of actual HDTV implementation.

This need for bandwidth reduction can be explained by the physical properties of HDTV RGB TV signals--20 MHz for every R,G and B totals 60 MHz while an NTSC studio quality signal (with 4.2 MHz for each RGB) presently totals 12.6 MHz of bandwidth. So the difference between the two types of signals is approximately 5 times greater bandwidth.

Furthermore, the NTSC studio signal being coded reduces this bandwidth by a factor of 3. So 12 MHz is transmitted as a composite of 4.2 MHz. If the same type of coding would be applied to a HDTV signal, the 60 MHz of HDTV would then require 20 MHz of base band spectrum. Such bandwidth is prohibited in the present TV broadcasting environment.

Presently assigned TV channels are 6 MHz wide. The Federal Communications Commission (FCC) hasn't shown any indications of favoring a significant increase in spectrum allocation for present television. So further reduction in spectrum is crucial.

A few HDTV approaches can be generically differentiated based on two major subsampling schemes. The first is the quincunx used by MUSE and the other is the HDS-NA concept which features a linear subsampling system [1].

The quincunx signal concept is based on the simple idea that only a quarter of the visual samples are transmitted in every field and full picture and full sample grid are recombined using samples from four fields.

Linear subsampling, on the other hand, is basically an extension of what's presently used as an interlace scheme. The actual subsampling also occurs within four fields. However, because of the way in which the samples are organized, it's not necessary to remember all the three fields prior to the field being displayed. Instead, it's possible to practically transmit, process, and display each field without memorizing other fields in the receiver at the expense of having artifacts in the diagonal direction (Fig. 1).

There are two basic differences between the quincunx and the HDS-NA approach in regards to memory. With quincunx, memory is required all the time but with the HDS-NA approach, memory is optional.

With linear subsampling and optional memory, unwanted visible artifacts can be eliminated in still pictures in diagonal directions. However, these artifacts can occur in the diagonal direction with motion.

On the other hand, since memory is absolutely necessary with quincunx subsampling this problem is far more prevalent. Motion artifacts are found not only in diagonal directions but also in the vertical and horizontal directions. These motion artifacts continuously accompany all moving sequences unless elaborate motion compensation systems are employed [2].

The linear subsampling method is friendlier to the human vision where the diagonal resolution of the eye tends to be slightly less than vertical and horizontal [3]. Also, the nature of the artifacts in this system will appear familiar to television viewers (even though artifacts will occur less frequently in the linear subsampling method than in conventional television).

On the other hand, the quincunx subsampling system is a completely new experience to the human eye. It can be described as uniform loss of sharpness when motion occurs. This conspicuous phenomena is readily detectable since it's very unnatural. Television viewers are not accustomed to seeing such drastic cuts in resolution on motion. Therefore, the motion artifacts that occur because of quincunx subsampling would inevitably prove more noticeable--and probably more annoying--to the average viewer.

Compatibility with NTSC

One of the biggest intangibles with regard to the introduction of HDTV is how the compatibility question will ultimately be addressed. As much a political question as a technical one, the issue will ultimately be decided when the Federal Communications Commission (FCC) makes its eventual standards decision.

With its ruling last fall to champion only an HDTV system compatible to NTSC, the FCC has set down a mandate. However, different HDTV proposals approach this compatibility requirement from different directions.

The varying approaches to compatibility can be categorized into three main methods:

* All extra information needed to improve present NTSC transmission is encoded within the existing 6 MHz NTSC channel by introducing additional artifacts.

* All information needed to be coded is separated and transmitted as an extra channel of 3 or 6 MHz (the so-called augmentation channel) and the existing NTSC channel isn't altered in a significant way.

* The HDTV channel is encoded in a separate 6 MHz channel and a simulcast transmission is used for compatibility.

Simulcast, the transmission approach favored in the last method, occurs when non compatible HDTV transmissions and compatible NTSC signals are transmitted at the same time.

Proponents of simulcast systems claim that in the future a large segment of the population will receive simulcast transmissions and that most of the consumers' homes will be specially equipped to see a noncompatible portion of the simulcast. Thus, the proponents reason, the compatible portion could eventually be phased out entirely, allowing the HDTV transmission to eventually be the most efficient means of transmission.

The heart of this proposal hinges on an earlier demise for conventional NTSC television than truly seems plausible. In actuality, it appears highly unlikely that such a phase out situation could occur in the next 15 or 20 years. NTSC hardware will probably continue to be sold in large volume over the next decade. NTSC television models have proven to be extremely reliable and shouldn't become obsolete in the next 10 or 20 years--at least.

For this reason, simulcast is actually a 6 + 6 (12 MHz total) transmission system and consequently becomes the least practical of the three scenarios for compatible HDTV (the other two being the 6 + 0, and 6 + augmentation approaches).

What are some valid criteria to help judge the effectiveness of compatible HDTV transmission? Three goals are pertinent:

1) The quality of the NTSC compatible portion of this transmission must be unimpeded. This preservation of quality is crucial to ensure a smooth introduction for HDTV and minimize hesitation and market wars.

2) The HDTV part of the terrestrial transmission should be good enough to compete with other HDTV transmissions possible from Direct Broadcast Satellite (DBS) systems, video tape recorders or other sources.

3) RF bandwidth utilization should be effective. Effective to produce best quality pictures and sound with the highest degree of image compression accompanied with reasonable complexity of receiver and low visibility of coding artifacts.

If these criteria are applied to each of the proposals, it's clear that the 6 MHz + 3 MHz of augmentation scenario comes closest to fulfilling all three. The HDS-NA proposal, for example, would provide 650 to 700 lines of vertical resolution and 500 lines of horizontal resolution and digital sound as well as a 16 to 9 aspect ratio. The picture would be free of motion artifacts with absolutely clean, unimpeded NTSC compatible transmission. Total RF bandwidth consumption will be 9 MHz.

The 6 + 0 transmission would have less quality for an HDTV receiver and could seriously impede the quality of NTSC transmission. The 6 + 6 transmission seems even more dubious when one notes that it may not even be possible to encode high quality HDTV transmission in 6 MHz. Even if it works, this system would present the least efficient use of the spectrum.

Weighing all these considerations, it appears that the 6 MHz (plus 3 MHz augmentation), NTSC unimpeded signal would provide the most balanced, and smoothest transition scenario for terrestrial and cable broadcasters from present NTSC to HDTV. It addresses the central issues of HDTV compatibility more effectively than either the 6 + 0 or 6 + 6 simulcast proposals.

Another factor that must be considered in evaluating HDTV proposals involves spectrum. The FCC has stated that there will be no additional spectrum allocation for terrestrial broadcasters. Because of this restriction, if the 6 + 3 scenario is chosen, the spectrum will have to be found within the broadcasting spectrum.

Linear Subsampling Method

The linear subsampling used in the HDS-NA concept is unique for two reasons. It provides very high quality, no motion artifacts, and a high resolution picture achieved with an inexpensive decoder. Secondly, it provides two kinds of expendability. First in the diagonal direction, if memory is used in decoder, and second in the horizontal, direction if more bandwidth will be available (better transponders, more bandwidth on CATV, more channels for terrestrial broadcaster, etc.).

The quality of the signal can be upgraded on a linear basis if more bandwidth is allocated in the future. This point is another basic differences between this system and the MUSE format where the spectrum is folded back around a folding point [1]. The basic disadvantage of that system is -- if a large bandwidth is eventually made available for such a signal, the designer must move the folding point to a different location. This means that a significant part of the system would have to be totally redesigned.

With the linear subsampling scheme used in the HDS-NA approach, if more bandwidth is allowed in the future the output filters could be stretched. If more bandwidth is allowed, the 2-dimensional spectrum can be linearly expanded to the right. This is illustrated in Figure 2.

Another advantage of the linear subsampling scheme is that it produces a high quality picture on the basis of a very simple 4-line memory decoder which will reproduce most of the resolution and subsample diagonal details. This decoder does not require any motion compensation or memories and it has the same response to noise signals as present television. In other words, there won't be any surprises in terms of implementation and quality for broadcasters and viewers.

Reference [3] demonstrates that the human eye instinctively notices resolution changes in vertical and horizontal directions more than similar changes in diagonal directions. It has been surmised that diagonal resolution can be slightly reduced from the perfect circular resolution without potential impact of the viewer.

However, in the HDS-NA subsampling concept, such downscaling of resolution quality isn't necessary. Internal memory functions make diagonal resolution fully recoverable for stationary pictures up to the full diagonal of the rectangular.

The approximation presented in the HDS-NA concept is an attempt to simplify two dimensional, digital signal processing hardware involved in the decoder. The result is very simple digital signal processing hardware which could be used for a simplified version of the HDS-NA decoder.

This hardware is utilized presently and its laboratory prototype is shown on Figure 3.

Implementation of this system can be easily achieved to put all necessary decoding hardware on a single card without very significant integration efforts.

Terrestrial Broadcasting, CATV, and Satellite Broadcasting considerations

The NTSC television standard was developed for use as a terrestrial broadcast signal and has demonstrated staying power. It's performed fairly well based on existing technology and it's proved its performance over the years as a terrestrial and cable format.

However, for a large scale introduction of HDTV to become a reality in North America, designing a direct exchange for NTSC is not enough. Other means of television delivery are just as crucial as terrestrial broadcasting. One of the delivery systems that poses the most growth possibilities is satellite destruction.

Presently, satellite delivery is used as a feeder signal. The majority of today's television is delivered to local, terrestrial and cable broadcasting facilities by satellite transmission (utilizing FM modulation). It's not enough to address all terrestrial and cable transmissions. In order to introduce the new HDTV format, provisions for satellite have to be made.

The NTSC signal, for the reasons of having the color subcarrier is an FM-unfriendly signal. Because of the inherent triangular noise resulting from FM transmissions, NTSC is acquiring noise in sensitive area around color subcarrier. The presence of the color subcarrier also makes NTSC more sensitive to threshold effects.

The subcarrier-based systems are generally not the best balanced signal for FM transmission. FM transmission is best utilized if a gradual roll off spectrum is presented as black and white or MAC (Multiplex Analog Component) signal. This gradual roll off spectrum are the optimal one to use in transmission because they accommodate the unique noise shape of FM communication in the most efficient way [4].

As discussed previously, spectrum is a scarce resource in terrestrial broadcasting and CATV. For this reason, any effective HDTV compatible transmission must be highly optimized for the prevailing AM environment. HDS-NA will utilize double side band (DSB) modulation in the effectiveness of single side band (SSB) modulation transmission. This unique feature of HDS/NA allows a more effective spectrum utilization, suppression of multipath type of channel artifacts, and a simple receiver architecture [5].

This kind of signal is poorly suited for FM transmission. So, the optimal situation for introduction of compatible HDTV is not one but two signals. Of these two signals, one must be optimized for FM and the other optimized for AM. Both signals must "talk" to each other in a friendly way and be transcodable from one to another presenting no additional artifacts.

This scenario is obviously the most balanced and responsible scenario of introducing compatible HDTV in North America. For this reason, HDS-NA is a two-signal concept.

The first signal is a highly optimized signal for satellite transmission which is fully capable of being transmitted over conventional C-band transponders or future DBS transponders [6]. This way the signal can be used either as a cable or terrestrial feeder signal or as a DBS signal with possible future applications.

The terrestrial/cable signal is directly transcodable from this format and then highly optimized for the AM environment to be transcoded in the highly compatible NTSC portion and augmentation portion for AM transmission. Both of these signals don't alter linear subsampling scheme in the HDS/NA format [7].

The transcoding from satellite signal to terrestrial cable signal is very simple and inexpensive. It also prevents the intrusion of transcoding artifacts.

In order to form terrestrial part of HDS/NA the portion of the signal extracted from the satellite format is then linearly compressed or expanded to form panels and centers respectively used for the augmentation of the NTSC channel. The panels are transmitted separately as augmentation signals. The centers are transmitted as an exact NTSC specified signal and on the main compatible channel.

Conversely, a system merely designed only for terrestrial and cable and which does not address the satellite situation seems like a potential candidate for a problems. In order to deliver compatible signals with such a system certain compromises would have to be made. Obviously, a system carefully designed to be most efficient for satellite, microwave, terrestrial, and cable TV presents greater possibilities and fewer limitations.

In the future, if fiber cables become a reality of television signal delivery, there could be two ways signals could be modulated on fiber: through a regular analog FM transcoder or digitally.

Utilization of "Taboo" Channels

The HDS-NA 6 + 3 system, as previously stated, consists of a standard 6 MHz wide NTSC channel and the 3 MHz-wide augmentation channel which will transmit all the additional information needed to augment the NTSC channel to help it become a HDTV transmission. This 3 MHz channel could be transmitted on a presently, unassigned television channel.

Unfortunately, such a luxury will not be afforded the terrestrial broadcasters. The only available terrestrial channel capacity appears to be the so-called "taboo channels" which are currently restricted for television transmission. However, if the amount of power transmitted could be decreased, these taboo channels could probably be utilized. See Figure 4.

Significant reduction of power would be possible with the HDS-NA system since there are strong indications that the augmentation channel could be transmitted digitally. This would allow for a significant reduction of transmitted power over the augmentation channel, subsequently reducing the amount of interference to other channels using the same frequency band [8].

Under this approach, the NTSC portion of the transmission--may be subjected to additional interference. However, this interference will be added to the NTSC transmission far away from the augmentation channel transmission. Furthermore, the antenna and receiving system will be optimized to receive the NTSC system from a totally different tower so the low power augmentation channel will be farther suppressed by the front to back rejection ratio of the receiving antenna.

This interference will decrease rapidly if the receiver location is closer to the main NTSC tower or the NTSC transmission of a compatible channel is in a different territorial location.

The 6 + 3 scenario only introduces possible interference in the specific areas and does not introduce any significant interference in other areas. Analysis of taboo channel utilization is given in [9].

It is a very difficult task to place all the information that has to be transmitted on the augmentation channel in the frequency space of 3 MHz. However, this problem is solved with some concepts inherent in the HDS-NA analog system.

The specific method in the analog system which makes efficient use of the AM spectrum. By using one line of memory, and expanding the circuitry, the AM double sideband augmentation channel transmission is as efficient as an AM single sideband transmission. The specific allocation of components to be modulated and inserted in the 3 MHz space is given in Table 1.

Digital Coding of Augmentation Channel

In an NTSC compatible HDTV transmission system such as HDS/NA, the NTSC compatible portion of the signal must be transmitted in analog form. There is, however, the possibility of digital transmission of enhancement information. The enhancement information consists of panels for 4:3 to 16:9 aspect ratio conversion, and extra horizontal and vertical detail information. In our study the HDTV signal is defined to be a 525 1:1 (progressive) or 1050 2:1 (interlaced) source with 59.94 frames/sec and 16:9 aspect ratio.

Digital transmission of video signals has the advantage that transmission at low power is possible, and that error correction schemes can be applied to improve the signal to noise ratio. The major disadvantage of digital transmission is that the transmission bandwidth will be higher than what is required for analog transmission of the same signal. This suggests that data compression of the source signal is a necessity if digital transmission is to be practical.

Extensive research in the area of transform coding has shown that two-dimensional (2-D) discrete block cosine transform (DCT) has superior performance compared to other transform encoding techniques while being practical for hardware implementation. However, the previous studies have traditionally dealt with standard television or teleconferencing signals and the results of these studies can not be readily extended to signals possessing mostly high frequency content. An investigation was done to evaluate the performance of the DCT encoding scheme for compression of enhancement components within HDS/NA.

The compression scheme currently used can be described as follows: The two-dimensional augmentation signal is partitioned into non-overlapping 8x8 or 16x16 blocks. Each block is then transformed using 2-D DCT and further encoded using quantization followed by variable-length coding (VLC). This process is complemented in the decoder. The example of a possible bit rate reduction for various components of digital augmentation channel is given in the Table 2.

"Taboo" Channel Characterization

Apparently, the taboo channel characterization has to be done in order to most efficiently utilize taboo channel capacity. There are strong indications that there are tremendous amounts of capacity within taboo channels. Some attempts to characterize this capacity, were done in [9] and indicated by the FCC replies to the Notice of Inquiry.

Any single party which attempts to characterize these taboo channels will not present the most objective point of view. This characterization should be done in an organized fashion and on the basis of what kind of augmentation or simulcast has to be transmitted.

For this reason, proponents who intend to use these taboo channels should be able to provide a test vehicle to examine potential requirements in three areas: transmission energy need, interference and channel characterization. This testing must be done in a centralized manner with statistical data available to system proponents.

Production Standard Issue

If HDTV signal transmission in North America is to be started in compatible way, then 1125 lines, 2 to 1 interlace, 60 Hz, HDTV production format parameters is not suitable.

If this type of standard is utilized for production material, the converters have to be employed continuously on all points of re-translation. This is an unnecessary expense from a user point of view, and will introduce unnecessary artifacts into the quality chain of HDTV transmissions. Regrettably, there will be plenty of chances to degrade quality by imperfect reception and transmission and it's completely unnecessary to introduce additional artifacts in the beginning of the chain.

Fortunately, NBC's recent proposal of a 1050 to 225; 59.94 Hz production format does not have this disadvantage. This format is completely compatible for transcoding into HDTV and transmitted on terrestrial, cable or satellite. It is quite obvious that this production format will be the best for the North American situation.

It is understandable why proponents of both the 6 + 0 compatible HDTV format or 6 + 3 augmentation format strongly agree with this statement.

Originally, when the 1125-60 format was approved, one of the main considerations was the availability of certain types of production equipment. Today no such shortages of equipment exist. Basic production equipment capable to work 1050/525 and 59.94 is now readily available from a variety of sources.

VTRs are available off the shelf from Sony which can easily be modified to work 1050/525 format. Telecine is available from Rank & Cintel in four different production formats by the manufacturer. The studio camera is available from BTS and is also offered in four different formats.

There are other types of telecines (CCD) that could be introduced also capable of a multi-production format standard (BTS). The monitors developed for 1125-60 have no problems adapting to the 1050/525-59.94 format.

For this reason, there is nothing missing that's needed to fully equip a contemporary production facility with 1050/525-59.94 format equipment. There is also no need to "fold back" to 1125-60 equipment, which would necessitate transcoders in order to transmit the signal compatibly.

Conclusion

In conclusion, the HDS-NA system can be summarized by a few important points. These include:

- compatibility on an even-handed basis between the three modes of television delivery--satellite, CATV and terrestrial broadcast.
- a linear subsampling technique that aids in effective bandwidth reduction and simple receiver implementation.
- low power transmission of the HDNTSC augmentation channel should allow to use the 'taboo' channel without introduction of interference to other TV channels using the same frequency band.
- main elements of HDS-NA system like linear subsampling method, separation and stitching panels and centers etc. has been demonstrated in actual hardware.

Acknowledgment

The author thanks the members of ATV systems department of Philips Research Laboratories, for help in conceiving and realization of HDS/NA system.

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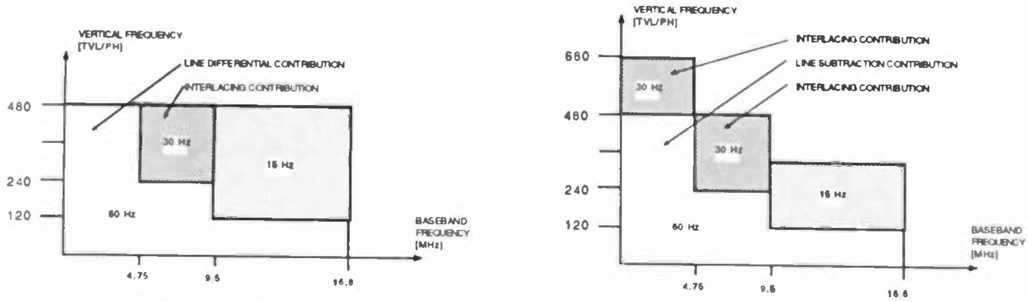

HDS/NA for 525; 1:1

HDS/NA for 1050; 2:1

FIGURE 1 HDS/NA LUMINANCE SPECTRAL DISTRIBUTIONS

HDS/NA for 1050; 2:1

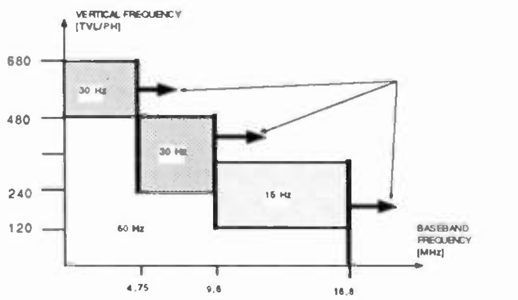

FIGURE 2. STRETCHING OF SPECTRUM IF MORE BANDWIDTH IS AVAILABLE

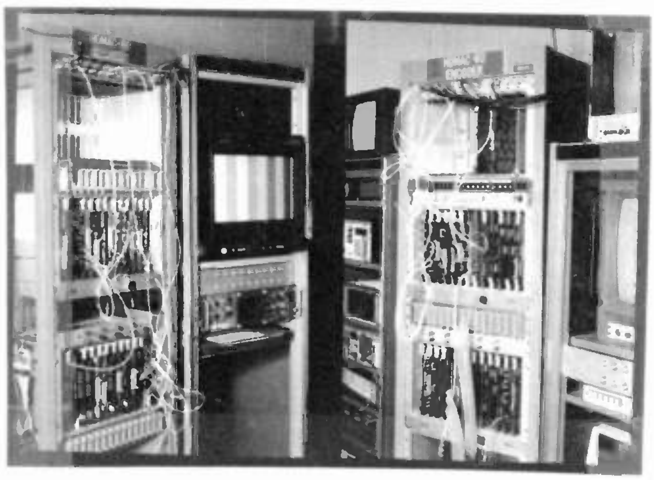

FIGURE 3.

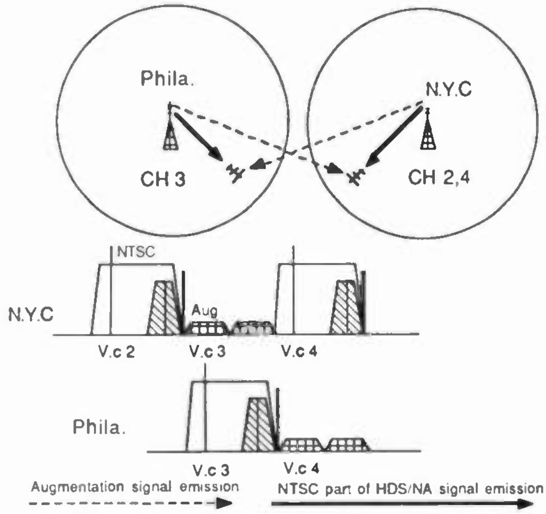

FIGURE 4. HDS/NA for Terrestrial Broadcasting

TABLE 1. TABULATION OF COMPONENTS FOR THE 3 MHz AUGMENTATION CHANNEL

COMPONENTS	ORIGINAL TIMEBASE & BASE BANDWIDTH	EXPANSION RATIO	TIMEBASE & BASE BANDWIDTH	
LDl2 + LDr2	7.1 μsec 3.3 MHz	5:2	17.7 μsec 1.3 MHz	modulated in quadrature; total bandwidth 2.6 MHz
LDl4 +LDr4	7.1 μsec 3.3 MHz	5:2	17.7 μsec 1.3 MHz	
Yh	26 μsec 5.6 MHz	2:1	52.00 μsec 2.8 MHz	
NTSC PL1 + PR1	7.1 μsec 11.2 MHz	8	56.7 μsec 1.4 MHz	modulated in quadrature total bandwidth 2.6 MHz
NTSC PL3 + PR3	7.1 μsec 11.2 MHz	8	56.7 μsec 1.4 MHz	
DSS	440 Kbps	QPSK	127.11 μsec 0.2 MHz (total bandwidth)	

TABLE 2. DIGITAL HDS/NA 'SUPER' LINE TIME BUDGET

	Time	Orig. B/W	fck	Bit rate
Pr + Pl -> Y	7μs	11.2 MHz	23 MHz	1b·23·7+127=1.26 mb/s
Pr + Pl -> Chr	7μs	11.2/4=2.8 MHz	5.7 MHz	1b·5.7·7+127=0.32 mb/s
		total for P3+P1 -> (1.26 + 0.32)·2 = 3.16 mb/s		
LD	26μs	3.3 MHz	6.7 MHz	0.5b·6.7·26+127=0.7mb/s
		total for LD2+LD4 ->		0.7·2 = 1.4 mb/s
Yh	26μs	5.6 MHz	12 MHz	0.5b*12*26/127=1.2mb/s
		total for Yh ->		1.2 mb/s
DSS	127μs			0.44 mb/s
(LDl+LDr; LDc encoded in quadrature with NTSC)	7μs	3.3MHz	6.7 MHz	(0.5b·6.7·7+127= 0.2mb/s)
total for digital -> ASP				6.2 mb/s (5 mb/s)
Transmission using digital modulation with 2 bit/hertz				3.1 MHz (2.5 MHz)

NTSC COMPATIBLE WIDE ASPECT EDTV

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INTRODUCTION

The EDTV system of NTSC-compatible wide-aspect images is under development as a high picture-quality television system for terrestrial broadcasting of the future.

In Japan, the HDTV system, primarily for satellite broadcasting, and the EDTV system, primarily for terrestrial broadcasting are being developed in parallel. The technical specifications of the first-generation EDTV system have already been difined and actual broadcasting operation is scheduled to begin this autumn.

The first-generation EDTV system, as a basic picture-quality improvement system, will be implemented by a combination of several techniques for the main purpose of compensation the deficiencies of the present NTSC system, while retaining aspect ratio of 4:3 as is. In this system, the anticipated picture quality is improved by approximately 1.5 in subjective assesment value ranking (7-grade comparison scale), allowing fine and crisp pictures to be enjoyed with large-size screens of about 30 inches.

Additionally, the transmitter and receiver devices are relatively simple, making the system preferable from the standpoints of cost effectiveness, technical feasibility, and social acceptance. Further, as a ghost canceller is employed, complex ghosts in urban areas are automatically eliminated. Thus this system is advantageous in cost/performance as a high-quality TV system for the presently intended terrestrial broadcasting.

But, even with the advent of expanded HDTV services in terms of service content and service areas via broadcasting satellites, CATV, and package services, the need for a wide aspect terrestrial EDTV system may well be increased because of TV audience requirements, TV program exchanges, and so forth.

OUTLINE OF THE SYSTEM

In order to realize wide-aspect images on NTSC=compatible EDTV receivers, one of the major problems is how to display images of different aspect ratios using the same screen. To deal with this question, there are basically two methods; one is to fully show the upper and lower image edges on display screens of 4:3-ratio NTSC TV receivers, and the other metheod to fully show the right and left image edges. In the former the right and left side panels are discarded, in the latter the entire images are squeezed with upper and lower portions in grey.

In the present broadcasting system, when wide-aspect movis pecture programs for cinema theaters are aired, a selection is made between the two methods described depending upon the situation. If the agreement of the program producer and the author are obtained, the program images are cut with proper trimming and televized.

However if no agreement is obtained for trimming, the program must be aired with squeezed image dimensions. It may also be necessary to consider similar questions for wide aspect EDTV system,however it is not a simple matter to decide because modification of the system itself is involved. To deal with this problem, we in Nippon Television Network Co. have been carrying out experiments to develop two types of transmission techniques to be able to cope with either of the two methods involved, and we will continue further studies on this matter.

Figure 1 shows the concepts of two Modes:Mode 1 is the method where the side panels are neglected, and Mode 2 the method of image squeezing. Tables 1 and 2 show the respective specifications of Mode 1 and Mode 2. For each system, the required functional levels can be selected below we present some descriptions of the basic systems.

Mode 1

Surveys have revealed that standard TV receivers generally have about 8 to 10% of overscanning in terms of screen size. In Mode 1, we assumed an effective screen area for NTSC TV receivers at 92% each for both the vertical and horizontal screen directions, with the side panel information transmitted over the overscanning areas of both the side edges which are hidden by the side frames. Thus, the extra side panel information is not visible on general NTSC TV receivers, but wide aspect images can be seen by EDTV-dedicatad TV receivers.

Figure 2 shows the relationship between screen size and pixels of Mode 1. The effective screen area includes 444 lines in the vertical direction and 684 pixels (4fsc sampling) in the horizontal direction. Over the remaining upper and lower 19 lines each and the 30 pixels at the right and left, the side panel signals are transmitted.

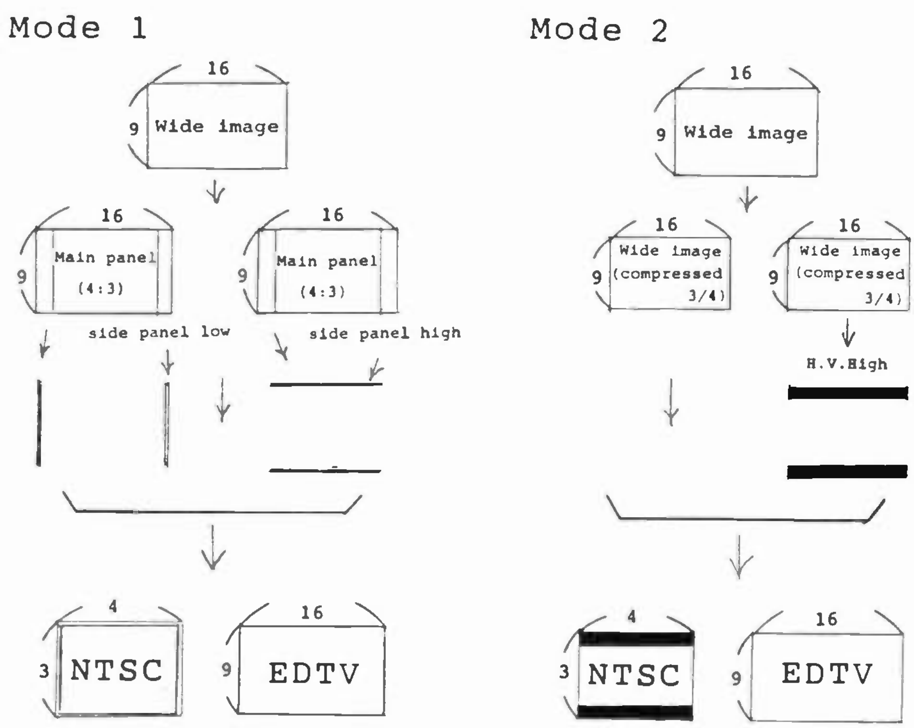

Figure 1. Concept of 2 Modes

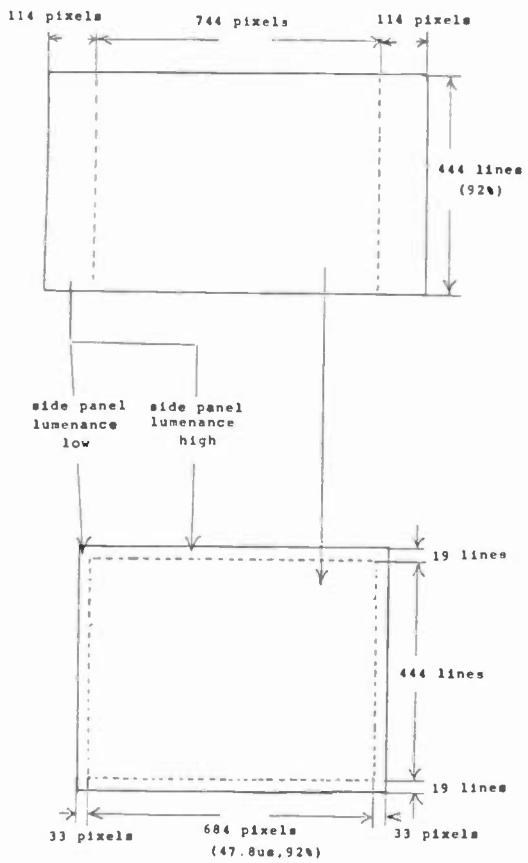

Figure 2. Relationship between screen size and pixels of Mode 1

Figure 3 shows the signal processing and transmission methods of the side panel information.

The number of side panel pixeis of the 16:9 ratio wide-aspect original images is 114 each on the right and left. As the available transmission channel area covered by the side frames is only the half of the area of side panel portions, some transmission techniques must be devised. Therefore, we selected a method to time-compress the low-band elements of the side panel information into one quarter and to transmit them over the screen side portions, while the high-band elements are transmittad over the upper and lower overscanning portions.

Since the frequency bandwidth is expanded when the signals are time-compressed the low-band elements of the side panel information of the original images corresponding to the 4.2-MHz bandwidth available for the transmission are in the range of 0 to 1.05MHz.

The high-band element information in the range of 1.05 to 3.15MHz to be transmitted over the upper and lower portione is frequency-shifted into the range of 0 to 2.1MHz, and its time axis is time-compressed into one half to producs signals in the range of 0 to 4.2MHz. The output data of the high-band elements for the right and left 8 lines are arranged in a single scanning line. The entire output data must be transmitted over the extra 19 overscanning lines at the upper and lower portions that can only afford half of the 444 lines. Therefore, the

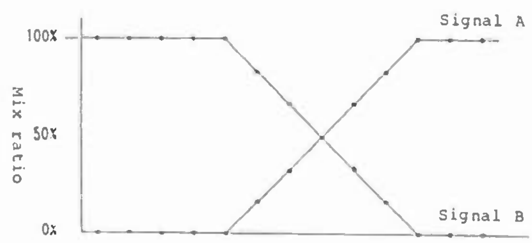

Figure 3. Method of overlapped splicing

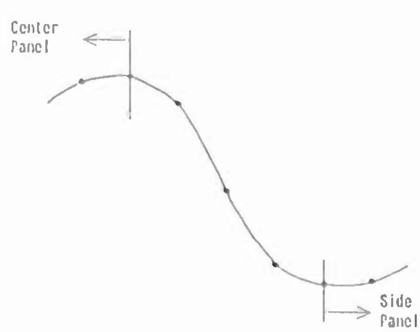

Figure 4. Concatination method of data

vertical resolution is reduced by half to conserve data volume, and the data for a single frame are transmitted but by using two frames.

As in Mode 1, which divides the side panels and the center panel to be sent over different transmission channels, the splicing lines may appear unnatural. In this system, to achieve smooth splicing of the center panel and side panel, they are overlapped by 5 pixels. We chose the number of 5 pixels, because improvement of picture quallty is recognized up to only 5 pixels according to a separately conducted experiment. Figure 3 shows the method of overlapped splicing.

On the one hand, when different data are arranged on a single line and transmitted, distortions may occur before and after the concatinated point, because of data discontinuity. To avoid this problem, signal processing is necessary and we employed a concatination method of data using a cosine curve as shown in Figure 4

Figqute 5 shows the configuration of the transmitter, and figure 6 shows a configuration of the receiver.

Mode 2

Mode2 is a transmission technique to compress the size of an entire image into three-quarters so that wide-aspect

Table 1. Specifications of Mode 1

1. Video signal
   1) Scanning rate 525/59.94/2:1
      a. Main panel 444 lines
      b. Upper panel 19 lines
         Lower panel 19 lines
   2) Aspect ratio 16:9
   3) Baseband width 4.2MHz
   4) RF bandwidth 6MHz
   5) NTSC compatibility yes
   6) Luminance signal bandwidth
      a. Horizontal bandwidth
         ·Center panel
            0 - 4.2MHz : Transmitted by main panel
         ·Side panel
            0 - 1.05MHz : Transmitted by left and rigft side of main panel (4X time compression)
            0 - 3.15MHz : Transmitted by upper and lower panel (frequency shift to 0-2MHz,2X time compression)
      b. Vertical bandwidth
         ·Center panel
            444/2 cph : Transmitted by main panel
         ·Side panel
            0 - 1.05MHz 444/2 cph : Transmitted by main panel
            1 - 3.15MHz 222/2 cph : Transmitted by upper & lower panel
   7) Chrominance signal bandwidth
      a. I signal
         ·Center panel
            0 - 1.5MHz : Transmitted by main panel
         ·Side panel
            0 - 1.5/4MHz : Transmitted by left and right side of main panel (4X time compression)
      b. Q signal
         ·Center panel
            0 - 0.5MHz : Transmitted by main panel
         ·Side panel
            0 - 0.5/4MHz : Transmitted by left and right side of main panel (4X time compression)
   8) Synchronizing signal same as NTSC

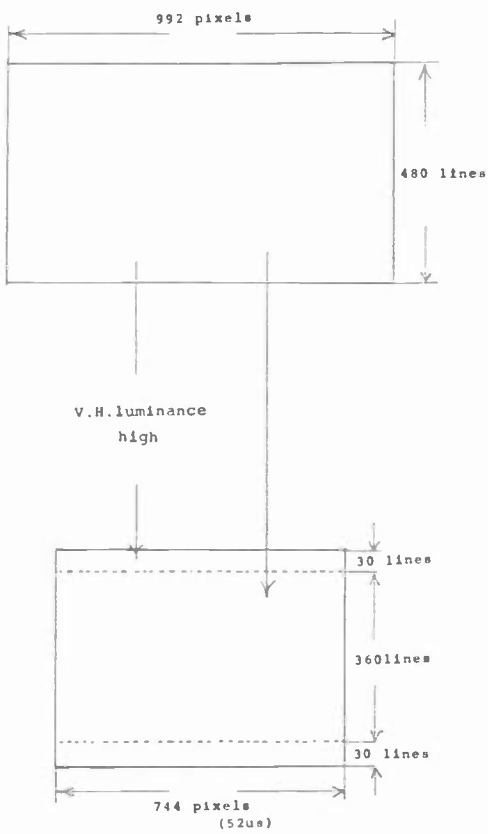

Figure 7. Relationship between screen size and line numbers in Mode 2

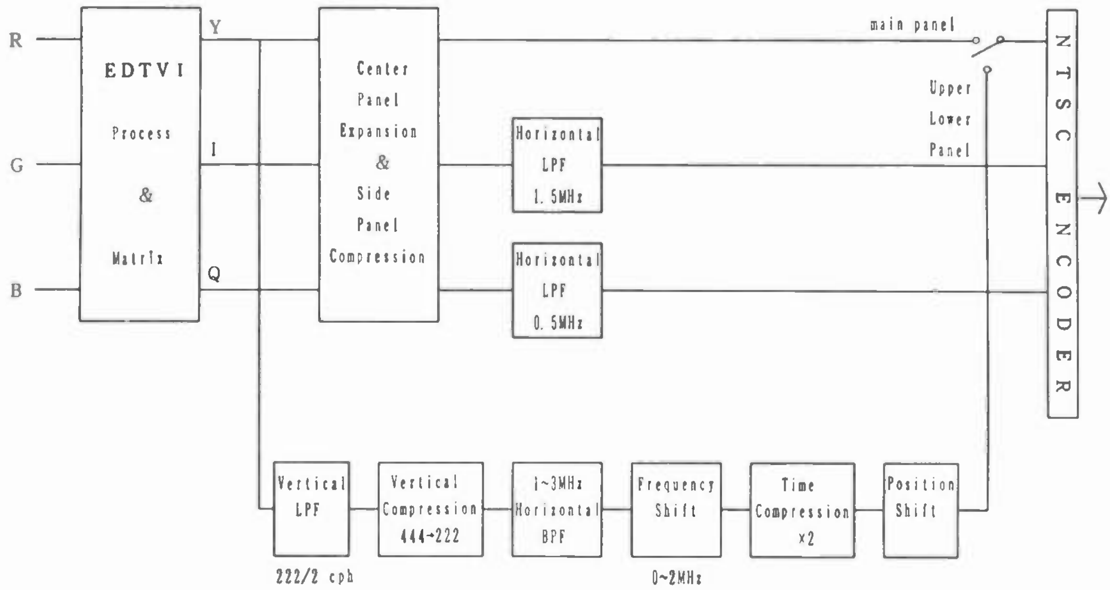

Figure 5. Configuration of the transmitter of Mode 1

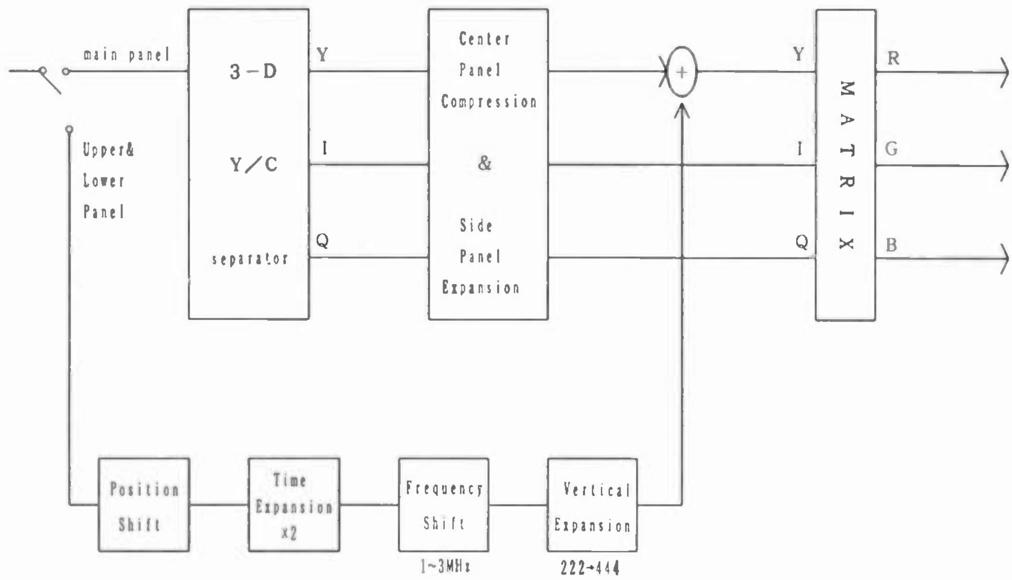

Figure 6. Configuration of the receiver of mode 1

images can be displayed aqueezed on an NTSC 4:3-ratio display. In this mathod, the images may appear smaller and thus the resolution may be degraded accordingly. However, seamless pictures can be regenerated with natural appearance on an EDTV receiver.

Figure 7 shows the relationship between screen size and line numbers in Mode 2. When 16:9-ratio wide-aspect images are aqueezed so that the right and left edges can be fully displayed on an NTSC display, the effective scanning lines number 360 in the vertical direction with the remaining upper and lower 60 lines each as unnecessary scanning lines. Those upper and lower portions correspondent to the blanking when wide-aspect movie picture programs are televised with the current TV system.

In Mode 2, the vertical and horizontal high-band elements are transmitted over the blanking portions. As the high-band element data level is relatively small, the blanking portions can be actually displayed in grey by means of an arrangement such as some pedestal.

Signal transmission over those portions will not be offensive to viewers of NTSC TV receivrs.

The frequency band of vertical 0 to 4.2MHz/2cph and vertical 0 to 360/cph are transmitted with the main panel in Mode 2. The remainig high-band elements must be sent over the upper and lower portions of the panel. As the data volume that can be transmitted over the upper and lower portions is equivalent to one-third the area of the main panel, an effective transmission technique must be devised. In this mode, high-band elements of 4.2 to 6MHz and vertical 360 to 480/2cph are transmitted over the auxiliary transmisson area to ensure the resolution by EDTV receivers.

Firure 8 shows the two signal processing methods used. The vertical high-band elements in the range of 360 to 480/2cph are band-limited to 2.1MHz

Table 2. Specifications of Mode 2

1. Video signal
   1) Scanning rate 525/59.94/2:1
      a. Main panel 360 lines
      b. Upper panel 60 lines
         Lower panel 60 lines
   2) Aspect ratio 16:9
   3) Baseband width 4.2MHz
   4) RF bandwidth 6MHz
   5) NTSC compatibility yes
   6) Luminance signal bandwidth
      a. Horizontal bandwidth
         0 - 4MHz : Transmitted by main panel
         4 - 6MHz : Transmitted by upper/lower panel (frequency shift to 2-4MHz)
      b. Vertical bandwidth
         0 - 360/2 cph : Transmitted by main panel
         360/2-480/2 cph : Transmitted by upper & lower panel
   7) Chrominance signal bandwidth
      a. I signal
         0 - 1.5MHz : Transmitted by main panel
      b. Q signal
         0 - 0.5MHz : Transmitted by main panel
   8) Synchronizing signal same as NTSC

in the horizontal direction, and then converted to the range of 0 to 120/2cph by frequency shifting. In contrast, the horizontal high-band elements in the range of 4.2 to 6.3MHz are band-limited to 120/2cph, and then converted to the range of 2.1 to 4.2MHz by frequency shifting.

Those two signals are frequency-muitiplexed and transmitted over the auxiliary transmission channel. We are also investigating transmitting the two signals after quadrature modulation

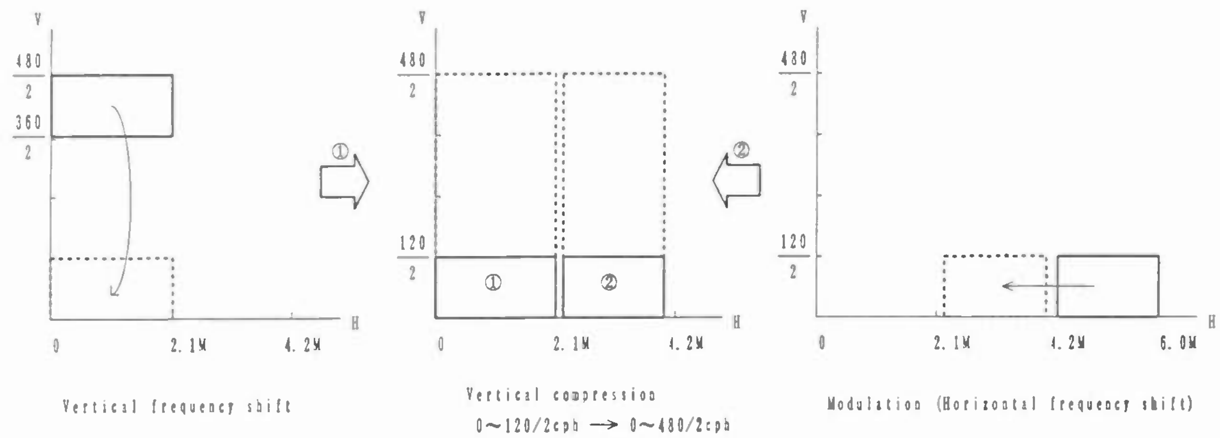

Figure 8. Signal processing method of Mode 2

using a 2.1MHz carrier frequency. It is expected that flickering of the upper and lower potions can be reduced by the modulation.

Figure 9 shows the configuration of the transmitter and figure 10 shows the configuration of the receiver.

PICTURE QUALITY AND COMPATIBILITY

Photos 1,2 and 3 show regenerated pictures of mode 1. Photo 1 is of the under scan mode condition of an NTSC receiver image using a studio monitor. It can be seen that side panel data at

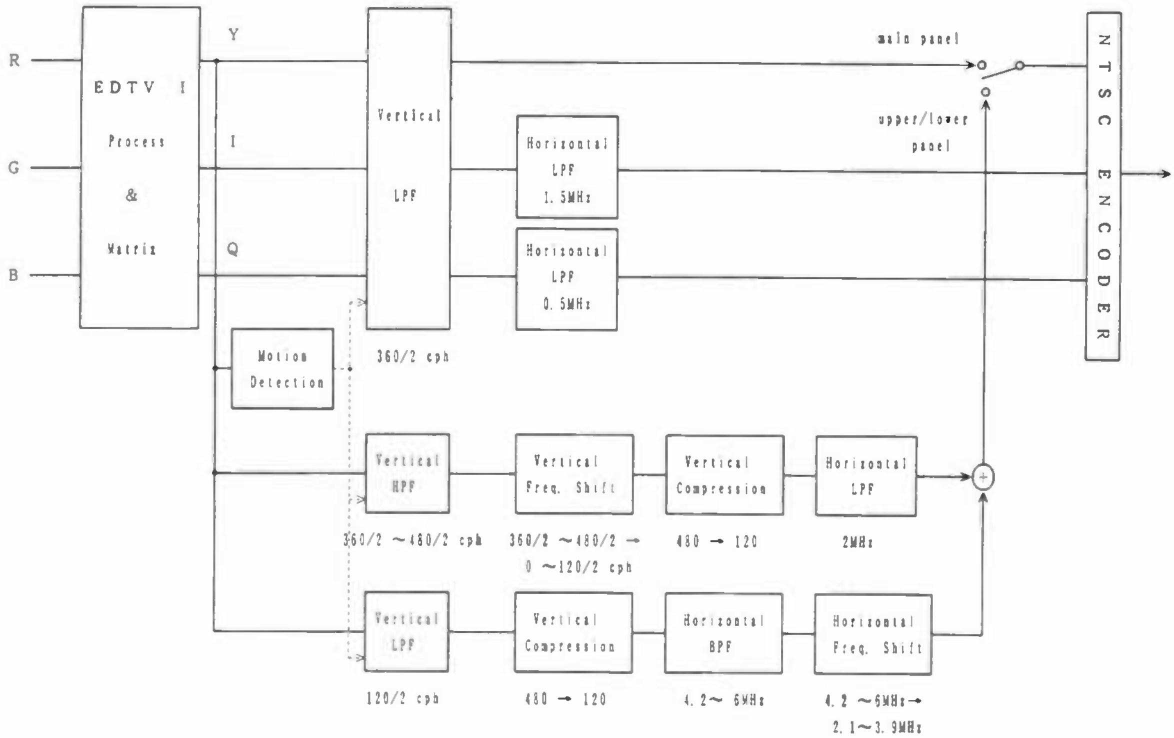

Figure 9. Configuration of the transmitter of Mode 2

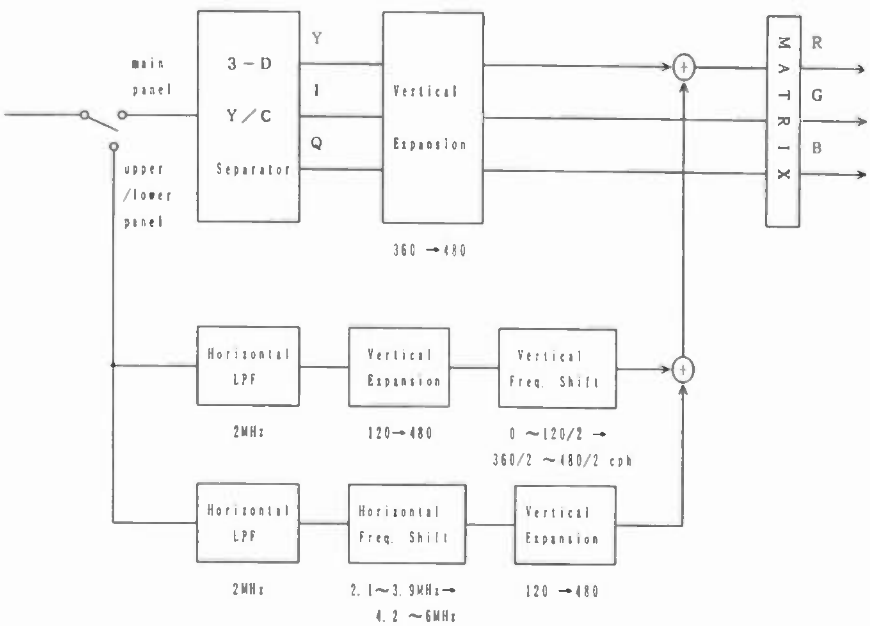

Figure 10. Configuration of the receiver of Mode 2

the edges are transmitted. Photo 2 shows the condition in which the same image is received by a customeruse TV receiver.

It is notable that the side panel data are almost invisible because of the overscanning. Photo 3 shows an EDTV-regenerated picture.
In mode 1, the picture quality of the main panel part is the same as that of the first-generation EDTV system, but it is slightly degraded at the side panel portions.

Photo 4 and 5 shows regenerated pictures of mode 2. Photo 4 is that of a regenerated picture by a customer use NTSC receiver. You can see the upper and lower portions of the screen are in grey and high-band auxiliary data interference is slight. Photo 5 shows an EDTV-regenerated picture. In mode 2, the picture quality is the same as that of the first-generation EDTV system or slightly better. Photo 6 shows a partially enlarged picture of Photo 5.

It can be noted that the vertical and horizontal resolution is improved by the auxiliary information transmitted over the upper and lower panels.

CONCLUSION

The NTSC-compatible wide-aspect EDTV system, which is being developed by Nippon Television Network Co., is based on the techniques of the first-generation EDTV system in Japan, and it is intended to further improve the picture quality of the NTSC system.

Mode 1 and 2 which we introduced here are the basic systems of the second-generation EDTV of the future, and complex signal processing such as signal multiplexing within the image area or spectrum expansion are not employed.

Therefore, the systems have high feasibily from the standpoints of device cost, signal transmission and compatibillty with the current system,

Table 3. Hierarchy of NATV (NTV's advanced TV)

Transmition Channel	Layer		Improvement technique	Level
1 + 0.5 (9MHz) **	7	More audio channel	Audio : PCM 4ch	Planning
	6	Enforcement of the picture quality	Chrominance signal bandwidth: expansion Horizontal resolution : expansion Vertical resolution : expansion	Planning
1.0 (6MHz) *	5	Enforcement of the audio quality	Audio : PCM 2ch	Planning
	4	Enforcement of the picture quality	Horizontal resolution: 850lines per wide picture width Vertical resolution : 480lines(moving picture 360lines) Chrominance signal improvement : Y3(chrominance phase adaptive)	Planning
	3	Compensation for the regradation of the picture quality by the wide aspect processing	Horizontal resolution : 640lines per wide picture width(moving picture 440lines) Vertical resolution : 480lines(moving picture 360lines) Chrominance signal improvement : Y3	Computer simulation & construction of experimental hardware
	2		Wide aspect processing(16:9)	
	1	EDTV-I	Higher resolution processing at the camera end(P1) Quasi constant luminance processing(Y3) Adaptive emphasis(S1)	Under experimental broadcasting

* Viewing distance 4 H, Display size 40 inches

** Viewing distance 3 H, Display size 55 inches yet selection of high-performance and high-functional levels is possible as show in table 3 It is also possible to combine the functional levels selection of discussed here in this paper with other techniques.

Nippon Television Network Co. has been engaged in development of core technological components of the first-generation EDTV system and will develop an advanced system of high-quality picture TV for the future with promising technical feasibilty, social acceptance and improved cost and performance, based upon the knowhow and techniques gained from our past experience.

A system of better cost and performance can only be possible by realizing a high picture-quality television system which is in common with the NTSC world, and we recognize that it is important to promote our studies based upon this consideration.

END

Photo 1. NTSC picture of Mode 1, (narrow scan mode on studio monitor)

Photo 2. NTSC picture of Mode 1, (customer use NTSC receiver)

Photo 3. Wide aspect EDTV picture of Mode 1,

Photo 4. NTSC picture of Mode 2
(customer use NTSC receiver)

Photo 5. Wide aspect EDTV picture of Mode 2

Photo 6. Partially enlarged picture of Photo 5

CHROMA CRAWL AND CROSS COLOR FREE HIGH RESOLUTION NTSC

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ABSTRACT

Two NTSC compatible technologies will be discussed in this paper:

1. CCF, (Chroma Crawl Free)™; by slightly modifying a few NTSC parameters, some of the most objectionable NTSC artifacts- chroma crawl, and cross color- are eliminated. CCF has been field tested during a four month period on two major cable networks. (HRS patent)

2. High Resolution Television, based on SPM (Synchronous Path Modulation), increases vertical resolution and eliminates vertical aliasing within the 525 line NTSC system. SPM is a natural extension of NTSC towards High Definition Television. SPM has been successfully tested in the laboratory. (HRS patent)

INTRODUCTION

High Resolution Sciences Inc has been involved during the past six years in developing techniques for improving NTSC television images. While much of the television industry has focused on NTSC incompatible HDTV systems, we gave ourselves the task of improving the picture on the 140 million NTSC receivers in use today in the United States alone.

Our first technology introduced in the field eliminates two of the most disturbing NTSC artifacts, chroma crawl, and cross color moire. It is called Chroma Crawl Free (CCF)™ technology. CCF is a system fully compatible with NTSC receivers, but with slightly modified NTSC parameters. The CCF technology was first introduced to the Cable Industry in the Spring of 1988. CCF was field tested between September 1988 and December 1988 on two major cable networks reaching 25 million households.

Based on the results of this field test, the FCC has now given its permission and support for experimental broadcasting of the CCF signal over the air. Negotiations with different Networks, Affiliates and Independent broadcasters are underway.

It seems therefore very timely to familiarize the TV Broadcast Community with these technologies.

BACKGROUND

When the National Television System Committee over thirty years ago conceived the NTSC color television standards, they did so under numerous technical and compatibility constraints including a huge population of B&W TV receivers from the pre-CTV era and vacuum tube technology. Therefore, we have re-evaluated in today's environment some of the NTSC system parameters that cause the well known inherent artifacts such as chroma crawl, cross color, vertical aliasing and resolution limitations.

Many of the constraints which forced the less then optimal choices of trade-offs which led to today's NTSC standards do not apply anymore. We have found that by slightly modifying some of the system parameters, it is possible to eliminate or reduce many of these artifacts and to improve significantly the quality of the images reproduced on unmodified, existing NTSC receivers

This paper will discuss two technologies;

1: Chroma Crawl Free, (CCF), which eliminates chroma crawl, and reduces cross color moire, and
2: Synchronous Path Modulation (SPM), which increases vertical resolution and reduces vertical aliasing.

CHROMA CRAWL FREE (CCF) SYSTEM.

The CCF principle

The National Television System Committee carefully chose trade-offs when they selected the color sub-carrier and the horizontal and vertical scanning rates in order to minimize interference with the sound inter-carrier and generate the least objectionable sub-carrier pattern on the screen, while still maintaining full compatibility with the existing B&W receivers.

The interdependence between the sound inter-carrier, color sub-carrier and the horizontal and vertical scanning frequencies is illustrated in ("Figure 1")

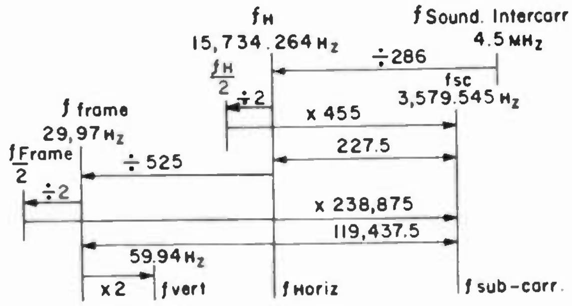

Figure. 1.

There are 227.5 color sub-carrier cycles in each line and 119,437.5 cycles in each frame. Therefore, the sub-carrier appears in each scanning line 180° out of phase from the preceding one.

There is a 180° shift from line to line ("Figure 2") in lines 1, 3 & 5 in the first field (solid line), or lines 2, 4 & 6 in the second field. (dashed-dotted line).

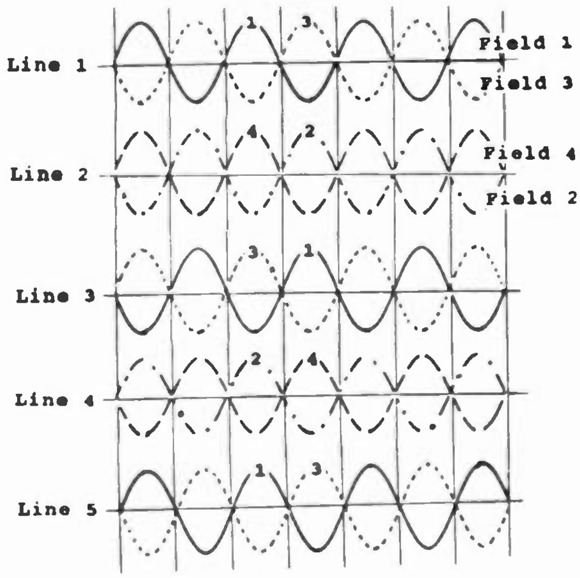

Figure 2.

The 180° shift from field to field can also be observed for instance between the last line in field 1. (solid line) and the first line in Field two (2nd scanning line). This inversion continues in Field 3, (dotted line) and Field 4 (dashed line). It takes four fields to accomplish an NTSC cycle, which introduces the very disturbing 15 Hz component.

In each consecutive field, the color sub-carrier will appear in the same phase, one line higher up. This causes the well known "Chroma Crawl" or "Dot Crawl" artifact. A better way to illustrate this is on ("Figure 3"), where the 4-3-2-1 pattern is more obvious.

4	2	4	2	4	2	4
3	1	3	1	3	1	3
2	4	2	4	2	4	2
1	3	1	3	1	3	1
4	2	4	2	4	2	4
3	1	3				

Figure 3

Now, if instead of containing 227.5 cycles, each line is shortened by half a cycle, so that a line contains only 227 subcarrier cycles, then the sub-carrier will remain in the same phase all the time. This would eliminate the 15 Hertz component and hence chroma crawl. However, if the sub-carrier is in phase in each line, it would produce a pattern of very objectionable vertical lines. This can be avoided if the length of the very last line in each field remains 227.5 cycles long. If so, the sub-carrier will appear from field to field in opposite phase. We have still a two-field cycle, completely free from chroma crawl and 15 Hz component, and the pattern of the vertical lines have been cancelled by the field-to-field inverted sub-carrier.

1		1		1		1
	2		2		2	
1		1		1		1
	2		2		2	
1		1		1		1

Field 1 = 3, Field 2 = 4.

Figure 4.

Furthermore, the interference between high frequency luminance components which pass through the chrominance filters and cause the cross color moire with the sub carrier will also cancel out. Since this happens at a rate of 30Hz rather then 15Hz, the cross color moire becomes practically invisible to the eye.

CCF Sync. Rate.

An NTSC sync generator can easily be modified to achieve CCF parameters. The Sub-carrier to Horizontal counter must be changed by a preset value, to divide by 454 during all but the last line of each field, when the preset value is being switched back to divide by 455 during that one last line in the field. ("Figure 5.")

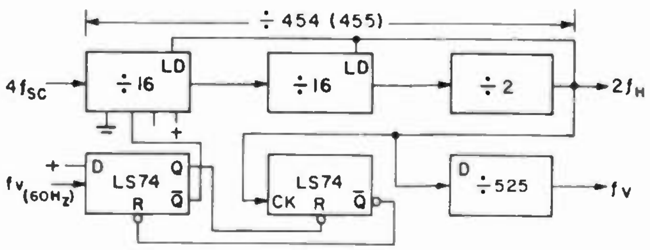

Figure 5.

A video camera, or any other original source of video signal gen-locked to such sync. generator, will produce chroma crawl and cross color free images.

NTSC to CCF conversion.

Most frequently, the video signal has been originated in equipment locked to standard NTSC sync. generators. In such cases there is the need for converting an NTSC signal into CCF. This poses two problems:

1: to minimize the chroma crawl and cross color inherent artifacts already present in the NTSC video signal,

2: to accommodate the time differential between the NTSC and CCF locked video signals.

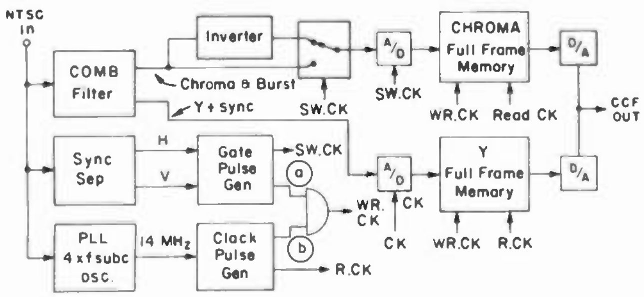

Figure 6.

The converter shown ("Figure 6") will provide these two functions. By separating Chrominance from Luminance in an effective comb filter, inherent NTSC artifacts can be substantially reduced.

Scince CCF line is 140 n.sec shorter then an NTSC line, there is consequently a continuously increasing time gap between the original and converted signal. To bridge this gap, the digitized NTSC chrominance and luminance signals are written into two full frame memories with a clock rate 4 times the color subcarrier.

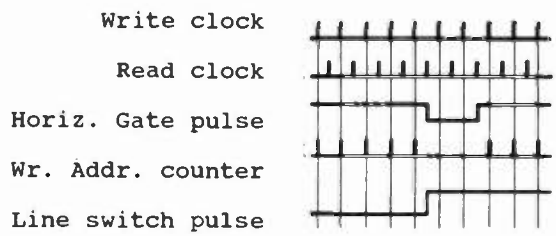

Figure 7.

At the end of each line, two write clock pulses are omitted. ("Figure 7.") When reading back the stored information, also at the same clock rate, the read-back line will be shorter by the equivalent of the two omitted write pulses or by 140 n.secs. As the reading back is faster then the writing, from time to time the reading catches up with the writing, and repeats reading the same field again. This event is normally unnoticeable to the human eye.

While cutting out 140 n.secs from the luminance signal has no significance, doing it to the chrominance signal reverses the phase of the color subcarrier from line to line.

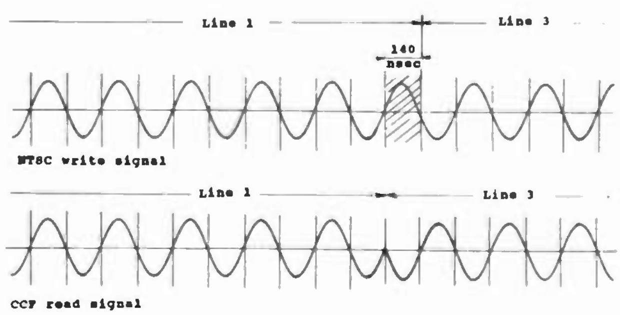

Figure 8.

To compensate for this, the polarity of the sub-carrier is inverted from line to line, prior to digitization. This way, the CCF chrominance signal after reading it from the storage and D to A converting, is continuous and in phase as the original signal was ("Figure 8.").

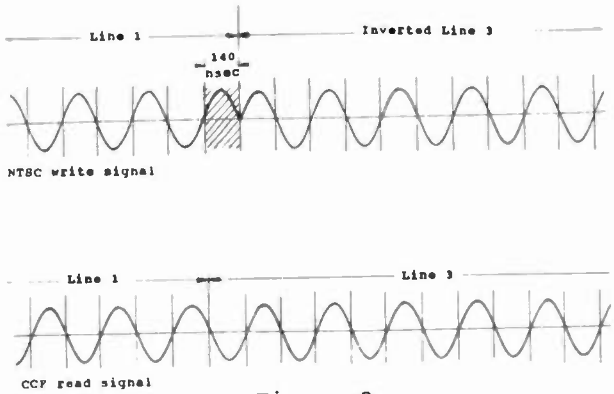

Figure 9.

However, by inverting the sub-carrier's phase from line to line changes the sub-carrier's phase in each second line relative the Luminance signal. This has no importance as long as the Chroma - Luma separation in the converter is effective.

However, if there is residual color sub-carrier in the luminance signal, this will be in phase with the new sub-carrier in each other line, and inverted in the lines in between. As a result a visible pattern of sub carrier will occur as the residual and new sub-carrier will ad up and cancel from line to line

To counter this ("Figure 10"), the Luminance signal is delayed each other line by 140 n.seconds. This way not only can the sub carrier be matched up with the luminance signal, but it can be kept always in inverted phase, causing the new sub-carrier always cancelling the residual old sub carrier:

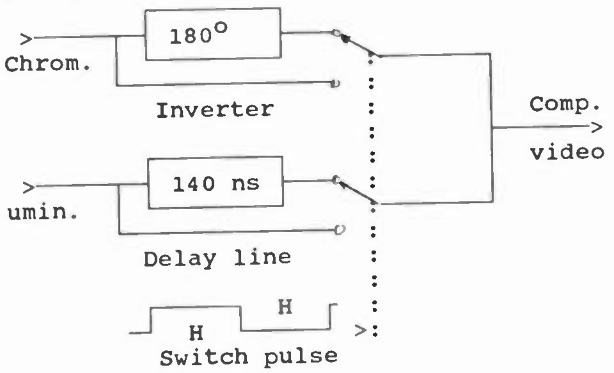

Figure 10.

There is also a more costly, but fully acceptable method which can be used by most standard converters for converting the NTSC signal into CCF, which is to demodulate the composite video signal to R-Y and B-Y and Y level, digitize the three video signals separately, write these into three separate full frame memories with the NTSC clock rate and read them back with CCF clock rate. After re-encoding, the composite video signal will be free of chroma crawl and cross color artifacts.

NTSC and CCF comparison.

In the following chart ("Figure 11") a comparison is given between NTSC and CCF. The color subcarrier frequency is the same for both systems, as well as the number of scanning lines. The horizontal and vertical frequencies are slightly different from NTSC. CCF also eliminates both chroma crawl and cross color.

	NTSC	CCF	DIFF.
Lines	525	525	SAME
Interlace	2:1	2:1	SAME
H. freq.	15,734.26	15,768.79	+0.21%
V. freq.	59.94	60.07	+0.21%
CS/c freq	3,579,545	3,579,545	SAME
cyc/line	227.5	227	-.021%
Chr.crawl	YES	NO	BETTER
cross col	YES	REDUCED	BETTER

Figure 11.

Chroma crawl free, without time shift.

CCF. II. is a method which eliminates chroma crawl without the CCF I time shift. By adding 140 nano sec to the first line of each other frame, and by shortening the other frames by the same amount of time, the average H and V frequencies will remain unchanged. Yet the sync to sub-carrier positioning will change to create the following raster: ("Figure 12")

1		1		1
2		2		2
	1		1	
	2		2	
1		1		1

Field 1 = 3, Field 2 = 4

Figure 12.

This pattern has no 15 Hz component and consequently no chroma crawl. However, as the sub-carrier in the adjacent fields is in phase, there is no cancellation effect of the cross color artifact.

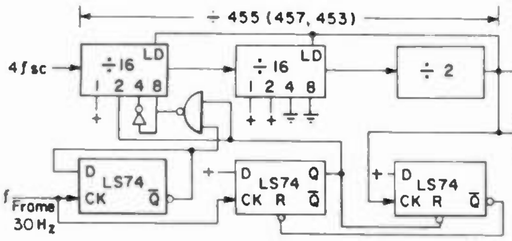

Figure 13.

Fig. 13. shows how a counter giving the CCF II raster, can be realized.

SYNCHRONOUS PATH MODULATION (SPM)

Background.

NTSC compatibility of a high resolution television video signal requires that a standard channel contain a full NTSC signal, while the additional high resolution information is transmitted in a way so that unmodified NTSC receivers are unaffected. It is expected that the NTSC image so received will be at least equal to a standard NTSC picture.

The same signal on a high resolution receiver will produce an improved image.

The horizontal resolution of a TV image is proportional to the video signals' bandwidth. (assuming unchanged number of lines and frame rate.). The vertical resolution depends on the number of horizontal scanning lines.

It is relatively easy to increase the video bandwidth to increase horizontal resolution, because the NTSC receiver will be "blind" to signals outside its video bandwidth.

To increase vertical resolution in a compatible way is much more difficult. Increased number of scanning lines are not readily compatible with NTSC receivers. A typical method is to generate a video signal at the camera with more then 525 lines, and then derive from this through digital interpolation a 525 line signal. This will then be the compatible NTSC signal. The difference signal carrying the horizontal and vertical high resolution information is transmitted to NTSC receivers by an "invisible" means, multiplexed into, or outside of the NTSC compatible signal to the high resolution receivers.

The Synchronous Path Modulation (SPM) offers a much simpler and more effective solution to generate a High Resolution signal. Without changing the number of scanning lines, it directly adds the additional horizontal as well as vertical information onto the video signal. The additional vertical information will be carried in additional bandwidth.

If, for instance, the speed of the scanning spot in the camera is doubled, it will pick up twice as many pixels during the 63.5 usec horizontal line, compared to the normal scan speed. However, instead of scanning two lines in that time period, its path is undulated around the original position. This way areas between the 525 lines will be scanned during the same 63.5 micro seconds.

An NTSC receiver will see such a signal as standard NTSC. The additional pixels (vertical and horizontal) will be above the receiver's bandwidth and will be filtered out.

A High Resolution Receiver using the SPM technology will process the additional information, and if the scanning path in the CRT is synchronously copying that in the camera, it will reproduce the high resolution image by replacing each pixel in its original position.

The principle is illustrated ("Figure 14") on a square wave modulated path. This will scan areas between the lines that were not reached before.

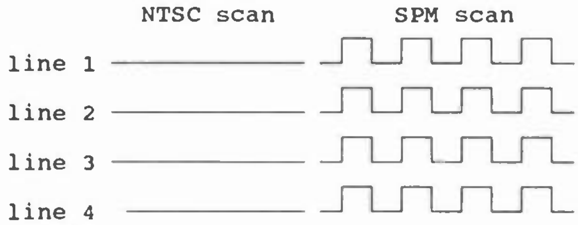

Figure 14.

A square wave form is not very practical for many reasons which will be discussed later.

An HRT receiver will have a similar scanning path, that is, instead of straight lines it will reproduce the shape of the scan lines in the camera, including the SPM deflection . The camera and receiver are phase locked to each other. Figure 15 shows a principal block diagram. The transmitted video signal includes a reference signal. As the color sub-carrier is already available in the receiver, it is most practical to use a sub-carrier derivation to synchronize the path modulator signal.

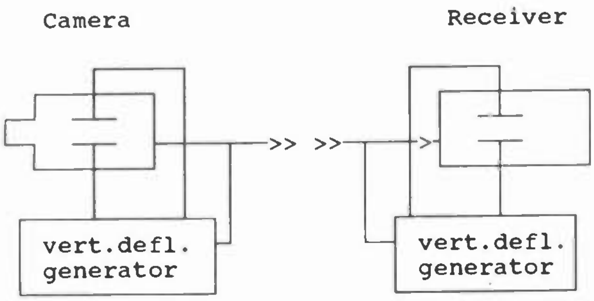

Figure 15.

```
         /\  /\  /\  /\  /\  /\  /\  /\
Line 1.    \/  \/  \/  \/  \/  \/  \/
         /\  /\  /\  /\  /\  /\  /\  /\
Line 2.    \/  \/  \/  \/  \/  \/  \/
         /\  /\  /\  /\  /\  /\  /\  /\
Line 3.    \/  \/  \/  \/  \/  \/  \/
         /\  /\  /\  /\  /\  /\  /\  /\
Line 4.    \/  \/  \/  \/  \/  \/  \/
```

Figure 16.

An NTSC receiver receiving this signal, not having vertical path modulating deflection, will ignore the vertical displacement of the pixels, and will simply project all pixels into one line.

The path modulating frequency

In the case of the square wave signal, the video bandwidth is unchanged (as long as the SPM frequency is within the video band). The additional vertical resolution has been achieved at the cost of the horizontal resolution, as now only half as many pixels in each line have been sampled.

Assuming, that the highest video component was 4.2 MHz, then if the SPM frequency is, for example 8.4 MHz, then, in case of square wave SPM, the original horizontal resolution has been re-established, while the vertical resolution has been doubled. It would require 4 x 4.2 Mhz square wave SPM frequency to double both horizontal and vertical resolution.

As mentioned earlier, it facilitates synchronization if the sub-carrier's second or third harmonic is chosen for path modulation. Laboratory tests indicate that the second harmonic is a good compromise for resolution and bandwidth efficiency.

Wave shape and amplitude

The shape of the SPM waveform will greatly determine the distribution of the gain in resolution in horizontal and vertical direction. The square wave was an extreme case since it has no vertical component. Therefore the total length of the scanning line is the same as that of a straight line. It is obvious that the longer the scanning line is, the more pixels it will pick up during its travel from side to side of the screen. Depending on the shape of its undulated path, these pixels will be vertically or horizontally displaced, increasing the vertical or horizontal resolution. ("Figure 16" illustrates this theory):

Using a sine wave ("Figure 17.") as an SPM signal has some advantages and some drawbacks. The major advantages are that it has no harmonics, which makes electromagnetic deflection easier, and reduces the risks of RFI and interference with different components of the television frequency spectrum.

However, the sine wave is a non linear function which will unevenly distribute the gain in horizontal and vertical resolution. At the peaks it will primarily contribute to vertical resolution.

The amplitude of the synchronous path modulation will, together with the wave shape, determine the contribution to horizontal and vertical resolution.

As mentioned before, the longer the scanning line, the more gain in overall resolution. However, there is an optimal amplitude. When the wobble sine wave crosses the original horizontal line at its inflexion point under an angle of 45^{o}, its contribution to horizontal and vertical resolution is about equal. At a given bandwidth, higher amplitude will contribute more to vertical resolution, and less to horizontal resolution. ("Figure 17").

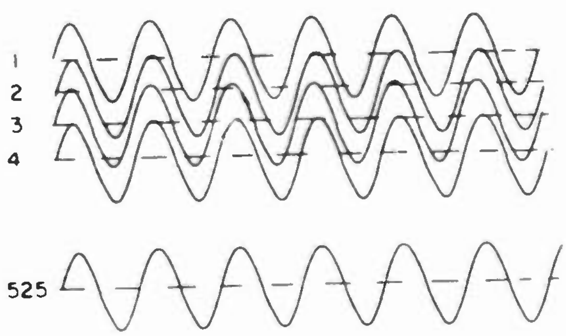

Figure 17.

The quality of the reproduced image will to a great extent depend on how accurately the shape of the path modulation in the camera is reproduced in the receiver, and how uniform the amplitude is over the entire screen.

Regardless of the shape and frequency, the scanning SPM must be in the same phase in every line and avoid crossing scanning lines, which would cause unnecessary redundancy and uneven distribution of luminosity. This is automatically achieved if double the sub-carrier frequency is used, as every NTSC line will contain 455 cycles of double sub-carrier frequency.

Vertical Aliasing.

Vertical aliasing is caused by a stroboscopic effect between the scanning lines and quasi horizontal elements in the picture (Venetian blinds), because the transition from line to line is only vaguely determined. The interlace adds a further flickering to this effect.

SPM practically eliminates this artifact, because the path deflection sine wave crosses the picture line under a more open angle, which gives better defined intersections. It has been found that even standard NTSC receivers show reduced aliasing with signals generated in SPM cameras.

Conclusions.

The CCF technology has been extensively field tested for long periods of time. It has been found to be very effective in eliminating the permanent jitter and flicker, which are always present in NTSC pictures, and it has shown significant improvement especially with programs normally rich in cross color moire (small details in picture), and chroma crawl, (saturated colors and letters).

Eliminating the sub-carrier related artifacts also makes it possible to improve horizontal resolution by reducing video band filtering, a practice which has been self imposed in many receivers to mask the NTSC artifacts.

High Resolution Television with the SPM technique has proven very promising in the laboratory by eliminating vertical aliasing and doubling vertical resolution. Compatibility of the SPM has also been proven. Practical implementation of uniform modulation in cameras and CRTs is in progress right now.

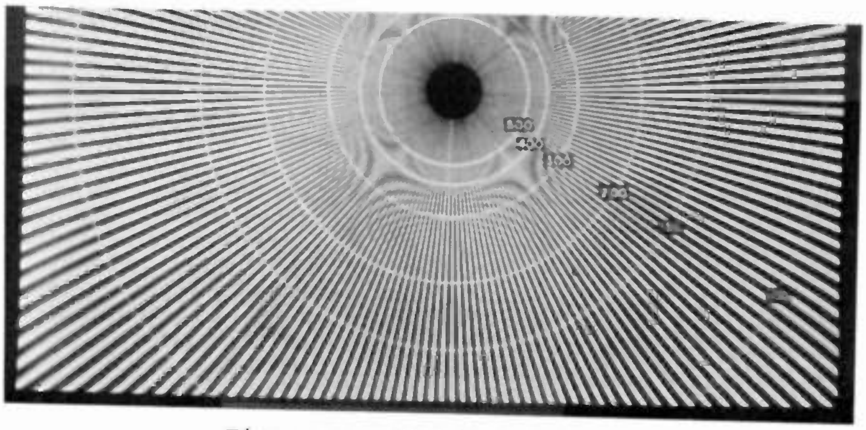

Figure 18. Standard NTSC

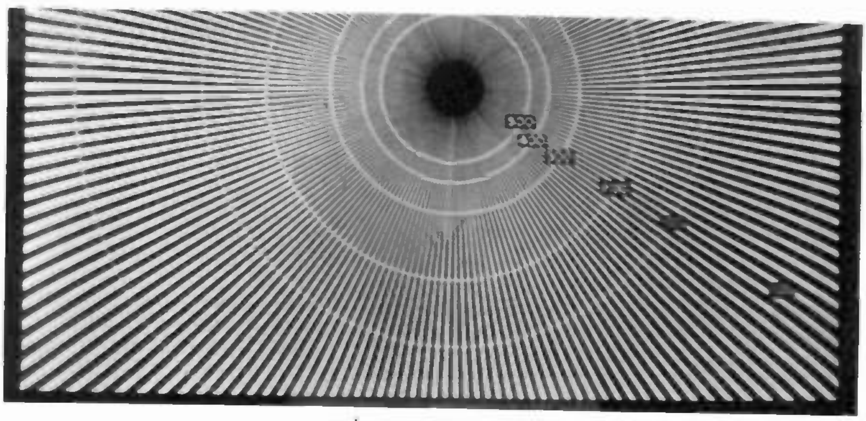

Figure 19. CCF

A FRIENDLY FAMILY OF TRANSMISSION STANDARDS FOR ALL MEDIA AND ALL FRAME RATES

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Abstract

This proposal is for a universal system of high-definition television *transmission* standards. The system is capable of accommodating all existing and a wide range of proposed formats. It allows each medium to configure its transmission format for optimum performance, considering the physical properties of the available channels. At the same time, it features exceptionally easy and inexpensive transcoding among widely varying transmission formats, requiring temporal interpolation only at the output of production systems and in receivers. It thus resolves the conflict between the need for exchange of programs among the various media and the desire of each medium not to be restricted in quality by the limitations of other media.

In the proposed transmission method, signals are divided into components, the data being grouped into 'packages' nominally 1/12 second long. Each component can therefore be thought of as a low-resolution, 12-fps progressively scanned picture. Components are combined to provide arbitrarily high spatial resolution as well as up to 60 progressively scanned (or even higher if desired) frames per second. The number and signal-to-noise ratio of the components varies from medium to medium; transcoding merely requires adding, deleting, and repacking components to achieve the highest quality given the physical characteristics of each medium. By a small variation of the duration of programs, "12" frames/sec becomes an integral submultiple of the frame rates of all TV and motion-picture systems used worldwide, thus simplifying the required temporal interpolation when converting to the display format.

The desirable characteristics of production, program-exchange, and display formats are discussed, and it is concluded that they should all be different. Thus, there are *four* different classes of formats that should be considered. The need for international uniformity of production and display formats is shown to be much less than for program-exchange and transmission formats. Current NTSC and PAL receivers must be served by today's signals. New receivers will have the capability of decoding a prescribed range of transmission formats, which, for a number of important reasons, will be decoupled from the display format.

Sections 1 and 2 are intended for all readers; the later sections are for TV specialists.

1. Background

Conventional TV systems use a simple raster scan and a video signal representing the point-by-point brightness of the input image, a scheme dating back at least to Bain's 1839 facsimile machine. (Similar — in some cases, actually more sophisticated — schemes were used even earlier to represent weaving patterns.) The key point is that, at the present time, camera, display, and video signal all use the same scanning standards in any one TV system, and that different standards are used in different countries. Still more standards are proposed for HDTV, and conversion from one to the other has become a serious problem involving both cost and quality, not to mention acrimonious debate. The preference for one standard or the other is neither entirely whimsical nor motivated solely by protectionist sentiments — there are real costs and potential quality problems associated with transcoding, particularly if it requires temporal interpolation.

Since 24-fps film plays such a central role in programming in all television systems, the friendliness of the TV system to film is very important. Systems that can make a one-to-one frame-to-frame conversion between film and television require less expensive equipment and get better motion rendition. This is the fortunate situation in the 50-Hz countries, which play film at 25 fps when used on TV. (Films made especially for TV are shot at 25 fps.) It is not surprising that Europe has resisted any 60-Hz production system, since that would entail giving up this important advantage without gaining any comparable benefit.

Motion rendition at 24 fps is not very good, but does not seem to be a source of dissatisfaction to today's viewers, even though they are exposed to much better motion rendition on video-originated subject matter. The 3-2 pulldown method gives even worse motion rendition than seen in the movie house. In principle, motion-compensated frame-rate conversion could give good motion when converting 24 fps to 60 fields/sec, but this has not yet been done commercially, and is likely to be more expensive than 3-2 pulldown.

The basic reason why transcoding is so difficult at present is because the video signals currently in use (and proposed for use in the NHK wideband system) are of the same directly displayable type as used by Bain so long ago. Transcoding such signals requires spatial and temporal interpolation, and the latter entails a tradeoff between smoothness of motion and sharpness of moving objects. While motion-compensated transcoding between PAL and NTSC has become technically quite successful, the transcoders are very expensive. Since they will never be made in very large quantities, they are unlikely ever to be cheap. This holds also for 1125/30/60[1] to PAL transcoding. On the other hand, cheap transcoders can readily be envisaged for receivers to be made in the millions.

From this argument, it is seen that easy transcoding is one of the most important characteristics of a transmission format, even within a single country. Although this paper is concerned primarily with transmission formats, careful consideration of such formats shows that their requirements are related to the particular uses to which they are applied. Clearly, other formats, such as those used for production, for international program exchange, and for display, are applied to quite different uses and therefore have quite different requirements. The program-exchange format is the only one that needs international agreement.

2. The General Idea

We first distinguish between directly displayable (DD) and sequential-component (SC) formats. The DD format is what is used today, in which the scanning standard is suitable for directly driving a display. Normal camera output is in the DD form. At present, NTSC, PAL, and SECAM are the DD formats in use, and many other DD formats have been proposed for HDTV. An SC format is one in which the various components are transmitted in time sequence, and must at least have their sequence altered and their time bases adjusted for display. MUSE is in SC format and so are most MAC systems. Temporal interpolation is often required to go from SC to DD format. In the SC formats herein proposed for transmission for new TV systems, the signal comprises data *representing*, but not necessarily simply related to, the original optical image on the focal plane of the camera. Furthermore, this data is transmitted in units ("frames") nominally 1/12 second in duration, ranging from 4/50 to 5/59.94 seconds. The significance of the 1/12 sec period is that it corresponds to a whole number of fields in all the systems under consideration. This is one of the elements that facilitates transcoding, since it eliminates the need for temporal interpolation. Each SC format includes a digital component. A large range of SC formats is permissible, with the relevant parameters indicated by a header word in the digital data. For convenience, a small number of the most common formats may be indicated by a very short designation.

An example of an SC format useful in this method is that of the MIT CC system; another is the Zenith Spectrum-Compatible System. In both of these systems, the signal is divided into a number of spatiotemporal frequency components, and these components are transmitted sequentially. Digital data representing audio, the lowest-frequency component of the video, including all or part of chrominance, and some additional auxiliary information is also multiplexed. To display this data requires separating out the components, interpolating each to the display line and frame rate, and combining them to form a DD signal. A typical transmission format is shown in Fig. 1. In this scheme, two baseband signals of equal bandwidth are used, derived from adjacent scan lines to quadrature modulate a single carrier in the center of the channel.

In the different media, it may be desirable to repack the components to get the best quality taking account of special properties of the transmission channel. For example, in over-the-air transmission in today's taboo channels, Zenith has proposed sending the digital data during the vertical blanking intervals of neighboring NTSC stations. In cable applications, interference is not a problem, and higher CNR[2] may be counted on than in terrestrial transmission. Therefore, digital data can be transmitted continuously by the 'data-under' method,[3] which permits much more digital data to be transmitted and which makes 100% of the transmission time available for analog information. It is easy to see that transcoding between these two SC formats requires only that the individual components be extracted and repacked. This requires multiplexing and perhaps time expansion or contraction, but not temporal interpolation. Of course, one signal may have more components and/or data and/or SNR than the other. In that case, some components must be dropped when transcoding from the higher- to the lower-capacity format, and some components may be absent when transcoding in the opposite direction. In both cases, the quality is limited by the information content of the poorer signal, but both signals are in precisely the correct format for further transmission.

It is seen from this example that the key to easy transcoding is that it is only *transmission* formats of the SC type that are involved, and not *display* formats. In transcoding, the signal is never put into DD form. Indeed, since the display format is independent of the transmission format, it is not necessary even to consider the display format during the transcoding process. Each 1/12 sec frame of information is rearranged into another 1/12 sec frame without any temporal interpolation. All that is needed is a frame store together with circuitry required to separate the components and to recombine them for retransmission.

The various components that comprise each 1/12-second data package can be thought of as independent still pictures. In transcoding, if the resolution of the components is changed, 2-dimensional interpolation must be performed. This may require reassembling the full image and again dividing it into components. Examples will be given of various schemes, some of which do, and some of which do not, require this operation.

3. Data-Packing Methods

In the SC format, various components are transmitted sequentially. In analog channels, such as terrestrial broadcasting, cable, satellite, and VCR's, most of these components are analog and some are digital. In digital fiber,[4] all components are digital. Unrelated digital and analog components can be combined for simultaneous transmission using the "data-under" method. Depending on the CNR and the required SNR for the analog signal, the digital data may be from one to four bits/sample. Two analog components can also be combined in similar fashion. One of the signals is coarsely quantized and the other reduced in amplitude so as to fit within one quantization step, and then added to the first signal. [1] This permits the CNR of the channel to be divided between two signals. When adaptively modulated, the highs components require only a very low CNR, so that it is quite practical to transmit two signals in a single channel at typical CNR's.

By these methods, the appropriate SNR can be achieved for each analog component while, at the same time, a substantial amount of digital information can also be transmitted. Of course, if desired, digital data can be transmitted only in one part of the frame, with the number of bits/sample chosen in view of the expected CNR. In the Zenith system, digital data is transmitted only during the vertical blanking interval of nearby NTSC stations so as to minimize interference. Where this kind of interference is not a a problem, digital data can be transmitted "under" all or most of the analog components.

In FM transmission, the appropriate SNR for each component can be achieved by using, in addition to the previous methods, appropriate time expansion/contraction. This operation changes the bandwidth, and hence the SNR, as a result of the relationship between modulation index and receiver SNR.

To maximize performance in imperfect analog channels, scrambling and/or adaptive modulation can be used on some or all of the analog components. [2] Adaptive modulation can be applied to any spatiotemporal component that is small in blank image areas. That involves multiplying the signal by a slowly-varying adaptation factor at the encoder and dividing it by the same factor at the receiver, thus reducing the effect of channel noise. Adaptation information is digitally transmitted. Scrambling involves transmitting the picture elements (pels) of each frame of a particular component in pseudorandom order. This results in dispersing the effect of most analog channel defects as random noise. When combined with adaptive modulation, the result is a ghost- and interference-free picture with vary large noise reduction in the blank areas where noise would otherwise be most visible. It is the use of these techniques that makes possible combining two components in one signal and still producing good pictures under typical channel conditions.

4. Some Examples Using Spatial Interpolation

In these examples, we shall allow spatial, but not temporal, interpolation during transcoding. This permits a wide choice of parameters for the components and makes it quite easy to devise systems optimized for various channel characteristics. However, it results in a more expensive implementation, since, in general, the entire luminance image must be synthesized into a single video signal and then redivided into components at each transcoding point. Although the process is straightforward, it loses the simple mapping between components and also may involve some quality loss in the filtering operations.

All the examples that follow are solely for the purposes of illustration. Other combinations of parameters are possible and may well be found superior with more experience in configuring the basic format for the different media, and as the properties of the media are better understood.

4.1 An Example of a Digital Transmission System

We first discuss a digital system since it is so easy to apportion channel capacity to each component. In Fig. 2, we have selected components to give a roughly diamond-shaped overall spatiotemporal frequency response, and we have assigned the SNR in inverse relationship to frequency. Thus the dc component has the highest SNR (59 dB, corresponding to 8 bits/sample) while the higher frequencies have the lowest SNR (23 dB, corresponding to 2 bits/sample.) We have taken two cases, 45 Mb/s and 90 Mb/s, in each case reserving 5 Mb/s for audio, adaptation data, and miscellaneous data. With equal horizontal and vertical resolution at an aspect ratio of 16:9, this gives 600x1064 and 872x1548 at 12 frames/sec for the two cases. Higher temporal resolution is provided at lower spatial frequencies, with chrominance rendered at 12 fps.

Note that this digital scheme is essentially uncoded. The high resolution achieved at moderate data rates is due entirely to using an efficient digital representation of the visual information. Naturally, statistical coding can be applied to get even higher efficiency. There is some possibility that a very simple-minded nonstatistical DPCM system using only temporal prediction would give higher SNR, particularly in the higher spatial frequencies of the stationary image areas, but that is not the topic of this paper, which is concerned primarily with transcoding issues.

In a digital system, the order in which the various components are transmitted within each 1/12 second "frame" is not important, as long as it is known. In all likelihood, error correction would be used. Note that conversion from one video format to another is very easy as long as each component is identified and no temporal interpolation is required. It would be quite feasible to have a very-high-resolution program-exchange format that had more components, or higher resolution in individual components. Transcoding would still be simple as long as the 1/12 second duration for each package of data were preserved.

4.2 An Example of A Cable Transmission System

In cable systems, all signals are of the same amplitude, and therefore all channels can be used. Receivers do not have to discriminate against adjacent-channel signals much stronger than the desired signal. Furthermore, there is no fringe area; all subscribers are guaranteed a certain minimum CNR, which we take here to be 36 dB. There is no need to bunch the digital data into the NTSC vertical retrace interval. We therefore use the "data-under" method in which each analog highs component is added to a multilevel "digital" signal, the number of bits per sample, and thus the number of levels, being chosen in accord with the required SNR for the added highs component. The digital data represents RGB lows, audio, adaptation data, and miscellaneous data. The highs components are adaptively modulated and scrambled before being added to the digital data.

Note that this "cable" system is also suitable for over-the-air transmission in those applications where it is not necessary to minimize interference to nearby NTSC stations to the maximum possible degree. This would be the case in an ultimate high-efficiency system used by all broadcasters after the phase-out of NTSC. In broadcasting, the CNR would not be defined as precisely as in cable, and it may therefore be advisable to include a system at the receiver for discarding analog components when their SNR becomes too low.

In Fig. 3, we show the number of bits/sample used in the data-under channel for each component. Eleven components have 3 b/pel, six have 2 b/pel, and 3 have 1 b/pel, for a total of 20 highs components, giving 48 bits for each sample of a typical component. Since the 6-MHz channel is nominally equivalent to 12 Msamples/sec, each component is 50,000 pels, giving a resolution of 166x300. The maximum possible data rate of the data-under channel is 166x300x48x12 = 28.8 Mb/s, and the maximum data required for RGB lows is 166x300x3x8x12 = 14.4 Mb/s. Both of these figures are too high. As a practical matter, it would be easier to transmit at a lower digital rate, and to do some moderate nonstatistical coding on the RGB lows.

The system has a resolution of 664x1200 at 12 fps. Although their spatial resolutions are not exactly the same, it is readily seen that there is a one-to-one correspondence between components in the digital system and those in the hybrid cable system, so that transcoding remains quite simple.

For cable or over-the-air transmission, double-sideband quadrature modulation of a single carrier by two signals of equal bandwidth is the most efficient method of transmission, since the basebandwidth is then equal to the rf bandwidth. This is the arrangement used in Fig. 1. To minimize defects caused by carrier phase errors, the two signals should be derived from vertically adjacent image points, i.e., from adjacent scan lines. If scrambling is used, the line pairs should be identically scrambled. If transcoding involves vertical interpolation, each component should be reassembled from its two halves beforehand.

4.3 A Zenith-Type Over-the-Air Broadcasting System

At 36 dB CNR, the standard deviation of noise is 1/64 of the peak value. Theoretically, this is equivalent to 6 bits/sample, but 4 bits/sample is more practical. In a 6-MHz channel, this is, theoretically, equivalent to 12 Msamples/sec, for a peak data rate of 48 Mb/sec. If the digital data is confined to the vertical blanking interval, assumed to be 8% of the period, then the average rate is 3.84 Mb/sec. If we arbitrarily reduce this rate by half to account for even lower CNR and/or for somewhat less than 6 MHz effective bandwidth[5] then about 1.9 Mb/sec is available. Assigning .5 Mb/sec each to adaptation data and to audio seems reasonable, leaving .9 Mb/sec for digital lows. Almost any simple DPCM system would give adequate quality with 2 b/sample for each of two chrominance components and 3 b/sample for luminance, for a total of 7 bits/sample. At 12 fps, this gives a digital lows component 77 pels high and 138 pels wide. Using a more powerful coding method, such as vector coding, the resolution of the digital components could be improved.

We could utilize 92% of the transmission time for the analog components, for an equivalent average rate of 11 Msamples/sec. Using the same scheme as that of the cable system, there would be 23 components (three more are needed to get adequate resolution for the color lows) giving a resolution for each component 150 pels high by 266 pels wide, as shown in Fig. 4. The overall resolution at 12 fps is 600x1064.

Note that in this scheme, the SNR of all the analog components is the same, which is not efficient. Even with a CNR of 36 dB, the SNR of the highest-frequency components is excessive. It may be possible to transmit some additional digital data "under" these components, or even to put additional analog components "over" them to get even higher spatial resolution. All such arrangements could be accommodated with equal ease and with no change in the overall scheme. Of course, whatever packing method is used must be unambiguously indicated in a header.

For transmission in an NTSC environment, the digital data is transmitted in bursts of 1.33 msec every 1/59.94 seconds. To minimize interference with PAL transmissions, the digital data would be in bursts of 1.6 msec every 1/50 sec. In that case, the nominal 1/11.988 second interval becomes 1/12.5 seconds instead. For nonreal-time transcoding, the 4% difference can be accommodated by changing the program duration (e.g., by running a VTR faster or slower) as in done in Europe with movies. For real-time transcoding, temporal interpolation is required as now used in PAL/NTSC conversion.

4.4 An FM Transmission System

Most TV programs, at some point, are transmitted in a satellite transponder channel using frequency modulation. In relay service, from point of origin to local television station or cable head end, large receiving antennas are used and the quality is very high. In DBS service, the emphasis is on small receiving antennas. Because of the rather sharp FM threshold, however, the CNR at the FM demodulator must be high enough to guarantee good-quality reception, with very little impulse noise.

FM has a "triangular" noise spectrum, so that the noise rises with frequency. This is quite desirable for the luminance signal, but undesirable for the color components if impressed on a subcarrier as in NTSC. In any event, since it is a one-dimensional signal that is modulated, the desirable frequency distribution of noise holds only in the horizontal direction. In the vertical direction, the noise is uniform. Subband coding can therefore be used with FM in order to achieve an overall noise distribution that is best from the perceptual viewpoint.

If a signal is divided into n subbands to be transmitted sequentially, then if each is time-compressed by the same factor n, all have the same SNR. By apportioning the relative time compression appropriately among the components, their relative SNR can be adjusted. Using a resolution of 150x266, each component has 40,000 pels, for an average transmission rate of 40,000x12 = 480,000 pels/sec. We elect to use the 45 components shown in Fig. 5. Note that 6 are devoted to data (audio, adaptation data, and dc component) and that we have, in this case, extended the chrominance temporal bandwidth to 24 fps. We assume a CNR of 15 dB and an rf bandwidth of 27 MHz. To achieve a difference of about 6 dB in SNR in adjacent components, we use time-compression factors as shown. The modulation index was computed using Carlson's rule and the SNR shown is therefore extremely conservative. Experience shows that much higher deviation can be used for the higher-frequency components and that we can expect an additional improvement in SNR in the blank areas of up to 24 dB by the use of an adaptive modulation index.[6]

It is believed that performance with these values will be excellent, although a simulation must be performed to be sure. With adaptive modulation, actual performance depends on image statistics. If the SNR is found to be too low, then the resolution or the number of components must be reduced; if higher than needed, which is likely in this case, then resolution or number of components can be increased.

Although not described here in detail, a format for magnetic tape recording would be similar to that for FM transmission by satellite. In both cases, the inherent CNR is low, the channel bandwidth available is fairly high, and the signal level is subject to substantial variation.

5. Some Examples Requiring Neither Spatial Nor Temporal Interpolation

If the components in all the various transmission formats have the same spatial resolution, no interpolation of any kind is required in transcoding. The formats would differ as to which components are digital and which are analog, and as to the number of components and their SNR. Synchronization methods might differ as well as the techniques used to avoid mutual interference with other transmissions. For the sake of this discussion, we shall assume that the resolution of all components is 150x266. Thus the 45-Mb/sec digital system, the Zenith-type system, and the FM system as described above are already of the right form.

5.1 A 90-Mb/sec Digital System

In Section 4.1, an increase in capacity from 45 to 90 Mb/sec was utilized by increasing the spatial resolution of each component but keeping the number of components the same. A simpler method of transcoding is possible by keeping the resolution of each component the same and increasing the number and SNR of the components. While this method is harder to think about, it is easier to implement in hardware since each component can be transcoded separately just by repacking. The components do not have to be recombined and reseparated with the synthesis and analysis filter banks that are required when the resolutions are different.

At a resolution of 150x266, there are 40,000 samples in each component. We previously reserved 5 Mb/s for audio and data, leaving 40 Mb/s for analog components. In 1/12 sec, this gives 84 components times bits/sample, for the distribution shown in Fig. 2, where there are 23 components ranging from 2 to 8 bits/sample. If, for a 90 Mb/sec rate, we reserve 10 Mb/s for audio and data, we have 168 components times bits/sample. Using the distribution of channel capacity shown in Fig. 6, we have raised the spatial resolution in the stationary areas to 750x1330 and have raised the spatial resolution at the higher temporal frequencies correspondingly. We have also raised the SNR of many of the components. It is safe to say that the picture quality of this configuration will be very high indeed.

5.2 A Cable System

For cable or for over-the-air use in an all-HDTV environment, where there is no need to take special precautions to avoid interference with nearby NTSC stations, the system shown in Fig. 7 can be used. In this case, we use the 150x266 component resolution, while the system of Section 4.2 and Fig. 3 uses the 166x300 resolution. At a data rate of 12 Ms/sec and with 40,000 samples per component every 1/12 sec, 25 analog components are permitted. In Fig. 7, we have 41 components, of which 6 are digital, leaving 35. Of these, 10 are doubled up, leaving 25. Digital data is transmitted 'under' all the components that are not doubled up, for a rate of 18.24 Mb/sec, which is on the optimistic side. Using DPCM for the digital components with the number of bits/sample as indicated in Fig. 7, we require 9.12 Mb/sec plus audio and data.

The net result of all this is that the resolution and SNR are about the same as the arrangement of Fig. 3. Experience may dictate some change in the total number of components in the interest of a more or less conservative design. As in the other examples, the numbers are for illustrative purposes only.

6. Production and Program-exchange Standards

From the previous discussion, we can see that it is possible to design very effective transmission systems once the requirements are clearly stated. In this case, the main requirements are bandwidth efficiency and easy, defect-free transcoding. These requirements can be met by the use of subband coding and adaptive modulation, together with appropriate choice of the resolution of the components. For good interference protection and encryption, we can add scrambling.[7]

By applying similar principles to production and international program-exchange standards, we quickly come to views that challenge the conventional wisdom. For example, we are now seeing a determined effort to get the 1125/30/60 system adopted as an international standard for these purposes. A little examination will show that the two applications — production and exchange — have quite different requirements and therefore the two kinds of formats should probably be different. A production system should readily be produced by a camera with appropriate performance, including spatial and temporal frequency response and sensitivity. The production signal should be suitable for post-production, including all kinds of special effects. It is quite clear[8] that interlace is a highly undesirable property of a production signal. In view of the fact that *all* HDTV transmission systems currently proposed discard diagonal resolution in three-dimensional frequency space, there is no need to include it in the production signal. This greatly eases camera design, in that the bandwidth can be reduced and the sensitivity thereby increased, by eliminating these components at the source. This can be done by offset (quincunx) sampling or by using separate camera tubes for the low-spatial, high-temporal components and the high-spatial, low-temporal components. The elimination of interlace in tube-type cameras will also raise the vertical resolution substantially, because of the equilibrium-discharge phenomenon coupled with the Gaussian beam shape. This is important, since no current interlaced TV camera has vertical resolution nearly good enough for 1000-line images.

While a simple video signal, much like that required to drive a display, is most appropriate for post-production, we have a totally different situation with respect to international exchange of programs. Whereas the product of the post-production process must be converted to the exchange format once only, the latter must be converted to the transmission format used in each medium repeatedly, at each point where the program is to be used. Hence transcodability is the most important characteristic of the exchange standard, just as it is for the transmission formats that we have discussed above. Furthermore, the exchange standard should as easily be converted to a transmission standard in the 50-Hz countries as in the 60-Hz countries.

Although bandwidth efficiency and interference performance are much less important for exchange standards than for transmission formats, a version of the SC format proposed here appears to be highly suitable. The spatial and temporal resolution and the SNR can be as high as desired by adding more components and/or by raising their SNR. This can be done as readily for digital, baseband analog, or FM versions of the format. By a small adjustment in time duration, 24, 50, 59.94, and 60 Hz systems can all be accommodated. It goes without saying that transcoding from this 'exchange' standard (which is actually a super-transmission format) into any of the likely transmission formats is very easy, entailing no temporal interpolation, and even no spatial interpolation if desired.

With this choice of exchange standard, the only places in the entire TV chain where temporal interpolation is required, from studio to the home, is after post-production and in the receiver. The cost of the former is implicit in all existing proposals to separate studio from transmission standards, and can readily be absorbed as part of the cost of production. Note that this has to be done once only. The cost of interpolation in the receiver can be low because receivers will be made in very large quantities and the necessary special chips will become practical and economical.

If a scheme like this were adopted, it is clear that international agreement would be required only for the exchange format. If desired, that could be in the form of rather flexible specification of components and packing methods, allowing for a range of quality levels with very little extra trouble. It is also clear that, except for the convenience of manufacturers, no international agreement at all is required for production standards.

7. Transcoding Between the Transmission Format and Camera or Display Formats

For displayable camera formats at 59.94 fps (nominally 60 fps) or at 50 Hz, the transmission components at 12 fps are first found by prefiltering and subsampling. For reconversion to the displayable format, the 12-fps components are temporally up-converted by integral factors of 5 or 4, as appropriate. Quadrature-mirror filter banks are the best implementation of the two operations at present, although there is some possibility that superior transient response can be obtained with other filters at the cost of some reduction in resolution. The particular filters used range from simple decimation and replication at one extreme to ideal low-pass filters on the other. To some extent, the filters implement a tradeoff between motion smoothness and the sharpness of moving objects.

When the input signal is from 24-fps film, there is no point using transmission components higher than 24 fps. In that case, a different selection of components is made, facilitated here because of the nominal 1/12 second 'frame' period. Higher spatial resolution is attained. Motion-compensated interpolation will eventually permit excellent motion rendition from film.

When transcoding to and from interlaced display formats, it is probably easier to interpolate in two steps, dealing with interlace/progressive conversion in one step and with the 12/60 or 12.5/50 fps conversion in a second step. It should be noted that a receiver-compatible EDTV signal such as ACTV-1 can be generated from one of the HDTV or EDTV transmission formats rather easily. Sufficient transmission components are used to achieve at least the desired EDTV receiver performance, and these are interpolated to 60 fps, progressively scanned. This signal is divided into the NTSC resolution image and the enhancement information. Vertical-temporal filters followed by vertical-temporal subsampling to get the interlaced format follows, and then the enhancement information is hidden within the NTSC signal or prepared for transmission in an augmentation channel as required by the particular system.

Conclusion

We have presented a scheme for configuring a range of transmission formats that feature very easy, defect-free transcoding between formats optimized for use in the various media. The considerations used in the analysis are applied to production and international-exchange formats, and it is concluded that the latter two have entirely different requirements. A production format should be progressively scanned and should discard diagonal components at the source. An exchange format can be a version of the transmission format described in this paper.

Acknowledgement

This work was supported by the members of the Center for Advanced Television Systems as well as by additional funding from Home Box Office. The specific proposals made in this paper are an outgrowth of the work previously done in designing both compatible and noncompatible HDTV systems for use in cable and terrestrial broadcasting in the United States.

References

1. W.F.Schreiber, A.B.Lippman, et al, "A Compatible High-Definition Television System Using the Noise-Margin Method," SMPTE San Francisco, 4 February 1989. MIT Report ATRP-T-99.

2. W.F.Schreiber and A.B.Lippman, "Reliable EDTV/HDTV Transmission in Low-Quality Analog Channels," SMPTE New York, 16 October 1989. ATRP-T-96R.

[1] In this paper, a/b/c designates a TV system with a lines/frame, b frames/sec and c fields/sec.

[2] We distinguish here between the carrier-to-noise ratio (CNR) of a signal in a channel and the signal-to-noise ratio (SNR) of a video signal that is to be directly displayed.

[3] This method entails superimposing a reduced-amplitude analog signal on top of an unrelated multilevel (typically 2- or 4-level) digital signal.

[4] It is also feasible to use analog transmission in fiber-optic cables.

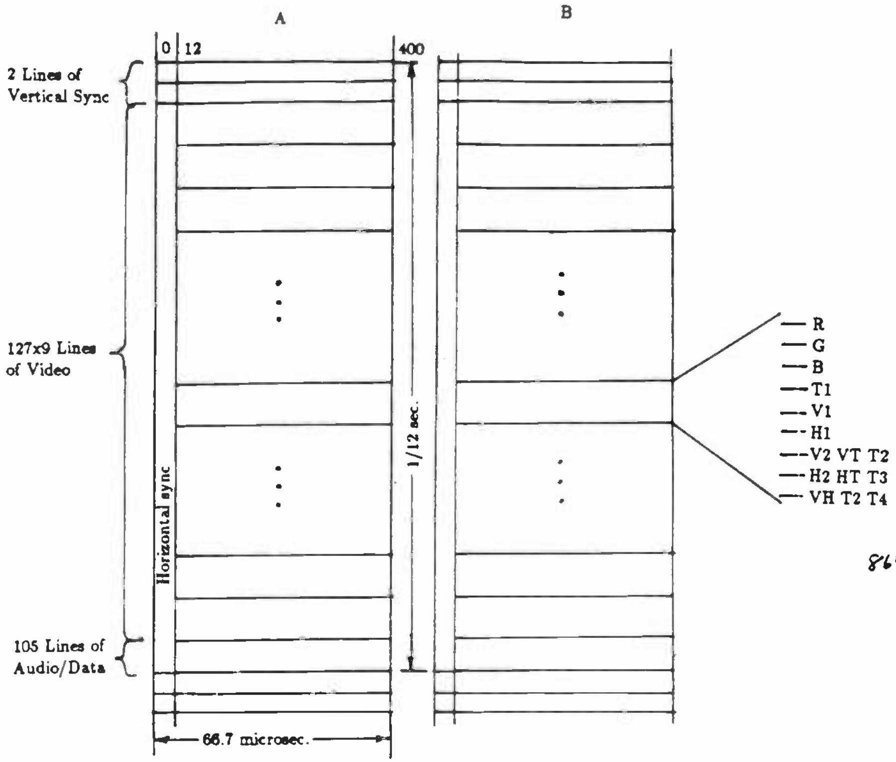
[5] Where adaptive modulation and scrambling are used, the interference characteristics are very good. In that case, there is no need to have very sharp-cutting filters to define the channels, and some overlap can be used. It is quite likely, in that case, that signaling at the full 12 Msamples/sec with a channel spacing of 6 MHz will be shown to be quite practical.

[6] These calculations were performed by Julien Piot.

[7] Note that scrambling can be used in any of the transmission formats. In transcoding, the scrambling can be kept, changed, or eliminated, as desired.

[8] At least to me. Everyone does not agree about this.

Figure 1. *Typical Transmission Format.* This shows how the data is packed for transmission in the MIT-CC system. Two 3-MHz analog signals are quadrature modulated onto a single carrier. Each signal comprises intervals 1/12 sec. in length, consisting of spatiotemporal components interleaved with sync and data.



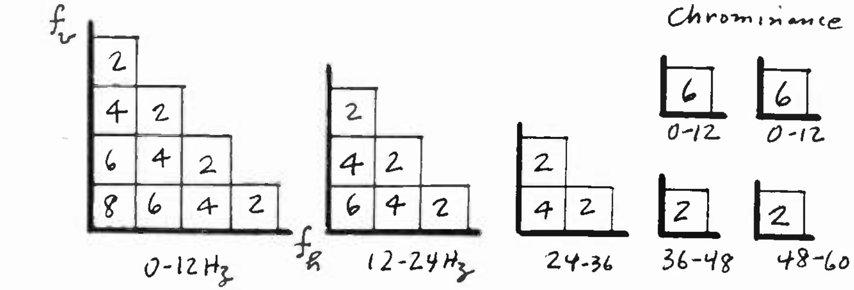


Figure 2. *Components in a Digital System.* At 45 Mb/s, each component is 150x266 pels; at 90 Mb/s, each is 218x387. The number in each component indicates the number of bits/pel assigned. See also Fig. 6.

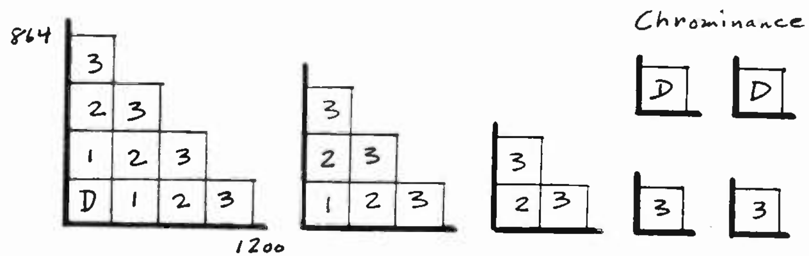


Figure 3. *Components in a Cable System.* This system uses "digital under" in which every component except RGB lows is transmitted in analog form with a digital signal underneath. The number in each component indicates the number of bits/pel in the "under" signal. D indicates that the component is transmitted digitally. In this version, the resolution of each component is 166x300. See also Fig. 7.

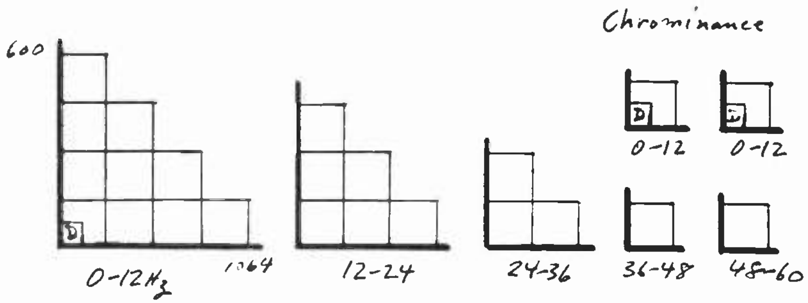


Figure 4. *A Zenith-Type Over-the-Air System.* In this system, all analog components are 150x266. In addition, there is a digital RGB signal 77x138. Digital data is transmitted in bursts synchronous with the vertical retrace interval of interfering conventional signals, either NTSC or PAL. See text at Section 4.3.

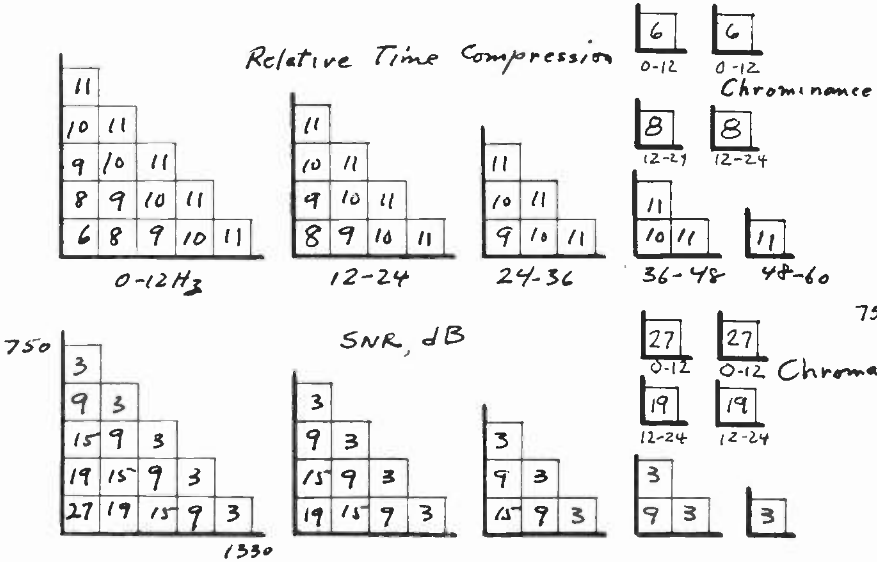


Figure 5. *An FM System.* There are 45 components, of which 6 are devoted to lows, audio, and data. The highs components are 150x266. The numbers in the boxes represent the relative time compression factors and the SNR for the respective components. The higher-frequency components are compressed more and therefore have a lower SNR. This permits a lower compression and higher SNR for the lower-frequency components. See text at Section 4.4.

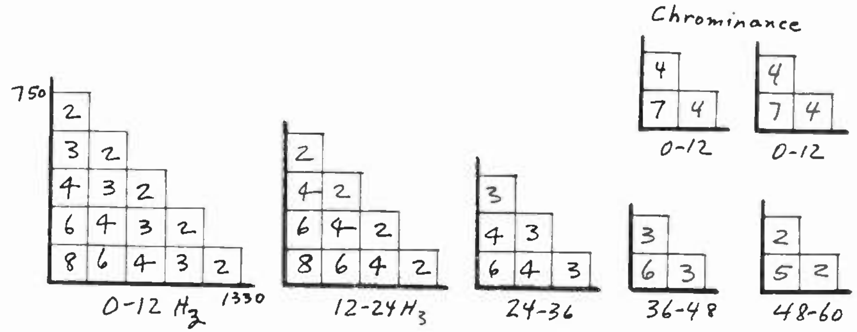


Figure 6. *A 90-Mb/sec Digital System.* This very high-quality system is configured to use the same 150x266 resolution as most of the others, for the sake of the easiest possible transcoding. Numbers show the bits/pel assigned. See also Fig. 2.

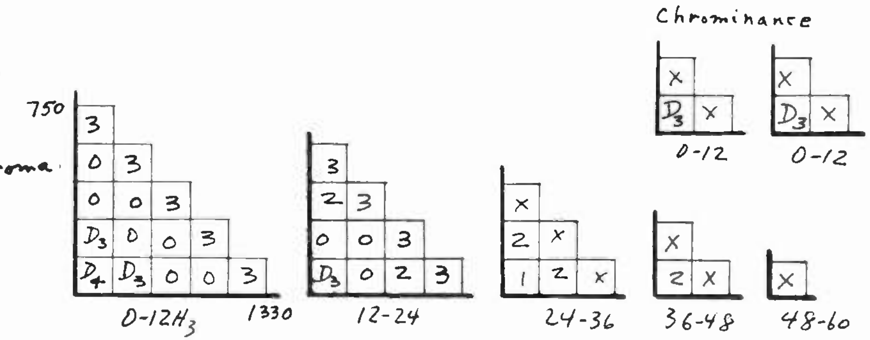


Figure 7. *A Cable System with 150x266 Components.* Like the system of Fig. 6, this one is also configured for maximum ease of transcoding. A number in a box indicates the number of bits/pel 'under' the analog signal. The character "x" means a component is hidden in a lower-frequency component marked "o." D_n means digital transmission (DPCM) at n bits/sample.

APPLICATIONS OF HIGH SPEED LOCAL AREA NETWORKS IN THE BROADCAST ENVIRONMENT

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ABSTRACT

Recent increases in the amount of broadcast equipment which is computer based has increased the broadcasters needs for control and data distribution systems. In order to meet these demands, Local Area Networks, or LANs are being utilized. These LANs, which are already in common use in the computer industry, make possible highly powerful and versatile control and data distribution systems.

This paper discusses LANs, their application in the broadcast environment, and strategies for successful LAN implementation. Specific areas of discussion include machine control, routing switcher control, station automation, and data distribution.

INTRODUCTION

The use of computers in broadcast equipment has increased dramatically over the past five years. Nearly all broadcast equipment being sold today is computer based. For example, nearly all modern VTRs are controlled by an RS-422 serial line. Most video effects devices and production switchers are controlled in a similar manner.

The large number of digitally controlled devices can present substantial problems in a large broadcast installation. The biggest problem presented is the interconnection of the control signals for all of these various devices. In many cases, it is sufficient to simply connect a cable from device A to device B. However, in most cases, it is necessary for the devices to be connected in different ways at different times.

For example, it is often necessary to control a VTR or a video effects device from studio A in the morning and from studio B in the afternoon. In these cases it is necessary to switch the control signals from each device in a similar way to which a routing switcher switches audio and video signals.

Current Methods

Sometimes a device similar to a routing switcher called a data switcher is used to connect all of these various signals. This device uses relays or solid state devices to physically connect one data device to another. However, data switchers have several disadvantages. The main disadvantage is that they can only connect devices which share a common protocol. In other words, they can only connect devices which would communicate properly if they were directly connected together by a wire. A second disadvantage of a data switcher is cost. They usually cost more than other alternatives. There are applications where a data switcher is the best approach. However, their use should be carefully considered and used only where really necessary.

AN INTRODUCTION TO THE LAN

The local area network, or LAN, can provide solutions to many of the problems discussed above. A LAN is a network of computing devices communicating with each other at very high data rates. LANs are characterized by three basic traits. They are:

1. Bandwidth or data rate.
2. Physical Configuration.
3. Traffic arbitration or protocols.

LAN Data Rates

Typical LAN data rates range from 1 million bits per second to 1 billion bits per second. The most common speed is 10 Mbits per second which is the speed used by Ethernet and all of the networks related to it.

Physical Configurations

LANs are used in one of two basic physical configurations. One is the star configuration shown below:

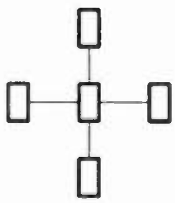

Figure 1.

In this configuration, the device in the center is called a hub. Each device must have a dedicated connection to the hub. All messages are transmitted through the hub to another device.

The second configuration is the inline configuration shown below:

Figure 2.

In this configuration, a single cable, usually called a backbone, is run throughout the length of the facility and is terminated on each end. The devices all connect to this backbone cable through a tap, which is usually referred to as a Medium Attachment Unit, or MAU.

The inline configuration is by far the most commonly used configuration because it is used by Ethernet and Cheapernet. However, both methods have advantages and disadvantages. In many cases the star configuration is easier to install in a broadcast environment.

Traffic Arbitration

The primary difference between a network and a simple serial data channel is that a network has many devices connected to it, any one of which can talk at a given time. This brings up the obvious difficulty of arbitrating whose turn it is to talk. Some LAN protocols rely on one master device who arbitrates the network. These LANs will not be discussed because they are generally unsuitable for a broadcast application. The bus arbitrator becomes a single point failure source of disastrous consequences.

LAN protocols which do not rely on a bus arbitrator are usually called peer to peer protocols because all devices on the LAN are equals and they can all talk to any other device.

In peer to peer networks, arbitration is usually accomplished in one of two ways, token passing and collision detection.

In a collision detection system, a device on the LAN can usually start transmitting whenever is has a message to send. However, before it can start transmitting, it must check the network to see if some other device is already using the network. If the network is in use, the device is must wait until the network is available before it can start it's transmission.

This alone solves most of the arbitration problems. However, there is still one condition which this approach does not deal with. This is the condition where two devices check the network for availability at exactly the same time, both find that it is available, and then both start transmitting at the same time. This is referred to as a collision. This condition is dealt with by hardware in each device which detects the fact that two devices are transmitting at the same time. When the device detects that a collision has occurred, it stops transmission, waits a random period of time, and then restarts transmission. The random period of time prevents the two devices involved in the collision from simply having another collision when they try to transmit again.

The token passing method of arbitration creates a special signal, or token, which is passed from device to device. Typically this token is a message which is sent from device one on the network to device two. If device two has something to transmit, it can do so as soon as it has received the token. If it does not, it passes the token on to device three. Thus the token is passed from device to device until one of them has reason to transmit.

Each of the two approaches has advantages. The collision detection approach is much more common because it is used by Ethernet. This approach is usually considered to be more reliable than token passing because the device which has the token at any time could be disconnected from the network or break down while it has the token. In some systems this can cause the network to completely quit operating. The token passing approach is usually considered better for networks which are extremely busy. When the overall traffic on the network exceeds 35% of the networks bandwidth, the collision detection system becomes inefficient because of excess numbers of collisions.

Along with these major characteristics, networks are also categorized by several minor factors, such as the type of cable and connectors they use, line impedances, and other factors.

ETHERNET / CHEAPERNET / IEEE 802.3

The LAN which is in the greatest use today is Ethernet. This network is a 10 Mbit/sec network which uses a 50 ohm coax as it's transmission media. It uses collision detection for bus arbitration and is physically configured using the inline configuration.

The transmitters are constant current devices which drive the line with 80 mA. Because the line is terminated at each end, this results in a DC load termination of 25 ohms. Since the drivers produce 80 mA, this results in a voltage of 2 volts on the line. If two transmitters are enabled at once, 160 mA will be produced, resulting in a voltage of four volts, which marks a collision.

The physical, electrical, and bus arbitration standards of Ethernet are described in IEEE standard 802.3.

Cheapernet is electrically identical to ethernet, except that it is carried in smaller 50 ohm coax, similar to RG-58, and it has lower isolation voltage requirements. Because of these differences, it is considerably cheaper then Ethernet, hence it's name. However, it is also more limited in distance, and it can run a maximum of 185 m. Because Cheapernet is so similar to Ethernet, one type of interface can easily be converted to another. They will therefore be discussed as one network throughout the rest of this paper.

Ethernet / Cheapernet is available in a very wide variety of computing devices. It is most heavily supported in the UNIX environment, but is also easily available for smaller machines such as PCs and for large scientific mainframes such as the DEC VAX family and the Hewlett Packard 800 family. It is also available on about 50% of the business computer systems in common use.

Networking Protocols

When Ethernet is used to connect two computing devices, a common set of software standards or protocols is needed. When IEEE defined the physical layer of Ethernet (IEEE 802.3), they also defined a software protocol to be used on this network. This standard is described in IEEE spec 802.2. However, this standard has not received wide recognition, and is not in common use. There is however a common set of protocols which are almost always used.

These protocols are the Internet Protocol (IP), the Transmission Control Protocol (TCP), and the User Datagram Protocol (UDP). These protocols were defined by the Department of Defense Advance Research and Project Agency (ARPA) and have become a widely accepted industry standard for networking.

The Internet Protocol (IP) is a protocol which provides several services to a user message. It's primary purpose is to allow delivery of a message, even if it has to be carried over several networks.

For example, consider the case of a large corporation with it's own nation wide computer network. A message may originate on a PC, be carried across a Cheapernet to a gateway, where it is transferred to an Ethernet, to be carried to the building mainframe. The mainframe would then route this message to a dedicated satellite link to be carried to a facility on the other coast. There it would be routed through a similar path to get to another PC. The Internet Protocol header would travel with the users message all the way through the various routes it may take to get from source to destination.

The Transmission Control Protocol (TCP) works along with the IP. It is used to provide segmentation, ensure ordered delivery, and arrange for retransmission of missing packets. For example, consider the example we mentioned above.

The PC on the West coast wants to send a 250 Kbyte file to the PC on the East coast. The maximum message length allowed by Ethernet is 1500 bytes. Therefore, the file must be transferred in at least 375 different messages. However, some of these messages may be lost by the satellite link due to noise problems. Messages can also get out of order if some are sent by satellite and some are sent by land line. The TCP on the transmitter splits the file up into acceptable size messages, and sends them to the receiver. The TCP on the receiver checks to make sure that it receives all of the packets, and requests retransmission of any which are missing. It also puts the packets back into the proper order if any are received out of sequence.

The User Datagram Protocol is a protocol intended to assist in sending messages directly from one device to another. Like the TCP, it works along with the IP. However, the UDP is much simpler than the TCP, and therefore, it is well suited to short messages which are not going to be segmented or routed between many networks. The UDP is a protocol which has many applications in the broadcast environment. For example, it is very well suited for delivering machine control commands to a VTR.

Higher Level Protocols

The three protocol mentioned above provide the underlying transport mechanism for a wide variety of higher level protocols. These standards have been defined by various organizations such as the UNIX divisions of AT&T, ARPA, and The University of California at Berkeley. These three organizations have been the leaders in networking standards, and have provided the industry with some excellent high level standards.

Examples include Network File Services (NFS). This software, which is available in PCs, UNIX machines and mainframes, provides a very sophisticated virtual file system capacity. It allows one computer to access files on a disk drive which is connected to another computer as easily as discs connected locally.

For example, this paper was produced on a word processor running on a PC using the MS-DOS operating system. However, the disk drive where the paper is stored is a 570 Mb hard disk connected to a UNIX mini-mainframe. The PC can access files on the mainframe as easily as it can access files on it's local disc.

Another example of a high level service is Berkeley Sockets, a system allowing a process running on one cpu to communicate with another process without even knowing which cpu the other process is running on.

BROADCAST APPLICATIONS

These high level protocols have made LAN usage, and specifically Ethernet, an integral part of most computing environments. How can we in the broadcast industry use these tools to solve the problems which we are faced with? Most of the applications in broadcasting do not involve the transfers of files or other large amount of data. Because of this, the low level protocols are the most useful in the broadcast environment. However, the high level protocols are extremely useful to us in a broadcast engineering environment.

Machine Control

Consider the data switcher we were discussing earlier in this paper. If we have a large number of serially controlled tape machines and several devices for controlling these tape machines, a data switcher could be used to connect machines to controlling devices. However, there would be some problems with this, the most important being that some of the machines probably use a different protocol than others. This is especially true if some of the machines are from different vendors. Therefore a system is needed which can not only carry the data from one device to another, but also translate it from one protocol to another. Modern microprocessor technology combined with a LAN provide an excellent solution to this problem. A device usually referred to as a gateway can translate the messages coming from the VTR or the controller to a common language, transmit that message on the LAN, and then another gateway can receive it from the LAN, translate it into the language of it's local device, and send it on to the local device.

For example, imagine a broadcast facility which has 8 Ampex VPR-3s, 8 Sony BVH-2500s, and 8 BTS D-1 machines. All three of these types of machine use a different RS-422 control protocol. This facility also has 12 machine control panels which use the ES-Bus protocol defined by the SMPTE and the EBU to control VTRs. Of course it is necessary to use any control panel with any machine at any time. How would such a control system be constructed? The following diagram shows one example of how this might be done. The diagram however, only shows one or two of each type of device, the rest would be connected in an identical way.

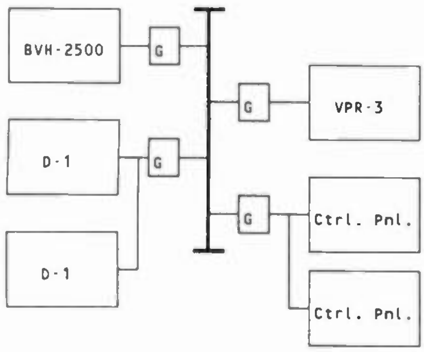

Figure 3.

As you can see from this diagram, all of the RS-422 lines from the devices connect to gateways, which in turn connect to the LAN. The gateways convert all of the different protocols into a common protocol for transmission on the LAN. In the case of the D1 machines and the Control panels, multiple machines can be connected to a single gateway. This is because they both communicate with ES Bus, which allows this type of connection. The most likely choice for the common protocol is ES bus. This protocol is not specific to any one vendor, and it is the most likely protocol to be supported in the future.

Now, to show the expandability of this concept, I would like to add some additional equipment to it. Suppose that this broadcast facility buys a Master Control Switcher and an automation System, both of which need to preroll these machines. If these devices are built to utilize a LAN, they can be connected as follows:

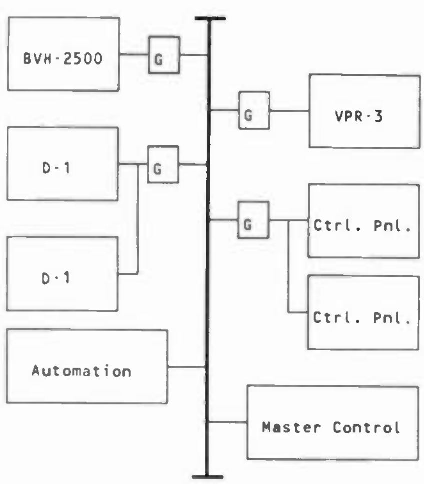

Figure 4.

The Automation system and the Master Control switcher can connect directly to the LAN. Because the control protocol used on the LAN is ES-Bus, the switcher and the automation system can communicate with the tape machines without regard for what type of machine they are. This allows equipment from many vendors to be easily integrated into a working system.

Of course, the gateways used to talk to the tape machines don't need to be limited to controlling one machine per gateway. The networking and computer hardware can be shared between several machines. Some gateways currently on the market will actually control 10 or more machines, depending on the protocol used by the machine and the capability of the system.

Routing Switcher Control

Once a system such as the one we have just discussed is installed for machine control, it makes sense to use it for other applications. One of the best applications of a system like this is for routing switcher control.

A routing switcher is typically installed in one central location and is controlled from many places. A LAN, running throughout a facility is very well suited to this. If we take the system shown in *Figure 4*, expand it to include routing switcher control, and use 10 channel gateways instead of one channel gateways, we arrive at the following system:

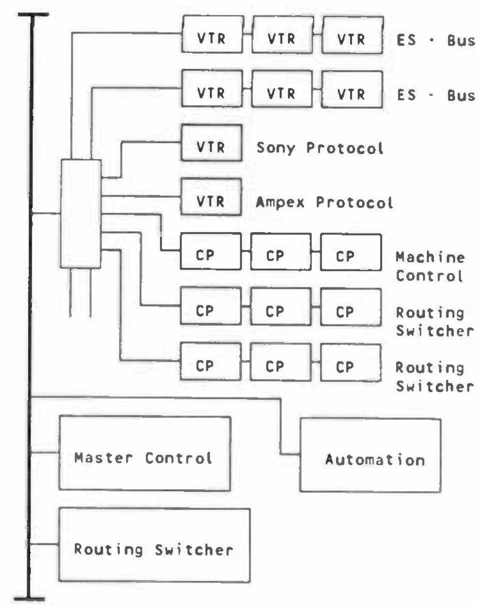

Figure 5.

Again, the routing switcher, like most other large pieces of equipment, connects directly to the LAN. The RS control panels and the machine control panels connect to the LAN through one channel of the gateway.

This configuration not only allows the automation system to control the tape machines, it also allows the automation system to control the routing switcher and the master control switcher. Of course, this concept can be expanded to many of the other parts of the broadcast facility as well.

For example, a system like this can be used to route signals between digital effects devices and the control panels that operate them. This allows a facility to install one digital effects device, install a control panel for it in each studio, and connect the desired control panel to the effects device.

This kind of a configuration also allows much more sophisticated routing switcher control systems then were previously possible. The speed and power of the LAN and the hardware which works with it make it possible to provide features which were previously not available

Configuration and Software Management

Obviously a system such as this with many gateways and other devices connected to the LAN can become quite complex. It is necessary to somehow tell the system how and where each device is connected and what type of device it is. Security considerations also need to be taken into account. Because of this, it is essential that a method be provided to easily configure the system.

Software Management is another important issue in a system like this. In many cases, each gateway will require slightly different software because of the complement of devices to which it is connected. However, software of this type is usually placed in EPROM memory, and is quite difficult to change or keep track of.

These issues could quickly make a system such as this totally unmanageable. Therefore a method must be found to solve these problems. Again, the LAN provides us with an excellent solution to these problems.

In the past, configuration information and program software was kept in EPROMs. This was done because the data needed to be in the devices at all times, even after a power failure. However, the data transfer speed of an LAN makes this unnecessary. Using the high speed of the LAN, it is possible to keep the program software and the configuration information in one central location and then download it over the LAN to the gateways and other devices.

The logical place to keep this data is on the hard disc of a central computer. A small multi-tasking computer such as a small UNIX machine is well suited to this task. In addition to storing and downloading data, the computer can serve as a central configuration terminal. By providing a central location where all configuration and software data is managed, the task of managing a network based system like the ones we have discussed becomes much easier.

The gateways and other network devices are then manufactured with very little EPROM and a lot of RAM. The EPROM which they do contain is merely a "boot" program which allows them to request their software from the LAN and load it into RAM. Once the software is loaded into RAM, it can be executed from there. The gateway can also be provided with a small internal battery to provide backup to the RAM in the event of short power failures.

Future Applications

As this discussion has been indicating, a properly designed LAN based control system can be versatile. When the gateways are based on RAM and not on EPROM, then it is simply a matter of putting a new disc in the central computer and downloading new program code into the gateway to make it do something previously not thought of. This makes a system like this very unlikely to become obsolete quickly.

As this sort of a control environment becomes wide spread, more and more equipment will be produced with Ethernet or Cheapernet ports on the rear panel. Even today, some VTRs currently on the market can be controlled directly by Cheapernet. This allows them to be connected directly to a system like this without having to use a gateway. As this becomes more commonplace, a system such as this will become even more useful.

FACILITY WIDE INTEGRATION

Many broadcast facilities are already using an Ethernet/Cheapernet type network. Often such a network is already in use by traffic, engineering, or data processing departments. There are many advantages to integrating a LAN based broadcast control system with other LAN based systems in the facility.

A LAN can be used to transfer schedules between a traffic and an automation. A billing department can gather information from the LAN to use when billing clients. All of these sorts of things are very practical and fairly easy to implement if a few precautions are taken.

1. The broadcast control system must be based on the Internet Protocol. If a control system does not use Internet Protocol, it will not only be impossible for broadcast devices to talk to other devices, it will probably not even be possible for them to coexist on the same network.

2. The physical installation of the LAN must be properly designed and implemented. The installation of a LAN system in a large facility can be moderately complex. In many ways it is like installing a master antenna system.

The best way to do it is to run an Ethernet or backbone to each major area of a facility. In a multi-story building it is usually best to run the backbone up an elevator shaft or a stairwell. Once the backbone is in place, devices called hubs can be used to tap off multiple Cheapernet runs to service the various devices.

A device called a bridge is available to isolate areas of heavy traffic from the rest of the network. This allows the devices within the area to communicate with each other without using up bandwidth on the facility wide network. However, they can still communicate with devices on the main network when they need to.

There are many experts available to assist with the installation of a network in a large facility. There is no need for outside assistance in a small network, but in a large, facility wide installation, it is usually best to seek expert assistance.

3. Deal with vendors who are willing to help. Often times it is difficult or impossible to connect two devices simply because one or both vendors won't give you information about how the device works. This situation can be avoided by dealing with vendors who express a willingness to work with you on projects such as this.

4. Try to purchase equipment based on industry standards whenever possible. It is always easier to work with a piece of broadcast equipment which uses an industry standard such as ES bus for it's communications protocol. It is also easier to work with a computer which uses a standard operating system such as UNIX or MS-DOS than one which uses a proprietary operating system. This problem becomes especially bad if the vendor won't tell you how the equipment operates.

5. Finally, don't be intimidated. The buzzwords and terms used in computer networking can scare people into thinking it's more difficult than it really is. It has been my experience that getting two devices to communicate over a network is considerably easier than getting them to communicate over a serial port. It just takes a while to get used to the environment.

CONCLUSION

The systems which can be built based on LANs have the potential to solve many of the problems faced by broadcasters. Modern computer technology has made it possible to build systems that would have been cost prohibitive just two years ago. For example, the 10 channel gateway I mentioned earlier can be installed for as little as $50.00 per VTR. This makes a system such as this quite affordable, even for the medium size or small broadcaster.

The power and flexibility of systems such this are unmatched. However, they could be much more powerful if there were more industry wide standards for data communications and LANs. We need to encourage the standards bodies which have been established to create the standards which we need. We must then encourage the vendors to follow them. By doing this, we can make a system which is already very powerful even more so.

THE EVOLUTION OF MICROCOMPUTER FILE TRANSFER PROTOCOLS

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NO MICRO IS AN ISLAND

The spread of personal microcomputers in broadcasting has heightened interest in reliable, fast, and convenient ways to transfer files over dialup phone lines. This paper discusses microcomputer file transfer protocols, how they evolved, and how they can affect your operations. You will know how to transfer data with a minimum of cost and a maximum of confidence.

XMODEM

Since its development in the late 1970s, the Ward Christensen **MODEM** protocol has allowed a wide variety of computers to interchange data. There is hardly a communications program that doesn't at least claim to support this protocol, now called **XMODEM**. Higher performance protocols were available, but microcomputer hackers with limited resources flocked to XMODEM because it was simple and easy to implement. XMODEM worked well transferring files at 300 "baud" over dial-up phone lines.

Changes in computing, modems and networking have spread the XMODEM protocol far beyond the environment in which it was designed.

XMODEM sends files in 128 byte blocks, waiting for an acknowledgement at each and every block. XMODEM's 90 per cent efficiency on 300 bps (bits per second) calls falls to twenty per cent or less on timesharing systems, high speed modems and packet switched networks.

XMODEM-1k

XMODEM was extended to 1024 byte data blocks in 1985 using a technique developed a few years previously by the *YAM* (Yet Another Modem) communications program. **XMODEM-1k** (sometimes erroneously called YMODEM) sends eight times as much data in a block, reducing XMODEM's protocol overhead by 87 per cent. The **JMODEM** protocol extends this to 8192 byte blocks, further reducing overhead under ideal conditions.

Unfortunately, these longer data blocks exact a heavy penalty when transmission errors from line noise or other problems require a retransmission. Some programs "fall back" to shorter block lengths when excessive block retries are required, but XMODEM technology does not allow the most efficient "fall back" to shorter blocks without compromising the safety of the file transfer.

XMODEM-CRC

The original XMODEM protocol used an 8 bit checksum to detect transmission errors. This proved inadequate, and today all but the most primitive communications programs support **XMODEM-CRC.** XMODEM-CRC replaces the 8 bit checksum with a 16 bit number which in not "fooled" into accepting corrupted data as often.

XMODEM Reliability

The CRC polynomial chosen for XMODEM is not the best (most reliable at error detection) possible 16 bit CRC. Unfortunately, this is not the weakest link in XMODEM's "chain" of operation. XMODEM uses single ASCII control characters to supervise the file transfer. These control characters were designed to be unique in the presence of the single bit errors typical of 300 bps modems, but they are easily fooled by errors common in today's modems. Proprietary Cybernetic Data Recovery™, logic enhancements in *Professional-YAM, ZCOMM,* and *DSZ* correct for some but not all of these weaknesses.

Kermit

Another problem with XMODEM arises from the very way it transfers data using all 256 combinations of 8 bits. Many mainframe computers cannot transmit or receive all 8 bit codes. Most have trouble with arbitrary control characters. Some require each input record to end in a carriage return or other special character. The **Kermit** protocol was developed at Columbia

University to allow file transfers under conditions that preclude the use of XMODEM techniques. Kermit uses a **quoting** technique to represent control characters and characters with the 8th bit set as multiple printing characters. For example, a hex 9D (¥ on IBM PCs) might be sent as #'[expressing one byte as three printable characters.

Properly implemented with optional reliability enhancements, Kermit transfers files reliably. Compression, Sliding Windows, and recently developed Long Packets have reduced Kermit's high overhead somewhat.

Kermit has two advantages over XMODEM besides higher reliability. Kermit transfers both the file name and the file contents, so the user does not have to type the file name twice, once to the sending program and once to the receiving program. Protocols that transfer file names are called **BATCH PROTOCOLS.**

Another advantage of Kermit over XMODEM is the ability to transfer file contents exactly, without adulterating the data with garbage added at the end of the file.

SERVER MODE supported by some Kermit programs allowed the calling program to request file transfers and other services without manual intervention.

The design of Kermit packets allows *Professional-YAM* and *ZCOMM* to recognize Kermit automatically, select the proper Kermit dialect (8 bit transfers or 8th bit quoting), and download files without extra keystrokes. Users of other programs sometimes fail to select the proper Kermit dialect, preventing file transfers.

YMODEM

The **YMODEM** protocol was developed in the early 1980s as part of the *YAM* program. Like Kermit, YMODEM provides batch transfers, allowing many files to be sent with one command. Minimal YMODEM sends the file name with each file. Full YMODEM sends the file length and date with the file name, allowing the receiver to reconstruct the file exactly. Full YMODEM also informs the receiver of the total number of files and the total length remaining to be sent, allowing the receiver to display an estimate of the remaining transmission time. Programs fully implementing the YMODEM specification may be certified by YMODEM's author as supporting **True YMODEM™** [1]

YMODEM uses XMODEM-CRC and XMODEM-1k technology. YMODEM shares XMODEM's reliability and performance limitations.

When completely error free links are available, **YMODEM-g** provides excellent throughput by eliminating the per block acknowledgements required to permit error recovery. YMODEM-g reduces protocol overhead to one half per cent, but a single transmission error will ruin an entire transfer.

Users of YMODEM and YMODEM-g have been hampered by some programs (Qmodem, Procomm, Crosstalk) whose authors did not care to follow the published YMODEM protocol specification.

X/YMODEM Variants

IMODEM, JMODEM, WXMODEM, Telink, SEAlink, and Meglaink are derived from XMODEM. Each improves on one or more aspects of XMODEM, and some of them are in common use to this day.

ZMODEM

In late 1985 Telenet was concerned by unsatisfactory results network users were having with XMODEM and Kermit file transfers. They funded the initial development of the **ZMODEM** file transfer protocol. ZMODEM is a language for implementing file transfer protocols, not just a single protocol good for a small set of applications. Since then, ZMODEM's speed, reliability and ease of use have made it the protocol of choice for more and more systems.

ZMODEM transfers files efficiently with buffered (error correcting) modems, timesharing systems, satellite relays, and wide area packet switched networks.

Optional ZMODEM compression provides even

1. Omen Technology Trademark

faster transfers on files with suitable data patterns. The speedup varies from 20 per cent for executable files to 1000 per cent for the *Personal Computing Magazine* ASCII test benchmark.

ZMODEM's 32 bit CRC protects against errors that continue to sneak into even the most advanced networks. Unlike XMODEM, YMODEM, JMODEM, et al., ZMODEM safeguards all data and supervisory information with effective error detection.

User Friendliness is an important ZMODEM feature. ZMODEM's design allows automatic file downloads initiated without user intervention. AutoDownload is a popular feature that greatly simplifies file transfers compared to traditional protocols. AutoDownload and security verified command download allow programs with integrated ZMODEM to operate in a server mode for reliable remote access.

Crash Recovery is one of ZMODEM's most popular features, allowing a transfer interrupted by a disconnect to be continued from were it left off. Crash Recovery is especially welcome when one trips over the phone cord just as a long file transfer is nearly finished.

ZMODEM provides security verified downloading and flexible control of selective file transfers. For example, one can transfer all files in a directory, skipping over files missing from the destination directory and skipping over files where the destination's copy is up to date.

CHOOSING A PROTOCOL

When attempting to transfer a file between two computers, one or the other of the communications programs may limit your choice to XMODEM or a poor Kermit implementation.

When a choice is available, ZMODEM provides unmatched reliability, convenience, and network compatibility while closely approaching the efficiency of YMODEM-g.

Many Kermit implementations provide reliable transfers for those willing to suffer Kermit's higher overhead.

The performance and reliability of XMODEM family protocol transfers varies greatly with the quality of the programs involved.

In continuation of a tradition made necessary by poor phone lines, Omen Technology conducts Protocol Stress tests from time to time. A special setup [2] generates moderately severe transmission errors under controlled conditions. In these tests ZMODEM and Kermit are the most reliable of the protocols tested. XMODEM and YMODEM with *Professional-YAM's* Cybernetic Data Recovery™, logic enhancements almost but not quite always succeed. Many XMODEM and YMODEM programs fail often; a few failed almost every time. Some even locked up one or both computers.

Typical File Transfer Speeds
(Without transmission errors)

Prot	ch/sec	Notes
X	53	CIS/Tymnet PTL 5/3/87
X	56	The Source PTL 5/29/87
SK	170	The Source PTL 5/29/87
ZM	221	CIS/Tymnet PTL
ZM	231	BIX/Tymnet 6/88
ZM	227	CIS PTL node
ZM	229	TeleGodzilla (local)
ZM	534	MNP Level 5 (text)
ZM	2561	pcbench.txt

Abbreviations:
K Kermit
SK Sliding Window Kermit
X XMODEM
ZM ZMODEM

2. Described in PTEST.DOC, available on TeleGodzilla BBS, 503-621-3746

ONLINE COMMUNICATIONS AND THE BROADCASTER

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Abstract

One major problem area for many broadcasters is the ability to communicate with each other and with manufactures between the yearly conventions and meetings. The Broadcast Professionals Area on CompuServe has been filling this need for over 5 years. The broadcast related areas on CompuServe have over 13,000 active participants from various segments of the industry. Participation from manufacturers and associations have been growing and making online communications an increasingly valuable asset to the broadcaster. The broadcast related areas on CompuServe include the Broadcast Professionals Forum (BPFORUM), the Journalism Forum (JFORUM) and the PR and Marketing forum (PRSIG), in addition to these broadcast specific areas, CompuServe also offers Weather, News, Stock market information and over 1300 other subject areas. CompuServe currently has over 500,000 active subscribers with over 500 local telephone numbers and worldwide availability via alternative communication networks.

Broadcast Related Areas on CompuServe

The Broadcast Professionals Forum is administered by John Hoffman, a broadcast engineer for over 25 years who has been with the National Broadcasting Company for the past 21 years.

Chris Hayes, Radio Engineer for KRLA/KLSX in Los Angles is the Radio Administrator.

John Reiser, Assistant Chief, Engineering Policy Branch, Mass Media Bureau, Federal Communications Commission, is the other main Associate.

Plus we have other personnel that are recognized for their expertise in specialized areas.

Broadcast Professionals Forum and InCue OnLine

The main interest areas that are available in the Broadcast Professionals Forum and InCue OnLine are:

- Television
- CATV/MMDS
- Technical - General
- Radio/TV Talent
- Strictly AUDIO
- SBENET (Society of Broadcast Engineers)
- AESNET (Audio Engineering Society)
- FCC Q & A
- Radio Technical
- Classified, Jobs
- Manufacturers Products and Support
- Cellular/LMR
- Radio & TV Promotions
- B E TECHNICAL Library
- INTERTEC Publications Preview

The Television area contains information on current television technology, concerns of the membership and discussions of the techniques used in various shows.

Our CATV area is moderated by Steve Johnson a Senior Project Engineer for ATC in Denver CO.

Jay Trachman, the publisher of "One to One" and a Radio Talent Consultant moderates our Radio Talent Area.

Jim Jordan, who is very active in live and recording audio moderates our "Strictly Audio" area.

We are also actively affiliated with both the Society of Broadcast Engineers and the Audio Engineering Society.

Facilities Available

The BPForum and InCue OnLine have multiple facilities available to support the broadcasting industry, a representative sample of these areas are:

- Special Announcement Areas for the immediate dissemination of information.
- Messaging areas for each of the above described areas, so that a person may read and reply to only those messages that apply to his/her specialty.
- Reference library areas for text files and programs related to our industry.
- Resumes for the perusal of prospective employers.
- Job postings and free-lance listings.
- Conference areas for public and private real time conversations.
- A special publication for on air radio talent written by Jay Trachman
- Publication previews and special articles provided through the facilities of Intertec Publications.
- The Broadcast Engineering Technical Library provided through the facilities of Broadcast Engineering magazine.

The Journalism Forum (JFORUM)

JIM CAMERON is President of Cameron Communications Inc., a New York City based consultancy specializing in radio news, program syndication, media training and public relations. He can be heard anchoring newscasts on the United Stations Networks several days a week.

DAN HAMILTON is Managing Editor of The Register, a small weekly newspaper in Yarmouth on Cape Cod, published continuously since 1836. He is a member of Investigative Reporters and Editors (IRE), Society of Newspaper Design (SND) and a former Porsche mechanic.

TONY RUSSOMANNO is a reporter for KGO-TV in San Francisco. Tony was also one of the four original filmmakers of the Oscar winning documentary "The Life and Times of Harvey Milk".

DAVE COHEN is the Midwest Bureau Chief for ABC News (radio and television). SCOTT HENRY is Associate Sysop for the Natl. Press Photog's Assoc.(NPPA) HOWARD FINBERG is Associate. Sysop [Society for Newspaper Design].

Facilities Available

- The Experts Index
- Stringers' File
- The Jobs File
- Journalism Tools
- Comment/Controversy
- Radio
- Television
- Print
- Photo - Video
- Ethics
- Freelance / Student
- Politics
- Features
- Soc. Newsp. Design
- DeskTop Publishing

PR and Marketing Forum

RON SOLBERG is the owner and operator of his own Chicago-area public relations and educational agency, EasyCom. The 1985-86 president of the Chicago Chapter/PRSA and 1985 chair of PRSA's national Communication Technologies Task Force.

HOWARD BENNER is president of OMNI Information Resources, Inc. DAVID COLMANS, Atlanta, GA, is Director of Communications, Management Science America Inc, (MSA), the nation's largest supplier of mainframe software applications.

DANIEL JANAL owns and operates Daniel S. Janal Public Relations, The High Technology Specialists, based in Fort Lee, NJ.

MICHAEL NAVER, is a writer and editor specializing in hi-tech for communicators. He is editor of the newsletter HI-TECH ALERT FOR THE PROFESSIONAL COMMUNICATOR.,

BILL WEYLOCK, is President of Weylock Associates, Inc., a New York marketing research and consulting firm with clients including leading public relations and advertising agencies and Fortune 1000 corporations.

Facilities Available

- On Your Own
- Resource Co-op/ebt
- Desktop Publishing
- Savtim
- Potpourri/products
- New Technology BooK
- Marketing/ama
- Pr Book
- High Tech Alert
- International
- Prl: Info/news
- PRL:Jobs Online
- PRL:Seminars/conf's
- Research
- PRL:Business
- Media Relations

Advantages of an on-line Resource

The main advantages of being associated with a national online service are the immediacy of information transfer that is available with this mode of communication and the quality of expertise that is available.

Those of us that are able to attend the major engineering conventions are able to talk directly to the various company representatives, see the new equipment available and ask questions about that equipment. The majority of individuals actually involved with the operations and maintenance of the equipment do not have the advantage of going to the conventions and seminars.

COMPLIANCE IN A DEREGULATED ENVIRONMENT

William M. Allison
Radio Management Systems
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DEREGULATION

A period of change

The decade of the 80's has been a trying one, even for the most savvy of broadcasters. Competition in the market-place along with advances in technology has meant increased investment in product, plant and people.

Remember the good old days of broad-casting when the morning show consisted of the morning person and a news person? Now we have the morning team. Remember the good old days of television when you could shop around for programming and make an offer? Now you have to bid on the product against someone who is bent on going bankrupt. Remember the good old days when you had to wait 3 to 5 years for the next new generation of remote control, or cart machine, or improved transmitter or video equipment?.

Today by the time you sign the purchase order, the piece of equipment is out of date. By the time it arrives, it is obsolete. Your only hope is that it falls off the truck, and the carrier has to replace it with the latest model.

Meanwhile, your sales people are out there standing in line, behind the other twenty or thirty sales reps trying to bring in some dough. By the time they get to see the prospective client, the event they were trying to sell has just aired its final commercial cluster. I could go on and on about the joys you face every-day but you know all about them.

FCC Relief

There is one area where you feel the world has been good to you. That was when the FCC threw out the rule book with deregulation. Here is something positive that has come your way in the 80's. At long last, the FCC has recognized that you are an honest hard working soul who deserves a break. So they relieved you of all those nit picking rules. Your people can now concentrate on what they are being paid to do -- sell, entertain and promote.

It's a great life, things are really rolling. Your engineering staff has been jumping, keeping all the phones in the sales cars going, setting up remotes and installing all of that new equipment and making sure you are the loudest thing on the dial. Yes sir, things have finally settled down and you can get on with taking care of business.

FCC still pays visits

Then one day you return to your office, via the rear entrance, about 2.30 p.m. after a three martini lunch. Your temporary receptionist informs you that some guy from the government has been in the front office for the past hour and a half. No one seemed to know what the Public File was, or if it existed where it was. The receptionist said the man became rude when he was not allowed to roam through the building.

By the time your day is over this guy from the government is mumbling something about EBS test problems, tower lights, modulation levels, power levels, meter calibrations, forfeitures. What ever happen to deregulation? The investment group that bought the station said the only thing you had to worry about was profit, return on investment, pumping it up for resale.

Deregulation and Compliance

More deregulation has not necessarily meant more compliance in the 80's. A more competitive marketplace has meant attention being directed at those things that are required to keep you competitive. In many cases it means fewer people having to cover more areas of responsibility. There has not been enough time to keep the FCC rules updated, much less check the station against the newly rewritten rule. Reality is what you must deal with day in and day out, and the reality is that most broadcasters have not seen the FCC for ten years or more.

The FCC has been faced with the same problems in the 80's as broadcasters. Budget constraints have left them understaffed. The advancement of technology has almost overwhelmed them in many areas. The broadcast area is apt to get attention only when there has been a complaint. While the FCC still does routine checks the numbers say that it may be another ten years before you see them. Reality for the FCC is that broadcasters share only a part of the spectrum that they must oversee and due to manpower limitations, they go where the problems are.

APPEARANCES

Appearances do make a difference

While deregulation has cut the number of rules broadcasters must comply with, the result has not been a greater level of compliance by broadcasters. For the most part, a low compliance level at a station is caused by the old problem of not being able to see the forest for the trees. To put it another way, familiarity breeds noncompliance.

If a higher degree of compliance is to take place at your station you must change the way you look at two different areas.

FIRST: Take a fresh look at all departments at your station. Are all areas well kept and organized? Does the equipment appear to be well maintained and the wiring neat? Are the required records readily available, up to date and properly reviewed? Are all required licenses, both station and operator, posted as required?

SECOND: Take a fresh look at the FCC rules. Read what the rule says, not what you want it to say. Do not read the first paragraph and then surmise you are in compliance? Read all of it and any sections that may be cross referenced by it. Do you more than cover the rule or do as little as possible and hope it is enough to cover the dang requirement? Too many broadcasters do selective reading when it comes to applying a particular FCC rule to their operation.

The new "As Necessary" rule

"As Necessary" is where most of the problems show up in compliance today. "As Necessary" can mean doing something weekly, daily, monthly, quarterly, annually or every hour. The change to "As Necessary" was intended to accommodate and recognize advancement in technology. How often a certain procedure should be done is dependent on the age of the equipment involved.

There is a need for routine checks and documentation for "As Necessary" rules. If a problem should be discovered at your station the next question by the FCC is always, "When was this last checked?" At this point documentation comes in very handy.

Get out of the office - take a walk

In walking through your facility, look at the way the equipment is being maintained. If you see wiring that looks more like the spaghetti you had the night before, than you know not enough time is devoted to maintaining the facility. Is wiring permanent or do you see clip leads being used? Too often temporary repairs or quick installations become permanent fixtures. It is one of those I'll fix it tomorrow projects, and you know what they say about tomorrow.

The problem with appearances is that we can get use to looking at anything. You can have a six inch hole in your office wall, but after a few months you do not see it anymore. When that visitor walks into your office it is the first thing he or she sees. When you make your appearance check, take the blinders off. Look at everything as if it were for the first time.

Appearances project an image about your operation. Not just to the FCC, but to everyone who enters your facility. If that image says efficient, it means the expectation is that things are in order. If that image says neglect, the expectation is that this place needs a good going over to find the problems. Good appearances are your best defense.

Take the offense. Improve the image you present to others.

BASICS

"As Necessary" basics

Before deregulation there were a lot of basics. Everything had a time table. You had to do this every thirty minutes, that every week, this every month and that every six months or annually, etc., etc. and ETC. Now with the highly publicized deregulation we have "As Necessary" requirements. A mushy phrase that no one seems to understand. So where do we go from here? Back to basics. Not the old basics, but some basics. Deregulation did not change most of the rules you must comply with, it only changed what you have to do to demonstrate compliance.

FCC rules no longer say meters must be read every three hours. They do say your system must be maintained within the parameters of your license. The rules further note that whenever you make a manual power adjustment you must log that adjustment. Well, if you have to log information regarding adjustments, it looks like you have to take readings. The question is, "How often?" The answer is, "As necessary." Sounds like a lot of double talk doesn't it?

Let me ask you this, "How old is your transmitter or remote control system?" If you have an oldie but a goodie, the answer to how often meters should be read, is apt to be at least every hour. Certainly not more than three hours between each set. On the other hand, if you have a new transmitter and remote control system along with a voltage regulator too, readings once per shift may be reasonable.

Checking your compliance level

Deregulation was intended to accommodate progress in technology, not broadcasters. When it comes to determining what "As necessary" means to your operation, look at the components that make up your transmission system. Be objective about the state of technology at your facility. Even if you have the latest piece of equipment out, if it does not perform as advertised and the stability of the unit leaves something to be desired, see that those "As necessary" rules are covered even if it means checking things hourly.

How do you check your facility to assure yourself that you are in compliance with a specific section of the FCC rules? I will cover some of the areas that almost every FCC inspection will involve. Operator licenses, EBS and antenna lighting requirements. The odds are the FCC, if they are on a routine inspection, will not be taking the time to go through an area with the same thoroughness as your inspection. However, when responding due to a complaint, the stations procedures will be checked in detail over a number of months, in the area of the complaint.

The results of your findings will point out the number of times that your facility has been vulnerable during the time period of your inspection. Steps must be taken to reduce the number of errors discovered.

OPERATORS

Do not overlook this one

While this requirement seems to be very elementary the problem is, too often improperly or unlicensed operators show up on station logs. Let us be sure we are all thinking of the same log when we speak of the "station log". The station log is where all EBS tests, both aired and received, are required to be entered along with tower light problems, meter readings, etc. This log is often referred to as the operating log or transmitter log. Section 73.1820 requires entries in the "station log", so I will refer to this log as the station log.

Operators must also be licensed when certain other required records are maintained and signatures are required for the entry. This applies to power calibrations, spurious and harmonic radiation data, quarterly tower inspections, those in charge of the transmission system and any person making an entry in the station log.

Also a valid license is needed by those who operate or maintain auxiliary stations covered by Section 74.100 (Experimental Broadcast Stations), 74.700 (Low Power TV and TV Translator Stations) and those operating stations covered under 74.1200 (FM Broadcast Translator Stations and Booster Stations).

Is that General Class License valid?

A licensed operator is a person who holds a restricted class license or a General Class, lifetime license, issued during 1985 and 1986. The "new" General Class license (form #758 EC, the one that looks like a verification card and states on the front to see the back) is not valid for broadcasting. To be a valid license it must have the FCC Seal. The license should be signed with a legal signature by the operator. A "legal signature" is a signature that can be used by the operator for signing checks and contracts.

To check for valid operators, go to the posting position and list all licenses that are posted. Check each for the FCC Seal and proper signatures. Carefully check all General Class licenses to be sure they do not have expiration dates. If they do, make sure they are still valid. The last of the regular General Class licenses, those with expiration dates, will expire this year. All General Class licenses now posted should be lifetime, no expiration date. If any temporary restricted licenses are posted, see that they are within their valid 60 day posting period. If this period has passed, you may not be guilty of having an unlicensed operator as stated in 73.1860, but you may still be in violation of 73.1230(b) for not having a valid license posted.

Temporary is just that

If a temporary Restricted license has

passed the 60 day time period check with the operator and be sure they can produce a valid Restricted license. I have run into cases where the operators were given the Restricted license form to fill out and send in, but they never got around to mailing it. In a couple of cases, the form never went to the FCC, the person mailed it to themselves. The only Seal on the license was a postage mark.

Today it is the licensee who issues the license. The FCC simply validates it. Use the following procedure to issue a Restricted license. Have the operator sign the form and the Chief Operator fill it out and mail it in for validation. The return address should be to the operator at the station. Do not have it returned to his or her home. Too often licenses are lost or misplaced when they arrive at a home address and are never brought to the station to be posted. By having the stations address as the return address you have a better chance of the license being posted on a timely basis.

If the Restricted license has not been received in four or five weeks, it is time to take action. Either issue another temporary, or reassign the person to a position that does not require a license, until one can be obtained.

Posting of licenses

A "posted license" is one located at the control point and not in some other office. After you have a complete list of all posted licenses you need to obtain six months of station logs. Start with the latest date and go day by day checking the operators on the log against your list. You may need help at times in identifying some of the operators by their signatures. Be sure signatures are legal. The operators must sign the logs, not initial them. Pay particular attention to weekend operators.

Check the date of any recently issued temporary licenses or Restricted licenses. As you go through the logs, make sure operators do not appear on logs prior to their temporary license being issued. The operator with a newly issued Restricted license should not appear on the logs more than four or five weeks prior to the date on the license. In no case should the new operator appear on logs 60 days prior to the new license being issued, the period covered by the temporary license.

Mistakes can be costly

Check each log. Do not go through the logs randomly. Note any dates that an operator may be on the log without a valid license posted. If the operator is licensed, but not posted, the error could cost the station $200 per occurrence. If same operator is signed on more than one date the costs go to $400 per occurrence, because the error is willful and repeated. If the operator is neither posted nor licensed the costs could start out at $400 and go to $800 per occurrence, because two rules have been violated.

Figures noted for forfeitures are examples. While failure to comply with either 73.1860 or 73.1230 have suggested forfeiture amounts of $200 placed on them, they may be adjusted for violations that are considered to be willful and repeated. You may be charged for each day the violation exists. For example: Five dates with an unlicensed operator could cost $1000 or more in forfeitures.

The actual determination of a forfeiture is up to the inspector. It will be based on a number of factors, appearances being one. The purpose of indicating a forfeiture amount is to give you some idea of the possible bottom line damage that can be caused by failure to give proper attention to required FCC Rules and Regulations.

Some licensees may look at a forfeiture as a cost of doing business. They play the odds that it is cheaper not to bother with the rules, as compared to the chances of being caught. Any forfeiture reflects on the licensees ability to operate the station within the FCC rules as required. This factor could be a costly one for a station facing a challenge for its license. When considering the costs of compliance, always weight it against the value of the property being protected. No licensee would consider going without fire insurance. Very few have taken steps to insure their license with a program that checks its compliance level. A high degree of compliance is the best insurance you can have to assure your stations renewal.

If you have passed the operator check with flying colors, Great! Now lets go on to EBS logging requirements.

EBS

EBS compliance

I will assume the following: (In my line of work you never assume anything, so check these out)

1. Your EBS unit is working properly.
2. Your Authenticator word list is up to date.
3. Your EBS check list is the new 1987 edition.
4. You are monitoring your "assigned" station, or have the required written

authority to monitor another.
5. Your EBS tones modulate the transmitter by the required 65-70 percent.

I will deal only with rule sections 73.1820, 73.932, 73.961 and 73.962. These sections outline the logging requirements of the various EBS tests that must be aired and received.

EBS tests must be noted, either in the station log or may be entered in a special EBS log. If a special EBS log is used it must be readily available and is considered a part of the station log. All station log requirements apply to the special EBS log.

Obtain station logs or EBS logs, for the past six months. Start your EBS compliance check by writing down all EBS tests noted in the records. Keep a separate record for those aired and received. For tests aired, note the time and date of each test. For each test received, note the date and source of the test. Also note any periods where EBS test problems may have been explained in the Chief Operator Review. If the EBS data is kept in a separate EBS log, be sure all entries have legal signatures by licensed operators. Once you have all EBS entries listed, it is time to check your success rate.

EBS schedules

The EBS test schedule may be broadcast on a regular calendar week (Sunday through Saturday) or broadcast calendar schedule (Monday through Sunday). The schedule must be consistent with one or the other. They cannot be mixed. Armed with a calendar for the period that you are checking, go through your list. See that dates logged fall within the required calendar weeks. For any weeks that do not show a test aired or received, see if the problem was explained. Make a list of the weeks that do not show the required EBS test and no explanation is noted.

For stations with a wire service, see that the required monthly tests have been noted in the station or EBS log. You should find at least one per month. If the station has two wire services, entries should be consistent for one of the services. If the station is a Network affiliate, you should find a notation regarding the CLOSED CIRCUIT TESTS run on a national basis. These tests are to be sent at least once per quarter, not more than one per month, by both Networks and wire services. The station should have a network decoder to alert them of these tests.

Importance of Chief Operator Review

EBS tests aired and received must be documented in the station log or EBS log. Tests not documented as required should have an explanation in the log. Tests not logged, or explained, are subject to a forfeiture of $300 per occurrence. If more than one test is missed, in a particular category, the error could be considered as willful and repeated and the station subjected to a forfeiture of $500 per occurrence.

The Chief Operator review process should be noting any EBS problems, along with an explanation. When a test is not logged as received, from the station you monitor and your receiver appears to be working OK, you are supposed to contact the station to see if they aired a test. Even if the explanation is, "I do not know why a test was not received", that is acceptable as long as you have checked all possibilities. This kind of explanation cannot be used for a number of weeks. Once you reach the second EBS failure, the FCC expects you to find the problem and correct it.

Section 73.922(b) implies that you are to air the audio from the station you monitor, in a national emergency. Check to see that the EBS receiver is available at a source on the audio console, or by patch arrangement.

Operators must know how to get the audio from the EBS receiver on the air. When an EBS test is received, operators must monitor the source for the test announcement, not kill the audio during the tones. EBS tests aired by the station must be aired between 8.30a.m. and local sunset. Check the times of EBS tests aired during your inspection. Updated rules require a station to air the EBS script even when you cannot air the tones due to equipment problems.

ANTENNA LIGHTING

Broadcast and Auxiliary

Any license issued to the station, whether it is for a Main or Auxiliary transmitter or antenna, or equipment covered by Section 74 of the rules; if it has lighting requirements you must comply with the requirements of 73.1820, 73.1213, Section 74 and 17 of the rules. When looking over these licenses see if more than one location is involved. All locations will require daily tower light checks, quarterly tower inspections, logging of lighting failures, notice to FAA as required, or the station must have on file a common tower agreement on the location. The common tower agreement must be with another licensee, showing them responsible for all logging requirements. Remember, a common tower agreement cannot be with a company or corporation that does not hold an FCC license.

For each license noted on the same tower, check listed coordinates to be sure they are the same. When a license is received from the FCC, they send along a notice to check all terms of the license carefully and notify them of any errors. Errors should have been covered by letter, as required.

Change in the status of a tower

All licenses with lighting requirements have a sheet explaining those for the tower listed. Lighting will be either beacons, and side lights, or strobes. Check each structure against this sheet.

Change in the status of a tower will often throw these requirements off -- For example: A main tower no longer used, other than for auxiliary equipment, after a new site has been built. This may be a former TV tower where the top TV antenna has been removed. The antenna section made up most of the top orange section. Now the tower has a white section at the top, or a very short orange section. When a change such as this occurs, the tower needs to be painted as soon as possible to conform with the rules. Lighting requirements may also be altered by a change on the structure. Both painting and lights should be checked.

Although FCC rules require daily tower light checks, they no longer state these checks are to be logged. As a matter of procedure, the station should require their logging to assure compliance. Quarterly tower inspections must not exceed three months, if they should the reason why should be noted.

FAA notification

Your station log must show that the FAA was notified regarding any top light or beacon indicated as out. The FAA number should be posted in the control room and all operators aware of its location. Notice to the FAA is again required when these lights are repaired. Light repairs should be made within a reasonable period of time. A top light or beacon should be repaired within 30 to 45 days, side lights 45 to 60 days. Tower light repairs should not take six months. The flash rate for beacons must be 12 to 40 per minute, strobes 40 per minute.

Paint on the tower should be clean and bright. If the paint is chipped, faded or deteriorated the station should have an agreement on file to have the tower painted, or at least be in the process of negotiating one.

COMPLIANCE

Deregulation does not mean: No compliance necessary

We've covered only three different areas of the FCC Rules. There are hundreds of rules, that cover dozens of areas. They need the same type of close examination as outlined in our three select areas.

Every station considers its compliance level to be good. Every station also has a failure rate in compliance. How bad is yours? Your business exists because you have been granted a license. You do not own the license, it is yours for as long as you serve the public interest AND comply with the rules that govern your class of license. A high degree of compliance can only be maintained through planning and hard work. The same kind of effort that is needed to hold market shares and obtain advertising dollars.

Back to basics

Basics are the foundation to any successful compliance level. You need basics in every area of your operation, whether it is programming, sales or compliance. If compliance routines have fallen by the wayside, get back to them.

Place the same effort into protecting the station license, as you do in sales, or advertising, or promotion. When you look at your P & L what does it say about your compliance efforts? Do you consider engineering costs or legal fees as your means of assuring compliance with FCC rules? Attorneys advise, engineers build, maintain and repair and there is little time left to make sure all the rules have been covered. The responsibility of FCC compliance rests with the licensee.

How should you cover the area of compliance? Develop a system, or buy a system and have someone within your organization responsible for keeping up with the rules. On a periodic basis, check the operation for its actual compliance level. Will this all take time? Yes. Will it cost money? Yes. If you are not sure it is worth the time and money, ask yourself this question: "How much would I spend to defend my license against a challenge?"

A healthy compliance level is a challenge

Take the challenge now, to improve your compliance level, and you greatly improve the odds of never having to face a challenge for your license.

Deregulation has not been the answer to a higher degree of compliance in the 80's. In fact it has had the opposite effect at too many stations. As the rules decreased, so has attention to those that remained. But the tide is turning as more and more licensees discover the best regulation of all, self regulation. Self regulation is a commitment to a level of compliance that is over and above that which is acceptable to the FCC. The self regulation adherent sets standards that border on perfection. While this level may never be reached, the act of trying for perfection keeps that station far ahead of the rest.

Self regulation is an attitude of wanting to be the best. The best in sales, programming, promotion AND compliance.

SOLVING FREQUENCY COORDINATION PROBLEMS AT THE 1988 POLITICAL CONVENTIONS

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Abstract

Regardless of whether 2 or over 1500 transmitters and receivers are being used in close proximity, each of these devices, though separate, becomes part of a larger, all-inclusive, communications system. Each part of this system is dependent on the others for interference free reception. At the 1988 Political Conventions cooperation among all RF users was one of the most important weapons in the fight against interference.

Introduction:

The instant I stepped into Atlanta's Omni Coliseum and passed six people with handi-talkies, not one of which was coordinated by the Committee, I knew all was lost 'RF-wise'. All of the interference analysis runs and all of the hard work that went into the Political Conventions Frequency Coordination Committee or the PCFCC was for naught.

Tempting fate, I decided to investigate further. Things only got worse. Political realities being what they are, the sheer number of law enforcement officers at these two political conventions was staggering. For the law enforcement types, it was as though there was an unwritten rule - "You're nobody without a walkie-talkie". For the broadcaster, there was another unwritten rule - "You're nobody without a substantial number of microwave links, fixed and portable, across the city". In this paper, we will discuss some of the procedures used for large scale frequency coordination, and possible improvements which can positively impact such coordination. We will also relate some anecdotes from the 1988 Democratic and Republican National Conventions.

Why Formal Coordination is needed.

From the standpoint of frequency coordination, one of the most difficult tasks is getting an accurate handle on existing, local frequency usage. Because frequency congestion in smaller markets is not always a problem, a licensee who had been previously coordinated for a particular channel, can sometimes 'fall off' frequency lists, without anyone noticing.

While a smaller market may live with the hit-and-miss procedure that we are all too familiar with, the frequency coordination database of a large market must, of necessity, be accurate. Every state has a frequency coordinator and most large cities also have their own coordinator. Obviously, these coordinators must maintain an active listing including frequency, user, contact person, power, path and location.

FOOTNOTES

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(The views expressed are those of the authors and do not necessarily reflect the views of the Commission.)

Unfortunately, more than once in both Atlanta and New Orleans, we were unhappily surprised by an emergency call from an STL operator complaining that we had permitted another broadcaster on an already-used frequency and path.

We recommend that every market, large or small, maintain a working database of all broadcast auxiliary stations in periodic or continuous usage, including maps of all fixed microwave locations and all local CARS locations. Table 1 shows one form of the local Atlanta Frequency Coordination Database, with just 4 out of 15 fields. Only 2 frequencies are shown, with some of the users.

Table 1 Local Atlanta Frequency Database

Station	City	Miles From Atlanta	Frequency
WDUN	Gainesville	50	161.67 MHZ
WMAZ	Macon	77	161.67 MHz
WLET	Taccoa	82	161.67 MHz
WZOT	Rockmart	43	161.67 MHz
WGST	Atlanta	--	161.67 MHz
WCON	Cornelia	73	161.67 MHz
WKEU	Griffin	35	161.67 MHz
WRFC	Athens	60	161.70 MHz
WYNX	Smyrna	Suburb	161.70 MHz
WKLS	Atlanta	--	161.70 MHz
WHNE	Cumming	35	161.70 MHz
WFOX	Gainesville	50	161.70 MHz
WVOP	Vidalia	156	161.70 MHz
WHIE	Griffin	35	161.70 MHz

44 Stations Using 950 MHz STL's
83 Stations Using the 450/455 Frequencies

More than 2000 frequencies were coordinated and listed by the PCFCC. There were a total of 25 fields of potential information for each frequency. Table 2 shows a portion of the database used for wireless mics. These mics were using frequencies in the TV12 and TV13 bands. Note the closely spaced channels and the note "Subject to Interference" (SI) on the TV 12 frequencies due to local use.

Table 2: PCFCC Frequency List

Frequency	User	Power	Area	Resolved
208.80000	UPI	.05	IO	Y-SI
208.8750	NBC	.05	IO	Y-SI
209.2000	CBS	.05	I	Y-SI
209.4250	CP	.01	PIO	Y-SI, A
209.4250	RNC	.05	I	Y-SI, NO
209.7250	WTVH	.05	IO	Y-SI
209.9250	WSB	.05	IO	Y-SI
210.0000	AP	.05	I	Y
210.6000	UPI	.05	PIO	Y
210.8000	BONN	.05	IO	Y
211.0000	NPR	.50	PO	Y

Table 2: A page from the Political Conventions Database. Not all of the fields were always printed because of space constraints. There were a total of 25 fields and about 2000 frequencies listed in our database.

During day-to-day operations and small events, there is usually enough flexibility in the microwave and communication frequency use listings to allow local users (and their affiliated users) to add slightly to their frequency allotment without any special coordination. During a large event, such as a political convention, the local broadcaster's frequency needs will certainly increase. But then, look out! Here come the networks and perhaps 100 more outside broadcasters, each with its own requirements. On the positive side, at the conclusion of such an event, the host city's frequency database is up to date.

Formal frequency coordination procedures are the only means of maintaining order in the limited available spectrum. Guidelines for coordination must be developed and adhered to. The guidelines of the 1988 Political Conventions Frequency Coordinating Committee are shown in Figure 1. Figure 2 is a copy of the FCC's Public Notice, suspending Section 74.24 of the Rules. A waiver of this Rules section is necessary for large events, in view of the fact that uncoordinated use of auxiliary broadcast stations on an 'automatic STA' basis might result in spectrum congestion and possible interference, causing less complete broadcast coverage.

Monitoring Equipment

Despite pre-planning and coordination, complaints of interference will occur and it is necessary to have the proper equipment to analyze, identify and resolve it. Inexpensive scanners should be avoided in highly

1988 FREQUENCY COORDINATING COMMITTEE GUIDELINES FOR OPERATIONS

1. FCC rules and regulations shall be complied with at all times unless a special waiver has been granted.
2. Each convention city primary frequencies shall be protected unless otherwise stated by the affected user based upon proper coordination, first by the user and then by the committee. The committee shall be kept informed at all times.
3. Each user shall initially select and request frequency assignments which are presently operated by, and deemed as primary channels by, the user to promote maximum use of existing equipment. Additional channel requirements will be satisfied by committee investigation and recommendation.
4. Each user will request and submit all notifications and STA applications for frequencies and associated equipment use to the FCC. File copies shall be provided to the committee.
5. All RF transmitting equipment other than microwave inside each convention hall shall be restricted to the minimum RF power output necessary for the intended operation which in no case shall be greater than 5 watts.
6. Sub-committees shall be formed to determine requirements, gather factual information, analyze alternatives and make appropriate recommendations to the committee on a needed basis.
7. Any conflicts during the convention period shall be initially worked out among the affected users. If no solutions are reached, the committee, upon request by any party, shall offer investigation and recommendations. The FCC should be notified only if no mutual agreement can be reached among all concerned parties and the committee.
8. It shall be compulsory for all microwave links, including satellite uplinks, outside the halls to be identified by visual means at all times when in use without actual program, that is by graphic ID over color bars or other signal.
9. Base stations operating with power output over 20 watts shall employ isolators/circulators with 3 stage cavities in order to reduce intermodulation products.
10. 2 GHz transmitters in the Convention Halls shall have their power output to the antenna limited to 200 milliwatts, by design or padding. Split channel operation shall be used with the lower half polarized clockwise and the upper half counter clockwise.
11. Users of Broadcast Auxiliary Service frequencies shall carefully test equipment to ensure that out of channel products that could contribute to harmful interference are not present.
12. When beginning operation, each station shall transmit the call sign of the associated broadcast station, or an appropriate organization identification, together with a unit designator, e.g. WWL-3, NBC-1, CNN-2, etc." add example.
13. Each organization intending to operate will furnish the committee with the name of a contact individual and telephone numbers to contact the individual at home base and at each convention city, and will notify the Committee upon arrival at each convention site.

FIGURE 1

congested frequency areas where split channels are used and active channels overlap. Spurious responses may be produced in these receivers, while a more advanced receiver may not exhibit these 'spurs'. The Political Conventions provided a good demonstration of receiver front-end quality, particularly the DNC in Atlanta's Omni. The construction of the Omni allowed RF signals to pass through its walls in both directions. Thus, while operating inside, not only were we concerned with signals generated and bouncing around inside but signals from the outside had to be dealt with also. In this venue inexpensive scanners, with their poor-quality front-ends, often demonstrated more problems and irregularities than actually existed.

In addition to programmable scanning receivers, one of the most useful devices was a high quality, communications service monitor. The device we used was a microprocessor-controlled, digitally synthesized unit with built-in batteries for portable use. It included an adjustable spectrum analyzer section, a signal generator section capable of generating modulated or unmodulated signals from 100 kHz to 1 GHz in 100 Hz steps, and a quadruple conversion, superheterodyne receiver section, capable of monitoring AM, SSB, and narrow and wideband FM signals from 300 kHz to 1 GHz in 100 Hz steps. The unit, though cumbersome due to the weight of its batteries, proved to be very useful in identifying and analyzing many types of problems.

At the DNC, a "power-up" test was held on the Omni floor, the day before the Convention began. In the center of the floor, I had set up the service monitor to observe the various bands. When the broadcasters powered up, I first thought that my monitor was out-to-lunch.

The bands that we were using for wireless mics were wall-to-wall signals. However, where problems existed, for example two mics operating co-channel, I was able to reassign them by picking out empty slots in the spectrum. This, of course, violates all the rules of intermod control. Responsible frequency users select their wireless mic frequencies in a pattern to control intermod. In Table 2 we saw a portion of the wireless microphone frequency configuration planned for the Conventions. By simply filling in the empty holes, all of the planning and analysis is wasted and all bets are off with regard to interference prediction.

Under adverse conditions, though, the sensitive spectrum analyzer couldn't handle the intense RF conditions and still differentiate between signals. The NBC Citywide EJ frequency in the 161 MHz band was coordinated many months before the DNC. Each afternoon, beginning five days prior to the Convention, NBC had tremendous interference on this all-important, citywide channel. Without this channel, field crews and other personnel had no communications.

The interference, an FM signal carrying a baseball game, was strong enough to capture the input to the repeater on the roof of the IBM Tower in downtown Atlanta. The signal was an almost continuous transmission of play-by-play from about 2:00 until 5 P.M. There were no other available frequencies in this band, but it seemed to be a problem which could be easily resolved. We were able to intermittently recapture the channel, so we continued using this frequency even during the times of interference.

The baseball game signal was fairly weak, too noisy to hear names clearly, yet it was persistent interference. Day after day this problem continued and grew more annoying. I spent one afternoon tracking down all of the American or National League games which might have been broadcast. I met with every local television and radio station in Atlanta to find out who was broadcasting a baseball game on this frequency. I had local stations for fifty miles in all directions monitor the frequency to try and determine the approximate location. All the efforts were to no avail.

Two days before the Convention began, solving this interference case took priority. I drove around Atlanta with the service monitor, various receivers, and spectrum analyzers to get a fix on the transmitter. After hauling all the equipment to the roof of the IBM building, I set up a monitoring station in between the scores of antennas temporarily set up for the Convention. The service monitor's front end, subjected to all of this RF, overloaded and was unable to operate. The rogue signal was strong, but none of the equipment was able to give a visual representation. Using a Yagi antenna,

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March 16, 1988

AUXILIARY BROADCAST FREQUENCY COORDINATING COMMITTEE FOR THE 1988 POLITICAL CONVENTIONS DESIGNATED AUXILIARY BROADCAST FREQUENCY COORDINATOR

The Commission has designated the Auxiliary Broadcast Frequency Coordinating Committee for the 1988 Political Conventions (Committee) as auxiliary broadcast frequency coordinator for broadcast coverage of the 1988 national political conventions to be held in Atlanta and New Orleans this summer.

The Committee is composed of representatives of broadcast networks, their owned and operated or affiliated stations, and numerous other group broadcast licensees. It requested to be designated coordinator for auxiliary broadcast frequency use for broadcast coverage of the conventions.

The Commission also granted the Committee's request to suspend Section 74.24 of the rules in the convention cities from July 1 through August 31, 1988. The period covering the suspension of Section 74.24 will allow for advance coordination of auxiliary broadcast frequency usage. To minimize harmful interference all Part 74 users, and all Part 21 and 78 licensees sharing Part 74 spectrum, are covered by the scope of this action. The affected areas are within 80 kilometers of Atlanta and New Orleans for terrestrial stations and 150 kilometers for any mobile stations onboard aircraft. Section 74.24 allows eligible broadcasters to operate auxiliary broadcast stations on a short-term basis, not to exceed 720 hours annually, without prior Commission authorization.

Abridged Version of FCC Public Notice

First three paragraphs

FIGURE 2

I was able to directionalize and approximate the location of the offending transmitter. Within minutes, I had the direction and received signal strength optimized, by listening to the'quieting' of the receiver. I then mounted a larger antenna to further optimize reception.

For the first time, the baseball game came in loud and clear, as though I was sitting in the announcer's booth. It was between innings and by a stroke of luck, the announcer was conversing with someone and happened to give the phone number of the studio. I quickly called on the cellphone, but the receptionist, the only one at the station, didn't know anything about the rebroadcast of a major league game.

The game resumed. I was puzzled by what I heard. It sounded like a bunch of little kids -- really little kids -- playing baseball. Quickly, because it was approaching the ninth inning of the game, I called the station back. The receptionist informed me that, yes, there was a little league game that the station was covering, but she didn't know where it was . She gave me four possible locations -- school yards in the East Smyrna area. With the help of another NBC employee, I quickly drove from one location to another. We were looking for a small schoolyard stadium with floodlights, because it was getting dark. We almost drove right past the stadium, but we spotted the cars parked nearby. A station logo was spotted on a small, glass-enclosed, anchor booth, by the field. We ran to the booth, holding various receivers, cellphones, headsets, and other equipment and almost scared the pants off the parents, the kids, and the station personnel. I quickly explained the situation to the announcer who was understanding and helpful. They were using a powerful, wideband transmitter to get to their studio, about eight miles away. There was only one game to go -- opening night of the Convention, but they graciously agreed to use a phone line. This particular incident attracted considerable attention, because of the interesting circumstances surrounding the baseball game.

Interference Control

Interference to two-way communications systems, low power RF microphones and repeater systems can originate from many sources. Industrial Heating, Ventilation and Air Conditioning (HVAC) units, welding equipment and medical (ISM) devices, atmospheric conditions and power lines are but a few of the possible sources. Such sources can be very difficult to control, so a site survey was performed about two months before each convention. A spectrum analyzer was brought to each location and the bands to be used were displayed.

Two other types of interference, which are usually easily controlled, are adjacent and co-channel interference. Either of these interference types can be misleading, because they may sometimes appear to be noise or intermodulation. Once recognized, their cure is straightforward, but not always painless. In the case of co-channel interference, lowering the power may help. Increasing the physical spacing between users may also help. Shut-down is the last resort. For adjacent channel or overlapping bandwidth problems, these methods may be tried, or one transmitter may be reassigned.

Intermodulation distortion ('intermod') interference distorts the 'wanted' signal with either a coupled "unwanted" signal or some combination of several "unwanted" signals. It can occur in any linear device, transmitter, receiver, or an external device, but the dominant form of intermod is caused by the mixing of signals in the power amplifier (PA) of a transmitter. This subject is more fully treated in the next section on interference analysis.

Many broadcasters, though using similar transmission systems, use antennas of varying gains. When a typical dipole or unity gain antenna is used in close proximity to a high gain antenna 12.5 kHz away, the unity gain system will captured, since receiver AFC circuits will lock onto the stronger signal. Therefore, antenna characteristics must be considered to fully understand the effects on the total system.

Three hours before the Convention began, we had an "inside the hall" microwave problem. Table 3 shows a list of "inside the hall" microwave users in the 2, 2 1/2 and 7 GHz bands. Frequencies were also used in the 13, 18, 23 and 40 GHz bands. Apparently, one user was either using too much power or their transmit frequency had drifted. We set up a test bed outside the CBS anchor booth and all users in this band brought their microwave equipment to be tested. Fourteen units were tested, and power was reduced on nine of them. The polarity was also checked and one user had accidentally reversed his polarity. The problem was resolved.

Spurious, overly wide, or overpower transmitters cannot be detected until they actually radiate, thus a pre-event screening is suggested to detect any obvious transmitter anomalies while there is still time to repair or replace a defective device. All RF equipment used to cover the Space Shuttle take-offs and landings is now being carefully checked by NASA or Air Force authorities. Equipment that has not been checked, or which does not meet spurious emissions requirements, is not allowed on the base. This check is done to assure that the media do not interfere with the Shuttle's safe operation, but we recommend testing every piece of RF generating equipment which will be operated inside the Convention Hall to identify possible interference sources.

"Bunching" of antennas is another common problem we encountered at the Conventions. The probability of receiver intermod's occurring rises, as does the coupling between the transmission cables, if the antennas are situated close to each other. Good site management dictates the use of circulators and cavities to minimize the interaction between systems. Further, the use of physically wide-spaced antenna sites will reduce these effects. The maintenance of the common power ground is necessary to reduce RF noise.

The coordination repeater, a 450/455 MHz pair, proved to be one of the most valuable pieces of equipment used to coordinate frequencies and determine causes of interference. In Atlanta, the FCC helped us in securing a location on a nearby tall Federal Building and also helped in its installation. The repeater performed flawlessly, city-wide. In New Orleans, where steel reinforcement of the Superdome prevented most signals from penetrating the building, the repeater

1988 POLITICAL CONVENTIONS

2, 2 1/2, and 7 GHz Use Inside Hall

Designation and Band	Active	Operating A (Minor)	Users Inside Hall	Active	Center Freqs. Center	Users Inside Hall	Micro (MHz) B (Plus)	Active	Users Inside Hall
A1 1990-2008	A1A ---	1994.50 ---	CNN	A1---	1999.00	Not Used	2003.50--	A1B----	Bonneville
A2 2008-2025	A2A ---	2012.25 ---	Mid-King	A2---	2016.50	Not Used	2020.75--	A2B----	NBC
A3 2025-2042	A3A ---	2029.25 ---	CBS	A3---	2033.50	Not Used	2037.75--	A3B----	Mid-King
A4 2042-2059	A4A ---	2046.25 ---	CBS	A4---	2050.50	Not Used	2054.75--	A4B----	Gannett
A5 2059-2076	A5A ---	2063.25 ---	Bonneville	A5---	2067.50	Not Used	2071.75--	A5B----	RNC* WXIA**
A6 2076-2093	A6A ---	2080.25 ---	Gannett	A6---	2084.50	Not Used	2088.75--	A6B----	CBS
A7 2093-2150	A7A ---	2097.25 ---	NBC	A7---	2101.50	Not Used	2105.75--	A7B----	WXIA** RNC*
A8 2450-2467	A8A ---	2454.25 ---	Not Used	A8---	2458.50	ABC	2462.75--	A8B----	Not Used
A9 2467-2484	A9A ---	2471.25 ---	Not Used	A9---	2475.50	ABC	2479.75--	A9B----	Not Used
A10 2484-2500	A10A --	2488.00 ---	Not Used	A10--	2492.00	Post	2496.00--	A10B---	Not Used
B5 6975-7000	B5A ---	6981.25 ---	Not Used	B5---	6987.50---	ABC	6993.75--	B5B----	Not Used
B6 7000-7025	B6A ---	7006.25 ---	Not Used	B6---	7012.50---	Gannett	7018.75--	B6B----	Not Used
B7 7025-7050	B7A ---	7031.25 ---	Not Used	B7---	7037.50---	NBC	7043.75--	B7B----	Not Used
B8 7050-7075	B8A ---	7056.25 ---	Not Used	B8---	7062.50---	NBC	7068.75--	B8B----	Not Used

* New Orleans only ** Atlanta Only Table 3

signal couldn't get through. Raising the power was contemplated, but we were fearful of causing interference to broadcasters ourselves. Instead, using hand-helds lent to us by CP-Communications (Yonkers, NY), we went simplex inside the Dome and used the repeater outside. The FCC Field Operations Bureau (FOB) monitored our coordination channel, whenever a problem arose, the Commission was instantly aware.

Interference Analysis

Intermod analysis is, in principal, a fairly straightforward operation. Since intermodulation distortion occurs whenever two or more signals are applied to a nonlinear circuit, it is unavoidable. For instance, two unmodulated signals, at frequencies f1 and f2, combine as follows[3]:

$$f_1 + f_2 \text{ -- } f_1 + f_2 + 2f_1 + 2f_2\ (f_1 + f_2) + (f_1 - f_2)$$

$$+ (2f_2 + f_1) + (2f_1 + f_2) + (2f_2 - f_1) + (2f_1 - f_2) + \ldots$$

The last four terms, containing the sum and difference frequency components, $2f_2 \pm f_1$ and $2f_1 \pm f_2$, are called the third order intermod products. There are, of course, higher order intermod products also. With three signals present, the third order intermod products become:

$$(f_1 \pm f_2 \pm f_3) + (2f_1 \pm f_2 \pm f_3) + (f_2 \mp 2f_1 + f_3) + (f_3 \mp f_2 \mp 2f_1)$$

The first term, the most significant term in amplitude, is called the triple beat, the remaining terms are simply permutations of the two signal case, extended to include three frequencies.

It can be readily seen that the number of third order intermods increases signficantly with the number of frequencies present. Tables 4 and 5 show some of the outputs of 3rd Order Interference Analysis runs we performed for the Conventions. The frequencies considered for the third order calculations used the same 10 wireless mic frequencies listed in Table 2. The Interference Analysis calculations for the Conventions were, of course orders of magnitude more complicated because a congested band had to be calculated for as many as 100 transmitters.

Table 4

2 Transmitters - 3RD ORDER PRODUCTS

TRANSMITTERS INVOLVED/PRODUCTS	AFFECTED USER	FREQUENCY
2 X 209.425 - 210 = 208.85	NBC	208.85 MHz
2 X 209.725 - 210.6 = 208.85	NBC	208.875 MHz
2 X 210 - 210.8 = 209.2	CBS	209.2 MHz
2 X 209.725 - 210 = 209.45	CP	209.425 MHz
2 X 209.425 - 208.875 = 209.975	AP	210.0 MHz
2 X 210 - 209.425 = 210.575	UPI	210.6 MHz
2 X 210 - 209.2 = 210.8	BONN	210.8 MHz
2 X 210.8 - 210.6 = 211	NPR	211.0 MHz

Table 5

3 Transmitters - 3RD ORDER PRODUCTS

TRANSMITTERS INVOLVED/PRODUCTS	AFFECTED USER	FREQUENCY
208.8 + 210 - 209.925 = 208.875	NBC	208.875 MHz
209.2 + 209.425 - 209.725 = 208.9	NBC	208.875 MHz
209.725 + 209.925 - 210.8 = 208.85	NBC	208.875 MHz

Only 9 out of 50 possible hits shown

When the fifth and higher order terms are also considered, the only possible way to analyze the intermod products of a large number of transmitters and receivers is with a computer. Tables 6 and 7 show the outputs of 5th Order calculations for Table 2's list. Again, these calculations are only for 10 transmitters and the results are only partial listings.

[3] The constant terms and sinusoidal functions are assumed.

Table 6

2 Transmitters - 5TH ORDER PRODUCTS

TRANSMITTERS INVOLVED/PRODUCTS	AFFECTED USER	FREQUENCY
3 X 209.425 - 2 X 209.725 = 208.825	UPI	208.8 MHz
3 X 210 - 2 X 210.6 = 208.8	UPI	208.8 MHz
3 X 209.725 - 2 X 210 = 209.175	CBS	209.2 MHz
3 X 209.2 - 2 X 208.8 = 210	AP	210.0 MHz

Table 7

3 Transmitters - 5TH ORDER PRODUCTS

TRANSMITTERS INVOLVED/PRODUCTS	AFFECTED USER	FREQUENCY
3 X 209.2 - 208.8 209.925 = 208.875	NBC	208.875 MHz
3 X 209.425 - 208.8 - 210.6 = 208.875	NBC	208.875 MHz
2 X 209.2 - 2X 209.725 + 209.925 = 208.875	NBC	208.875 MHz
2 X 209.725 - 2 X 210 + 209.425 = 208.875	NBC	208.875 MHz
2 X 209.725 - 2X 210.8 + 211 = 208.85	NBC	208.875 MHz
2 X 209.925 - 2X 210.8 + 210.6 + 208.85	NBC	208.875 MHz

6 out of 60 possible hits shown!

We are aware of at least five programs which can perform interference analysis. Each of these programs performs the calculations after the manual loading of frequency use and bandwidth data for a site. We had to modify one of the programs in order to be able to properly calculate the large numbers of users in each band. In the 450/455 MHz band, for example, every channel and every split channel was used. With every change, I reran that particular band. In order for this data to be useful and reliable, large-scale cooperation was required by all of the users.

FCC Field Operations Bureau

Broadcasters generally dread the idea of FCC involvement in their affairs, since it has traditionally meant a 'pink ticket'. In this age of privatization of government services, it is amazing that the FCC's Field Operations Bureau (FOB) still provides an interference (IX) resolution service at many large events. Generally, the FCC will be present at large events, especially political conventions to support other government agencies, such as the Secret Service or the White House Communications Agency (WHCA). They have interference problems, especially with itinerant operations, just as broadcasters do. This is their primary function at the event. However, if you have exhausted the other means available to resolve an interference problem, the FCC may be able to help.

Security

At any event where many politicians are involved, security must be tight. At the Political Conventions, security takes on a whole new meaning. If a frequency coordinator at an event of this kind is also charged with determining interfering sources, a slew of credentials is needed. By the end of both Conventions, Joe Russo of ABC, who shared the burden, and I each had over 20 credentials around our necks. Credentials included those from the networks, CNN, the political party, from any organization who wanted us to have quick access, venue perimeter and hall and floor passes for each day. One morning during the Conventions, the two of us, armed with enormous amounts of radio gear and conflicting credentials, were checking on frequencies that the Omni security personnel were using. Unknown to us, the Omni security had been temporarily taken over by the Secret Service. When we walked in to their secure area, they didn't believe the credentials were valid. We were held by two Atlanta State Troopers until the FCC came in person to vouch for us. Needless to say, after this episode we allowed the Secret Service to use whatever frequencies it wanted.

On to 1992

For any large media event of National importance, particularly those of such a large magnitude, more and more people will require communications, wireless mics and RF cameras. Cooperation between RF users is imperative. A good coordination process can arrest many of the possible sources of interference which can disrupt news gathering operations.

The guidelines for the 1988 PCFCC (Figure 1) can provide a good base for smooth operations. However, there are a number of changes that can be implemented in the frequency coordination procedure to help prevent some interference and isolate other interference quickly. These additions are mentioned below:

1988 POLITICAL CONVENTION COORDINATION COMMITTEE
INTERFERENCE SUB-COMMITTEE
FREQUENCY REQUIREMENT FORM
REPUBLICAN NATIONAL CONVENTION
NEW ORLEANS, LA

Name ____________________ Address ____________________
Organization ______________ Phone No. ____________________
Number of Stations _______ TV ___ Radio ____ Print Media ____

Times/Dates of equipment use: (Note: All times must be EDT)
Start up/Finish in city ______________
Test Times (Dates, Hours) ____________
Transmission: ______________________

Have you applied for a skybox? Yes _______ No _______
Have you applied for a stand-up area? Yes _____ No _______

Please fill in the form below where appropriate. The column marked Choice 1, 2, 3 is reserved for those who are not frequency agile. Do NOT use this column if your equipment is frequency agile or you can recrystal.

Example:

	TV	Band	Freq. Agile?
No. of RF Cameras	2	7GHz	Yes

TV = TV, R = Radio, P = Print Media

WHERE USED	Inside / Outside Hall	TV	R	P	BAND	FREQ. AGILE?	RECRYSTAL POSSIBLE?	CHOICE 1, 2, 3	USE
	No. of RF Cameras								
	No. of RF Talents Mics								
	No. of RF Camera Control								
	No. of Communication Ch.								
	No. of IFB Channels								

USE CODES
- P = Program Material
- I = Interrupted Feedback
- C = Control / Cue
- H = Handi Talkies
- V = Vehicular
- M = Mobile Base
- B = Base
- O = Operational Communications
- T = Technical Communications
- R = Radio Microphone
- K = Wireless Camera
- D = Data

Write in any other use

WHERE USED CODES
- I = Inside Hall
- O = Outside Hall
- P = Perimeter of Hall
- H = Hotel
- R = Remote Pickup
- L = Return Link

Figure 3

1. Starting now, before a large news event comes to your area, make certain that an up-to-date list of broadcast auxiliary frequency use is available.

2. Pre-event testing of all RF equipment used inside the venue.

- Handi-talkies should be tested for proper output power and spurs.

- Wireless mics should be tested for power, bandwidth and spurs.

- RF (microwave) cameras should be tested for power levels, bandwidth, proper offsets, polarity and antenna type.

- IFB systems and any other transmission systems which operate continuously warrant special attention due to their high power and continuous duty cycle.

3. Pre-event Publicity to all concerned parties:

- An information packet should be sent to all local broadcasters (radio and TV), and all other broadcasters and cable organizations interested in participating in the event. Prior to this, preliminary information should be disseminated as widely as possible, through magazines, press releases, wires and public notices, to form a master list of participants. All local broadcasters should be included, regardless of their potential level of activity. One week before the DNC began in Atlanta, a local radio station informed us that their news operations had planned for remote broadcasting from the convention area, just outside the Omni, all four days, eight hours per day! Needless to say, plenty of equipment juggling had to be done to fit them in.

The information packet should include:

A. Proposed guidelines for broadcasters.

B. Coordinating Committee contacts.

C. Frequency/Channel Requirement forms (see Figures 3 and 4 A, B - the forms used for the 1988 Political Conventions).

D. FCC Contacts

E. Preliminary schedule of meetings and walkthroughs.

4. Conduct an RF site survey to determine the ambient RF level on frequencies of interest.

5. Bring portable analysis equipment (spectrum analyzers and receivers), in case you have to go on a 'fox hunt'.

6. Conduct a 'power up' exercise to be certain that there are no surprises.

7. Avoid the use of wide-band equipment, particularly on the lower bands.

8. Establish and maintain a power limit on all bands.

9. Establish a pool of backup channels on each band. Rather than each user having his own backup channels, it is more efficient to have a common pool available, since it is unlikely that all users will need their back-up channels at the same time.

10. Utilize 30 degree isolation of microwave antennas where possible.

11. Utilize cross-polarization of adjacent microwave channels where appropriate.

(See instructions before filling these boxes. Please print in large bold letters.)

Band Designation C or Ku	Identifier	Convention City

Subcommittee on Satellite Frequency Coordination

1988 Political Conventions Frequency Coordinating Committee

This form should be used to register all satellite uplinks carrying traffic associated with the 1988 political conventions. The convention frequency coordinator will use this information to verify frequency coordination for uplinks requiring frequency coordination, and will use this information to prevent, or rapidly resolve interference issues caused by the unusually high satellite loading associated with the political conventions. A separate copy of this form should be completed for each uplink in each city. Please complete this form as soon as possible even if some information is not yet available, then file an amended version when site information is complete.

1. Organization Name: ________________________

2. Contact Person(s): ________________________

3a. Telephone number before convention: ________________________

b. Telephone number at convention: ________________________
c. Telephone number at uplink: ________________________

4. Convention city this form is for. Please enter here and in the box at the top of this form in bold print: ____________

5. Convention city Hotel: ____________ Telephone: ____________

6. If this uplink will be used by a cooperative, or group effort formed for the conventions please enter the group name: ____________ Contact telephone: ____________

7. Uplink location in convention city: ________________________
(If uplink will be used at more than one location, please state all expected locations.) ________________________

8. Physical description of uplink: ________________________
(Truck type, Logos, fixed facility name, etc.)

9. Enter the uplink band that this uplink will use in the box at the top of this form. Use either "C" or "Ku".

10a. For C-Band uplinks enter the organization that has performed the uplink frequency coordination for this uplink at this site:
____

b. Contact person at above organization: ________________________
c. Contact person telephone: ________________________

Figure 4A

11. Enter a description of the type of transmission that will be employed: ______________________
(Such as "Video", "SCPC Analog audio", etc.)

12. Transmission bandwidth: ______________________

13. How will this uplink identify its transmissions: ______________

14. Please enter in the box at the top of page one a short identifier that you will use when the uplink is on the air, but not sending program material, such as "ABC 5" or "Conus 2".

15. Maximum available transmitter power in Watts: ______________
Typical power in Watts: ______________________

16. Please fill in the table below showing your expected use of satellites and transponders. You may also submit your schedule.

Parameter	Satellite #1	Satellite #2	Satellite #3	Satellite #4
Satellite Name				
Satellite control center phone #				
Uplink Frequencies that are likely to be used: (Enter in MHz followed by an H or V to designate uplink polarization)				
Schedule Information				

Thank you for your help and good luck at the conventions.

Figure 4B

Thanks

Thanks is given to all who actively contributed their time and effort to insure that all broadcasting communications went "on air" without a glitch.

Special thanks is given to Martin Meaney of NBC. Martin's decades of RF experience went into the planning of the entire committee. Special thanks is also given to Stan Baron of NBC, whose typical "day" "lasted from 7:00 A.M. to 1 A.M. the next day.

Thanks also to Committee co-chairs Joe Russo (ABC) and Rich Harvey (CBS); Atlanta Frequency Coordinator Ernie Watts (WSB); New Orleans Frequency Coordinator (WWL); Jim Durst, Alan Schneider, Angelo Ditty, and Jim Hawkins (FCC), and to Russ Abernathy (WYES).

I would also like to acknowledge the work of the following NBC personnel whose help went into planning of the Democratic and Republican National Conventions Frequency Coordination.

Tony Cassano - for his software development.

Maryanne Kollmann - for maintaining the contact list for all the broadcasters, cable and government personnel.

Ellen Royce - for maintaining the coordination database for the entire political convention process.

OPERATIONAL CONSIDERATIONS OF SATELLITE NEWS GATHERING

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ABSTRACT

Satellite News Gathering has made the transition from being a new phenomenon to being one of the many established tools on which a news organizations relies. Although there are many factors to consider in choosing, operating and staffing a SNG system, these factors are fairly well defined. The issues fall into various categories including technical, business and political. Several of these are discussed here.

TECHNICAL

The technical considerations of SNG revolve around the ability of the equipment to do the job and ability of the technician to operate the equipment as designed.

Link Budgets

For each end-to-end satellite path, the person planning the satellite transmission must prepare a link budget to determine what kind of equipment is necessary to produce the desired performance levels. The link budget takes several factors into account: the power of the transmitter, the size of the uplink antenna, the characteristics of the satellite, the size of the downlink antenna, the capabilities of the receiver, and many other miscellaneous factors.

Saturation

One of the key issues of the link budget analysis is the question of whether the uplink can "saturate" the transponder. Satellite transponders consist of a receiver, a frequency converter and a transmitter. As the strength of the signal received by the transponder increases, the strength of the signal transmitted increases, but only to a point. This point is called saturation which means that the transponder's transmitter is at maximum power. Beyond this point, no matter how much stronger the received signal gets, the transmitted signal can get no stronger. It is common to calculate link budgets so that the transponder is operating at, or just below saturation. This is done to get maximum efficiency from the satellite portion of the link.

Signal to Noise Ratio (S/N) The bottom line of a link budget (done for a video signal) is the signal to noise ratio of the picture. From this it is possible to determine whether the equipment and satellite are sufficient to provide a picture of the desired quality. In some instances the link budget may dictate the use of a larger receive antenna or a narrower bandwidth than was originally planned. In some SNG situations, it may be difficult to saturate the transponder with a small antenna and transmitter (i.e. when using a fly-away). It is possible to attain an acceptable video signal to noise ratio without saturation if other factors in the end-to-end link compensate.

Equipment Compliance

The SNG equipment must meet various criteria in order for the SNG operator to be able to achieve proper performance and acceptable levels of interfere with other users on the same and adjacent satellites. Some of the key areas of concern here are antenna sidelobes, intermodulation products, and equipment alignment.

Antenna Sidelobes The transmitting antenna must focus as much energy as possible on the desired satellite and as little as possible on other adjacent satellites. The specific requirements which must be met are prescribed by FCC CFR 25.209 for domestic satellites (Intelsat requires its users to meet the same standards). 25.209 requires that the energy radiated from the antenna not exceed the 29 - 25 log Θ curve close to boresite, with other requirements further off axis. Each antenna should be tested and certified to comply with the specifications.

Intermodulation Products The transmit electronics must not create any spurious emissions on frequencies outside of the assigned range. Any non-linearities in the uplink chain of equipment could cause components of the desired signals to interact with each other and produce unwanted intermodulation signals at frequencies on other transponders on the satellite.

Equipment Alignment

In order to achieve acceptable performance from the SNG equipment, the SNG operator must align and operate the equipment in compliance with the specifications provided by the manufacturer.

Antenna Alignment The satellite antenna must be aimed to the proper satellite and adjusted properly. The operator can determine the proper aiming direction by using a known algorithm or by consulting a simple conversion chart. This calculation or chart takes into account the antenna's longitude and latitude, as well as the west longitude of the satellite. This will produce the azimuth and elevation directions in which to aim the antenna. The operator then uses a compass and inclinometer (manual or automated), to find the area of the sky where the satellite is.

The next step is to determine a frequency on which there is a known signal being transmitted by the satellite, and then to monitor that frequency with a spectrum monitor or some type of receiver. The operator then "peaks" the azimuth and elevation positions by sweeping the antenna back and forth and up and down alternately until the signal appears and is at its strongest. During this peaking operation, it is necessary to move the antenna beyond where the signal fades (in all directions) to insure that the antenna is not receiving on a sidelobe. This step is crucial to get the optimum performance from the system for both uplink and downlink. If the antenna is not peaked properly, crosspole performance will not be optimal, the power needed to saturate the transponder will be greater, and interference to adjacent satellites may exceed acceptable levels.

Crosspole Domestic US satellites employ linearly polarized transmissions. This means the microwave signal transmitted to and from satellites is oriented in a plane. These planes can be thought of as horizontal and vertical. If there is a transmission in the vertical plane there will be no energy in the horizontal plane which is perpendicular or orthogonal. It is possible to transmit two different signals between the same two points on the same frequency if the signals are orthogonally polarized to each other. It is also possible to orient microwave signals clockwise and counter-clockwise as is done on some Intelsat satellites. Both of these approaches are called frequency reuse.

When setting up an antenna it is important to orient it so that the signals in the desired polarization are received while those in the other polarization are rejected. This is usually done by rotating the feed manually or remotely while observing a spectrum monitor or receiver. The optimum approach is to rotate the feed until the signals in the unwanted polarization are lowest or nulled. Once this is done it is possible to further peak the polarization by transmitting to the satellite and monitoring the return signal while in contact with the satellite operator.

Deviation The satellite video signal is Frequency Modulated (FM) and thus the signal swings or deviates back and forth around a center frequency. It is possible to cause this deviation to be greater or less for the same signal. There are advantages either ways. If the deviation is greater, more information can be transmitted (i.e. higher resolution in a video picture). However, to receive the increased amount of information (and thus bandwidth) the receiver must be set to a wider IF noise bandwidth. This allows the receiver to see proportionally more random noise in addition to the signal. The result is lower signal to noise ratio. The bandwidth and deviation are taken into account in the link budget and can be compensated for by other factors. It is advantageous if an operator can use a fixed deviation every time he uses a certain type of transponder or half transponder.

Subcarriers

The audio which accompanies the video is added to the video signal as a subcarrier above the highest frequency of relevant components of the video. Some commonly used frequencies for subcarriers are 5.7, 6.2, 6.8 and 7.5 MHz. In addition to the frequency, other considerations include the power and deviation of the subcarrier and the bandwidth of the subcarrier demodulator. The power and deviation of the subcarrier are figured into a link budget for the subcarrier and for the main signal (which is degraded by the amount of power and deviation the sub carrier takes from it). Often more than one subcarrier is placed on a video signal for more audio channels or data. Every time a subcarrier is added the main video signal is degraded.

Dual Video

For many reasons it has become desirable to put video signals from two SNG uplinks on one transponder at the same time. It is possible to do this and has been done for some time by Intelsat which sells Half Transponder Channels. Because of the AM-PM characteristics of the non-linear transponder there tends to be chroma interferences from each of the signals on the other. This is much more prevalent on C band and is almost unnoticeable on Ku band. The situation can be corrected on C band by genlocking or pseudo-genlocking the signals together.

Power Sharing When two or more signals pass through one transponder, the available power in the transponder is shared between the signals. As the input power to the transponder of one signal is increased, its output power will increase while the output power of the other signal or signals will decrease. This is known as small carrier suppression. It is, therefore, very important that each of the two parties using a dual carrier transponder adheres to a pre-determined uplink power level so as not to degrade the signal of the other.

Back Off In a power sharing situation it is necessary to operate the transponder slightly below saturation (backed off). This is due to the fact that the transponder's non-linear characteristics become much greater as saturation is approached. Non-linear characteristics are those which cause multiple signals to produce unwanted intermodulation products. The exact amount of back off depends on the characteristics of the transponder and the signals passing through it.

Access During Other Carrier Operation When one SNG uplink is already using a transponder and an other wants to begin operating in the other half of the transponder, it is necessary to have established the correct power levels and verify that the first signal is at the assigned level. Then it is possible to add the second signal to the transponder starting at very low levels with no modulation and increasing the power gradually while monitoring and adjusting the cross pole until both signals are at the same level. This is done by using the video averaging setting on the spectrum analyzer or monitor.

SNG EQUIPMENT CONFIGURATIONS

There are various ways in which SNG equipment can be configured. The most common of the approaches is to mount the equipment in a vehicle, put it in flight cases or both.

Vehicle

Equipment is usually mounted vehicles, and these vehicles vary greatly in size and features. Generally there is an antenna on the top or back of a truck with racks of electronics inside. The size of the vehicle can determine its utility: a large truck can hold more equipment but may not be able to reach remote areas off road.

Coverage Area A truck can cover an area with a radius of one to two hundred miles by ground with a response time of less than four hours. If a greater coverage area or a faster response time is necessary, it may require that the truck be loaded onto an aircraft and flown to or near to the event site.

Editing and Other Facilities Vehicles generally have some room for additional equipment, such as videotape editing equipment. This allows the SNG crew to post-produce a story that is not covered live and send only the finished product by satellite. The savings in satellite time can be substantial over sending the raw footage and having it edited back at the newsroom. It is also possible to outfit a truck with microwave gear so that an SNG truck can double as ENG when close enough to the newsroom for a microwave shot.

Fly-away

An other configuration for SNG equipment is in flight or shipping cases. This configuration is known as fly-away.

Configuration A fly-away generally consists of three to six cases of electronics with two to six cases for a segmented antenna and its mount. Once the fly-away has arrived at the location of the event to be covered, the electronics cases are stacked, the antenna is assembled, and the electronics and antenna are connected. A tent or awning is often used to protect the electronics and operators from the elements.

Redundant It is possible to include more electronics in the fly-away package to provide redundancy of critical equipment. This allows the transmission to proceed even if a transmitter has failed. The redundant equipment can also be configured to allow simultaneous feeds on different transponders or to increase the power of the transmission to the satellite.

Power The output power (EIRP) of a fly-away is generally less than that of a truck or other SNG configuration with a larger antenna. This is because, generally, larger antennas have higher gain figures and thus can deliver more power to the satellite. It must be remembered that it is not always necessary to saturate the transponder to get an acceptable video signal, especially if the downlink antenna is large.

Size and Weight One of the key factors in determining the usefulness of a fly-away is the size and weight of the largest shipping case. The airlines place restrictions on the size and weight of articles they will handle. They also sometimes charge a fee for items larger than a specific size. It is imperative that the largest of the shipping cases not exceed the weight or height limit. It is worth checking with the airlines likely to handle the fly-away in question, but the general limits are 100 linear inches and 100 pounds. It is often possible to slightly stretch the airlines limits given the right motivation to the people responsible for baggage. The criticality of these factors can not be stressed enough since the fly-away is totally useless if it can not be gotten to the event to be covered.

Transportation The fly-away can be transported by plane, train, automobile, aircraft, helicopter, boat, donkey or people. The advantage of the fly-away is that it can go to places where a truck can not go. The disadvantage is that the equipment and operators have little or no shelter from rain or sun.

Convertible

It is possible to design an SNG package to fit into either a vehicle or a fly-away configuration. Such a system is known as a convertible. The convertible has the advantages of both worlds. The vehicle provides protection for the equipment/operators and also provides extra space for additional equipment (i.e. editing). The vehicle can be driven if there is an event to cover within driving range, or the equipment can be removed and put in flight cases if the event is in a location beyond the truck's timely reach. It is also possible to outfit a vehicle in such a way that the redundant equipment can be removed for the fly-away and two single thread systems can operate simultaneously.

COMMUNICATIONS

The SNG operator needs to communicate with other key parties such as the satellite operator and the newsroom. It is often not possible to get landline telephone service to an event in time or at all. For this reason, it is desirable to have a communications channel through the satellite. SNG consortiums and satellite operators have made provisions for these needs by putting telephone communications systems on satellites used for SNG. These telephone systems can be accessed by SNG operators who have the interface equipment in their SNG vehicle or fly-away. Some of the systems have a pre-assigned frequency used to make initial contact with the satellite operator. Once this initial contact is made, the satellite operator assigns the SNG operator another frequency to use during the feed. Other systems are automated and use features such as automatic demand-assignable multiple-access frequencies.

Remote PBXs

It is desirable to have multiple telephone communications channels available to the SNG operator at the event site. Lines may be needed for engineering, IFB, local directors, reporters, producers, and others. To meet these needs it is desirable to have a PBX included in the SNG package. The PBX should be able to connect to the satellite telephone channels as well as landlines and use either or both simultaneously.

SATELLITE ACCESS AUTHORIZATION AND BUSINESS

It is necessary to make prior arrangements in order to use a satellite transponder. Generally there two kinds of arrangements, technical and business. Each satellite owner and/or operator will require that all uplinks accessing their satellite meet certain technical criteria. It is necessary to make arrangements with the satellite operator or a transponder resaler to book, use, and pay for the satellite time.

Domestic

Domestic US satellite owners and/or operators will often require a test transmission during which the uplink operator must demonstrate compliance with technical criteria. Once the uplink operator has successfully completed the test, some satellite operators will require that the user complete a brief test prior to each transmission. Others have less stringent requirements. It is particularly difficult for an SNG operator to test each time before he transmits if the whole transmission may be five minutes or less. The entire transmission would be taken up with the test. Often testing can be completed earlier in the day if the SNG operator is on site. A shortened version of the test can be completed in one or two minutes before the transmission if the satellite operator is familiar with the SNG user.

International

When in other parts of the world, it will probably be necessary to use satellites other than US domestic. If US satellites are used from nearby countries such as

Canada and Mexico, a Transborder agreement is necessary. The Transborder agreement allows international use of a domestic satellite and must be obtained from the FCC which makes arrangements with the appropriate agency in the other country. There are various international satellites and transponders available including Intelsat, Brightstar and Pan-Am Sat.

Intelsat The most prevalent international satellites are owned by Intelsat and are over the Atlantic, Indian and Pacific Oceans. The Atlantic and Pacific Ocean satellites can be used to get a story back from many parts of the world in one hop. That is, it is possible to feed from a SNG uplink in an other country to the Intelsat satellite and downlink the signal at the television station or news agency in the US.

SSOG Intelsat requires (in all but the extenuating cases) that the uplink pass a Satellite System Operations Guide (SSOG) test. This test certifies that the uplink meets all of Intelsat's criteria for performance and interference. Once this is done the uplink is added to a list of uplinks approved for use on Intelsat satellites. One area that the Intelsat is concerned with (that some of the other satellite operators are not) is equalizing the uplink to compensate for group delay in the transponder.

TV Bookings Intelsat time can be booked for any uplink which has passed the SSOG test by contacting Intelsat TV Bookings.

Signatory Arrangements In many instances a country has a government agency which is their Intelsat Signatory. This agency may have the exclusive rights to transmit to an Intelsat satellite from that country. It will then be necessary to make arrangements with the signatory for permission to transmit from that country. These arrangements can range from a verbal approval over the telephone, to paying a fee, to giving the equipment to be used to the country or their signatory. It is best to begin working on these arrangements as soon as possible.

Foreign Satellite Operator

It may sometimes be necessary to use a foreign countries' domestic satellite either with a Transborder agreement directly back to the US or as the first hop to an international earth station. In either case, it is necessary to contact and make technical and business arrangements with the foreign satellite operator. If it is necessary to use the foreign satellite as the first hop to an international satellite, it will also be necessary to arrange for downlink at the international uplink.

PTT Arrangements

Often the Intelsat signatory will be the Public Telephone and Telegraph (PTT) and it is with them that arrangements must be made for uplinking to the Intelsat satellite. It is also with the PTT that arrangements are often made with to backhaul the feed from the source to an international uplink by land line or microwave. In some countries the PTT has a monopoly and all telecommunications into, out of, or inside the country must be sanctioned by the PTT.

C vs Ku Band Coverage and Availability

There is satellite coverage over a good part of the surface of the earth. The coverage consists of service from international and domestic satellites of many countries and also of C and Ku band coverage. Some of the less developed parts of the world are served only by C band satellites or have no coverage at all. SNG users may not be accustomed to using C band satellites or equipment. Generally C band equipment is bigger and requires more power. An advantage is that the C band signals are less susceptible to degradation/interruption by rain. There are many configurations of C band equipment which are suitable for SNG. There are large tractor trailers with built-in electronics and antennas as well as smaller trucks pulling antennas on trailers. The latter configuration can, in many cases, be put on a cargo plane and flown to an airport close to a news scene. There has been discussion of outfitting airplanes with equipment to report from news scenes or even from the air over the event.

Spot Beam Up and Down

The antennas on satellites are directional to varying degrees and can be aimed to specific points on earth. Many of the US domestic satellites are equipped with antennas which allow them to receive and transmit to the entire continental US. There are also domestic and international satellites which have much smaller coverage areas such as the north east part of the US. These smaller coverage areas are usually called spot beams. In many cases it is possible for the satellite operator to direct these spot beams to various locations within a larger area. In this way powerful coverage and sensitive reception are available to remote areas as long as it is not needed in more than one place at a time. If there is reason to cover an event in a remote location which does not have regular satellite coverage, it is advisable to check with Intelsat or the satellite operator for the area about the possibility of getting a spot beam swung over to the location. It is possible to have a spot beam on a location which allows transmission from the site but no reception at the same satellite. In this case it would be necessary to locate the satellite by a beacon or some other signal on the satellite and it would not be possible to watch a return of the video signal transmitted.

POLITICAL

Many of the factors that effect SNG are political and thus not as easily controllable as the technical and business issues.

Customs

When entering a roreign country with SNG equipment (or any sophisticated and expensive equipment), the local Customs will want to know what is intended. In some cased (i.e. with eastern block countries) the US Department of State will have questions and regulations. It is usually possible to get an ATA Carnet which allows goods to enter a country with the understanding that they will leave again within a fixed amount of time. If a Carnet is obtained there are usually little or no duties to be paid.

World Conditions

Many of the news events which would dictate the use of SNG equipment in other parts of the world are in the form of some kind of disaster, natural or man-made. Although it seldom is possible to consider not going to cover the event, it is possible to consider the safest way to go.

Natural Disasters In the case of an earthquake or flood it useful to have a means of getting supplies to and from the site since food and fuel may not be available locally. It is best to plan to establish a communications channel, possibly on the same satellite being used for the news feeds. Bring things that the type of disaster dictates, boats to floods and shelter to earthquakes.

War, Terrorism and Military Overthrow

War, terrorism and military overthrow are politically inspired events and the motivations should be evaluated when making plans to cover them. If, for instance an American airliner is hijacked and the passengers are being held hostage it would probably be best to find an SNG crew of non-Americans to cover the story. This will reduce the danger to the crew and allow the event to be covered without shaping the news as would happen if one of the crew was captured and became another hostage. The US Department of State maintains a database on political conditions for various spots around the world and the implications for Americans traveling there.

Nationalization of equipment Some lesser developed countries may be motivated to allow the SNG equipment into the country but refuse to allow it to exit at the completion of the coverage. The best way to avoid this is to document the movement of the equipment into the country and if necessary consult the US Department of State.

CONSORTIUMS AND NETWORKS

There are many consortiums and networks of television stations and news organizations which cooperate on Satellite News Gathering. The advantages to the individual user include negotiation for bulk transponder time from the satellite operator, ability to share stories and facilities without each station sending equipment and crew to an event, and a single communications system specified and operated by the consortium for all. For instance, some television stations have multiple affiliations with a network and a regional cooperative. In this case they may have one type of communications equipment to work with each group.

INDUSTRY ORGANIZATIONS

It could be beneficial for an industry group to be formed that reports back to the National Association of Broadcasters, Society of Satellite Professionals International (SSPI), Satellite Operators and Users Technical Committee (SOUTC) and possibly others to establish recommended engineering practices and operations procedures for satellite uplinking. This would be particularly useful since satellite users cross the boundaries between many industries from broadcasting to banking and each has its own set of requirements and problems. It would be advantageous if there were a standard minimum level of knowledge that could be assumed for all satellite users. Another potential is a universal plan to readily identify and contact a user of one satellite who may be interfering with a user of an adjacent satellite.

Recommended Practices

There may be some benefit to setting some industry-recommended practices such as deviation of uplink signal, bandwidth of the receiver for full and half transponder operation, and frequency of the subcarriers. There are already some fairly standard procedures but decisions have to be made on how to do things each time a new consortium is started or truck is built. There should be some recommendation on which to base the decision with fairly universal compatibility being one of the factors.

User Training and Certification

As part of the deregulation that has taken place at the FCC, the operator licensing requirements for broadcasting and satellite uplinks have been dropped. At present the NAB and SOUTC have just established a training certification program for technicians and operators in the Satellite Uplinking field. However, this program is currently voluntary. This leaves the door open to the possibility that SNG operators may not all possess the knowledge and skills necessary get optimal performance from a system or avoid interfering with an other satellite user. It may be worth considering a mandatory program.

CONCLUSION

The arrival of Satellite News Gathering greatly expanded the ability of news organizations to reach out across the globe. However, this new capability brings with it new challenges and demands on broadcasters and news organizations. There are many concerns that must be appreciated and addressed collectively by the satellite community if these new tools and those of other industries are to work safely and at their fullest potential.

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NECESSARY ENVIRONMENTAL CONCERNS FOR BROADCASTERS

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ABSTRACT
We work in a world of regulations. Regulations can easily exist today which are not known but must be followed. As broadcasters we are well aware of the FCC and FAA rules which companion our day-to-day business activities. What is not as apparent are the federal EPA and OSHA rules which affect broadcasters. Most states, many larger cities, and even some counties have parallel rules which often contain additional requirements. Fines can be imposed for rule violations, even though the rules are not well publicized and therefore unknown. This brief overview lists rules, agencies, dates of compliance, and where to find more information.

DID YOU KNOW that the Environmental Protection Agency (EPA) and/or the Occupational Safety and Health Administration (OSHA) have final rules which affect broadcasters, covering:

- Asbestos,
- Air Contaminants,
- Underground Storage Tanks,
- Hazard Communications Standard,
- Electromagnetic Radiation (EMR),
- Occupational Noise Exposure, and
- Polychlorinated Biphenyls (PCBs).

Did you know that OSHA has proposed standards which will affect broadcasters including:

- Electrical Safety-Related Work Practices,
- The Control of Hazardous Energy Sources (Lockout/Tagout), and
- Generic Standard for Exposure Monitoring.

Additionally, you can reasonably expect OSHA to propose rules concerning video display tubes and the EPA to promulgate rules regarding the use of halon.

The goal of the OSHA Occupational and Health Standards is to make the workplace safe and the EPA rules are designed to protect the environment. These rules are not specifically aimed at the Broadcaster, but are for industry/business in general. The broad rule application is part of the problem. The EPA and OSHA have not gone out of their way to inform. For example, in the case of PCB's, the EPA rules name the power and food industries as PCB users but no mention is made within the rules of the broadcast industry. It wasn't until a PCB transformer was broken open in Oregon that broadcasters became aware of the PCB problem. Funding may be part of the notification problem, considering there are so many businesses affected and not enough staff assigned for the purpose of informing and enforcing. To help spread the word, the media will probably receive requests to outline OSHA and EPA rules as well as report fines for rule violations.

INFORMATION SOURCES are available that publish the EPA and OSHA Rules, but they are not generally distributed and usually must be bought. Obviously the rules will not even be sought if it is not known that they exist. As an excellent secondary source, NAB has provided member stations with useful information concerning underground storage tanks (UST's), asbestos, PCB's and EMR. Some law firms are now notifying their clients of compliance requirements. Newspapers also serve as a secondary source, but since the articles are usually written to provide general information it is easy to miss what isn't specifically directed toward the broadcast industry. Not knowing that rules exist will not prevent being fined if a penalty is applicable.

The primary source of the OSHA and EPA rules is the Code of Federal Regulations (CFR) which is the general and permanent collection of rules published by the Federal Register (FR). The Code is divided into 50 titles (example: title 29 for Labor), each title is divided into Volumes (example: title 29 has eight volumes), each title is divided into chapters which usually bear the name of the issuing agency (example: CHAPTER XVII Occupational Safety and Health Administration, Department of Labor) and each chapter is further subdivided into numbered parts (example: volume 5 is title 29, contains parts 1900 to 1910 and is all CHAPTER XVII).

Each volume of the Code is revised at least once each calendar year. If you purchased at $17.00, volume "47 CFR PARTS (70-79)" dated October 1, 1988 so you would have the Federal Communications (FCC) Rules Part 73, you know it is not up to date. So you also subscribe to a service that sends you FCC updates; you hope to learn of major changes from NAB; you subscribe to the Federal Register (FR) at $340 a year, or some combination of the above.

And so it is with the EPA and OSHA rules. After paying $28.00 for title "29 PARTS 1900 TO 1910", Revised as of July 1, 1988 you don't have all of it because "29 CFR Part 1910 Air Contaminants; Final Rule" was published in the Federal Register January 19, 1989....all 651 pages.

If you want to know if any changes have been made to the Code, you can subscribe to the "List of CFR Sections Affected" (LSA) for $21.00 a year. The LSA is included with a subscription to the FR. Upon learning from the LSA that a change had been made, it would be necessary to pursue the change in the Federal Register. Some local libraries subscribe to the Federal Register.

It would be very helpful if someone published an all-encompassing compliance manual. I am not aware of one, which is why this information was collected. From the number of topics, you know that I am obviously not an expert. With so much environmental and occupational information being released, the dissemination of environmental information within most companies is fragmented because no one is in charge of the whole picture and there usually aren't volunteers. Some businesses have only learned of a regulation by chance and sometimes that exposure has been expensive.

At the end of each topic discussed here, references are listed so that you may pursue more sources of information. The best we can do presently is keep alert and question any hint that a compliance requirement applies to some aspect of our business.

ASBESTOS removal and disposal are both regulated by the EPA. and OSHA regulates the exposure to asbestos in the workplace.

The National Emissions Standards for Hazardous Air Pollutants (NESHAPS) require that asbestos material be removed when it is friable, that is, crumbled or in any form which could free it into the air. Asbestos is one of eight known naturally occurring carcinogens. It is second, only to smoking, in causing cancer.

Asbestos does not need to be removed if it is enclosed or encapsulated. However, eventually asbestos will need to be removed and many property sales are being conditioned on "no asbestos present".

Asbestos was used commercially as fire-proofing as early as the 1890's, as insulation in early 1900's, and at the height of its popularity in more than 3000 items, including such widely distributed products as brake linings and hair dryers. Although you can easily identify some material forms of asbestos, by color and consistency, final identification should be done by a lab which can also measure the quantity in a sample. Removal of asbestos must be handled by a licensed contractor and it is expensive. We deal with contractors who are experienced, have certified trained workers and supervisors, carry proper insurance, and have submitted a satisfactory work plan. We require the air to be monitored during removal. It is also a good idea to have occurrence type liability insurance for asbestos related work.

We demand certification of the disposal of asbestos in an EPA licensed landfill. We also inform our employees of what is going on. I believe employee notification is required under the OSHA Hazard Communication rules. And it doesn't end at the Federal level. Some States had regulations before the EPA or OSHA did. Not handling asbestos abatement properly can result in a health hazard, the delay of job completion, and fines. The OSHA asbestos standard states that medical surveillance must be provided to any employee exposed to an airborne concentration of asbestos in excess of 0.1 fiber per cubic cm of air calculated as an eight-hour time-weighted average. Further, medical records on employees exposed must be maintained by the employer, indefinitely if necessary, even after employment ends.

For additional information see:

"Dealing with Asbestos in TV and Radio Stations", NAB Info-Pak/December 1988.

"Occupational Exposure to Asbestos, Tremolite, Anthophyllite, and Actinollite;" Final Rules; Amendment, 29 CFR Parts 1910 and 1926. OSHA, FR September 14, 1988, pgs 35610-29.

"Asbestos, tremolite, anthophyllite, and actinolite", OSHA, 29 CFR 1910.1001, July 1, 1987, pgs 682-723.
"Asbestos", EPA, 29 CFR 763, July 1, 87, pgs 251-344.
"Consumers patching compounds containing respirable freeform asbestos ban", Consumer Product Safety Commission, 16 CFR 1304.
"Occupational Exposure to Asbestos, Anthophylite, and Actinolite", OSHA, FR, June 20, 1986, 29 CFR Parts 1910 and 1926, pgs 22612-790.
Contact;
Office of Toxic Substances, EPA, 401 M St. SW, Washington, D.C., 20460 202-554-1404.

AIR CONTAMINANTS are defined by OSHA in a final rule, (all 651 pages), and this rule became effective March 1, 1989 with the start-up date for compliance September 1, 1989. This rule modifies many of the existing Permissible Exposure Limits (PELS) of substances, establishes PELS for substances for which no previous exposure limits exist and includes Short Term Exposure Limits (STEL) to complement the 8-hour Time Weighted Average (TWA) limits There are 428 substances listed in the rule which include such items as wood dust and carbon tetrachloride. It is estimated that every 20 minutes a new and potentially toxic chemical is introduced into industry. Thus, we can expect this list of 428 substances to be increased as substance studies continue on new and existing substances. For example, the PELS for fibrous glass and mineral wool, two popular insulation materials, were not given in the 428 substance list. They are named as having the PELS delayed.

This current revised rule on Air Contaminants shouldn't have a significant impact on the broadcasting industry since there would not normally be exposure to many of the substances listed or at least in amounts to trigger concern. However, the only way to be sure is to review the list of 428 substances and determine if a substance is being used in significant amounts. As an example, wood dust is listed as causing respiratory effects and (except for Western red cedar) carries a TWA of 5 milligrams per cubic meter. This TWA can be met with proper ventilation. More serious are those sustances which can cause cancer, kidney or liver effects and other long term physiological concerns. It is obviously beyond the scope of this brief review to list the 428 substances listed and their health effects. Under the Hazard Communications Standard, discussed herein later, you should already be aware of hazardous substances being used in the workplace.

For additional information see:
"Air Contaminants", Final Rule, OSHA, 29 CFR Part 1910, FR January 19, 1989, pgs 2332-2983.
Contact;
OSHA Office of Public Affairs, Mr James F. Foster, Room N-3647, Department of Labor, 200 Constitution Avenue NW., Washington, DC, 20210. 202-523-8151
Copies of the Air Contaminants document may be obtained from the OSHA Publications Office Rm. N-3101 at the above address 202-523-9667, or any OSHA regional or area office.

UNDERGROUND STORAGE TANK REGULATIONS were published in final rule form September 23, 1988, in 165 pages of the Federal Register. An EPA Form 7530-1 or a similar State Certificate of Notification should already be filed with the environmental agency of the State. Beginning December 22, 1988, the owner/operator of an Underground Storage Tank (UST) needed to begin keeping records which include the date and nature of inspections, repairs, and replacement of UST's in use. Within a ten year period of December 22, 1988, depending on the age of the tank, the UST's must be made corrosion resistant. After October 24, 1988, any person who sells a tank to be used as an underground storage tank must notify the purchaser of the requirement to report bringing the tank into use.

An underground storage tank is defined as a tank and piping which has at least 10 percent of its volume underground. There are a few EPA exclusions (not necessarily State exclusions) to the definition of UST's, which at least presently include;

- -Farm or residential tanks of 1,100 gallons or less capacity storing motor fuel for noncommercial purposes.
- -Tanks storing heating oil for use on the premises where stored.
- -Septic tanks
- -Storage tanks situated on or above the floor in basements
- -Tanks that have a capacity of less than 110 gallons (not excluded in the Ohio State regulations for example)

The UST rules define new tank design, construction, and installation. Also defined are the requirements for leak detection, spill prevention, record keeping, reporting, tank closure, and the corrective action for spills.
Under the Superfund Amendments and Reauthorization Act of 1986, SARA, the EPA is given authority to clean up petroleum releases or to require owners or operators to do so. If the EPA cleans it up, don't expect them seek a low bid and you can expect an invoice with a fine.

An EPA study indicated that 42 percent of the tanks which were 15 to 20 years old and 30 per cent of the 10 to 15 year old tanks were leaking. Release detection at existing UST systems must be added. Older tanks which are unprotected from corrosion must have release detection within one year from December 22, 1988, with the newest tanks that are protected from corrosion to have release detection within 5 years. A protected tank is one with proper cathodic protection or a tank made of an acceptable material, such as fiberglass over steel. Periodic tank tightness testing every five years combined with monthly inventory control is allowed at new tank installations for ten years after installation, but after 10 years, monthly release detection is required. Monitor wells are a method of leak detection and are required by some states and may be in addition to whatever other method is used.

Tank owners and operators must report actual and suspected releases. Owners and operators of leaking UST systems must follow measures for corrective action. Cleanup levels will be established on a site-by-site basis as approved by the implementing agency. Tanks over 10 years of age must be either internally inspected or lined to meet upgrade requirements. This procedure may not be practical for small tanks. It may be more practical to replace an older tank with a new protected one. When a new tank is installed, the installer must certify that the UST system was installed according to the rules. All substandard existing UST systems must be closed, replaced, or retrofitted with corrosion protection within 10 years of December 22, 1988 which is also the years of grace before adding overfill protection to all tanks. The required overfill protection leaves much to the discretion of the operator but to wait 10 years before installing protection burdens the environment and could burden the operator financially if there is a spill. Protection can be as simple as automatic supply shut off when the tank is 95 per cent full. At some locations overspill protection, such as a retaining basin, may be appropriate. If more than 25 gallons are spilled, the spill must be reported. No exceptions here.

In all cases, the spill or overfill must be cleaned up immediately, and if not, even spills under 25 gallons must be reported.

This is only a small sample of the regulations. In addition to the EPA rules, there are State/City regulations which additionally may impose requirements not contained in the EPA rules. Many States had rules which were in effect before the EPA wrote theirs. For example, although tanks associated with emergency generators are presently deferred from release detection by the EPA, they are not deferred in Florida. It really is not wise to defer meeting the requirements of the new rules since the consequences of a spill or leak can be severe to both the environment and the bottom line. The cost to replace a 1000-gallon storage tank in Cleveland, including monitor wells, is around $10,000. If it would be necessary to remove tons of contaminated soil, the costs could be unaffordable.

If a tank is leaking, it must be replaced and any contaminated soil removed. It may be a better choice to replace aged tanks rather than wait until a leak starts. If a tank is buried and not leaking and its removal would be too expensive or not practical, it may be emptied by removing liquid and sludge, and then filled with inert solid material. The choice of inert fill material is not specified. The EPA accepts sand and concrete, some states do not accept both. However, concrete could cause a future construction problem.

Civil fines for violation of the regulations can be up to $10,000 a day with criminal fines topping out at $25,000 a day. Corporate officials face fines up to $250,000 with up to 15 years in prison, with corporate fines running to one million. Additionally, the cleanup costs of an underground water system could be astronomical.

The EPA addressed financial responsibility in the October 26, 1988, issue of the Federal Register. All owners and/or operators with a tangible net worth of 20 million dollars or more must comply by January 24, 1989, all others with lower net worth by October 26, 1990. This responsibility requires owners, such as broadcasters, to maintain financial assurance of at least five hundred thousand dollars per occurrence, 1 million aggregate, to insure that damage resulting from a leak will be paid for by the owner.

For additional information see:
"New EPA Requirements for Underground Storage Tanks", Counsel from the Legal Department, NAB Info-Pak/January 1989.
"Underground Storage Tanks Containing Petroleum-Financial Responsibilty Requirements and State Program Approval Objective", 40 CFR 280 and 281, EPA, FR, October 26, 1988, beginning page 43322.
"Underground Storage Tanks; Technical Requirements and State Program Approval;" Final Rules, 40 CFR Parts 280 and 281, EPA, FR, September 23, 1988, pgs 37082-37247.

Contact;
RCRA/Superfund hotline 800-424-9346 and in D.C., (202) 382-3000.

OSHA HAZARD COMMUNICATIONS STANDARD is a rule with compliance required by May 23, 1988. It applies to non-manufacturing industries and can even apply to white collar offices with a copy machine that uses a chemical which is listed among the more than 400 substances considered to be hazardous. Because of the disaster at Bhopal, India, Congress enacted Title III, the Community Right To Know Law under the Superfund Amendment and Authorization Act (SARA) of 1986. SARA was originated to cover the clean-up of hazardous waste sites under the Comprehensive Environmental Response, Compensation and Liability Act, (CERCLA). The EPA is also involved since it lists the CERCLA-defined hazardous substances and, among other things, also generates reporting forms. If all of this seems confusing, it is. It isn't apparent that there is any effort to consolidate and/or communicate all of the necessary reporting requirements to the millions of businesses which are affected. Most do not know they need to comply, or how. At Scripps Howard, we developed a written Hazard Communications Program at all of our stations and we believe we are in compliance. To comply, an employer must develop and maintain a written Hazard Communications Program, which explains and accomplishes the following:

1. Identify and list all of the hazardous chemicals in the workplace
2. Obtain, (usually from whom you bought the product) the material safety data sheet, MSDS. Any manufacturer who manufactures a product which contains one or more of the chemicals listed by CERCLA and EPA must prepare an MSDS. (The Office of Management and Budget under the paper work reduction act has exempted drugs regulated by the FDA in the non-manufacturing sector.)
3. Make the MSDS sheets available to the employees.
4. Identify those workers who should be trained and provide training to include the dangers of the substance and the correct and safe use.
5. Know what to do if there is a spill, fire, or personal contact.
6. All substances should be properly labeled and stored.
7. Appoint someone in charge of the program.
8. Any company required to have available MSDS's must submit copies of the MSDS's, or a list of MSDS chemicals to the State Emergency Response Commission, the Community Emergency Planning Commission, and the local fire department. Under some instances, an EPA-designed inventory report form must also be sent to the before mentioned three entities. Those were due March 1, 1988.

There was for a period of time a consumers product exemption which suspended the above reporting requirements if a product was not used differently in a business than in a home. This exemption has, at the time of this writing, been eliminated by the Third Circuit Court.

By July of 1988, 25 States had OSHA-approved state plans for their own hazard communication rules. There are Federal and State fines which may be imposed for some infractions.

For additional information see:
"List of Hazardous Substances and Reportable Quantities", CERCLA, FR, 3/16/87, pgs 8150-71.
"List of Extremely Hazardous Substances and Their Threshold Planning Quantities", EPA, FR 4/22/87, pgs 13395-410.
"List of Toxic Chemicals", FR, 6/4/87, pgs 21169-77.
"Hazard Communications; Final Rule", 29 CFR Parts 1910, 1915, 1917, 1918, 1926, and 1928, OSHA, FR 8/24/87, pgs 31852-86.
"Emergency and Hazardous Chemical Inventory Forms and Community Right-to-Know Reporting Requirements; Final Rule", 40 CFR 370, EPA, FR 10/15/87, pgs 38344-77.
"Toxic Chemical Release Reporting; Community Right To Know; Final Rule: 40 CFR Part 372, EPA, FR 2/16/88, pgs 4500-54.
"Extremely Hazardous Substances List", Final Rule, 40 CFR Part 355, EPA, FR 2/25/88 pgs 5574-5 and FR 12/17/87 pgs 48072-4.
"Hazard Communication; Display of Office and Management and Budget Control Numbers Assigned To Collection of Information", 29 CFR Parts 1910, 1915, 1917, 1918, 1926, and 1928, OSHA, FR 12/4/87, pgs 46075-80 and also FR 4/27/88, pgs 15033.
See 40 CFR 702, 704, 710, and 717 for more information.
Contact:
TSCA, Toxic Substances Control Act Assistance Office 202-554-1411.

ELECTROMAGNETIC RADIATION has been covered adequately in other forums. By now it is well known that the EPA has suspended efforts to establish a standard for human exposure. OSHA was depending on the EPA, although OSHA has a general population standard based on the current ANSI standard. If this regresses into another "let the market place decide" situation, broadcasters could find themselves dealing in a myriad of local, county, and state standards.

For additional information see:
"Evaluating Compliance With FCC-Specified Guidelines for Human Exposure to Radiofrequency Radiation", OST Bulletin No. 65, October 1985, Office of Science and Technology, Federal Communications Commission.
"A Broadcasters Guide To FCC Radiation Regulation Compliance", (includes above), National Association of Broadcasters.
"Request for Declaratory Ruling; Radiofrequency Radiation Compliance" General Docket 88-469;FCC 88-291, Federal Register, October 19, 1988.
"Power Line Fields and Human Health", February 1985, IEEE Spectrum.
"Biological Effects of Electromagnetic Fields", May 1984, IEEE Spectrum.

OCCUPATIONAL NOISE EXPOSURE rules became effective in 1983. A disc jockey probably won't turn down his headphone volume to be in compliance with the rules, even though the sound level exceeds the limit. If someone does listen to sound levels which exceed the OSHA levels, it would be a good idea to limit the power available to the headphones. A broadcaster who is asked to set up an earth-shaking PA system may want to become familiar with them. While normally operating audio systems would not be in violation, it is important to know that the rules exist and what the limits are.

For additional information see: "Occupational Noise Exposure", OSHA, 29 CFR 1910.95, 7/1/88.

POLYCHLORINATED BIPHENYLS (PCB's) are EPA listed as toxic and persistent. Based on animal data, the EPA concludes that in addition to chloracne, PCB's may cause reproductive effects, developmental toxicity, and tumors (oncogenicity) in humans. It doesn't matter if you think the case against PCB's may be overstated, under EPA regulations, get rid of PCB's by paying a licensed firm to dispose of them. Should a capacitor explode all over the inside of a transmitter, the spill must be cleaned up, all of the cleanup material disposed of properly, and the site must be certified or sampled depending on the quantity of the spill. A broadcaster could be off the air during cleanup if an uncontaminated standby facility is not available. And if PCB's are in a fire, the use of the building could be lost. Similarly, the EPA can impose severe fines for not following the rules under the Toxic Substances Control Act (TCA) which was passed by Congress in 1976. Any authority not granted to the EPA under TCA is given under the Resource Conservation and Recovery Act, (RICRA).

The final EPA transformer rule was published July 19, 1988, in the Federal Register. The owners of PCB transformers which are in or near a commercial building need to take action. As of October 1, 1990, all radial PCB transformers and those with secondary voltages below 480 volts, not located in sidewalk vaults must be equipped with electrical protection to prevent high current faults. As of October 1993, all transformers below 480 volts in sidewalk vaults in use near commercial buildings must be removed from service. The use of PCB transformers and capacitors is still permitted in TV and radio transmitters if the PCB's are contained, should even a leak occur, and there is no reasonable risk of injury to health and the environment. However, as stated earlier, there is risk in using, and as time goes on expect to find it more difficult and costly to dispose of PCB items.

Several recent articles have been published concerning the extensive rules pertaining to PCB's, and rather than repeat that information here, please refer to those articles since they are readily available.

For additional information see:
PCB Alert, NAB Today, September 22, 1986.
PCB Prohibition Deadlines, NAB Info-Pak, November 1987.
"Polychlorinated Biphenyl Spill Cleanup Policy; Amendments and Clarifications", Final Rule; amendment and clarification of policy statement, EPA, 40 CFR Part 761, Federal Register October 19, 1988, Pgs 40882-4.
"Polychlorinated Biphenyls in Electrical Transformers;" Final Rule EPA, 40 CFR Part 761, Federal Register July 19, 1988, Pgs 27322-29.
"Polychlorinated Biphenyls (PCB's) Manufacturing, Processing, Distribution in Commerce, and Use Prohibitions. EPA, 40 CFR Part 761, July 1, 1987. pgs 194-246
Brad Dick, "Managing The PCB Risk", Broadcast Engineering, October 1988, pgs 68-94.
Beth Jacques, "The Letter Of The Law", BME, September, 1988, pgs 58-9
Jack G. Pfrimmer P.E., "Identifying and Managing PCB's in Broadcast Facilities", Proceedings 41st Annual Broadcast Engineering Conference. NAB, 1987.
Contact:
Office of Toxic Substances, 401 M St. NW, Washington, D.C., 20460 800-424-9065, and in D.C., (202) 554-1404.

ELECTRICAL SAFETY WORK PRACTICES were proposed by OSHA November 30, 1987. In the proposal, electrical and electronic technicians and engineers were included in the category of employees facing a higher than normal risk of electrical accidents. Under paragraph 1910.333 of this proposal, the application of lockouts and tags on deenergized circuits is covered. Under another proposed rule-making, specifically titled, "The

CONTROL OF HAZARDOUS ENERGY SOURCES (Lockout/Tagout)", which was released April 29, 1988 of this year, lockouts/tagouts are covered in paragraph 1910.147 of that proposal. Obviously there is duplication.

Broadcasters will find themselves answerable to these rules. However, there doesn't seem to be anything unusual since we now observe many of the proposed practices anyway.

For additional information see:
"Electrical Safety-Related Work Practices", Proposed Rule, OSHA, 29 CFR 1910, FR 11/30/87 pgs 45530-45549.
"The Control of Hazardous Energy Sources (Lockout/Tagout); Proposed Rule", OSHA, 29 CFR 1910, FR 4/29/88 pgs 15496-528.
Contact:
James Foster, U.S. Department of Labor, OSHA, 200 Constitution Ave. NW, Washington, D.C. 20210, 202-523-8148.

GENERIC STANDARD FOR EXPOSURE MONITORING is a proposed rule which was published September 27, 1988. A generic standard is one that addresses a health related issue rather than a substance. These issues cover such topics as initial monitoring, whether to use personal or area sampling, or whether to use full shift or grab sampling. The generic standard is tied in with workplace medical surveillance. OSHA is questioning whether to require medical surveillance programs for all toxic substances in the workplace. The impact of these two proposals of exposure monitoring and medical surveillance could be felt by many businesses.

For additional information see:
"Generic Standard for Exposure Monitoring" and "Medical Surveillance Programs for Employees", Advance Notice of Proposed Rule Making, OSHA, 29 CFR 1910, FR 9/27/88 pgs 37591-96.
Contact:
Office of Information and Consumer Affairs. 202-523-8184.

IN SUMMARY, this brief collection of information and references has been pulled together to alert the broadcaster to the EPA and OSHA rules which affect the industry. This information is far from complete. Lengthy seminars are often given on just a single subject. Rule changes can be expected as well as expansions of the rules. Rules on subjects such as halon, video display tubes, and electric power generation from small generators can be anticipated from the EPA and/or OSHA.

PERSONAL SAFETY CONSIDERATIONS WITH BROADCAST TRANSMITTERS

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ABSTRACT

Personal safety must be a very important consideration in the design, operation, and maintenance of transmitter equipment containing high voltages, currents, and large amounts of energy storage. The equipment should incorporate adequate safety protection against accidental direct exposure to these dangerous potentials. This paper discusses the various aspects of personal safety, as well as different methods and devices used in a typical radio broadcast transmitter to achieve the desired safety level.

INTRODUCTION

Most people are concerned with safety one way or another in our daily lives and are generally safety conscious. This is particularly true in the broadcast industry. Yet sometimes safety is taken for granted. The question of safety gets little or no attention until the occurrence of a major safety related accident. Much of the responsibility related to safety rests in the hands of broadcast station engineers.

Personal safety must be a very important consideration in the design, operation, and maintenance of broadcast transmitter equipment containing high voltages, currents and large amounts of energy storage. The equipment should incorporate adequate safety protection against accidental direct exposure to dangerous potentials. More importantly, the broadcast engineering staff should be aware of the possible hazards and follow good electrical safety practices. This is especially important in today's highly competitive radio station environment where technical expertise is depleting at an alarming rate. This paper discusses the various hazards which may be encountered, the safety requirements for transmitting equipment including standards and the protective circuits, devices, and methods used in a typical broadcast transmitter to achieve the desired safety level.

SAFETY HAZARDS

The safety hazards which are of primary concern to broadcast staff are described below.

Electrical Shock

Current rather than voltage is the most important parameter which affects the intensity of electric shock. Three factors that determine the severity of electric shock are: (1) amount of current flowing through the body; (2) path of current through the body; and (3) duration of time the current flows through the body. The voltage necessary to produce a current dangerous to life is dependent upon the resistance of the body, contact conditions, and the path through the body [1]. The resistance of the human body varies with the amount of moisture on the skin, the muscular structure of the body and the voltage to which it is subjected.

Studies of adult human body resistance have indicated that under normal dry skin conditions hand-to-hand resistance varied typically from 6,600 ohms to 18,000 ohms and hands-to-feet resistance varied from 1,550 ohms to 13,500 ohms [2]. The body resistances of children were found to be generally higher. Higher voltages have the capability to breakdown the outer layers of the skin thereby reducing the resistance. In judging a product for safety against electric shock, Underwriters Laboratories (UL) uses a resistance value of 1,500 ohms under normal dry contact conditions and a resistance value of 500 ohms under wet conditions [2,3]. Based on research of Charles F. Dalziel, Professor Emeritus, University of California, Berkeley, the effects of 60 Hz AC (alternating current) on the human body, are illustrated in Table 1 [4]. The safe "let-go" currents generally accepted for 0.5 percent of population are approximately 9 and 6 mA for men and women repectively [5]. The "let-go" current is the maximum current at which a person is still capable of releasing a live conductor by using muscles directly stimulated by that current. Currents only slightly in excess of one's let-go current are said to "freeze" the victim to the circuit.

The maximum safe current specified by the International Electrotechnical Commission (IEC) is 2 mA DC (direct current) and 0.7 mA peak AC measured in a non-inductive resistor of 2,000 ohms connected between the part containing voltage in excess of 72 volts peak and ground [6].

Sufficient current passing through any part of the body will cause severe burns and hemorrhages. However, relatively small current can be lethal if the path includes a vital part of the body such as the heart or the lungs. The duration of current flow also affects the severity of injury. The effects of electrical current and time duration on the human body is illustrated in Figure 1 [4]. The current range previously noted in Table 1 which causes "freezing" to the circuit is also illustrated. It is obvious from Figure 1 that a 100 mA current flowing for 2 seconds through a human adult body will cause death by electrocution. Considering a minimum value of hands-to-feet resistance of 1,500 ohms, a current of 80 mA can flow if both hands are in contact with a 120V AC source and both feet are grounded. If this condition persists for more than 2 seconds, it may cause electrocution. The above data provides insight into the hazards of electrical shock.

Electrical and Radio Frequency Burns

Electrical burns are usually of two types, those produced by heat of the arc which occurs when the body touches a high voltage circuit, and those caused by passage of high current through the skin and tissue. In the latter case even the low voltage source(s) containing large amounts of energy can cause severe arcing or overheating if accidentally short-circuited with the possibility of injury to personnel and the risk of fire. This can occur when a metal part in contact with the skin such as jewelry or tool provides path for high short-circuit currents.

Radio Frequency (RF) burns are caused by the flow of RF currents through the skin when it is exposed to an RF energy source. The energy is absorbed by the resistance of the skin. The severity of burns will depend on the area of exposed surface, the amount of current flow, the voltage level, the frequency and the time duration.

Harmful Radiation

The two types of harmful radiation which may be encountered in and near the transmitting equipment are: (1) Non-ionizing Radiation and (2) Ionizing Radiation.

TABLE 1. THE EFFECTS OF 60 Hz ALTERNATING CURRENT ON THE HUMAN BODY

1 milliamp or less	-	No sensation, not felt.
More than 3 mA	-	Painful shock.
More than 10 mA	-	Local muscle contractions, sufficient to cause "freezing" to the circuit for 2.5 percent of the population.
More than 15 mA	-	Local muscle contractions, sufficient to cause "freezing" to the circuit for 50 percent of the population.
More than 30 mA	-	Breathing difficult, can cause unconsciousness.
50 to 100 mA	-	Possible ventricular fibrillation* of the heart.
100 to 200 mA	-	Certain ventricular fibrillation* of the heart.
Over 200 mA	-	Severe burns and muscular contractions; heart more apt to stop than fibrillate.
Over a few amperes	-	Irreparable damage to body tissues.

*NOTE: Ventricular fibrillation is defined as "very rapid uncoordinated contractions of the ventricles of the heart resulting in loss of synchronization between heartbeat and pulse beat". Once ventricular fibrillation occurs, it will continue and death will ensue within a few minutes. Resuscitation techniques, if applied immediately, may save the victim.

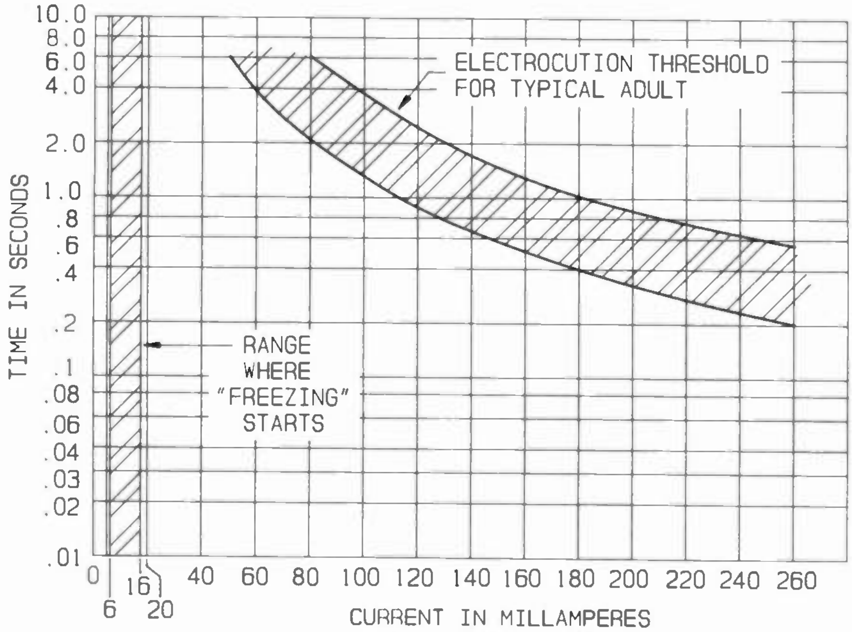

FIGURE 1. THE EFFECTS OF ELECTRICAL CURRENT AND TIME ON THE HUMAN BODY

Non-ionizing radiation may exist due to poor shielding of the transmitter equipment operating at high power levels or due to the proximity of antenna. Exposure to excessive non-ionizing radiation of radio frequency electromagnetic fields in the frequency range from 300 kHz to 100 GHz will cause heating of the body which in turn may have adverse biological effects. Studies have shown that whole-body-averaged absorption rates approach maximum values when the long axis of a body is parallel to the E-field vector and is four tenths of a wavelength of the incident field. At a frequency near whole-body resonance, which is about 70 MHz for the Standard Man, the absorption of RF energy is maximum [7]. Under 3 MHz, most of the energy will pass completely through the human body with little attenuation or heating effect. The dangers of non-ionizing RF radiation are most severe at UHF and microwave frequencies. Human eyes are particularly vulnerable to low-energy microwave radiation and blindness can result from overexposure. Cardiac pacemakers may also be affected by RF radiation.

Ionizing X-ray radiation may exist near high power tube transmitters depending on the work-function of the materials that the tube is constructed with. Typically X-rays are emitted from the copper anode at high voltages. As operating voltages increase beyond 15 kilovolts, power tubes are capable of producing progressively dangerous X-ray radiation [8].

X-ray levels should be checked at regular intervals for possible changes due to tube aging. Exposure to excessive ionizing X-ray radiation may damage human body cells with resultant biological changes due to dissipation of energy in body tissues. The levels of radiation, the exposure rate, and the length of time over which exposure occurs are closely connected with the nature and extent of any damage. The effect of ionizing radiation on matter is to release charge either by direct ionization or by the liberation of ionizing particles [9].

High Temperatures and Fire

The transmitting equipment parts may attain high temperatures under normal conditions. The external surface of power tubes operates at high temperatures (up to 200 degrees to 300 degrees centigrade). All hot surfaces may remain hot for an extended time after the the transmitter is switched off [8]. Thermal burns may result if the body skin comes in contact with hot surfaces. Hot water lines used for tube cooling in some transmitters may present a similar hazard. The temperature rise of some components under fault conditions may be excessive so as to cause injury to personnel. Staff should keep away from hot surfaces and should be aware of any possibility of fire or its spread and take necessary precautionary measures.

Other Hazards

Personnel should be aware of components which may cause danger due to implosion or explosion. These apply to components such as cathode-ray tubes, vacuum power tubes, electrolytic capacitors or glass fuses. Accidental breakage of vacuum tube glass envelope can cause an implosion, which will result in an explosive scattering of flying glass particles and fragments. This may cause serious personal injury [8].

Beryllium oxide ceramic material (BeO) is used as a thermal link to dissipate heat away from a tube or transistor. BeO dust or fumes are highly toxic and breathing them may result in serious injury endangering the life [8]. Polychlorinated biphenyls (PCBs) used in older oil-filled power transformers and high voltage capacitors are also hazardous. The Environmental Protection Agency (EPA) has established regulations (40 CFR Part 761) regarding the use and disposal of electrical components containing PCBs.

Care should be taken to prevent injury due to contact with moving mechanical parts such as fans, gears. Sharp projections or edges should be avoided to protect from cuts or abrasions. Exposure to excessive noise can cause damage to hearing and to the nervous system.

SAFETY REQUIREMENTS FOR TRANSMITTING EQUIPMENT

Safety Standards

Safety standards related to broadcast transmitter installations are found in the following publications:

a) International Electrotechnical Commission (IEC) Publication 215 contains the safety standard for radio transmitting equipment [6]. This is the only standard which specifically addresses the safety requirements for transmitting equipment.

b) The general safety standard used widely for reference purposes is the Military Standards, "MIL-STD-454K: General Requirements for Electronic Equipment, Requirement 1, Safety Design Criteria - Personnel Hazard" [1]. This standard establishes safety design criteria and provides guidelines for personnel protection.

c) Safety Standard which deals with permissible levels of human exposure to RF electromagnetic fields is contained in the American National Standards Institute document ANSI C95.1-1982 [7].

d) The National Electrical Code (NEC) is a comprehensive document that details safety requirements for all types of electrical installations. The National Electric Code or The National Electrical Code Handbook is published by the National Fire Protection Association (NFPA) [10].

e) Another NFPA publication titled "NFPA 79: Electrical Standard for Industrial Machinery 1987" provides detailed information for the application of electrical/electronic equipment, apparatus, or systems supplied as part of industrial machinery which will promote safety to life and property [11].

f) U.S. Department of Labor, Occupational Safety and Health Administration (OSHA) Safety and Health Standards 29 CFR 1910 contains design safety standards for electrical systems, safety-related work practices, safety-related maintenance requirements and safety requirements for special equipment [12].

Other publications related to safety are given in the reference section of this paper including the addresses to order any of the publications listed.

Safety Requirements

The principal design and construction requirements for safety of personnel during the installation, operation and maintenance of broadcast transmitters are discussed below. Major differences between existing standards are also highlighted.

(a) Protection against electrical shock and burns, including RF skin-burns.

(1) An effective grounding system is essential to prevent the possibility of electric shock. The equipment grounding is necessary to insure that all the external metal parts, surfaces, shields are bonded together and then connected to a safety ground by a low-impedance conductor of sufficient capacity to carry operating and fault currents. System grounding is required to connect one of the primary AC conductor and service equipment to ground, which then completes the ground-fault loop. Proper grounding also protects equipment from damage caused by AC line disturbances.

(2) A reliable main power disconnect switch for cutting off all power to the transmitter should be provided. The switch should plainly indicate whether it is in the open (off) or closed (on) position. Live conductors shall be protected against accidental contact. A fused type disconnect is preferred over circuit breakers by some broadcast engineers.

(3) Type of protection required to prevent accidental contact with different voltage levels is given in Table 2. Protection requirements specified by NFPA 79, MIL-STD-454K, and IEC 215 are also shown in the table. Voltages in excess of 30 volts (per MIL-STD-454K and NEC) or 50 volts (per IEC 215) should not be directly accessible under normal operating conditions.

(4) A grounding stick with an insulated handle and a rigid conducting hook connected to ground by means of a flexible stranded copper wire (covered with transparent sleeving) should be provided as an additional safety measure.

(5) Transmitter output terminals or transmission lines with RF voltages should be protected from accidental contact by guards or screens. MIL Standards require protection against RF voltages in the same manner as for AC voltages in the 70 to 500 volt range. IEC 215 Standard requires that RF output connection has provision to drain off any static charge build up. It should also be protected against RF voltages pickup due to coupling from other transmitters operating on the same site.

(6) Low voltage/high current parts such as tube filament supplies, large filter capacitors, and high-capacity batteries should be protected against accidental short-circuits. This may be accomplished by the use of mechanical guards with warning signs or safety devices. MIL Standards require protection for all power busses supplying 25 amperes or more.

(b) Protection against harmful radiation.

The transmitter construction should have adequate shielding so that there is no danger to personnel from any stray or cabinet radiation.

TABLE 2. TYPE OF PROTECTION RECOMMENDED TO PREVENT ELECTRICAL SHOCK

<table>
<tr><th rowspan="2">VOLTAGE RANGE (rms or dc)</th><th colspan="3">TYPE OF PROTECTION</th></tr>
<tr><th>NFPA 79</th><th>MIL-STD-454K</th><th>IEC 215</th></tr>
<tr><td>0-30</td><td colspan="3">None</td></tr>
<tr><td>> 30-50 Volts</td><td colspan="3">Provide doors or covers to protect from direct accidental contact under normal operating conditions.</td></tr>
<tr><td>> 50-70 Volts</td><td rowspan="5">Doors permitting access to voltages 50 volts or more should be interlocked to disconnect power when opened.

Exposed voltages should be discharged to 50 volts within one minute after disconnecting power.</td><td>Exposed high voltage circuits and capacitors should be discharged to 30 volts or less within 2 seconds after disconnecting power.</td><td rowspan="2">Protective covers not removable by hand.</td></tr>
<tr><td>> 70-250 Volts</td><td rowspan="2">Parts exposed to dc, ac or rf voltages should be guarded from accidental contact with a "CAUTION" sign. Bypassable interlocks required.</td></tr>
<tr><td>> 250-500 Volts</td><td rowspan="2">Current limit in a 2K ohm test resistor connected to ground is 2mA dc or 0.5mA ac.</td></tr>
<tr><td>> 500-700 Volts</td><td rowspan="2">Exposed parts should be completely enclosed with a "DANGER" sign. Access door or cover should be interlocked to remove power when opened.</td></tr>
<tr><td>> 700 Volts</td><td>Exposed parts should be grounded by "fail-safe" grounding switch when access door or cover is opened.</td></tr>
</table>

(1) Non-ionizing radiation at radio frequencies: MIL-STD-454K specifies the requirements of the American National Standards Institute (ANSI) C95.1-1982 Standard with respect to human exposure to RF electromagnetic fields in the frequency range from 300 kHz to 100 GHz. ANSI Standard recommends specific absorption rate (SAR) below 0.40 watts per kilogram as averaged over the whole body over any 0.1 hour period. "SAR" is the time rate at which RF energy is imparted to an element of mass of a biological body. Radio frequency protection guide for whole-body exposure of human beings in terms of the equivalent plane-wave free space power density measured at a distance of 5 cm or greater from the transmitter part as a function of frequency is illustrated in Figure 2. The limit on the power density between 30 to 300 MHz is 1 mW/cm^2 (milliwatts per square centimeter). A 10 mW/cm^2 per 0.1 hour average level has been adopted by OSHA as the radiation protection guide in the frequency range of 10 MHz to 100 GHz [12]. The IEC 215 Standard recommends a power density limit of 10 mW/cm^2 over the frequency range 30 MHz to 30 GHz.

MIL-STD-454K requires that shields, covers, doors, which when opened or removed allow microwave and RF radiation to exceed the above, should be provided with non-bypassable interlocks.

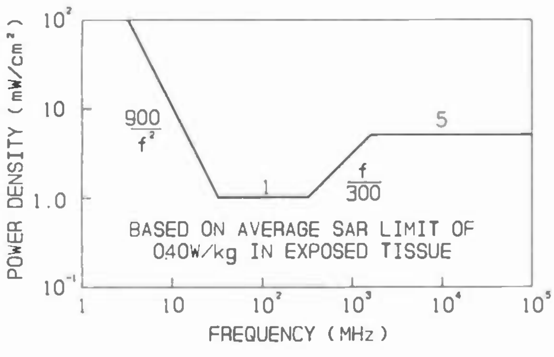

FIGURE 2. RADIO FREQUENCY PROTECTION GUIDE FOR WHOLE-BODY EXPOSURE OF HUMAN BEINGS

(2) Ionizing radiation of X-rays: For X-rays an exposure that releases a charge of 0.258 coulomb per gram of dry air is defined as one roentgen. MIL-STD-454K specifies limit of radiation levels to less than 2 mR (milliroentgens) in any one hour and 100 mR in any consecutive 7 days. Shields, covers, doors which allow X-ray radiation to exceed the limit when removed should be provided with non-bypassable interlocks. The IEC 215 Standard specifies a limit of radiation level to less than 0.5 mR per hour per kilogram.

(c) Protection against high temperatures and fire.

MIL-STD-454K specifies the temperature rise limit to exposed parts including enclosure of the equipment to 35°C (degrees centigrade) maximum and those of front panels and controls to 24°C rise at 25°C ambient. The IEC 215 Standard requires that temperature rise of accessible parts be limited to 30°C under normal operation and 65°C under fault conditions at 35°C ambient temperature, to prevent injury to personnel [13].

The electrical insulation or mechanical strength of equipment parts should not be impaired by the temperature rise. No part of the equipment shall reach high temperature so as to cause danger or fire or the release of flammable or toxic gases. The use of flammable material should be avoided and the possibility of fire and its spread should also be minimized.

(d) Other Hazards.

Components prone to implosion or explosion under fault conditions should be protected against danger to personnel. The safety valve of the components such as electrolytic capacitors should be clearly marked and oriented so as not to endanger the personnel in the event of its operating.

Moving parts such as blowers, motors, fans, gears should be adequately guarded to prevent possible injury. Mechanical design should minimize the possibility of injury from sharp edges, protruding corners, release of springs or accidental pulling out of drawers or assemblies. Attention should also be paid to minimize the generation of acoustic noise.

Permissible noise exposure limit specified by OSHA regulations in a full work day of 8 hours is 90 dB(A) of sound level when measured by a precision sound-level meter [14].

PROTECTIVE CIRCUITS, DEVICES, AND METHODS USED IN A TYPICAL RADIO BROADCAST TRANSMITTER TO ACHIEVE DESIRED SAFETY LEVEL

The protective circuits and safety devices typically used in radio broadcast transmitters manufactured in the United States will be discussed to illustrate safety design considerations.

Protective Circuits

Incoming AC primary power source should be connected to the transmitter through a fused main disconnect switch so that all power may be cut off quickly and reliably, either before working on the equipment, or in case of a fault condition or an accident.

The equipment enclosures, chassis, or frames, including ground terminals of power supplies, are connected to the cabinet or rack ground strap. Typically, a two-inch wide copper strap is routed inside each transmitter rack or cabinet. Straps of individual racks are then bolted together and connected to the ground terminal as shown in Figure 3. The ground terminal is provided for connection to the station earth ground and the system ground. The ground strap has sufficient current carrying capacity and provides a low impedance path for equipment ground fault currents.

A simplified primary AC control diagram of a Broadcast Electronics FM transmitter is shown in Figure 4. The primary AC input to the transmitter is distributed to the low voltage and high voltage supplies through separate properly rated circuit breakers and contactors. The transmitter design incorporates a safety interlock circuit to disconnect primary power from contactors when access doors or panels are opened. Contactor coils are de-energized by specially designed optically coupled relays (OCR) which are in turn operated by the transmitter controller logic level commands. The contactors cannot be re-energized to restore the power without first closing the interlock circuit and then manually resetting the transmitter turn-on sequence.

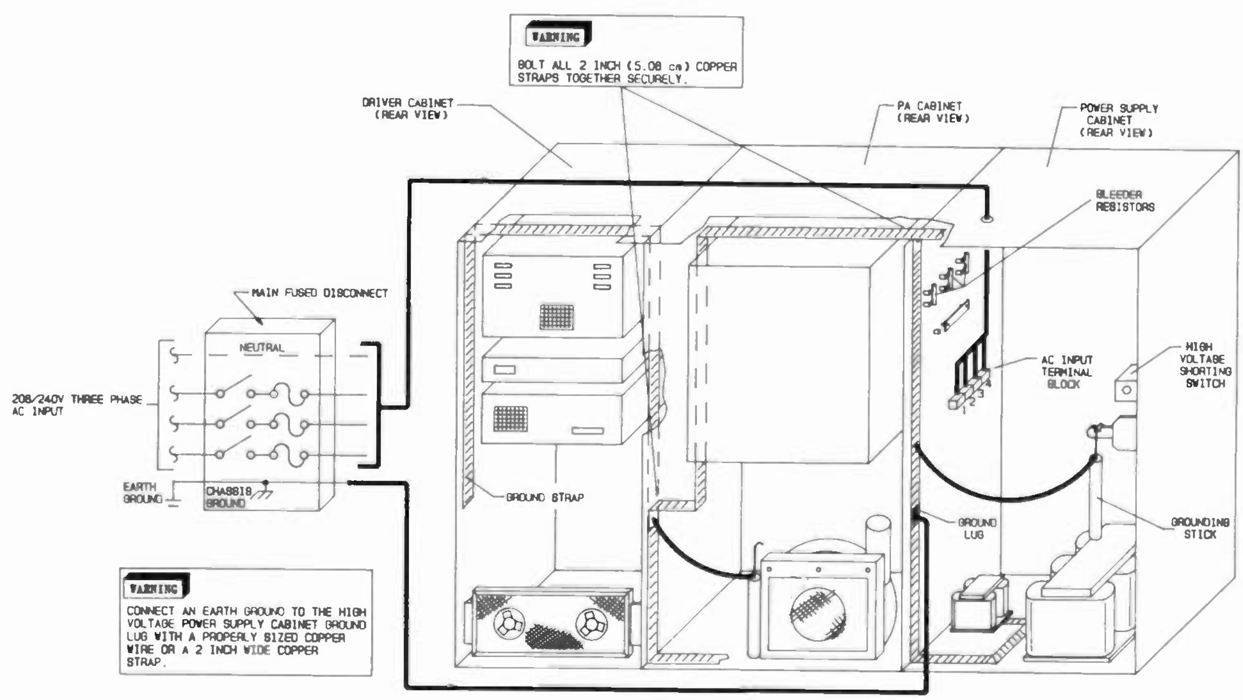

FIGURE 3. TRANSMITTER CABINET GROUND STRAP CONNECTIONS

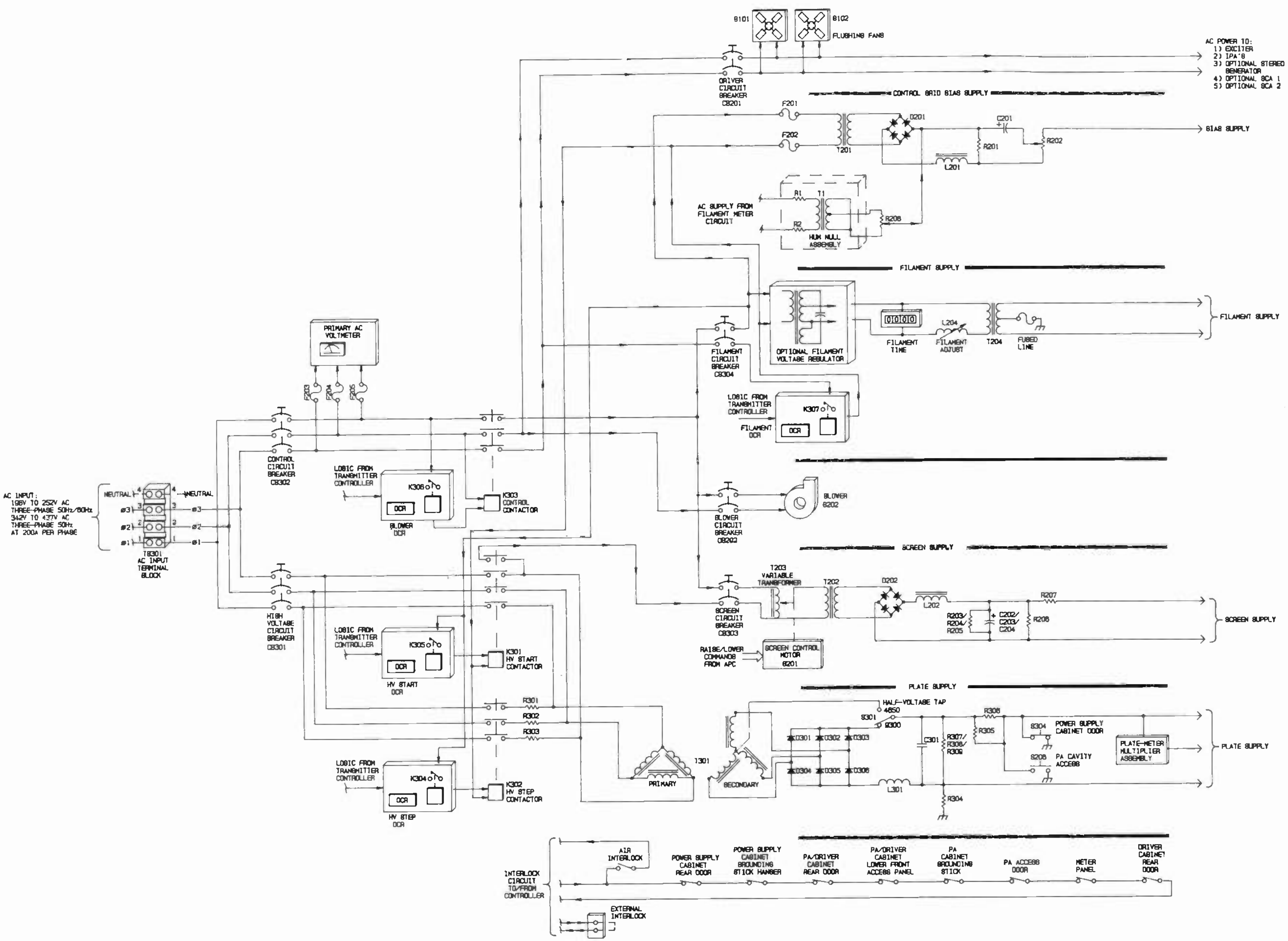

FIGURE 4. SIMPLIFIED PRIMARY AC CONTROL DIAGRAM OF A BROADCAST ELECTRONICS FM TRANSMITTER

This feature eliminates the possibility of turning on the transmitter due to accidental closing of doors during maintenance. An external interlock circuit such as for a test load or remote control fail-safe connection is also provided to disable the high voltage supply when opened. A positive going control voltage of +15 volts DC is required to complete the interlock circuit in the transmitter controller. This is a "fail-safe" feature because any ground fault in the interlock circuit wiring will make the circuit fail in the "safe" condition, thereby eliminating the possibility of turning on the high voltage.

Grounding sticks and high voltage shorting switches are also interlocked to prevent the transmitter from turning on if these safety devices are not in the normal operating position. The transmitter control circuit design allows the blower to run for few minutes after turning off the filament supply so that the tube may cool down. This safety measure will help prevent accidental burns, if the tube anode radiator is touched by maintenance personnel after the cool-down period.

In addition to personal safety devices, the following additional circuits are provided to protect the equipment and its parts:

(a) Component stress at power-on is reduced by a step-start circuit which limits inrush current in the high voltage power supply.

(b) An air interlock circuit to insure adequate differential air pressure and flow for the tube before the filament voltage is applied.

(c) The step/start circuit is interlocked through contacts of filament contactor to assure that the filament voltage is applied to the tube before a high-voltage-on sequence can be initiated.

(d) The RF drive to the tube cannot be applied without turning on the high voltage.

(e) Any fault condition causing circuit overloads due to high plate current, screen current, grid current, or high VSWR will be interrupted to protect the equipment from possible damage.

(f) Solid-state intermediate power amplifiers have built-in temperature sensors to shut down the transmitter when the heat-sink temperature exceeds the maximum limit.

Protective Devices

(a) Bleeder Resistors.

Bleeder resistors provided in all power supplies function as the first level of protection against dangerous voltages. The bleeder resistors discharge residual voltages from all components with stored energy when the primary power is switched off. The rate at which the voltage discharges depends upon the nominal voltage and the R-C time constant of the power supply. The resistor values should be chosen to allow the voltages to decay to a safe level within the specified time interval after turning off the power. The voltage drops to 0.37 times the initial or nominal voltage in one time constant interval (RC seconds, where R is in ohms and C is in farads). This is shown in Figure 5.

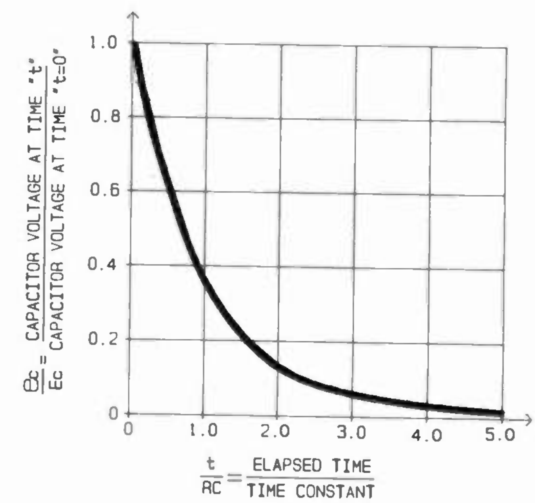

FIGURE 5. CAPACITOR VOLTAGE DISCHARGE WITH TIME

(b) Safety Interlock Switches.

Safety interlock switches typically used in broadcast transmitters and their construction are shown in Figure 6. Figure 6A shows switches with an activating lever which closes the interlock contacts when the grounding stick is properly secured. This type of switch is also used to insure that the high voltage shorting switch remains in the open state when the access door to the RF enclosure is closed.

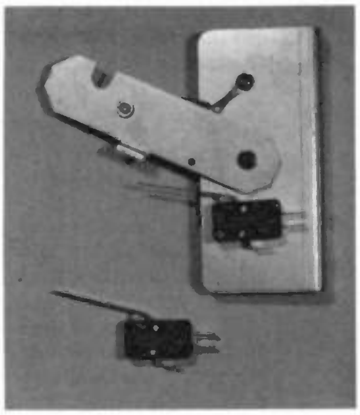

FIGURE 6A

FIGURE 6B

SAFETY INTERLOCK SWITCHES

The switch shown in Figure 6B is used extensively to interlock cabinet doors, enclosure access doors, panels, or covers. The switch is designed in accordance with the "fail-safe" principle to keep the contacts open when the mechanical spring is expanded in its natural state.

(c) High Voltage Shorting Switches.

High voltage shorting switches provide a back-up system to the safety interlock switches described in "(b)" above. This philosophy provides two independent safety systems to protect personnel from accidental exposure to high voltages.

A high voltage shorting switch design based on the "fail-safe" principle, is shown in Figure 7. The insulated rod with a built-in spring mechanism and the block for mounting contact plates are all integral parts of the switch which remain in or go to a "safe" condition to provide protection to personnel in the event of a fault within the device. These positive acting, highly reliable devices are actuated by mechanical release when the door is opened. High voltage is short-circuited to ground due to the closure of contact plates. The insulating rod and housing material has been chosen such as to allow smooth, unrestricted movement from "safe" to "unsafe" position or vice versa. The switch cannot be bypassed without deliberate action violating the safety rules. The high voltage shorting switch shown in the above-mentioned figure is used for short-circuiting the high voltage when the RF enclosure access door is opened. The built-in interlock switch contacts open first to remove primary power just before grounding the high voltage.

The switch is designed such as to prevent any corona discharge under normal operating conditions and to withstand breakdown voltage at least twice the nominal high voltage level.

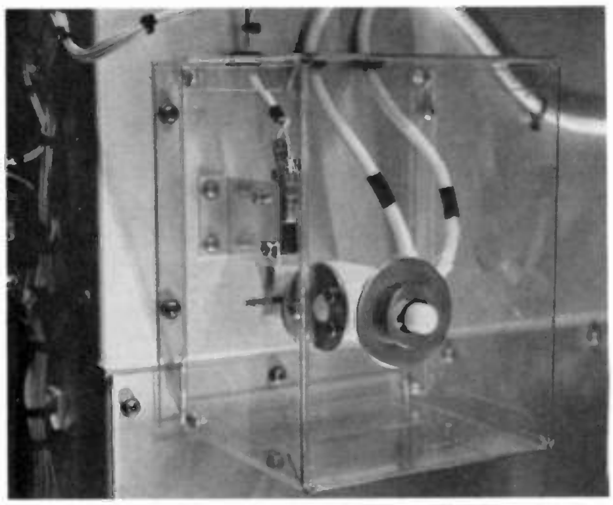

FIGURE 7. HIGH VOLTAGE SHORTING SWITCH

(d) Fail-Safe Solenoid.

The fail-safe solenoid shorts the high voltage circuits to ground and provides a back-up system to the safety interlock switches described in "(b)" above. This philosophy provides two independent safety systems to protect personnel from accidental exposure to high voltages.

A fail-safe solenoid is shown in Figure 8. This safety device is actuated by an electrical solenoid such that the plunger drops to short the high voltage terminal to ground when the transmitter cabinet door is opened. In addition, this device will short the high voltage circuits whenever power is removed from the blower and cabinet flushing fans.

The solenoid design, as the name implies, is based on the "fail-safe" principle. It will remain in or go to a condition which provides protection to personnel in the event of a fault within the device. As soon as the door is opened, the power to the solenoid coil is interrupted and the plunger will drop due to its weight and the mechanical release of spring, thereby shorting the high voltage. This device cannot be disabled without deliberately violating the safety rules. The spacing between the contacts is selected to eliminate possibility of any corona discharge or dielectric breakdown.

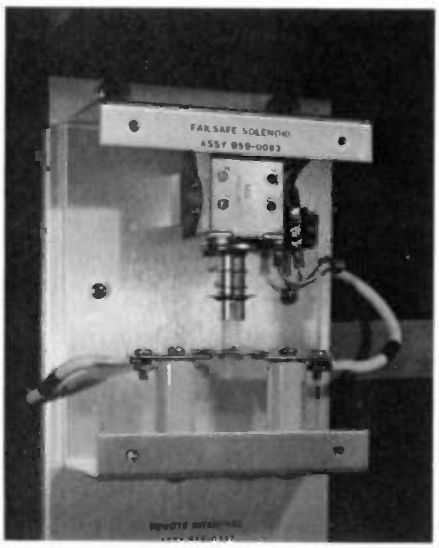

FIGURE 8. FAIL-SAFE SOLENOID (WITHOUT COVER)

(e) Grounding Sticks.

The purpose of a grounding stick is to remove residual voltages from exposed parts of the transmitter before working on it. It is essential to discharge voltages remaining in the equipment after turning off the transmitter, because residual voltages in the energy storage components may be dangerous to personnel safety, particularly if the other safety devices did not function properly.

The grounding stick is a mandatory safety device in all transmitters containing dangerous voltages.

A typical grounding stick is shown in Figure 9. It consists of an insulated handle appropriate for the voltages in the equipment, with a rigid metal hook at one end. A flexible stranded copper wire of adequate size connects the hook to the cabinet ground strap. A transparent sleeving is used as an insulation for the wire to allow visual verification of the ground wire integrity. The grounding stick is permanently secured in the transmitter to make it readily visible and accessible by means of either a ground stick hanger or a pair of clamps with built-in interlock switch to insure its correct placement as shown in Figures 9 and 10.

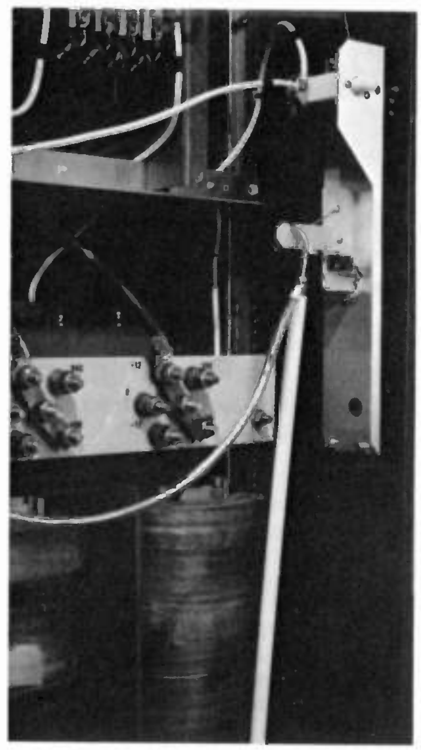

FIGURE 9. GROUNDING STICK

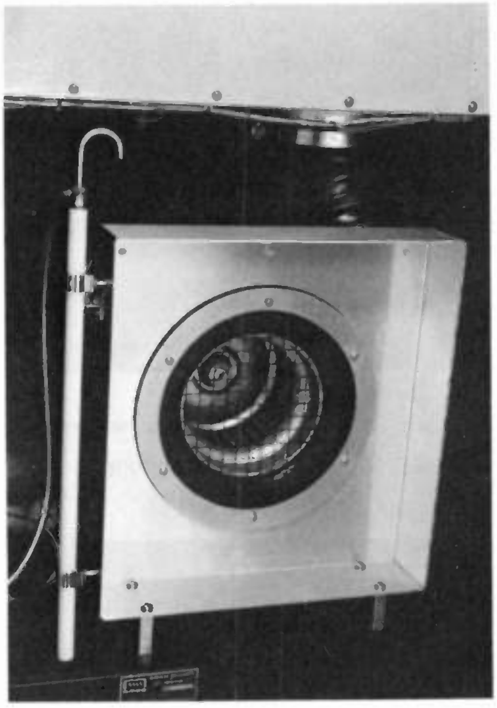

FIGURE 10. GROUNDING STICK AND BLOWER SAFETY SHIELD

FIGURE 11. GUARD FOR AC TERMINAL BLOCK

(f) Protective Covers, Guards, Shields and Markings.

All contacts, terminals, and conducting parts having voltages higher than 50 volts (per IEC 215 Standard) and 70 volts (per MIL Standards) with respect to ground when exposed and exhibit safety hazard are guarded from accidental contact by personnel. A guard for an AC terminal block is shown in Figure 11. High voltages are guarded by protective insulator or metal shields as shown in Figure 12. Low voltage components with large amounts of stored energy and conductors carrying high currents are also guarded where necessary by protective covers with proper markings. An example is shown in Figure 13.

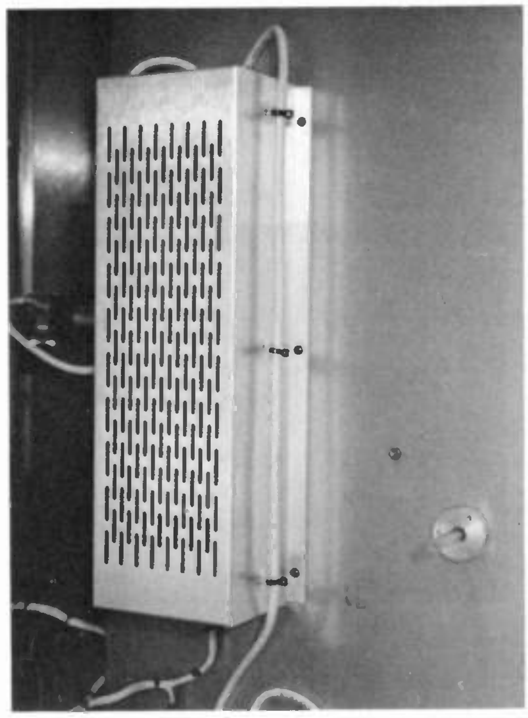

FIGURE 12. HIGH VOLTAGE METAL SHIELD

Protective shields with warning signs are also provided to prevent contact with moving mechanical parts such as fans and blowers. Figure 14 shows a protective shield for fans with warning labels. Blower safety shield can be seen in the Figure 10 mentioned above. The cabinet doors are provided with appropriate markings as shown in Figure 15.

FIGURE 13. PROTECTIVE GUARD FOR CAPACITOR TERMINALS

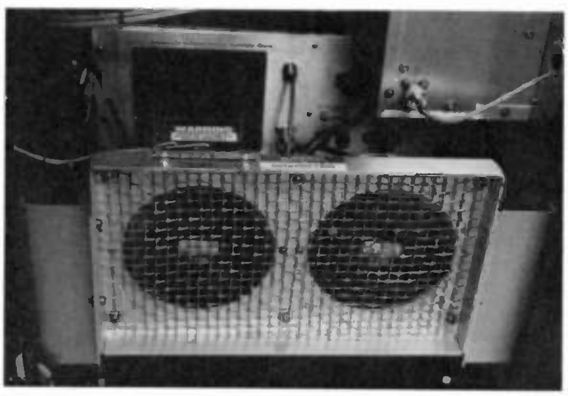

FIGURE 14. PROTECTIVE SHIELD FOR FANS WITH WARNING LABELS

Safety protection against RF radiation is provided by proper shielding to reduce RF leakage from doors, vent holes, air inlet and exhaust openings. Conductive finger stock and special aluminum shield cell honeycomb panels are used to provide adequate shielding as shown in Figures 16 and 17. An instrument for measuring the RF radiation levels to OSHA recommended limits is available from Holaday Industries, Inc. Broadcast Electronics uses this instrument to insure that the residual RF leakage from the transmitter is below the safe limit.

FIGURE 15. CABINET DOOR WARNING LABEL

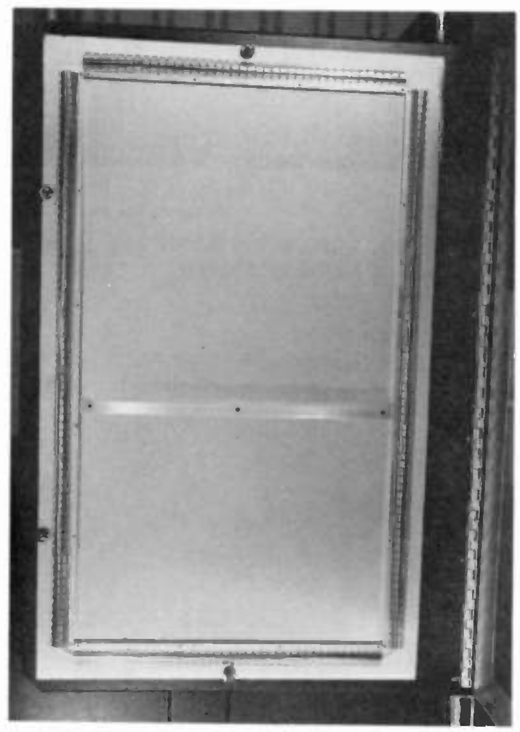

FIGURE 16. CONDUCTIVE FINGER STOCK TO REDUCE RF RADIATION

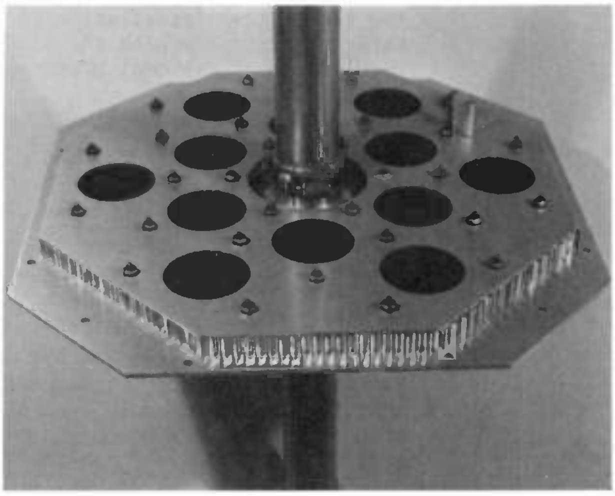

FIGURE 17. ALUMINUM SHIELD CELL HONEYCOMB TO REDUCE RF RADIATION

(g) Circuit Breakers, Fuses and Contactors.

Main primary circuit breakers used in the transmitters are equipped with a thermal as well as magnetic trip elements in each pole. Smaller size breakers have magnetic trip elements. The breakers used conform to applicable UL, National Electrical Manufacturers Association (NEMA), and IEC Standards. The circuit breakers have adequate making and breaking capacity and are selected to protect the equipment against excessive steady-state or instantaneous (less than one cycle) fault currents. The thermal trip protects against high temperature rise.

Circuits or assemblies which do not have individual breakers are protected by properly rated enclosed fuse elements. A fusible link in the center tap of the filament transformer secondary provides overload and safety protection for the filament supply wiring if a short-circuit to ground develops in either leg of the filament supply.

Contactors rated for maximum load and which have adequate making and breaking capacity are used for primary AC control of the transmitter in conjunction with the interlock circuits and the controller unit. The contactors remove the power from accessible areas when the interlock circuit is broken due to opening of doors, panels, or covers.

(h) Smoke Detectors, Fire Alarms, and Fire Extinguishers.

These devices are not part of the transmitter equipment and will not be discussed here. However, it seems prudent to note the following:

Appropriate type and number of smoke detecting devices and associated fire alarm circuits should be installed in the transmitting station. A reliable fire extinguishing system should also be provided to protect the personnel and equipment from fire hazards. Halon 1301 based systems are very effective and will not damage electronic equipment [16]. Automatic fire extinguisher systems should be interlocked with the transmitter control system to turn off the transmitter when the fire alarm system is activated.

Protective Methods

(a) Safety Protection Levels.

Protective methods used to provide different degrees of safety levels can be summed up as follows:

(1) Primary safety level is accomplished by providing doors, panels, and covers with warning signs or labels to avoid direct access to dangerous voltages. In addition, bleeder resistors are provided to discharge residual voltages from energy storage components such as capacitors which may be hazardous to safety even after the equipment is switched off.

(2) Secondary safety level is established by providing contactors together with mechanical and/or electrical interlock systems to insure that the primary power is removed when access doors, panels, or covers are opened without switching off the equipment.

(3) A third safety level is insured by providing shorting switches to short high voltages to ground when the door is opened and also by providing shields and guards which require tools for their removal.

(4) Ultimate safety level is achieved by providing good grounding system, by removing primary power from the equipment with a main disconnect switch, and by using a grounding stick to short out all residual voltages. An external voltage measuring instrument may be used to verify the absence of voltage. When the transmitter is equipped with primary AC metering, it may also be used for this purpose.

(b) Safety Protective Methods.

(1) Safety Program.

A good safety policy should be established by the station management and a comprehensive safety program should be developed and implemented as part of the regular business activity to insure that the facility is operating safely. Safety standards, rules, and guidelines should be developed and enforced. Safety hazards should be identified and necessary precautionary measures taken to eliminate or control them. All the broadcast staff and particularly those staff who have access to the transmitter facility should be properly trained in safety practices, including cardiopulmonary resuscitation (CPR) techniques and in the use of personal protective equipment if required. An adequate first aid kit with training should also be provided.

The United States Department of Labor, Occupational Safety and Health Administration (OSHA) regulations and guidelines contain safety requirements. Necessary information can be obtained from OSHA to start a safety program or to seek the services of a consultant [15].

(2) Safety Practices.

Basic electrical safety practices are described in various standards, regulations and other publications which are listed in the reference section of this paper. Some key personal safety precautions to be considered are high-lighted below:

- Thinking safety and ensuring that the transmitter installation is safe in accordance with the OSHA regulations or National Electric Code.

- Taking time to be careful and using common sense.

- Turning off all power circuits before touching anything inside the transmitter.

- Eliminating the possibility of someone else turning on the equipment (by local or remote control methods) while working on equipment.

- Discharging all the voltages to ground, particularly from energy storage components.

- Avoiding bodily contact with any grounded object when working on the transmitter.

- Avoiding unnecessary exposure to RF radiation.

- Using safety tools and equipment.

- Ensuring that all the safety circuits and devices function correctly.

- Avoid working alone or when tired.

CONCLUSION

Safety is an important factor in the design and development of broadcast transmitters. However, it is not uncommon to find safety taken for granted in today's highly commercial broadcast station environment with fewer trained and experienced technical staff. The management and staff in the broadcasting business should give a high priority to the matter of personal safety because it concerns with the protection of personnel against injuries which may endanger the life. The cost of failure to recognize this fact may far exceed the small initial investment required in implementing a sound safety program.

Various hazards as well as the industry standards and the safety requirements related to transmitting equipment have been reviewed. Design considerations for numerous types of protective circuits, devices, and methods used in broadcast transmitters to achieve the desired safety level have also been discussed.

It is hoped that this paper will serve to stimulate greater awareness of personnel safety among broadcasters, equipment manufacturers, as well as equipment users at large and provide motivation to implement one or more positive action plans to make the broadcast station environment a "safer" place to be.

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NEW TOWER STRUCTURAL STANDARDS ANSI/EIA 222D

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ABSTRACT

This paper presents the responsibility of The Tower Purchaser to specify and verify various parameters for tower design and rating of existing towers in accordance with current American National Standards Institute (ANSI), Electronic Industries Association (EIA) and Telecommunications Industries Association (TIA) 222-D(E) standards. The standards are found in the Structural Standards for Steel Antenna Towers and Antenna Supporting Structures.

The objective of the standards is to provide Minimum Criteria for specifying and designing steel antenna towers and antenna supporting structures. The standards are not intended to replace or supersede applicable codes.

The standards shall apply to antenna towers and antenna supporting structures for all classes of communications service.

INTRODUCTION

A tower represents a sizable economic investment. If ever an industry demanded quality, surely it is the communication company's service to market. Towers usually support many antennas which transmit signals not only to station listeners and viewers but to private or public agency receivers. The tower's important service is sometimes taken for granted by all but the station's engineering staff.

The average service life of a communications facility is approximately 15 to 25 years. Life expectancy of the tower depends on the structural design, quality of construction and construction materials used. You can evaluate structural adequacy by looking at three critical factors: 1) Design Loads, 2) Environment Conditions and 3) Compliance with National and Local Codes. A design report for tower and antenna criteria should be prepared in advance of the final structural design or analysis.

The tower purchaser and/or owner is now responsible for specifying and/or verifying certain design criteria in accordance with the following appendices of ANSI/EIA-222-D, adopted October 1986 and effective June 1, 1987.

APPENDIX A [1]

PURCHASER CHECKLIST

Reference Section

*2.1.2 A. It is the responsibility of the purchaser to **specify** appropriate ice loads for locations where ice accumulation is known to occur.*

*B. The standard does not **specify** ice-loading requirements since ice accumulation may vary substantially within a given geographical area.*

*C. It is recommended that a minimum 1/2 in. [12.7 mm] of solid radial ice be **specified** for locations where ice accumulation is known to occur.*

2.1.3 A. *It is the responsibility of the purchaser to* ***verify*** *that the wind loads and design criteria specified meet the requirements of the* <u>*local building code*</u>*. If other loading criteria are required, they shall be provided to the designer.*

2.3.3 A. *For bidding purposes it is recommended that the purchaser* ***specify*** *the basic wind speed (V) to obtain designs based on identical criteria. Wind speeds* ***specified*** *for use with the standard shall be fastest-mile wind speeds at 33 ft. [10 m] above ground level.*

B. *The basic wind speed map, the equations for the exposure coefficient* (K_z) *and the gust response factor* (G_h) *are based on wind conditions in open, level country, and grasslands.*

C. *The equations* ***specified*** *for* K_z *and* G_h *result in conservative design wind loads for urban and wooded areas.*

D. *It is the responsibility of the purchaser to* ***specify*** *basic wind speeds and appropriate equations for* K_z *and* G_h *in hurricane, mountainous, and coastal areas, in the special wind regions indicated in Appendix A, and where local conditions require special consideration.*

E. *The purchaser shall* ***identify*** *the elevation of the base when the structure will not be placed at ground level.*

F. *The purchaser shall* ***identify*** *the relative elevations of the guy anchors with respect to the structure base and shall* ***identify*** *the maximum and minimum permissible guy radii.*

G. *The basic wind speed map provided in Appendix H corresponds to an annual probability of 0.02 (50-year recurrence interval). If the purchaser requires another probability, the basic wind speed shall be provided to the designer.*

2.3.5.1A. *The purchaser shall* ***specify*** *which appurtenances may not be equally distributed on all faces of the structure.*

2.3.15 A. *Due to the low probability that an extreme ice load will occur simultaneously with an extreme wind load, wind load has been reduced 25 percent when considered to occur simultaneously with ice (equivalent to 87 percent of the basic wind speed.)*

B. *For basic wind speeds based on a 50-year recurrence interval (.02 annual probability), the reduced wind load approximately corresponds to a 5-year recurrence interval.*

C. *It is the responsibility of the purchaser to* ***specify*** *other critical wind and ice loading combinations for locations where more severe conditions are known to occur.*

5.1.1 A. *Galvanizing is the preferred method of providing corrosion protection for towers. Alternate methods of corrosion protection, such as epoxy paint, chlorinated latex paint, plating, electrogalvanizing, etc, may be used only when* ***specified*** *by the purchaser.*

7.2.2 A. *When standard foundations and anchors are utilized for a final design, it is the purchaser's responsibility to* ***verify*** *by geotechnical investigation that actual site soil parameters equal or exceed normal soil parameters. If the purchaser elects to accept the normal soil foundation for construction, he accepts the* <u>*responsibility*</u> *and* <u>*liability*</u> *for the adequacy of the subsurface soil conditions.*

B. *It is the responsibility of the purchaser to* ***verify*** *that the depths of standard foundations are* *adequate, based on the frost penetration and/or the zone of seasonal moisture variation.*

*11.2 A. The purchaser shall **specify** the operational requirements when the <u>minimum</u> standard does not apply.*

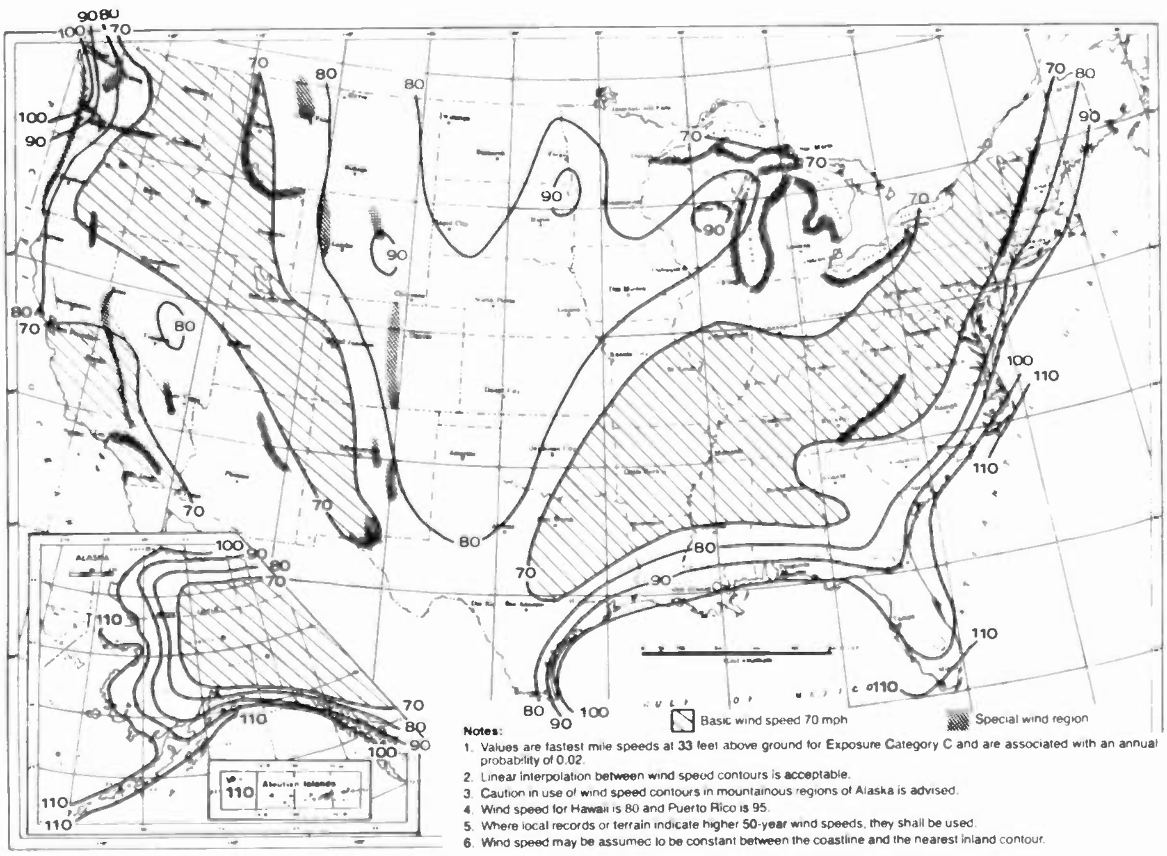
*12.2 A. The purchaser shall **specify** other grounding requirements for conditions where the <u>minimum</u> standard will not be adequate.*

*13.2.1 A. The purchaser shall **specify** requirements for climbing and working facilities, hand or guardrails, and climbing safety devices.*

<u>WIND AND ICING</u>

In Tower design, the structural engineer designs a structural system capable of resisting gravity and lateral loads. The gravity loads can be very accurately determined, while lateral loads such as wind are not as well defined. Wind induced loads do not have a constant value which can be entered into a static analysis. The wind load is variable by its turbulence, occurrence, direction, gust, height above ground, exposure and dynamic properties of the structure.

The wind contour map of the United States (Figure 1), gives wind speed for a 50 year return period or a 2 percent chance of occurrence in any one year. For special wind regions or tower locations, a probability analysis should be performed. The wind speed data is compiled from decades of records by the National Weather Service (NWS) at 129 locations in the United States. Most large airports have a NWS office. The wind data from the NWS can be analyzed statistically to arrive at site specific wind criteria. The criteria shall not be less than those shown in Figure 1.



Notes:
1. Values are fastest mile speeds at 33 feet above ground for Exposure Category C and are associated with an annual probability of 0.02.
2. Linear interpolation between wind speed contours is acceptable.
3. Caution in use of wind speed contours in mountainous regions of Alaska is advised.
4. Wind speed for Hawaii is 80 and Puerto Rico is 95.
5. Where local records or terrain indicate higher 50-year wind speeds, they shall be used.
6. Wind speed may be assumed to be constant between the coastline and the nearest inland contour.

FIGURE NO. 1—BASIC WIND SPEEDS IN MILES PER HOUR

Atmospheric icing of radio and television towers has long been a source of problems for broadcasters. These problems vary from complete tower collapse to distortion of transmission signals.

Ice loads due to rime and solid icing can and have destroyed tall towers. The weight of ice combined with wind are major design considerations for all towers in cold environments. [3]

In evaluating criteria for icing on tower structures, it is important to treat free-standing and guyed towers differently. Ice loading for free-standing towers is easier to calculate than ice loading for guyed structures.

To evaluate the effect of wind and ice on a guyed tower, one must recognize the potential of varying load distributions.

A load distribution for a guyed tower is more asymmetrical due to the collection efficiency and level of the guys.

Certain conditions of freezing precipitation place more ice on the windward guys than on leeward guys. The wind will add to the load of the windward guys, and unload the leeward guy by uplift action.

The action of ice and wind on a guy therefore pulls the mast into the wind, opposing the drag of the wind on the mast. Consideration should be given to the loading of ice on the windward guys.

The combination of wind and ice loading on a guyed tower will have impact on the dynamics or frequency of vibration. [4] It can lead to buckling of the mast.

Icing can also have a serious effect on the dynamic response of a guyed tower to gusty wind.

Under certain conditions of ice and wind, loss of stiffness may occur due to changed dynamic properties. In such instances, energy may be transferred to components which may get overloaded.

The widest range of icing conditions occurs in mountainous areas. Coastal areas and prevailing on-shore winds, stabilized by the ocean temperature, exhibit relatively stable climatic conditions.

Since actual conditions seldom match expectations, it is prudent to be conservative to ensure reliability.

During the past couple of years, SWMB has performed wind and icing studies for planned tower projects in Washington and Oregon (see Figure 2). Projects included a 900′ freestanding tower in Seattle, Washington, a 500′ guyed tower near Rooster Rock, Washington (at 3,500′ elevation), and a 600′ free-standing tower in Portland, Oregon, at a base elevation of 1,000′.

FIGURE 2

Location	Elevation[1] Average Sea Level	Radial Ice[2] Thickness in Inches	Wind[3] Fastest Mile at 30 Feet	Hill Speed-Up Percent
Seattle, WA	1,350	1.0	35	15
Rooster Rock, WA	4,000	4.0	30	80
Portland, OR	1,600	3.5	55	30

[1]Top of Tower [2]On Smallest Member [3]50 Year Return Period

In each study, the work was coordinated with a meteorologist from the Department of Atmospheric Sciences at the University of Washington to evaluate icing and wind data collected by the National Weather Service. The data was analyzed statistically and represented a minimum of 25 years or 250,000 hours of observation for each site studied.

In each of these three studies, the wind and icing criteria results were significantly different. Although the wind speeds for the critical load combinations were lower than the commonly used 87% of the full design wind speeds, ice thickness could be rather large.

Another important consideration was the possible speed-up of the wind due to local terrain conditions or hill speed-up.

The icing and wind criteria report for a tower project is the result of a combined effort by the owner, engineer and meteorologist.

The icing of free-standing and guyed towers will require different analysis procedures. Towers are complicated structures that require static, dynamic and non-linear evaluation. Current computer technology can help perform a static, dynamic and nonlinear analyses.

SITE CONDITIONS

The tower purchaser is responsible for providing a site survey for the designer of a tower project. The site survey shall be done with the services of a professional land surveyor and documented accordingly. Also, the foundation design parameters and alternatives shall be furnished by a professional geotechnical engineer.

TOWER FABRICATION

The tower structural components shall be fabricated in accordance with the latest edition of the American Institute of Steel Construction (AISC) and the American Welding Society Structural Welding Code (AWS). Particular attention shall be given to the quality assurance programs for material, workmanship, welding, erection and corrosion protection. See Responsible Tower Quality for additional discussion of this subject. [5]

OPERATIONAL & SAFETY

The operational equipment such as vendor broadcast antennas, waveguides, cable(s), feedlines, microwave antenna, and lighting etc. shall be taken into account for tolerances of twist, sway, displacement and dynamics of the tower structure.

The safety equipment such as lighting, marking, ladders, elevators, icing mitigation security, etc. shall be carefully reviewed with the regulations of the Federal Communications Commission (FCC), Federal Aviation Administration (FAA) and the Occupational Safety and Health Administration (OSHA).

APPENDIX F: [1]

CRITERIA FOR THE ANALYSIS OF EXISTING STRUCTURES

Periodic revisions to this standard are made by the EIA-222-D Committee based upon comments received from the industry.

The committee does not intend that existing structures be analyzed for each revision of the standard; however, structural analysis of existing structures should be performed by qualified professional engineers using the latest edition of this standard when:

a) There is a change, in antennas, transmission lines and/or appurtenances (quality, size, location or type)
b) There is a change in operational requirements (twist and sway)
c) There is a need to increase wind or ice loading.

To perform the analysis, the following data is required:

a) Member sizes, dimensions and connections
b) Material properties
c) Existing and proposed loading; antennas (size, elevation, and azimuth), transmission lines and appurtenances.

Data may be obtained from the following sources:

a) Previous stress and rigidity analysis (structure and foundation)
b) Structural and detail drawings (design and as-built)
c) Specifications
d) Construction records
e) Field investigation

NEW TOWER EQUIPMENT ADDITIONS

A continuing demand exists for installation of new and larger equipment on towers. Many times, permitting agencies are reluctant to allow mounting of additional equipment on existing towers without further analysis.

After completing the review of existing tower conditions, field inspection, tower period of vibration, measurements, plans, specifications and equipment operational requirements, a structural static and dynamic analysis can and should be performed for a load rating report.

If, during the load rating analysis, it is found that the tower structure requires strengthening, then the current ANSI/EIA-222-D criteria shall be used for such retrofit upgrading.

RESPONSIBLE TOWER QUALITY [5]

In reviewing the preceding responsibilities, we find a checklist is necessary for assisting the tower purchaser in specifying and verifying requirements for a specific structure. The quality of the specific structure includes features, attributes and characteristics of the tower facility or workmanship that bear on its ability to satisfy a given need and fitness for an intended purpose.

The owner, as the originator of a tower project, is responsible for leading and directing the project team. This team consists of the owner, the vendor, the designer and the constructor. Each member of the project team is usually a team in itself. When members of the project team are competent and work together, chances for quality are greatly increased. All team members, indirect and direct, benefit from quality in construction.

Poor quality invariable results in a higher total cost. Though the initial cost may be less through the use of sub-standard materials or as a result of poor workmanship, the total cost to the user or owner over the life of the project will be greater. Since poor quality eventually leads to repairs, breakdowns, and a shorter useful life for the facility, resources are improperly allocated.

Poor quality damages and degrades the quality of life for the project's users, often results in injury to people and property, and frequently leads to conflict and litigation.

Owners must recognize that both quality and total life-cycle cost of a facility are most influenced early in the project life. See Figure 3 for an illustration of this concept. At times, expectations of cost and quality may appear to conflict, particularly when the time of completion is important. An owner or investor requiring early returns on an investment may insist on a completion time that will not allow for sufficient quality, which will result in shorter useful life for the facility. Such time pressures often reduce the time available for design and construction, resulting in a conflict between production and quality.

Since quality is defined as meeting the requirements and the requirement is early completion, excellence in performance might be seen as meeting the time schedule. In such a case, the owner or investor must be aware of the affect of this requirement on the useful life or even the total life of the facility and on the efficiency of operation.

A team effort is based on mutual trust and understanding among all parties. The project, as well as all participants, should fulfill their obligations to society, not only in individual actions but in the concept and content of the project.

CONCLUSION

The quality of a tower project requires an understanding of the requirements of current standards and regulations by the purchaser and project team. The appendices of ANSI/EIA-222-D as referenced and discussed in the paper are minimum standards and should be reviewed with a qualified professional engineer.

ACKNOWLEDGEMENTS

The Author is Ramon D. Upsahl, Vice President, SKILLING WARD MAGNUSSON BARKSHIRE INC., 1215 Fourth Avenue, Suite 2200, Seattle, Washington 98161, (206)292-1200. He would like to acknowledge and thank Ms. Debra Anthony for the typing the manuscript.

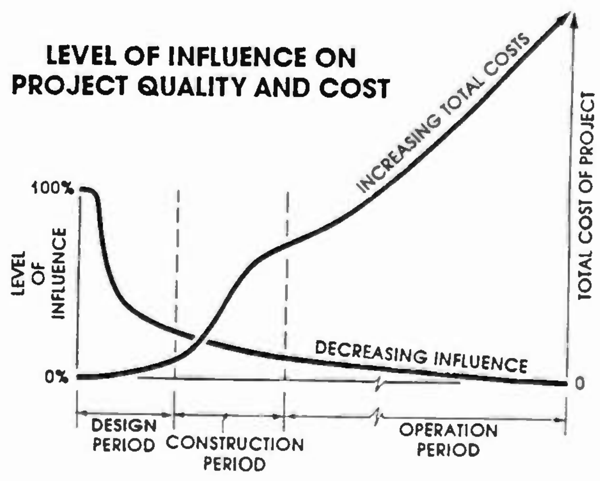


FIGURE 3

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GUIDELINES FOR VIBRATION CONTROL OF TOWER GUY CABLES

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ABSTRACT:
This paper concerns the guidelines to be used in obtaining vibration protection on steel guy cables for towers. It explains the two major types of vibration. These are identified as high amplitude, low frequency galloping, and low amplitude, high frequency vortex vibration. Various research results are summarized pertaining to the development of the SANDAMPER(R) method of gallop control. Wind tunnel testing, laboratory testing, and field testing are described. The parameters that determine the destructive forces in galloping cables are the first two natural frequencies of the vertical motion.

In the case of high frequency vortex vibration, the large number of vibration modes makes vibration control difficult. A wide band damper is described. Tests of vibration from actual field data illustrate the wide range of frequencies that must be controlled. Laboratory tests of the AR DAMPER are compared with laboratory tests of a stockbridge damper.

INTRODUCTION:
The vibration of tower guy cables has been a problem for many years. It has become an increasing concern recently because of increased costs of maintaining hardware in good working order and because of the perceived higher risks of failure due to vibration induced stresses. The question of when to use vibration protection and the means by which vibration reduction can be achieved are addressed in this paper. There are two primary kinds of vibration that occur in tower guy cables. The first kind is the high frequency vibration or vortex vibration due to the passage of a steady wind perpendicular to the cable. The wind speed range is from three miles per hour to fifteen miles per hour. Above the higher wind speed, the vortex trail behind the cable tends to become obliterated by turbulent mixing. This means the aerodynamtic forces on the cable change from regularly periodic to irregularly random. The frequency is from 5 to 40 Hertz. The second kind of vibration is the low frequency or galloping due to the formation of ice along the cable. The ice acts like an airfoil creating lift leading to dynamic unstable motion. The motion may be composed of movement in both a horizontal and a vertical plane, such that a cross section of the cable moves in an irregular ellipse. The frequency is 1/5 to 1/2 Hertz. In the first kind of vibration, the amplitudes are less than the cable diameter but the stress in the cable at the ends due to bending are potentially capable of fatique damage and failure. Further, vibration passed to tower can shorten the useful life of strobe lights, end fittings and hardware. In the second kind of vibration, the amplitudes are nearly equal to the cable sag and the dynamic changes in the cable tension that will occur are capable of destroying the tower structure itself. In assessing the cost and benefits of any vibration control system, it is possible to examine the known cost for similar towers that do not have any control system. Such costs include maintenance and repair, replacement of components and in extreme cases replacement of the tower itself. Insurance costs that relate to casualty loss should not be ignored as a component. Some executives in the insurance industry have expressed a view that premiums could become higher on those towers that employ no vibration control devices. Further, business costs associated with outright loss of the facility should not be ignored. While no guidelines are included now in any structure design standards for the inclusion of the dynamic loading or for vibration components, I believe it is only a matter of time until such standards do begin to support existing loading standards. In that event, the use of damping control devices can be beneficial and especially those that will limit the dynamic loading of the structure.

Part One - The Control of Galloping or Low Frequency Vibration

For proper orientation, we begin by a view of a typical guy attached to a tower in Figure 1. Notice that the guy makes a plane with the tower and we will identify two types of motion. In the plane AOC for example, the guy can move across this plane. Commonly, this is referred to as horizontal vibration. We refer to this type of vibration as type I. Contrary to that in plane "BOC" the vibration lies in the plane. This type of vibration is commonly referred to as vertical. We here consider it to be type II. Depending upon the direction of the wind in general, both type I

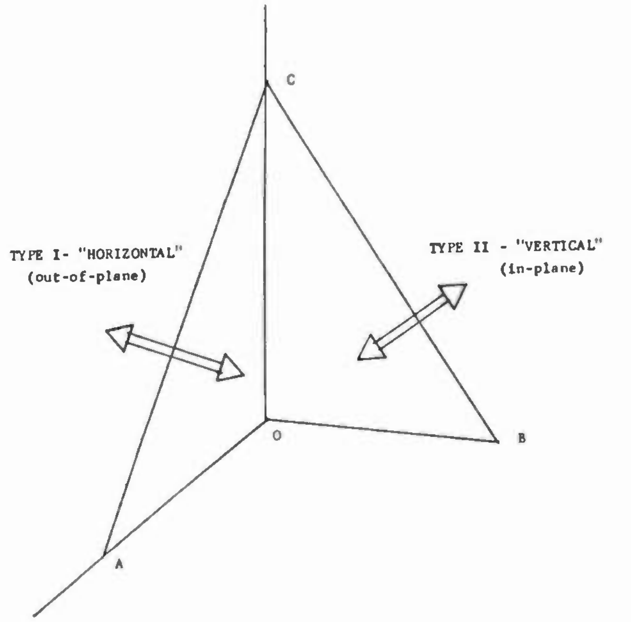

FIG. (1) TYPES of GUY MOTION

and type II vibrations occur in any tower guy system. In general, these vibrations occur together, that is, they are mixed in and one standing off and observing would be hard put to say just how much of each vibration exists. An example of a type of breakdown in the components of the vibration is seen in Figure 2. In Figure 2a for example, a horizontal component is shown and a vertical component is shown. Each having the same magnitude of motion. In other words, these compose both type I and type II vibration. Notice that there is a small phase shift between the two. Even though the two amplitudes are the same in Figure 2b the result in space is an ellipse. In this particular case, the ellipse is tilted at 45 degrees and the magnitudes already mentioned are equal in both type I and type II.

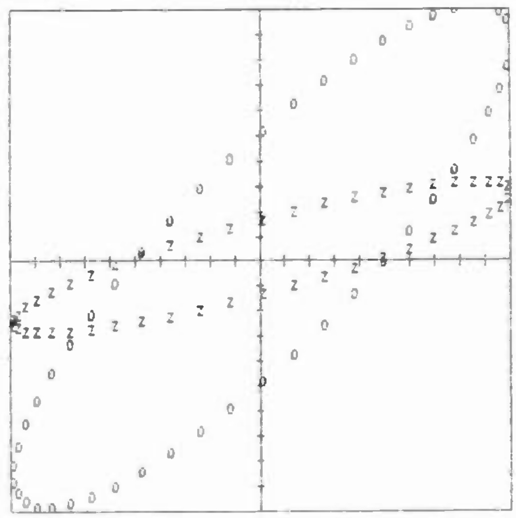

Fig. (2b) Elliptical Motion

On the other hand, if type I motion is less than type II or vice versa then the flattened ellipse in Figure 2 would be appropriate. Further, if type II were greater than type I the ellipse would take on a more vertical orientation. Obviously then, in a general situation, the motions of tower guy cables are composed of the two major components already identified and the precise mixture often is impossible to determine. Now the explanation just given is applicable to any vibration of a tower guy whether it's a gallop low frequency type or whether it's a high frequency type. It makes no difference. Each type of vibration is governed by the same component parts already identified. Now in the case of galloping motion, it's a matter of observed fact that this motion rarely occurs, but when it does occur

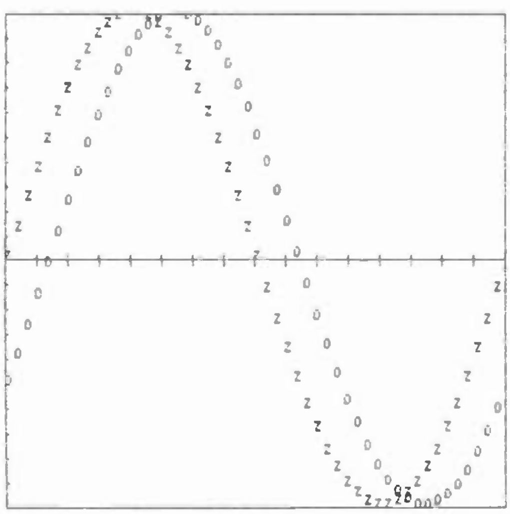

Fig. (2a) Elliptical Motion Components

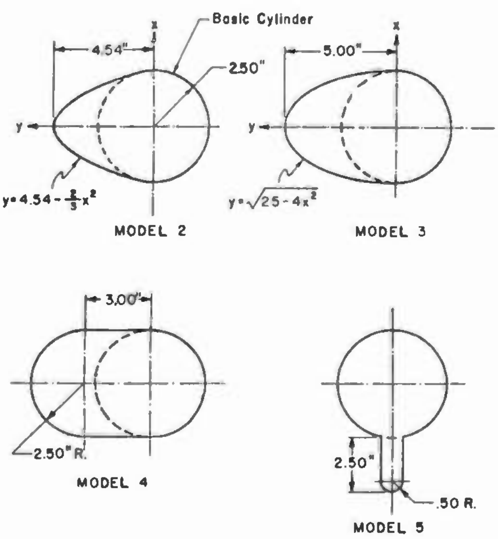

FIGURE 3
MODELS FOR WHICH LIFT, DRAG, AND MOMENT VERSUS ANGLE OF ATTACK DATA HAVE BEEN OBTAINED

normally it has a coating of ice or sleet that is distributed the entire length of the guy. Several years ago, we did some research at M.I.T. and one of the first things that we did was to examine the actual aerodynamic forces that occur on what could be called typical ice formations. Figure 3 is an example of the ice formation that can occur and has been observed on guy wires and overhead power line conductors. Our research project at M.I.T. put as an objective the wind tunnel testing of these various forms or simulated ice shapes. In the upper left hand corner of the graph, one sees what is probably a more common type of ice shape; namely, a tear drop shape having the relatively thin layer of ice that forms on the windward side of the cable. When the angle of that shape changes with regard to the wind; namely, by way of a measured quantity called the angle-of-attack, then all of the forces and moments on the cable itself will vary. Figure 4 shows a setup in a wind tunnel at M.I.T. that employed a wooden model of one of the ice shapes. This was tested in the wind tunnel. The results of the test of model number two are seen in Figure 5. Test conditions are noted at the top of the graph the dynamic pressure of 1.82 lbs. per sq. ft. corresponds to the wind speed of about 26 miles per hour. The diameter of the test model in this case was 5 inches. It was that size so as to have high accuracy in the measurement of forces and moments. From that, coefficients could be calculated that can then be applied to any diameter of cable, a well-known principal that has been applied for many years in the field of aeronautics. Looking first at the lift scale on the left, one sees a variation of the curve following a variation of the angle-of-attack seen in the lower left from a zero angle to an angle of 180 degrees, turned completely around with the rear facing the wind. Over that range, the lift is seen to rise at a steep rate from zero to about 1.4 lbs. per foot. Then at an angle of 20 degrees the shape is stalled, the flow separates, the lift suddenly drops to a value of about .6. Thereafter it continues to drop at a gradual rate, then at about 105 degrees it reverses and begins to sharply increase positively again until the angle of 160 degrees is reached when it sharply drops again. Finally, coming to zero at 180 degrees where the shape has its axis aligned with the wind. Looking now at the drag, it begins with a force of .6 lbs. per foot falls off slightly to about .5 lbs. per foot as the angle changes from zero to 20 degrees then begins a gradual rise to its maximum value of about 1.2 lbs. per foot at 90 degrees angle. The shape there is broadside to the wind, and then it continues beyond that value and falls gradually, assuming a final value at slightly above .65 lbs. per foot at 180 degree position. The pitching moment or moment that is applied as the result of the aerodynamic forces follows the lift up until it reaches the stall point at 20 degrees. It rises from a value of 0 to a maximum value of 3.2 inch lbs. per foot. The moment is measured about the center of twist of the cable. Thereafter the moment follows roughly the same type of variation as previously explained.

Well what does all this mean? In the jargon of the aeronautic wind engineer, it means that there is an unstable situation after the air flow separates from the shape around 22 degrees or so

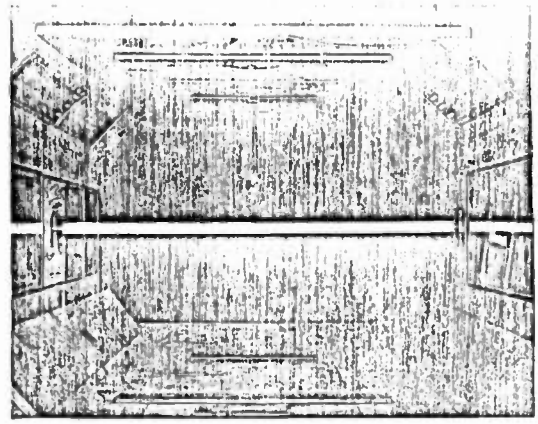

Figure 4

MODEL MOUNTED IN WIND TUNNEL

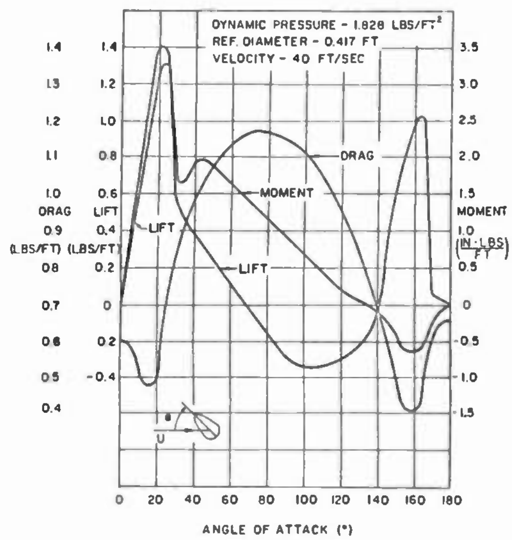

Figure 5

MODEL NO. 2 WIND TUNNEL MEASUREMENTS

and that is associated with the very steep drop. That basically is what's behind the galloping phenomenon itself. Turning to Figure 6, a curve is plotted showing again the angle-of-attack and a parameter which is referred to as the damping parameter. This is made up of the log decrement or the damping ratio, relative density of the cable-the weight of the cable in comparison to the weight of air-and the frequency of the cable-the vibrational frequency. The diameter of the cable, and the wind speed are also included in the parameter. Looking at the cross hatched area, one sees in the range of about 22 degrees angle-of-attack to about 38 degrees an area that has zero damping. Then if the damping is included, the limit of the instability is reduced to the inverted parabolic curve seen in the same angle range. It is from this system of charts that one can examine in detail the amount of damping that is needed in a cable to guard against the occurance of destructive galloping. Notice that throughout the entire range of angles from zero to 120 degrees seen on this graph there are only 2 relatively narrow ranges where galloping is possible. However, the forces of nature are such

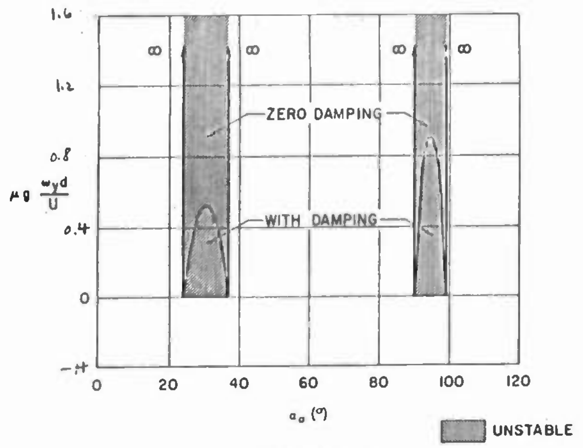

Figure 6

STABILITY BOUNDARY FOR D-SECTION BASED ON SINGLE - DEGREE - OF - FREEDOM (VERTICAL) ANALYSIS

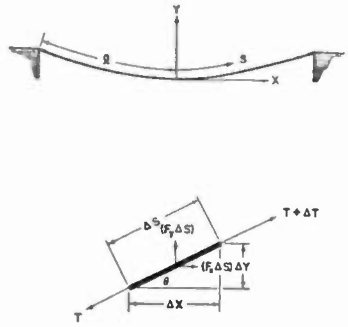

Figure 7 CATENARY GEOMETRY

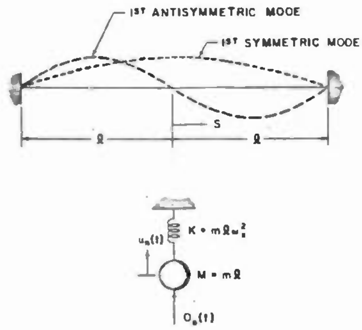

Figure 8 DYNAMIC RESPONSE OF STRING IN TERMS OF NORMAL COORDINATES

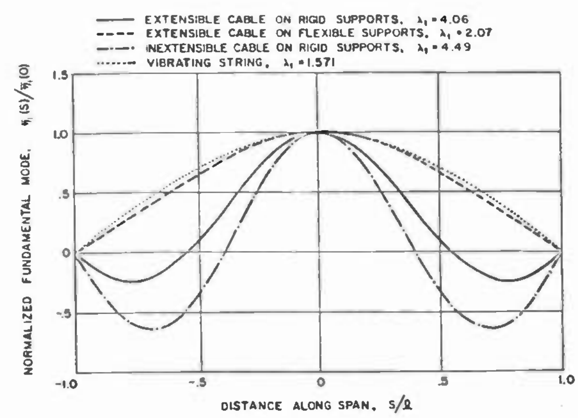

Figure 9 FUNDAMENTAL SYMMETRIC "VERTICAL" MODES FOR TRANSMISSION LINE WITH SMALL SAG

that very often these angles become available to a cable sitting with ice, and often when the wind is in the range of 20 to 30 miles an hour it will become unstable and it will gallop. Other studies were made from the aerodynamic studies that were based on the actual mechanics of the cable and in Figure 7 and Figure 8 we see the basic dimensions of the models that we used. In Figure 7, the cables considered to be anchored rigidly at each end, and to have a coordinate system measured in the X and Y orientation. The little figure below the catanary shows the differential forces including the air forces and tension forces that are present in the cable itself. These models have led then to the determination of the type of vibration that is possible and these are described as modes. In the figure, we see the shape of mode in correspondence to the first symmetric mode, and corresponding to the first anti-symmetric mode. Each mode has its own frequency. In the little photo shown, one sees the mathematical model that is used to calculate the motions and the forces that are associated with the modes. In Figure 9, a more detailed breakdown of the mode shape is seen. Here we see in the upper part of the curve, a set of dash and dotted lines which are identified as the modes of a stretched string. I mentioned each mode has its own frequency that is dependent upon the sag of the cable, tension, the mass and other factors. At the bottom of the figure, it is noted that the solid curve and the broken curve are below the zero position. These modes are having frequencies that exceed the second frequency that was seen previously on the Figure 8. In other words, even though the mode in Figure 9 is the fundamental, if the cable sag is large enough, the span is long enough, its frequency can actually be higher than the frequency of the anti-symmetric or two loop mode. In that case the first mode will have 3 loops as seen here. These modes which are type II modes are very dangerous. The reason they are dangerous is because when the shapes begin to approach 3 loops as seen, then the dynamic tensions that occur at the ends of the cable become very high and can actually destroy a tower. Next, is shown on a table is a formula that may be used to compute the amount of dynamic tension that is applied to a tower when it gallops. The table shows the keen dependence that exists upon the ratio of the first two vibrations. These forces are extremely high as may be seen. In fact when

TABLE 1 Dynamic Tension - 1st Mode Gallop = Sag

DT/To = 1/2 [π f1/f2] COS (π f1/f2)

Frequency Ratio	Dynamic Tension Ratio
.55	.23352
.6	.54898
.65	.94655
.7	1.4213
.75	1.9628
.8	2.5551
.85	3.1768
.9	3.8016
.95	4.3988
1	4.9348
1.05	5.3736
1.1	5.6789
1.15	5.815
1.2	5.749
1.25	5.4522
1.3	4.902
1.35	4.083
1.4	2.9889

DT = dynamic tension when gallop equals sag
To= initial static tension
f1 - first symmetric "vertical" mode frequency
f2 - first anti-symmetric mode frequency

we first calculated these forces we didn't believe them. And so subsequently we began to examine test data from other investigators, and we found a series of test results seen in Figure 10. These tests were made in Japan on galloping overhead power transmission cable attached rigidly at each end to steel towers. Measurements were made of the peak vibration tension applied to the tower ends, and its associated change in the angle of the deflection of the cable at each end, seen on the left of the graph. Theortical results are seen in various solid and dash curves. Notice in particular the maximum level of the tension in the vibrating cable. Tension is in the range of four figures, and in some cases exceeding five figures pounds, double amplitude. In other words from these data, clearly it is possible to have several thousand pounds dynamic tension occur as a result of the galloping motion. One other thing that concerned our attention several years ago was how to control the gallop, and we find that there is very little that can be done because the frequency is so low. Dampers that would utilize automobile type shock absorbers, or rubbing friction type dampers, or mass type dampers simply would be too big. We deveoped a device that is known as a SANDAMPER(R). A photo of the prototype unit is seen in Figure 11 and consist of a rotating drum

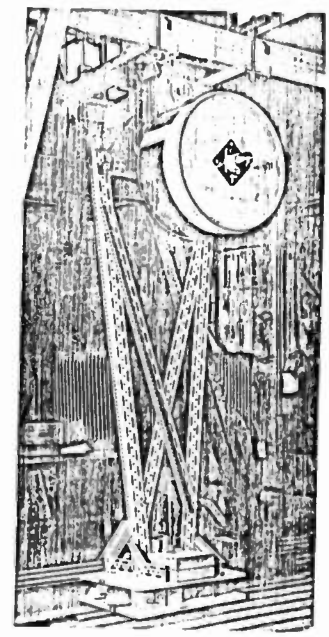

Figure 11 PHOTO OF DAMPER IN LABORATORY TEST

partially filled with sand. In the laboratory tests, a suspension arrangement about five feet long supported a dead weight of lead of 411 lbs. and this dead weight was swung as a pendulum driving the gear box that in turn drove the sand drum itself on a ratio of about ten to one rotation. Results of the test, seen in Figure 12, show that at the top when there is no drum the weights swinging back and forth as a pendulum require approximately 8 cycles to damp out. If the drum is added, the number of cycles the damp is reduced to 4. This is due mainly to the fact that the gears inside the gear box are creating damping and reduce the vibration. Then progressively amounts of sand from 6, 12, 18 and 24 lbs. were added, and as seen with 24 lbs. in the drum, the vibration is reduced in about 1 cycle or less. While the drum rotation causing that runs in the range of one full rotation peak to peak. So, here with a simple device, one could have an effective damping arrangement. Then the system was placed on an actual two span transmission line. Tests were made at the ALCOA Laboratory in Massena, NY and each span was approximately 800 feet in length. Each cable approximately 1-1/2" in diameter, and at the

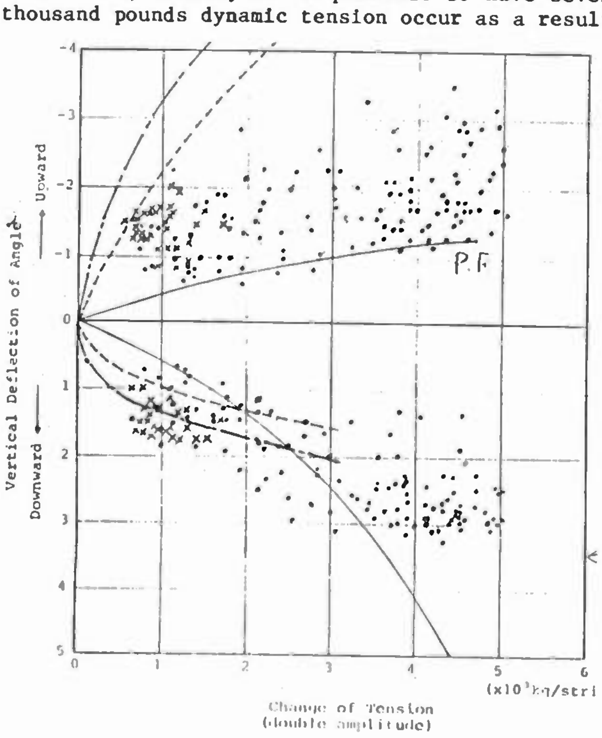

Fig. 10 Vertical Deflection and Dynamic Tension

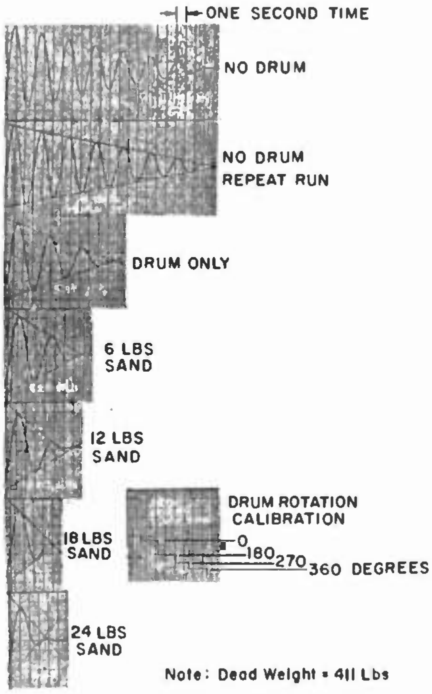

Figure 12 SAMPLE RECORDS OF LABORATORY DAMPER TESTS

center tower the SANDAMPER(R) drum was located so it would take out the tension variations in each of the spans. Galloping was induced by pulling at the mid sag position with a rope. First, the measurements are seen without any damper attached, and these are shown in Figure 13. Plotted on log scale the damping paremeter known as g which is about twice the damping ratio, can be calculated. In this case, the g value is a little over 1%. Then the tests were repeated using 48 lbs. of sand in the drum and again the data plotted as seen in Figure 14. Here the g parameter is found to be about 7-1/2% an increase of 5 to 6 times the damping that is available in the line by itself. These results were also confirmed by calculations seen in Figure 15. So it was possible then to arrive at a design basis for the SANDAMPER(R) . The calculated results are shown as compared to the measured results. Comparison is very good. Calculations were then made as to the total amount of energy that could be dissipated within the rotating drum. Here, we see energy dissipated in foot-pounds per cycle, ranging from zero to about 800 foot pounds per cycle. The mid-span gallop

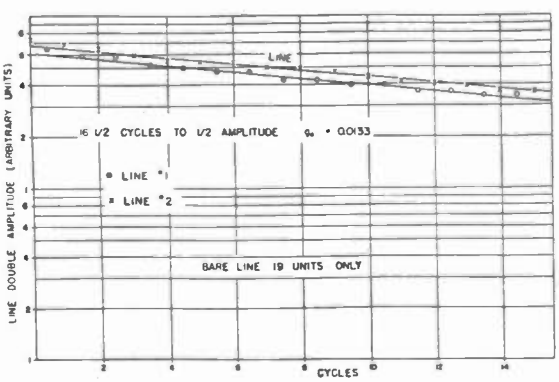

Figure 13 SAMPLE OF DAMPING CHARACTERISTICS OF BARE LINE

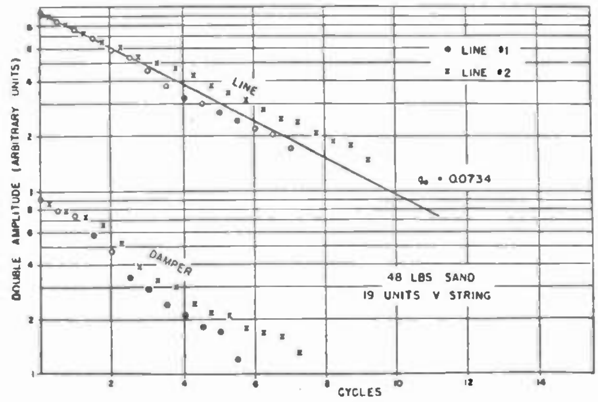

Figure 14 SAMPLE OF DAMPED CHARACTERISTICS OF DAMPED LINE

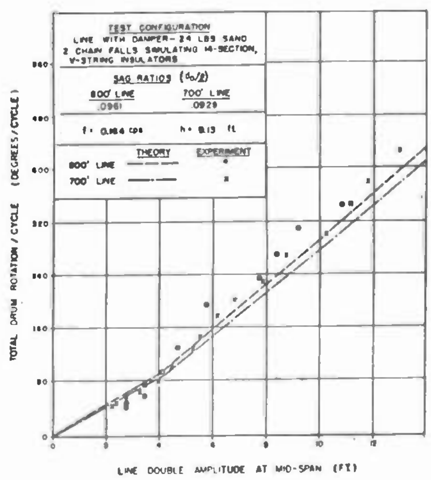

Figure 15 DAMPER vs LINE MOTION FOR A TWO-SPAN TRANSMISSION LINE

amplitude of the 800 foot line is from zero to 12 feet double amplitude. The various test conditions are shown in the upper left and the X's illustrate the experimental results whereas the solid curves illustrate the theory. Again good agreement is found, Figure 16.

The ultimate results of several design changes in the SANDAMPER(R) resolved in a unit that is more compact, lighter weight, and easily attached to a guy cable. Its attachment is made simply by rolling a wheel partially filled with sand up to an altitude of above 150 to 200 feet above ground at the anchor end of the cable. Figure 17 shows a underside view of 1/2 of the SANDAMPER(R) casting itself which is manufactured of aluminum. One sees the size in relation to the persons who are holding this. Two of these units are back-to-back bolted together. After the sand is filled, the halves are separated by a lead washer or a special plastic washer and sealed against the elements so that water cannot penetrate. Final assembly on the guy wire is seen in Figure 18 where the tower may be seen in the background. This particular unit happens to be located on the WTVD tower for Durham, NC and the unit seen here is manufactured by Kline Iron & Steel. Other SANDAMPER(R) units

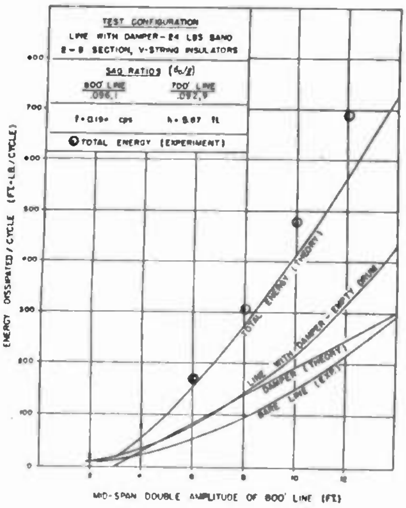

FIGURE 2.38 ENERGY DISSIPATED BY DIFFERENT COMPONENTS OF TWO-SPAN TRANSMISSION LINE

Fig. (17) View of the SANDAMPER(R) Drum

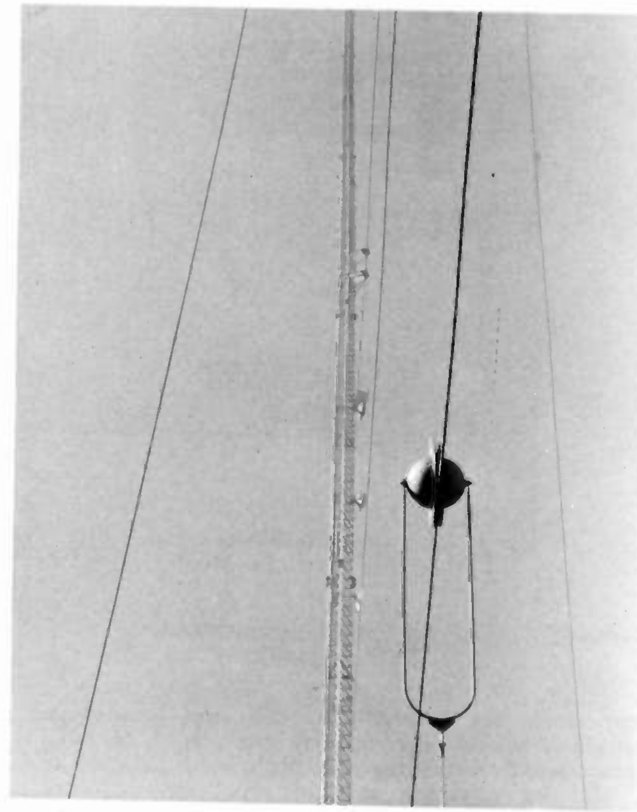

Fig. (18) Title: SANDAMPER(R) on Television Tower Guy Cable

have been licensed for manufacture by LeBlanc Royale Telcom, and Structural Systems Technology. This tower has a particular interest because I was contacted last summer by the United States Geophysical Research Laboratory in Bedford, MA. They told me that certain measurements were made on this tower that took exception to Newton's Law of gravity regarding the variation of gravity intensity as one leaves the earth's surface. The Air Force person, Dr. Donald Eckhard, director of the Earth's Sciences Division, told me about a paper published by his associates Dr. Romaids and Dr. Sands, who actually carried out the experiments. The experiments consisted of transporting a gravity meter up the tower by an elevator to the top—some two thousand feet above ground—and measuring the amount of gravity. However, these instruments are very sensitive and they can detect gravity very accurately. By that means, they did prove the fact that Newton's Law was somewhat in error. I believe that the findings are being reported, this month, May, on television's educational channel by the research folks in the National Geographic's Society. At any rate, it was explained to me that the reason, primarily that these scientist could make their measurements was because the tower itself was so quiet. A detection of milli-G levels of accleration normally is impossible on towers subject to winds and other vibration features because, after all, vibration is nothing more than another form of induced gravity, so one needs a very quiet environment. Apparently, they found the use of the SANDAMPER(R) on that particular tower to be a definite factor enabling them to make this key finding.

Part Two: High Frequency Vibration

The second kind of vibration that must be dealt with on the tower guy cables is identified as the high frequency vibration. This is caused by the rapid vortex pulsations trailing off behind in the wake of wind. It seems that the low winds are worse than the high winds. To the uninitiated, that may seem like a very strange phenomena, and it is to some extent. The reason the low winds, I am speaking in terms of three to eight or nine miles per hour, is more troublesome is because the vortex rate-even though it is slower-is more regular. In other words, at low winds the vortex all along the span seems to get together and become coherent. At higher winds in the range of 12 to 15 miles an hour, the vortex trail become more irregular even though vibration is present, forces tend to drop off. Another reason that the high winds and therefore the high frequencies are less troublesome, say up to 20 miles an hour or above, is that the cable itself provides more damping. It is a peculiar trait of stranded cable that the internal friction damping increases as the frequency increases, and therefore the need for additional, or external, damping devices is diminished. The proper design of a damping system on a tower guy system takes into account the assistance that can be provided by the cable. There have been some reported cases, few, but not negligible, where large diameter cable having a smooth surface-as a plastic coating-vibrate even more intensely than a corresponding stranded cable of the same diameter. Here again, since this has been observed mainly at low frequency, the answer is found by an examination of what the vortex trail does on a smooth body as opposed to a rough body. Clearly the flow around a smooth body is more laminar. A sample of the high frequency vibration is seen in Figure 19. These are

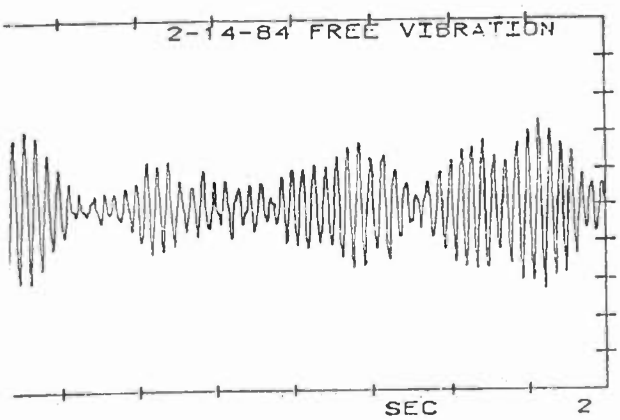

Fig. (19a) Measured High Frequency Vibration

measurements that we actually made ourselves in the field on a cable whose diameter was 1.2 inches. The upper graph shows the vibration on a scale of 0 to 2 seconds, and one can easily determine frequency of the vibration by counting the number of cycles in one second. Full scale on the graph is 2 G's from the mid line of the graph. In other words, the maximum plus 2 G's and the minimum is minus 2 G's. This vibration happens to have a peak value slightly less than 1 G. It isn't a severe vibration, nevertheless it is one that would be of some concern. A typical character of the high frequency vibration, as seen here, is certainly not a pure motion. It is not even a motion having beats as would be the case of a mixture of 2 or more sine curves. No this motion is best characterized as narrow band noise, and that is precisely the best mathematical model that we have found in dealing with high frequency vibration. The cable itself becomes an oscillator, and because the damping is low its frequency response a narrow band pass filter.

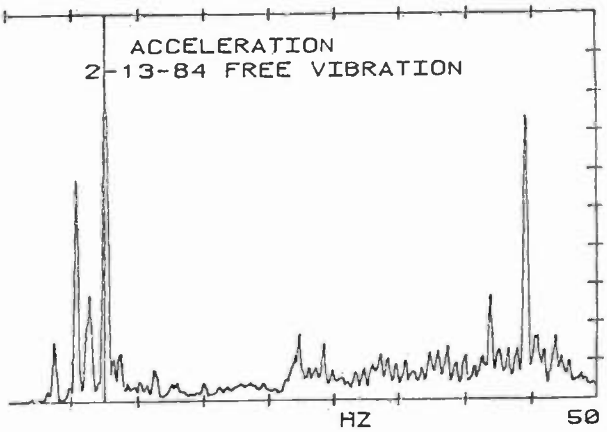

Fig. (19b) The Spectrum of Vibration over a Period of Three Hours.

When the wind reaches the correct value to excite a particular mode, it does so at the frequency of the mode. Now, the difficulty in controlling this in a cable is because there are so many modes that can be excited in the wind speed range of from 3 to 15 miles per hour. What we need to say here is: let's take a 2 inch diameter of the cable, then in that speed range one would have at least 50 vibration modes to be excited-not all at once, of course. It depends on the wind speed. If the wind speed varies from 3 miles an hour, let's say, to 5 miles an hour the vibration would follow that wind speed from 4.5 cycles per second to 6.5 cycles per second or if you prefer 4.5 Hertz to 6.5 Hertz. Now within that band of 2 Hertz on a cable whose length is, typically, 1200 feet there will be as many as 5 modes per Hertz. Hence, in that small speed range mentioned, more than 10 modes would have been excited. There is some mixing with nearby modes. Therefore, long spans tend to concentrate high frequency vibration at the low end of the scale which makes the control of it rather difficult. On the other hand, consider that same wind speed range of 3 miles per hour to 5 miles per hour, on the cable's diameter of 1 inch instead of 2 inches. Then the range on the frequency would be 9 Hertz to 19 Hertz. Because the mode density of the same span length hasn't changed; namely, 5 per Hertz, on the small diameter we will have excited 50 modes. Thus, there are many modes and many frequencies to be concerned with. This is illustrated in the second part of Figure 19 where we have measured on the cable previously indicated, 1.2 inch diameter, over a three hour sample time period. Notice the large number of spikes in the spectrum which runs from 0 Hertz to 50 Hertz. Notice the maximum peak indicated at 12.6 Hertz accompanied by several other frequencies lower than that down to 6 Hertz range. Above that 12.6 Hertz there are several modes excited although not quite as much until in the range of 40 Hertz there is another dominate mode. Why the void between the low frequency 12.6 Hertz and 40 Hertz? That is not easily explained. Over a 3 hour period of time, it could be as simple as the wind direction changed to account for it. Well, one method that we have found to deal with the vibration is illustrated in Figure 20. In the upper half of the figure, one sees a vibration damper known as the AR DAMPER located at the close end of the anchor point on a

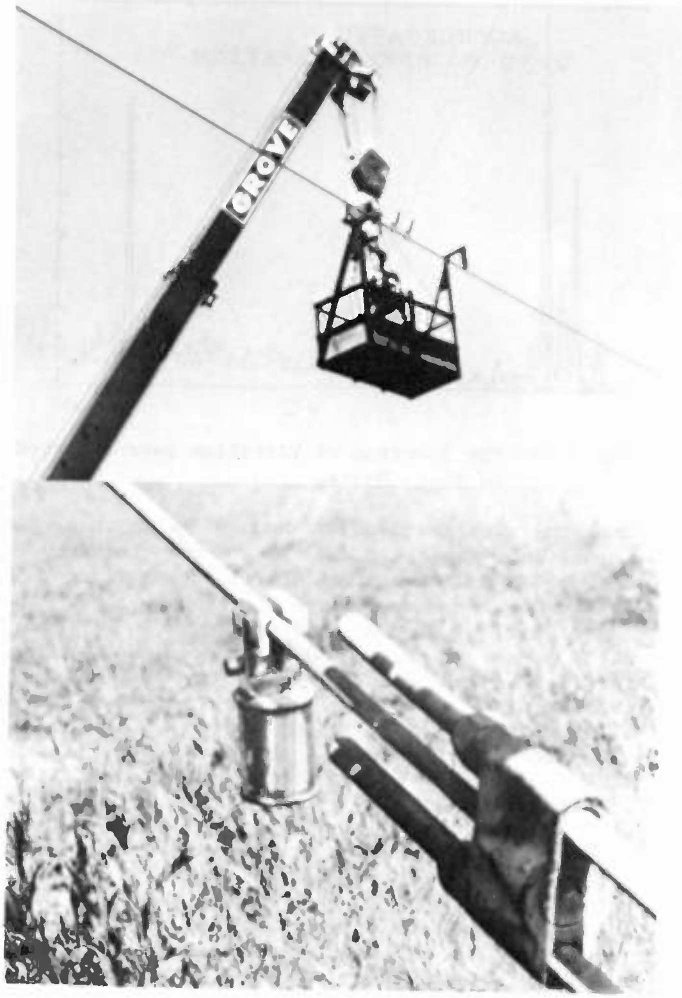

Fig. (20) The AR DAMPER Type of Vibration Damper

steel guy cable. In the lower half of the figure, the same kind of dampers are being installed farther out on the span on the guy cable by means of the bucket truck. Notice that there are two dampers being installed. Ordinarily the treatment of high frequency vibration calls for a minimum of 2 dampers per cable, one at each end. One at the lower end is identified as the low frequency damper one at the high end being identified as the high frequency damper. Depending upon the length of the cable and the diameter of the cable there may be two at the bottom, two at the top, three at the bottom or as many as a total of six or even seven dampers on a span. These are normally mounted in a V pattern with one damper on each side of the guy plane as seen in Figure 21. The distance for locating each group of dampers varies accordingly to diameter and tension of the guy. That becomes part of the general specifications for the system. Mounting the dampers in a vertical V is particularly useful. If for some reason the bolt is not fully tightened during installation, the damper clamp loosens. Before it can become really loose and cause damage, the damper will then swing down to a bottoming position which can then be easily detected and the situation quickly remedied. A closer view of the

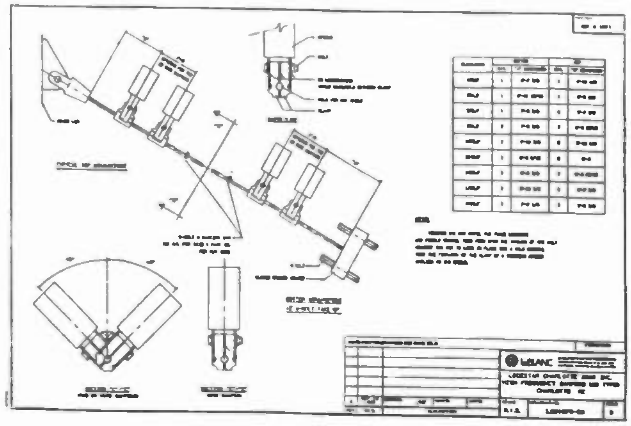

Fig. (21) Damper Installation

damper is seen in Figure 22 which illustrates the manner in which its assembled and put together. In the upper right is the system of aluminum brackets that are used with two holes in each. These are slipped into the damper weight with a curved slot arrangement that is seen in the accompanying sketch. The aluminum clamp seen at the bottom are then used to grip the cable, and fastened with a single steel galvanized bolt. The damper weight itself is aluminum. There are four models that are available. The weight itself goes from about 5 lbs. to 16 lbs. Clamps are individually matched to the cable from 1 inch diameter to 2.75 inch diameter in matching steps as close as 1/8 of an inch. This assures a tight grip to the steel cable by means of the aluminum clamp. The two bolt positions are either top or bottom. Top as shown in this figure provides for the gripping of the clamp combination with the weight in such a way that the weight is retained

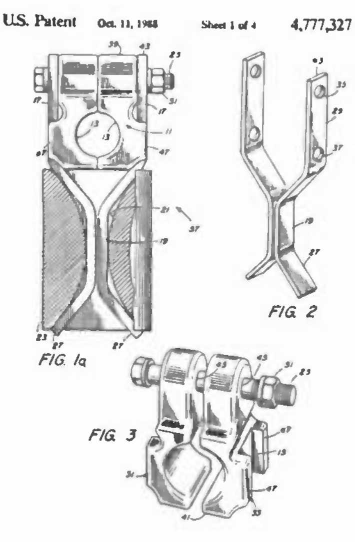

Fig. (22) AR DAMPER Detail

rigidly. In the arrangement previously mentioned of the vertical V when vibration occurs the offset weight twists the cable and through its internal hysterisis creates a damper necessary for control. A more common arrangement is one where the second bolt hole is used and the clamp is reversed from that seen here that the weight is loosely retained. The vibration creates an impact between the bracket and the weight, and the impact of the weight and the bracket provides the dissipated effect. Energy is taken out of the vibration. Limits have been tested in a laboratory setting seen in Figure 23 where the V arrangement is clearly seen and two models of weights shown, one with a smooth body and the other, not. Both weights are retained in the manner last described so that they would be loosely disposed for vibration damping. What is seen here is the top view of a vibration electrodynamics shaker, and then the accelerometer devices are on the weight, and on the shaker, on the cable, and on the clamp. Various tests were conducted on this assembly to measure the accelerations. In Figure 24, a more common type of damper is shown and the accelerometers are also shown on the same shaker and the same cable and the same clamps. The aim here is to compare the efficiencies of the AR DAMPER with the efficiency of the more common type damper. One type of test conducted is a random vibration test where the frequency ran from 5-40 Hertz. Figure 25 illustrates the input to both types of damper assemblies which was programmed by computer to the electrodynamic shaker. The overall acceleration input was 1G rms total over the full band width of 35 Hertz. The average level was in the range of about .03G's square per Hertz. The results of the test for the common damper and for the AR DAMPER are shown for the AR DAMPER in Figure 26, and for the more common damper type seen in Figure 27. Notice how the output spectrum for the AR DAMPER stays well above the output spectrum for the stockbridge damper. The overall root-mean-square response for the AR DAMPER was found to be 1.77G's rms and the overall response for the stockbridge damper was found to be 0.38 G's rms. These results illustrate the superior performance of the AR DAMPER.

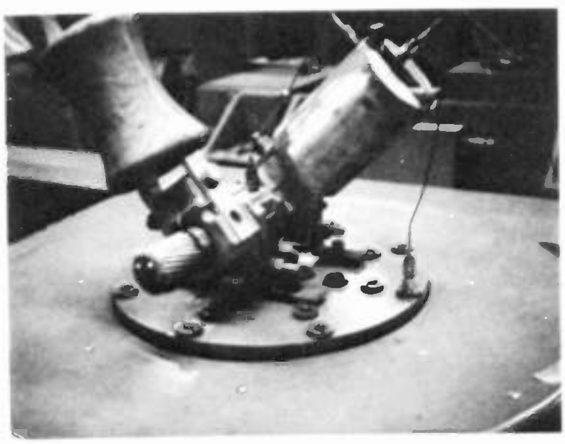

Fig. (23) Vibration Test of AR DAMPER

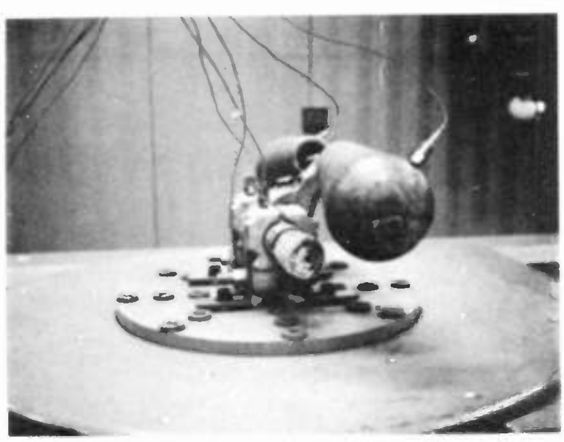

Fig. (24) Vibration Test of Stockbridge Damper.

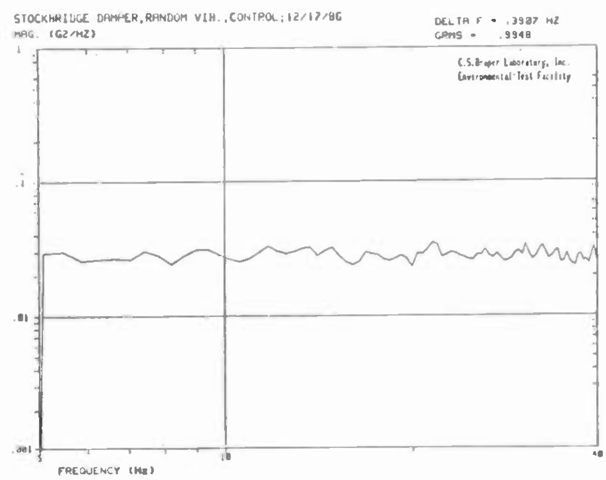

Fig. (25) The Acceleration Spectrum for the Laboratory Tests.

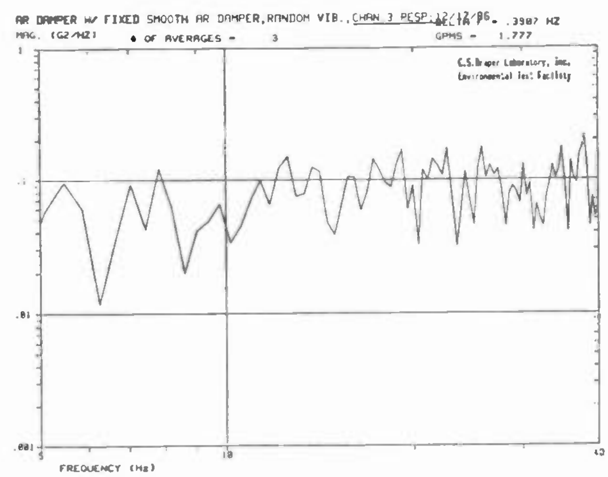

Fig. (26) The AR DAMPER Test Results

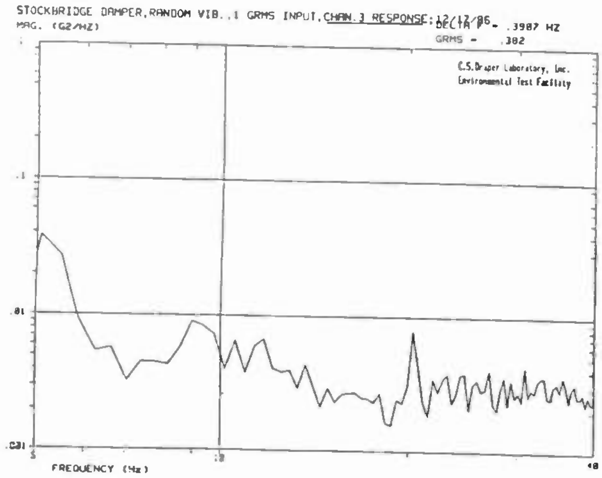

Fig. (27) The Stockbridge Damper Test Results.

CONCLUDING REMARKS:

The study and the analysis of vibration on guy wires and power conductors have been reviewed. Research over the past twenty-five years has been highlighted. In the case of low frequency vibration - galloping - the cause of it has been identified as aerodynamic lift force acting on airfoil/ice shapes. The remedy for it has been identified as a device that will prevent the dangerous buildup of destructive dynamic tension forces ocurring when the cables move in a vertical direction. The shape of the vibration on the span requires that the device be located well above the guy anchor. A device that provides both a snubbing action and a damping action has been identified as the SANDAMPER(R) Anti-gallop Damper for Tower Guy Cables.

In the case of high frequency vortex vibration, the vibration is seen to comprise a great many modes of vibration in the cable. However, at any one time only a few of these modes become excited. Field test results confirm the existance of many vibration modes over a wide range of frequencies. The remedy for high frequency vibration is the installation of several damper devices on each tower guy cable. Two type of dampers are identified for that purpose, a Stockbridge type of damper, and an AR DAMPER type. Laboratory tests were performed on each type under identical simulated conditions of high frequency vibration from 5 Hz to 40 HZ. The results of these tests are shown for comparison.

Both high frequency control methods and low frequency control methods are available commercially. The prudent broadcast executive will examine the needs that he has in his own situation, compare the devices that are available on the basis of his needs, and then decide if vibration protection will meet those needs effectively, and economically.

THE EMPIRE STATE BUILDING COMMUNICATIONS FIRE

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Overview

The word "Fire" is one of the most feared words heard by anyone especially a Broadcast engineer. On August 12,1988, "Fire" was heard by over a dozen Broadcasting and Telecommunications engineers that operate facilities at the Empire State building in New York City.

As a result of a massive fire in one of the main transmission line chases affected the majority of the broadcasting facilities operating in the building.

This paper will address this catastrophe from the broadcasters point of view, illustrate the conditions prior to the catastrophe and outline the work necessary to return the facility to normal operation.

Description of the Empire State Facility

The Empire State building, one of the world's tallest buildings, rises at an elevation of 1,454' above sea level. It is the second tallest building in Manhattan. Housed within the upper floors of the building one can find a critical number of Broadcast and Telecommunications facilities.

The technical data for the building is shown in Table 1.

In addition to housing the majority of New York City's FM stations, the building is a prime relay location for a number of ENG and TV programming microwave repeaters. Listed in Table 2 are some of the broadcasting facilities that were operating in the building on the day of the fire.

Table 1

Location	350 Fifth Avenue New York, NY
Height	1,454' AMSL
Floors	103
Tower	Skeletal steel 204'
Mooring Mast	16 floors
Broadcast Facilities	81st through 103rd Floors

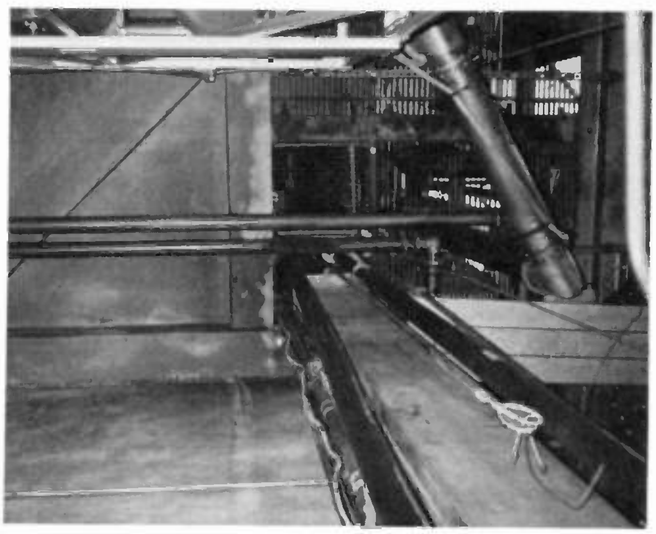
Various broadcast antennas are located on the 204' tower which is situated atop a decorative structure referred to as the "Mooring Mast". Atop this mooring mast is located the Master FM antenna which is circularly located around the 102nd and 103rd floors.

Table 2

WBLS	FM
WXRK	FM
WRKS	FM
WYNY	FM
WPLJ	FM
WNCN	FM
WQXR	FM
WLTW	FM
WNSR	FM
WBAI	FM
WEVD	FM
WCBS	FM
WHTZ	FM
WNEW	FM
WNBC TV	ENG Microwave
WABC TV	ENG Microwave
WHSE TV	Transmitter & Microwave Repeater

The Master FM transmitter facilities are located between the 81st and 85th floors. To route the numerous transmission lines from these facilities, a number of vertical chases exist throughout the building and the mooring mast. The various Master FM diplexers are located on the "E" levels within the skin of the Mooring mast structure.

The "E" levels comprise the 88th to the 100th floors. As shown in Picture 1, the steel grated floors are supported by the internal access stairway and the structural steel of the mooring mast. There are no internal walls other than the elevator shaft and the fire stairwell.



Picture 1

The fire of August 12, 1988 was contained in a vertical access area which runs from the 85th floor to the 103rd floor. This access area, or chase, is exposed throughout the vertical run with the exception of an enclosed portion from the 85th to the 88th floors and from the 101st to the 103rd floors. Picture 2 shows the portion of the 88th floor chase that is believed to be the origin of the fire. As an aside, the tear in the WR-1150 sidewall is the result of a fire-fighter's mistake in ventilating what he thought was heating duct.

The Transmission Line Run

The transmission lines shown in Table 3 are closely arranged within that portion of the chase that penetrates the 85th to the 88th floors. This grouping is also shown in Picture 3.

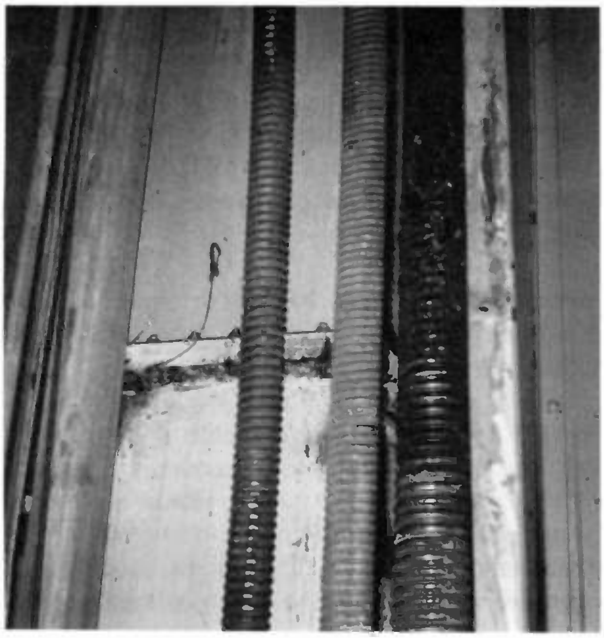
Table 3

WXRK FM	Main 3 1/8" line Standby 1 5/8" line
WHSE TV	Main WR-1150 waveguide EW-63 waveguide
WCBS FM	Main 4 1/8" heliax
Master FM	Diplexer interlock conduits

It may appear unlikely, from the relatively small number of items listed in Table 3, Although Table 3 indicates a small number of lines it appears unlikely that a fire in a single chase could wreck such havoc on virtually all of New York Cities FM facilities, a major TV station and various microwave repeater operations.

While the fire destroyed the transmission lines of only three stations, it completely ruined all the diplexer interlock wiring contained within the Mooring mast. As a result, the loss of this critical interlock wiring prohibited the undamaged Master FM stations from immediately returning to the air.

Picture 2



Picture 3

The Fire

The exact cause of the fire still remains unknown. Unfortunately, due to an continuing insurance investigation, several aspects of the fire are presently beyond the scope of this paper.

However, the source of the fire has been tentatively isolated to an area located in the chase between the 85th and the 88th floors as shown in Picture 2. A point to consider is that the most intense heat was generated within the two chase areas; the 86th to 88th floors and the 101st to 103rd floors.

The various coax and the EW-63 waveguide transmission lines in the chase were encased in the industry standard black polyethylene jacket. It is our opinion, that once the fire started within the lower chase area, it's confined space was responsible for the concentration of the heat necessary for the ignition of the polyethylene jacket material. The ignition point of this material is 660° Fahrenheit or 349° Centigrade. We also believe that, once ignited, the jacket material served as the primary fuel for the fire. As a result, the fire raced out of control up the entire internal length of the Mooring mast.

As the blaze passed each technical floor (or "E" level) the enormous heat ignited a number of sympathetic materials. In addition, a significant amount of electrical wires located immediately adjacent to the burning transmission lines were destroyed. Investigation disclosed that the wiring for both the Master FM diplexer interlocks and the diplexer blower assemblies was destroyed.

As the fire reached the 101st floor the chase again served to concentrate the heat within it's confines. This accounted for the extensive amount of damage to the 102nd floor observation deck area shown in Picture 4.

From this point on, we determined that the total vaporization of the polyethylene jacket also contributed to the extensive amount of airborne contaminants throughout the interior of the mooring mast. Extensive tests have been conducted to determine the extent of future damage to the RF components from acidic combustion byproducts. These tests indicated that no appreciable amount of contaminants were present on any of the RF components located within the Mooring mast.

Picture 4

Damage Assessment

Damage assessment was delayed until approximately 6:45 PM, at which time the NYC Fire Department released the building for occupancy. About that time, the majority of the electrical power was restored and the various engineers were able to begin determining the exact cause of the fire.

Damage assessment was hampered by the lack of elevator service in the mooring mast, loss of electricity in a number of the technical areas and the extraordinarily high temperature throughout the mast. The temperature that day, as recorded by the National Weather Service, was 94° F; however, we estimate that the internal temperature of the mooring mast was considerably higher.

One of our immediate concerns was to return to the air any of the undamaged facilities without causing a new fire. Since the exact cause of the fire was unknown at this time, the Chief Engineers were reluctant to take further action.

As soon as electrical power was restored the Master FM participants were able to resume broadcasting at a reduced level, the facilities with standby antennas returned to the air at full power and the various microwave facilities resumed operations.

Surprisingly, a relatively small number of broadcasters suffered extensive damage to their facilities from the fire. We have shown the severely damaged facilities in Table 3.

Master FM

To add to the consternation, AC power to the upper floors of the building was immediately shutoff by the firefighters. This action resulted in the loss of the Master FM interlocks, the antenna protection circuits and the diplexer cooling fans. As a result, the loss of these circuits prohibited the Master FM facilities having no standby antennas from immediately returning to the air.

On the succeeding day, tests were conducted to determine the exact damage to the master FM diplexers and antenna. They indicated that no damage was caused by the fire to any of the RF components. We should mention that the Master FM participants were not able to resume transmitting at full power because of the damage to the interlock and diplexer blower fan wiring which was replaced over the next several days.

Fortunately, the majority of the stations were able to return to full power by Wednesday August 17, 1988.

The FM facilities shown in Table 4 switched to back up antennas thereby resuming broadcasting within a short period of time. However, WHSE TV and WRKS FM were unable to resume broadcasting for some time because of the extensive damage to their transmission lines.

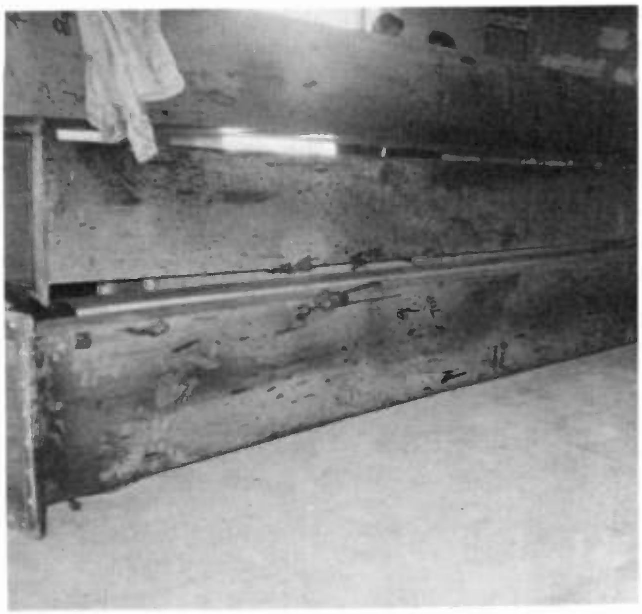
Table 4

WCBS FM	25 minutes
WHTZ FM (Master FM)	2 minutes
WNEW FM (Master FM)	38 minutes

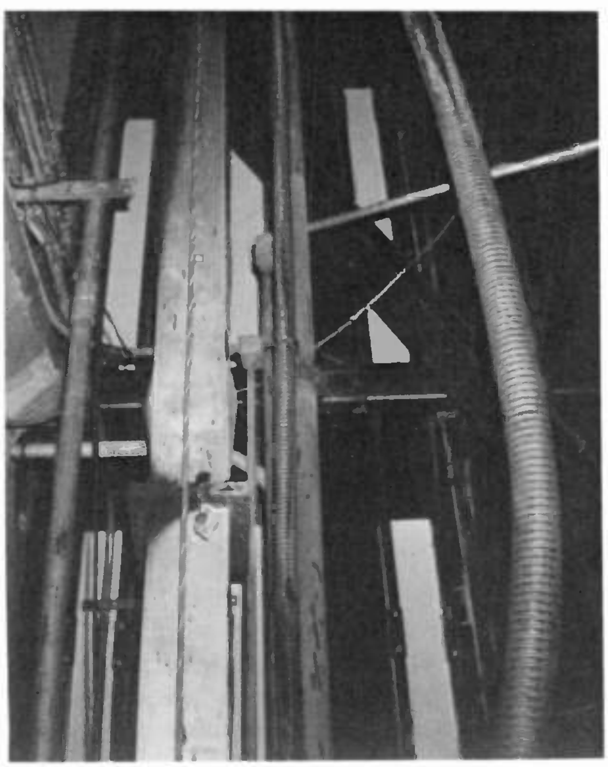
WHSE TV

The WR-1150 waveguide for WHSE was totally destroyed by the fire. Picture 5 shows the damage to several of the waveguide components and is indicative of the damage to the entire waveguide vertical run. The tremendous heat, from the burning polyethylene jackets on the coaxial and microwave transmission lines, resulted in the loss of tension in one of the steel support cable (the striped cable shown in Picture 6) serving as the vertical support for the WR-1150 waveguide run. The loss of this support system, along with the extremely high temperatures, resulted in the warping and buckling of all the vertical waveguide.

Damage to the vertical run of EW-63 microwave waveguide (shown in Picture 7) prohibited the continuation of programming on the Long Island facility serviced by the Empire State inter-city relay. A visual inspection of the waveguide by one of our engineers determined that no significant electrical damage was evident. Programming service was restored to the Long Island facility later that evening.



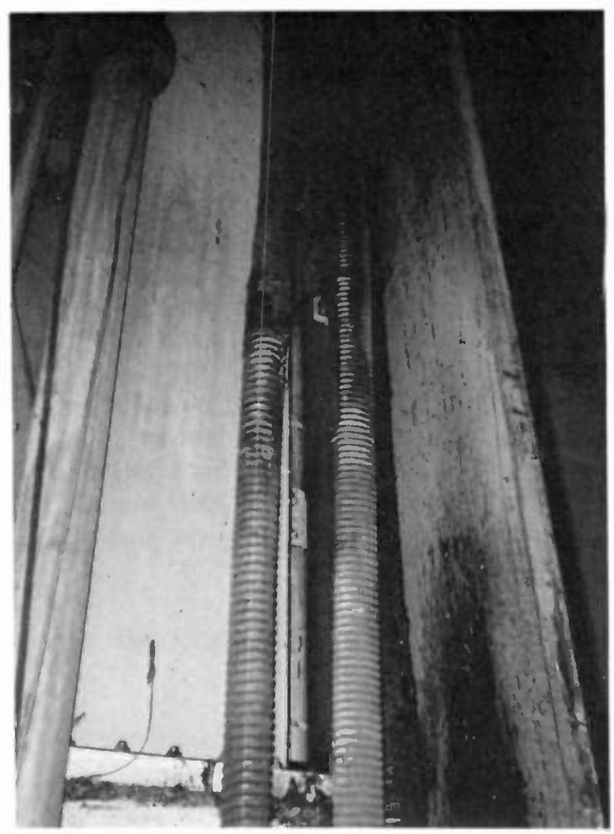
Picture 5



Picture 6

The replacement of the WR-1150 involved a considerable amount of time to repair. A visual inspection of the line indicated the extent of the damage. It was immediately evident that a significant portion of the vertical run of WR-1150 waveguide was destroyed. To replace the damaged sections calls were made to the manufacturer of the line (Microcommunications, Inc.) to determine the availability of suitable replacements. This was difficult to accomplish since sufficient quantities were not immediately available. Fortunately, WHSE had a number of pieces of WR-1150 at their Long Island facility. A truck was dispatched to transport them to the Empire State building for inspection by an MCI representative who determined that they were suitable for reuse.

Simultaneously, a second crew was already in the process of removing the damaged line in preparation for the installation of the new components. In picture 8 a section of line is shown being removed from one of the "E" levels.



Picture 7

Picture 8

Another point that we should not neglect is that as the damaged line was removed, a serious problem was encountered with the remaining undamaged components: water was found in all of the remaining line sections. All these sections had to be removed and cleaned before the station could resume broadcasting. To complete the task, the entire transmission line run from the transmitter RF switch output to the 103rd floor transition had to be removed and cleaned and/or replaced.

WHSE TV was able to resume full power service on Saturday August 20, 1988.

WCBS FM

WCBS FM was fortunately able to switch to their standby transmission line and antenna. Broadcasting was resumed at full power within a relatively short period of time.

During the damage assessment it was found that the primary run of 4 1/8" coax, from the 85th to the 103rd floor, was destroyed. Picture 9 shows the damage to the coax at one of the splice joints on the "E-2" level. In addition to the destruction of the outer jacket, the heat completely destroyed the inner polyethylene components of the coax.

Pictures 10 and 11 show the damaged 4 1/8" and 3 1/8" coax and several sections of WR-1150 after it's removal from the chase.

Picture 9

WRKS FM

WRKS FM, by far, suffered the worst damage of any of the FM facilities. In addition to the loss of it's primary 3 1/8" transmission line, the backup 1 5/8" transmission line and the diplexer interlock wiring were also damaged beyond repair.

Picture 10

Picture 11

Simultaneously, Mr. John Lyons, Chief Engineer of WRKS, obtained a temporary replacement transmission line. This replacement line was installed to bypass the damaged section from the 84th floor to the stations antenna changeover switch located on the "E-3" level. As a result, WRKS was able to resume broadcasting at 3:23 AM Saturday morning.

Work was already in progress to replace the destroyed main and standby transmission lines. By Wednesday, a third crew had completely replaced the damaged 3 1/8" and 1 5/8" transmission lines and the station was again operating at full power.

Conclusion

Many people were involved in repairing the extensive damage caused by the fire. We are grateful for their assistance and cooperation in bringing a potentially serious conflagration into check and returning operations to normalcy within a short period of time.

We cannot prove what caused the fire or prevent it's recurrence but this report shows what can be accomplished by a group of dedicated individuals.

ELECTROMAGNETIC INTERFERENCE TO AVIATION RECEIVERS—FAA CONSIDERATIONS IN THE BROADCAST SITE SELECTION PROCESS

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INTRODUCTION

The Federal Aviation Administration (FAA) has recently added a new computer tool to assist in the determination of potential electromagnetic interference (EMI) hazards to airborne receivers from FM broadcast station proposals. The EMI evaluation is made as a part of the FAA's obstruction evaluation process in determining whether proposed FM broadcast station facility changes will be a potential hazard to air navigation. This paper will discuss the implications of the FAA EMI policies and provide examples of the results obtained through use of the FAA computer model.

HISTORY OF THE FAA EMI PROGRAM

It has long been recognized that FM and television broadcast signals can cause interference with aeronautical receivers used for aircraft communications and navigation. Unlike most land based communications systems, interference to aeronautical radio systems presents pervasive safety implications.

The FAA evaluates new broadcast station construction proposals involving possible aeronautical obstruction hazards. Historically, that evaluation encompasses physical hazards to aircraft in flight. For some time, the FAA has included routine evaluation of potential electromagnetic interference (EMI) to aeronautical radio systems.

In the past, this EMI evaluation was made primarily to consider the need for modifications to FAA facilities or procedures as a result of the broadcast proposal. As the radio spectrum and airspace have become more crowded, fewer options are available to the FAA for relocation of affected facilities and procedural changes necessary to accommodate new proposals. The volume of new facility proposals generated by FCC Docket 80-90 has strained the limited FAA staff. Furthermore, the extensive exposure which the mass media have given to air safety concerns provided the FAA with further impetus to tighten its evaluation of broadcast proposals.

As a result, the FAA has been issuing more objections to broadcast proposals citing both physical considerations and EMI concerns. Even in those cases where "no hazard" determinations are issued, they are sometimes conditioned to limit operation strictly to the frequencies and power levels originally proposed.

Under the Communications Act of 1934, the Federal Communications Commission (FCC) is given authority to regulate non-government radio transmitting equipment. Other laws provide the Federal Aviation Administration with the responsibility to develop a safe and efficient air transportation system. Most aeronautical communications and navigation frequencies are controlled by the FAA. Assignment of FAA frequencies are handled by the Interdepartmental Radio Advisory Committee (IRAC), which provides spectrum allocation functions for the Federal Government. Recommendations and requests for particular aeronautical frequency assignments are made by the FAA.

In 1987, Public Law 100-223 clarified for the first time the authority of the FAA in regulating EMI to airway facilities. Congress directed the FAA and the FCC "to work together in reviewing broadcast applications and tower proposals."

The FAA recently contracted with Ohio University to develop a computerized model which would simulate the interference which could be caused by broadcast station proposals to aeronautical facilities. This has become the new method by which the FAA conducts EMI evaluations.

To date, the only information which has been provided is the program (and accompanying databases), a users manual, and a technical reference manual. Two separate versions of the program have been provided; the available documentation covers only the earlier version. Complete background data on the program have not been released to date, despite requests from the broadcast community.

Due to the limitations noted above, this paper is based solely on analysis of program runs, comparisons of individual test calculations, and readily available reference material. The program is subject to change at any time without notice to the broadcast community.

The FAA uses some terminology which is not common to communications engineering. Where possible, standard communications terminology will be used, with a brief explanation of the FAA terminology as appropriate.

FAA INTERFERENCE CONCERNS

The FAA considers any form of interference to airborne communications and navigation systems as a hazard to air safety. Broadcast FM and TV operations are a particular concern with respect to two blocks of frequencies, those used for navigation aids (NAVAID) and those used for communications (COM).

In recent years, the FAA has been particularly concerned with interference to operations in the 108 MHz to 137 MHz band from FM broadcast stations. The frequencies between 108 MHz and 118 MHz are used for NAVAIDS, with the remaining frequencies being allocated for air to ground communications. VHF Omni Range (VOR) and Localizers (LOC) share the 108 MHz to 118 MHz spectrum. VOR systems are used to provide in-flight navigation guidance, while localizers are employed in instrument landing systems (ILS).

The Airspace Model currently supports analysis of localizer systems only.

Potential applicants for broadcast facilities are cautioned that standard FAA procedures continue to require evaluation of proposals for potential interference to communication frequencies and non-localizer NAVAID facilities using existing evaluation standards. It is possible that the airspace computer model may be changed in the future to incorporate VOR or COM evaluations.

THE LOCALIZER

A localizer is an integral part of the standard ILS used by aircraft. 90-Hz and 150-Hz modulated signals are generated to provide the aircraft pilot with fly-left or fly-right instructions on a cockpit course deviation indicator (CDI). The CDI provides lateral guidance to the runway centerline. The localizer is normally paired with a glideslope transmitter (operating in the 330 MHz band) to provide the pilot with guidance as to the aircraft's vertical position on runway approach. Marker transmitters operating in the 75 mHz range provide indication when specified points are reached along the landing approach.

The US standard service volume (SSV), which defines the limits of the ILS runway approach, is "keyhole" shaped, extending 18 nautical miles from the localizer between angles of ±10 degrees from the approach centerline and 10 nautical miles over a span of ±35 degrees from the approach centerline. An expanded service volume which extends as far as 25 nautical miles is used at some less congested airports.

FAA standards require that the available signal strength at the receiver terminals be no less than -86 dBm at any point within the designated service volume. As with any VHF radio system, the localizer signal is subject to multipath and terrain effects. In practice, the actual available signal strength may be less than -86 dBm at some points within the service volume.[1/]

At present, the FAA is concerned with interference inside the specified NAVAID service volume for a given airport. It is clear, however, that the FAA has authority to consider the impact of broadcast facilities on future NAVAIDS for which the service volumes have not yet been defined.

TYPES OF INTERFERENCE

The FAA evaluates broadcast proposals for three potential types of interference to airborne receivers. These are:

° Receiver overload/desensitization. The FAA refers to this as type B2 interference.

° Adjacent channel. The FAA refers to this as type A2 interference.

° Intermodulation. This is referred to as type B1 interference.

These forms of interference types are not unique to aviation receivers. All receivers, including consumer grade broadcast receivers, are susceptible to one or more of these forms of interference.

In general, receivers used for aviation must be capable of tuning a wide range of frequencies. There are three basic groups of aviation receivers, navigation and communications and a hybrid combining both functions, referred to as NAV, COM and NAVCOM, respectively.

Each type of receiver is capable of tuning to any channel within its respective block of frequencies. In order to provide wide frequency coverage, these receivers have limited front end selectivity. This increases the probability of interference from unintended and undesired signals. As with other receivers, the front end design is a major factor in interference susceptibility.

SIGNAL STRENGTH CALCULATIONS

The FAA and aviation community calculate the available signal power using free space propagation equations, without correction for terrain and earth bulge effects. The signal strengths are typically given in decibels above one milliwatt (dBm) at the receiver terminals, Distance units are given in nautical miles.

The standard conversion factor from dBm to microvolts at a 50 ohm receiver is given by:

$$E = \text{antilog}\left[(P + 107) / 20 \right] \qquad [1]$$

where
E is the voltage in microvolts
and
P is the power in dBm.

The free space propagation employed for aeronautical analyses is given as:

$$P = EIRP - 20 \log [D * F] - C - LR \qquad [2]$$

where
P is the received power in dBm,
EIRP is the effective isotropic radiated power in dBm,
D is the distance in nautical miles,
F is the frequency in MHz,
C is the free space constant of 37.8,
and
LR is the receiving antenna loss factor at frequency F.

The EIRP in dBm is calculated by:

$$EIRP = 10 \log [ERP] + 62.2 \qquad [3]$$

with ERP being the effective radiated power in kilowatts.

Nautical miles may be converted to the more familiar kilometers or statute miles by using a multiplication factor of 1.852 and 1.1508, respectively.

The antenna loss factors vary depending on the type of analysis performed. For Venn diagrams (see below), the antenna factor for FM broadcast facilities is:

$$LR = 3.0 + 108.0 - F, \qquad [4]$$

with F in MHz. For the airspace computer model, the corresponding antenna rejection is

$$LR = 3.5 + 108.0 - F \qquad [5]$$

for frequencies less than 108 MHz and greater than 100 MHz, and

$$LR = 11.5 + 0.5\,(100.0 - F) \qquad [6]$$

for frequencies below 100 MHz, with F in MHz.

EVALUATING POTENTIAL INTERFERENCE

The Venn Diagram

The Venn Diagram was developed for the FAA as a method of evaluating the possible impact of high powered broadcast signals on airway facilities. Extensive bench and flight testing over the years by the FAA Technical Center, ARINC, Transport Canada, and others empirically determined the field strength levels required to cause overload, adjacent channel and intermodulation interference in aviation receivers.2/3/

Based on these tests, it was determined that overload interference could occur in aviation receivers when the broadcast facility signal strength exceeds -10 dBm. For high-end FM stations, adjacent channel interference may also occur within the -10 dBm contour. Intermodulation interference is possible where one station (designated prime) exceeds -20 dBm (for NAV frequencies) and a second station (designated secondary) exceeds -30 dBm. The locus of points resulting from the intersection of these field strength contours is the area within which intermodulation might occur. For three-product mixes, the prime signal level remains at -20 dBm. For both secondary stations the signal level is -30 dBm.

Since uncorrected free space propagation equations are employed, the signal contours are concentric circles centered on each broadcast facility. The pertinent contours are plotted on a map, along with the NAVAID service volume. If any part of the interference area falls within the service volume of the NAVAID, the FAA will conclude that the NAVAID is likely to be affected by interference. An objection to the broadcast proposal will be made.

Examples of Venn Diagrams are shown later in this paper.

The Venn diagram represents a simple approach to determining possible interference from broadcast facilities. It is based solely on empirical data, rather than intermodulation equations or receiver theory. The Venn Diagram does not account for the front end response characteristics of the aviation receiver, and may overestimate interference involving low end FM stations (and underestimate interference from upper end FM stations).

For these reasons, the judgments made by means of this method are only approximate. To date, however, the method has provided a uniform means of evaluating broadcast proposals for their impact on aviation facilities.

The Airspace Model

The Ohio University computerized interference model is intended to automate and improve the evaluation of broadcast proposals.

The program is designed to predict the occurrence of A2/B2 (overload and adjacent channel) and B1 (receiver intermodulation) interference within localizer service volumes. VOR interference calculations are not yet implemented.[4/] The program operator can choose interference calculations for the standard service volume, extended service volume, or the back course at each facility.

The earlier and present versions of the program are nearly identical; the later version includes FAA NAVAIDS which might enter into intermodulation products along with broadcast stations.

For each type of interference examined in the model, receiver tests were conducted to establish a threshold interference level.[5/] This threshold level forms the basis of the equations employed in the interference model.

The program developers conclude that the new test data for overload and adjacent channel interference correlates quite well with previous data obtained in tests conducted by the FAA, FCC and Transport Canada.[6/] Assuming that the test results are accurate, this correlation supports the equations, but only at the limits of the service volume.

No such correlation exists for intermodulation, and the documentation fails to explain this discrepancy. A review of the available test data tends to support a conclusion that inadequate tests were conducted in the development of its interference model.

The airspace model attempts to predict interference based on the theoretical intermodulation equations. This method will provide accurate results if the receiver transfer function is properly developed.

The power level of an intermodulation product is related to the components in the following manner:

$$P_0=2P_1+P_2+TF \qquad [7]$$

for the two frequency case and

$$P_0=P_1+P_2+P_3+TF \qquad [8]$$

for the three frequency case, where
P_0 is the power of the product in dBm,
P_1,P_2, and P_3 are the powers of the components in dBm, and
TF is the transfer function of the receiver.

The transfer function is frequency related and dependent on the front end characteristics of the receiver.

In the airspace model, the transfer function is defined as threshold level at which interference occurs in test receivers. The concept of a desired-to-undesired (D/U) signal ratio is not employed, except as a factor of the interference threshold. There is no explicit D/U ratio calculated or used.

The airspace program documents describe the test procedure used to develop the interference criteria. In brief, a simulated localizer signal is introduced into the receiver at a constant level of -86 dBm. The interfering signal is simultaneously introduced, and its level increased until the CDI is deflected 9 microamps, the maximum acceptable course deviation error established by the FAA. For intermodulation measurements, the simulated FM signals are introduced at equal signal levels, along with the -86 dBm localizer signal. The simulated FM signal levels are increased simultaneously until 9 uA of CDI deviation is noted. The threshold level for interference is simply the signal level of the component frequencies. Previous work by RTCA concluded that this was a valid method of evaluating intermodulation susceptibility 2/.

The receiver transfer function for intermodulation was then determined by plotting the interference threshold levels against the "log frequency product" of the signals, defined by the equation

$$\log((f_L-f_1)*(f_L-f_2)*(f_L-f_3)) \quad [9]$$

where

f_L is the frequency of the localizer in MHz

and

f_1,f_2,f_3 are the mix components in MHz (for the two frequency mix, $f_1=f_2$).

A best fit line was drawn for the worst three receivers at each set of test frequencies to define the equation of the receiver transfer function. Since the simulated FM signals levels were equal, the "equisignal threshold level" is established by intermodulation equations [7] and [8] as 3 times the signal level of the component threshold level.

This transfer function will be independent of the signal level of each intermodulation component, provided that the receiver front end operates in a linear fashion. In practice, receivers are not perfect; however, the non-linearity is negligible, except when the receiver is overloaded. The Airspace Model computes overload separately; therefore the intermodulation calculations assume that the receiver is not overloaded.

The test results provided tend to validate the threshold equations for intermodulation combinations with a log frequency product of approximately +0.5 and greater.

There are inadequate test results to establish the validity of the transfer functions at lower log frequency products. In the two frequency case, only one test point below 0.5 was made (at approximately -0.35), and the lowest log frequency product used for the three frequency test was at approximately +0.3.

For log frequency products below +0.5, the model appears flawed.

The receiver intermodulation equation is expected to be different for very small (and negative) log frequency products. Log frequency products below 0.5 occur when the frequencies producing the intermodulation are close to the product frequency. For airspace analysis, the product frequency is also the desired or adjacent localizer frequency. The front end frequency response of a typical aviation receiver offers less rejection to FM signals near 108 MHz than to lower FM frequencies, which are on the skirts of the front end selectivity curve. Since the transfer equation was developed using a best fit of log frequency products which lie on the selectivity skirts, it is expected that the equation will not adequately model receiver performance at higher FM frequencies. A linear equation factor in this region may not be appropriate, and may produce far greater protection ratios than necessary.

This is best shown by example.

For purposes of demonstration, assume that an airborne receiver tuned to 108.3 MHz is also subject to two FM signals, one at 107.9 MHz, and one at 107.5 MHz. In the worst case, the receiver front end offers no attenuation of the two FM signals compared to the NAVAID (this assumption is reasonable only for FM frequencies close to the localizer frequency). The full FM signal levels will be present at the device which produces the intermodulation product (an RF amplifier or mixer). The actual level of the intermodulation product is calculated as $2P_1 + P_2$ using equation [7]. For these frequencies, the FAA model predicts an interference threshold value of -157.3 dBm (equivalent to .003 uV). Since perfect front end response has been assumed, this will be the actual value of the interfering intermodulation product at the localizer frequency of 108.3 MHz. Using the FAA minimum localizer signal level of -86 dBm (7.7 uV), an intermodulation product at the threshold level established by the model provides a D/U signal ratio of 68 dB on the localizer frequency. In other words, the FAA equation predicts that interference will be caused for products which are greater than 68 dB below the localizer level. Such a low level intermodulation product will likely be well below the receiver noise floor. For typical receivers, front end filtering would attenuate FM broadcast signals and provide an even lower intermodulation product (and higher D/U ratio).

The FAA's own frequency allocation guidelines specify that a 23 dB D/U signal ratio is required at the edge of the service volume to avoid interference between co-channel localizers.

This analysis shows that even under the worst conditions the Airspace Analysis Model offers far more protection than is required under FAA criteria.

The effect of the FAA model is easily demonstrated. Using the case shown above, the model predicts interference for equal FM signal levels of -52.4 dBm. If the primary FM station were to have a signal level of -20 dBm (that used in the Venn Diagram), the secondary station would be required to be less than -117 dBm (0.30 uV) to avoid interference under the FAA model. For a 23 dB D/U ratio, the secondary level would cause interference only if it were greater than -69 dBm (79 uV).

The maximum interference threshold used by the program should never be greater than-109 dBm, using the FAA's 23 dB localizer D/U protection ratio.

The Airspace Model predicts greater interference than the Venn Diagram for log frequency products of 2.41 (two frequency intermodulation) and below. At log frequency products above 2.41, the airspace model predicts less interference than the Venn Diagram. Antenna factors were not considered in these calculations.

For NAVAIDS near 118 MHz, there is less likelihood of interference from FM stations. However, inclusion of FAA NAVAIDS as mix components can produce a finding of additional interference due to the low thresholds levels involved. Note that the FAA does not routinely consider intermodulation products to other NAVAIDS when assigning frequencies to facilities.

Unlike the Venn Diagram, normal operation of the model employs vertical plane radiation characteristics for FM antennas. A "generic" antenna is normally employed, which assumes main beam response for elevation angles of ±5 degrees. For other angles, the response is -14 dB. Several other antenna types may be selected by the user (it appears that other NAVAIDS included in the broadcast list are calculated using the generic antenna).

The model does not account for terrain or earth bulge correction. This will not normally be a factor, except in those cases where very low intermodulation thresholds are involved, and the stations considered in the products are located far from the affected service volume.

Another concern is the protection of a uniform -86 dBm localizer signal. The FAA justifies this by stating that the -86 dBm level is a target value, and may not actually be present at all points in the service volume. It might be more appropriate to employ a higher signal level at locations near runway end of the service volume since the aircraft will be lower and close to the localizer. With the plane at a lower altitude, terrain and earth bulge effects will come into play and the localizer signal will be stronger due to the aircraft proximity to the transmitter.

Reported "Bugs" in Airspace Model

Several users of the Airspace Model have reported errors in the airspace model and documentation. These errors include typographical errors in the constants for the two-frequency intermodulation equation shown on Page 21 of the technical reference manual, and calculation errors involving the two-frequency intermodulation model. The reported calculation errors are related to the computation of field strength at each test point.

The FM database is derived directly from the FCC database. Errors and missing information are frequently noted in the FCC database.

As detailed above, the model appears to

suffer from an inadequate development of the intermodulation transfer equations.

Of more importance is the FAA's representation of the service volume. In several cases studied by this author, the FM antenna is flagged as being located within the service volume even though the antenna is located on a mountain top which itself extends into the service volume. Other users have noted the lack of accounting for terrain and existing man-made limitations to the service volume.

Comparison of Analysis Results

To evaluate the Airspace Analysis Program for this paper, test runs were made for MacArthur Airport in Islip, New York. The author has some previous experience with the FAA involving interference caused to the localizer at Islip.

There are two localizers at Islip, both use 108.3 MHz. For this paper, the localizer identified as ISP was analyzed. This localizer provides guidance for aircraft approaching the runway at a heading of 45 degrees true. A standard service volume was used.

Station WEBE, Westport, Connecticut was chosen as the proponent. WEBE operates on 107.9 MHz, the top of the FM band. This is close enough in frequency that adjacent channel and overload interference would be of concern.

As expected, the Airspace Model showed substantial interference from WEBE. However, the interference was solely from predicted intermodulation; brute force overload and adjacent channel interference was not a concern. Of the numerous interference predictions produced by the model, two were chosen for comparison analysis by Venn Diagram and spot checks by hand to determine the accuracy of calculation.

Figure 1 shows the Venn Diagram and Airspace analysis for the two frequency product involving WEBE (107.9 MHz) and WBLS (107.5 MHz). The Venn Diagram shows no overlap between the WEBE -20 dBm contour and the WBLS -30 dBm contour. Under the Venn diagram analysis, no interference would be predicted to occur.

The Airspace model provides different results. The program output shows that interference would be caused over the entire service volume. For this combination, the interference threshold was computed to be -157.3 dBm.

Interference calculations were made by hand at the 5 outside corner points of the service volume. Due to the distance from each station, the elevation angle from the broadcast antenna was close to zero, and the full effective radiated power was used.

The calculations indicate that the threshold interference level would be exceeded across the entire service volume, despite the relatively low power of WBLS and the distances of the stations from the service volume.

Figure 2 shows the comparison for the product resulting from WEBE, WVIP, Mount Kisco, NY (106.3 MHz), and WBLI, Patchogue, NY (106.1 MHz). The low power level of WVIP is apparent in the small Venn Diagram radius. Here again, the Venn diagram does not predict interference.

The Airspace Model predicts interference over nearly the entire floor of the service volume. Again, hand calculations at the corner points of the service volume confirm the plotted results.

Similar test runs performed for other airports revealed that the Airspace Model predicts severe interference to NAVAIDS from one or more FM stations. It is safe to assume that nearly all FM stations in this country could not meet the criteria set forth by the Airspace model, particularly those operating above 100 MHz.

The latest version of the airspace model reviewed in this study includes FAA NAVAIDS in the FM stations listing. Intermodulation calculations are performed including these stations. The interference threshold is evaluated using the same equations. There is no consideration that the frequencies involved are above the localizer of interest. Again, no documentation is provided that these equations continue to hold for these frequencies. In fact, an even greater interference threshold is produced for the combination of 109.0 MHz, 108.8 MHz and 107.9 MHz than is produced for the two-frequency intermodulation involving WEBE and WBLS. To compound the problem it is apparent that the generic FM station antenna is employed for the VOR facilities. This is not a true representation of the performance of the VOR antenna.

CONCLUSIONS

The development of the FAA airspace model is a move towards mathematically correct means of analysis of interference to NAVAIDS. The fundamental concepts of frequency-corrected intermodulation equations and brute force calculations offer better protection to navigation receivers, and more accurately represent real world receiver performance. Use of vertical antenna patterns and directional antennas for these analyses provide a more realistic estimate of real world interference conditions. The broadcast industry should welcome the development of this form of analysis.

The present airspace model, however, is severely flawed in its representation of intermodulation interference. For certain frequency combinations, the model offers far greater protection to NAVAIDS than the FAA's own co-channel criteria. There is no supporting evidence that receiver performance tracks the model equation for log frequency products of less than 0.5. The inaccuracy of the model is particularly evident when other NAVAIDS are considered as part of an intermodulation mix. This is the single greatest flaw in the model. If the current model were correct, nearly every airport in the country would presently experience debilitating interference to the ILS from one or more FM stations. To date, no evidence is available to support this conclusion.

Several flaws and "bugs" exist in the Airspace Model. The most serious flaw is the apparent problem with the intermodulation model. The net result is that the program overstates the interference which will actually occur. Several of the operational "bugs" have been reported to the program developers, and a replacement version of the program is expected in the near future. It is quite likely that the FAA will not voluntarily release the updated version to the broadcast community.

The FAA has, apparently, decided to forgo the usual public comment and joint development of standards for evaluating broadcast proposals. There has been no outside scrutiny of the test results which led to the development of this standard. Of particular concern is the lack of correlation with long standing tests of receiver intermodulation performance without explanation.

The best route to resolution of interference between broadcast facilities and NAVAIDS is cooperation between the aviation industry, the broadcast industry, the FAA and the FCC. It would not be unreasonable to include representatives of the common carrier and land mobile industries who may find the FAA analyzing their higher power fixed stations. Ultimately, the standards applied must be technically correct, well tested, and appropriate to achieve the goals of broadcasters and the aviation community.

1/ FAA Airspace Analysis Manual Technical Reference Manual.

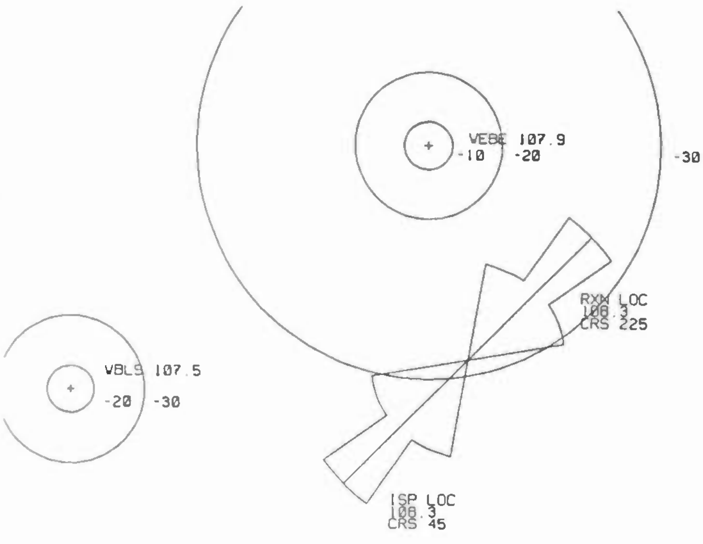
2/ Spectrum Management Regulations and Procedures Manual, Federal Aviation Administration, September 1987 (Appendix 1, P.12).

3/ FM Broadcast Interference Related to Airborne ILS, VOR and VHF Communications, Document RTCA/DO-176, Radio Technical Commission for Aeronautics, November 1981.

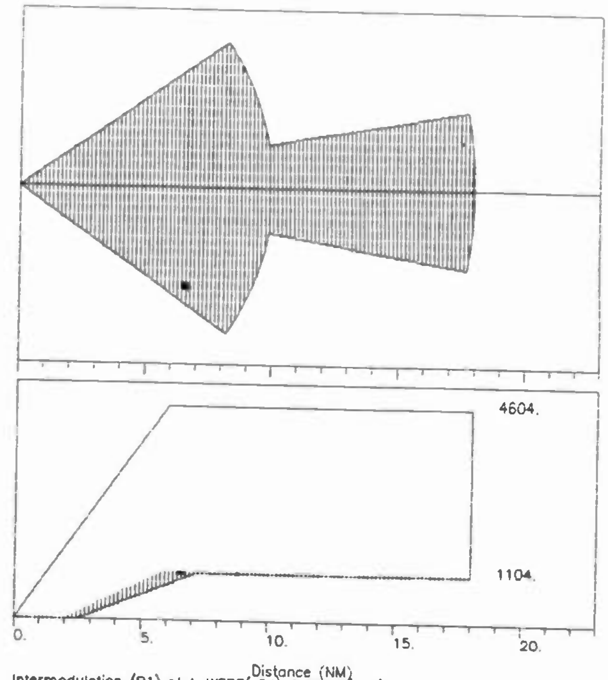
4/ FAA Airspace Analysis Manual Technical Reference Manual.

5/ Ibid.

6/ Ibid.



Venn Diagram

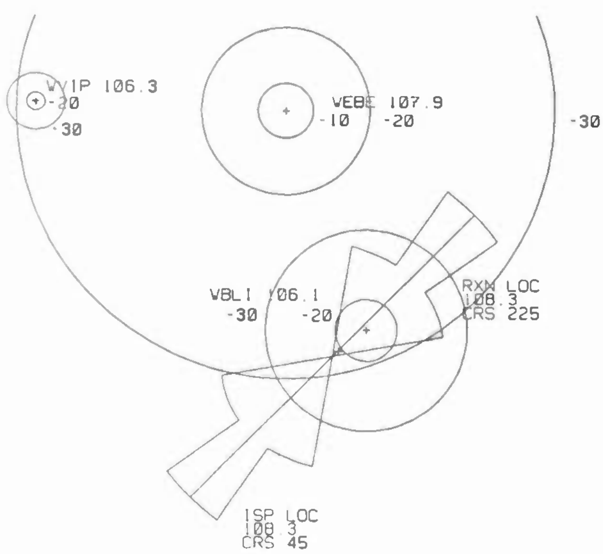


Intermodulation (B1) plot: WEBE(94) & WBLS(93)
ISP 108.30 MHz
Heading = 45.0
Runway Altitude (Ft MSL) = 104.
Latitude = 40 – 48 – 16 N Longitude = 73 – 5 – 30 W
Frequencies: WEBE = 107.90 MHz WBLS = 107.50 MHz

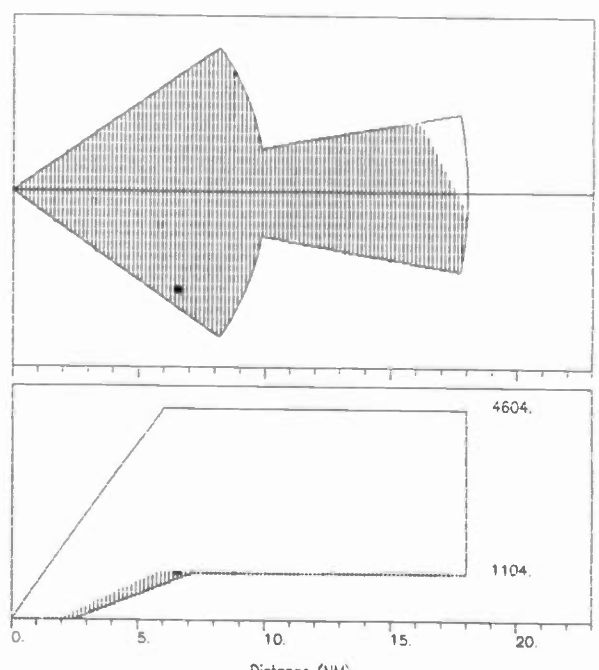
Airspace Analysis Program

Figure 1

EMI ANALYSIS COMPARISON FOR
WEBE AND WBLS TO ISLIP LOCALIZER



Venn Diagram



Intermod (B1) plot: WEBE(94), WVIP(89), & WBLI(87)
ISP 108.30 MHz
Heading = 45.0
Runway Altitude (Ft MSL) = 104.
Latitude = 40 – 48 – 16 N Longitude = 73 – 5 – 30 W
Frequencies: WEBE = 107.90 MHz WVIP = 106.30 MHz WBLI = 106.10 MHz

Airspace Analysis Program

Figure 2

EMI ANALYSIS COMPARISION FOR
WEBE, WBLI AND WVIP TO ISLIP LOCALIZER

DEALING WITH RADIO FREQUENCY INTERFERENCE COMPLAINTS

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When it comes to dealing with radio frequency interference complaints, many stations do not have the relative "luxury" of a mountain top or rural transmitting site. Growing suburban areas are engulfing formerly sparsely populated antenna sites and with the electronics industries pouring new, oftentimes hardly R.F.I. suppressed devices into the market, the licensee is typically faced with the occasional radio frequency interference complaint. What is the station required to do in this instance? Dealing with the complainant is a part of the job of the broadcast engineer and dealing with these complaints may be required under current F.C.C. rules. The station must be involved in this, whether by providing information and advice, by being a liaison between complainant and manufacturer or even by making a house call. Before exploring this further, let's review our current regulations.

F.C.C. RULES

The A.M. regulations are quite straightforward: "the licensee of each broadcast station is required to satisfy all reasonable complaints of blanketing interference within the 1 V/m contour." This contour can usually be found in the station's technical files [1].

The F.M. rules, however, are a bit more involved. The F.C.C. has determined that the 115 dBu (562 mV/m) contour is the area to be protected from blanketing interference. Since most stations do not have this information handy, this area can be determined by either of the following formulas :

$$D \text{ (in kilometers)} = 0.394\ (P_{KW})^{1/2}$$

or

$$D \text{ (in miles)} = 0.245\ (P_{KW})^{1/2}$$

After calculating the 115 dBu contour, the station can determine its obligation. The licensee must satisfy all complaints within this 115 dBu contour, for a one year period following any new construction; resolution shall be at no cost to the complainant. However, there are several specific problems that the licensee is not obligated to correct including malfunctioning or mistuned receivers, improperly installed antenna systems, use of high gain antennas or antenna pre-amplifiers, mobile receivers and non-R.F. devices such as tape recorders, hi-fi amplifiers, phonographs, C.D. players and telephones. After the one year requirement is fulfilled, the station must provide technical information or assistance to the complainant [2].

The F.C.C. rules do allow you to determine your obligation. However, just because the complaint is outside of the protected area, the station cannot ignore the fact that it is a part of the community. The station is a neighbor to many people; by providing a bit of concern and assistance the station can be a good neighbor.

THE INITIAL COMPLAINT

The complaint usually occurs as a telephone call to the evening or weekend announcer. "Your radio station is coming in on my T.V. set!" Most announcers are not particularly adept at correcting this type of problem: the best move is to forward this complaint to someone who can. Do not have them call back next week--instead, get their name, address, phone number and a description of the problem that they are experiencing. Let them know that the Chief Engineer will be receiving their complaint and will get back to them the next business day. Should this complaint have occurred during the business day, have the receptionist get this information from the complainant if the Chief Engineer is not available. Be sure that these complaints are treated as such and that the complainant is treated courteously, as in any other business. The complaint should then be forwarded to the Chief Engineer, or in his absence to the General Manager, for the return call. On occasion, the station may receive a complaint letter. It will be handled the same as the telephone complaint, but be sure to place a copy of the letter in the public inspection file, unless otherwise requested [3].

The most important aspect of handling a complaint is the initial contact by the Chief Engineer or the General Manager. Many times the complainant is upset because the radio station is "causing" all of the problems that are being experienced with his equipment. It is best to be a good listener at this time, and let them explain exactly what is happening, with what type of equipment, when does this occur, etc. Let them know that you are concerned, that the station is concerned, and that you understand. By having a management level individual take care of their problem, their complaint is important; by listening to their complaint, it should help reduce any animosity that might have existed.

The station must now determine the course of action it plans to take. If the complaint is occurring within the blanketing contour, the station is required to take some form of action, as previously mentioned; should the complaint be outside the blanketing contour, you should at least be prepared to offer advice and information. If you are not able to provide the necessary assistance or information at that time, set up another call with the complainant, and keep this commitment. Also, be sure to document any complaints and follow up actions.

CORRECTIVE ACTION

There are several areas which can be addressed. The approach taken will vary according to the situation. The first thing to be done is to ascertain whether the station is operating as authorized, including power and modulation within limits; also, make sure no spurious emissions are being generated. This is a good place to start, especially when a rash of complaints occurs. Once the station has determined proper operation, the complaint itself can be addressed.

After reviewing the complaint, many times the station will be able to provide information to the complainant they may be unaware of. A good example of this is someone who's hobby is DXing--trying to receive distant signals can be done, but it is much more difficult with a several kilowatt broadcast station nearby. There are just certain things that do not work like everyone would like them to. You will have to explain these type of situations in terms that they can understand.

Simple remedies can be as easy as moving the stereo to a different location in the room. Try another electrical outlet. If it is a stereo system, shielded speaker wires may help, especially if the R.F.I. is not noticed in the headphones. Component systems are usually grounded together by the shielded audio cables--adding dedicated grounds between components, and perhaps earth ground, may eliminate the problem. Also, the lengths of these audio cables may be a factor, as many times they make fine F.M. antennas connected into a phono preamplifier. Antennas are another source to look at. If the antenna is outdoor, is it installed properly and oriented properly? If it is old, is the co-ax or twin-lead making proper connection?

These simple remedies do not work every time and sometimes more elaborate measures may be required such as filtering. If the source of interference is suspected to be the A.C. line, simple plug-in filters are available. An overload to the T.V. antenna input can be usually corrected by installing an F.M. trap, or high pass filter for A.M. problems. F.M. interference in an F.M. tuner would be reduced by a tunable trap. However, other signal problems may result in interference--if the complainant is using "rabbit ears" to pick up T.V. signals from several miles away up to the fringe areas, an outdoor antenna may be required to get a decent signal, and if necessary add a trap. Telephones may also require filtering to eliminate problems. Some local telephone companies will provide filtering on the line, and some telephone vendors sell filters for R.F.I. protection. While a capacitor or choke may be all that is required, the lines themselves are F.C.C. regulated. You will not just be able to hang a component or two on the line--stick to the commercially available units.

If you have exhausted all of the easier remedies, you may be faced with the most involved, and usually most expensive, corrective measure: the internal equipment modification. The addition of capacitors, chokes and ferrite beads into the troublesome stages or inputs should reduce or eliminate problems when all other methods fail. Modifying the complainant's equipment yourself is not recommended for a variety of reasons, the most important being safety. Also, most equipment warranties will be voided by these modifications; and, should any other problems occur in the modified unit at a later date, the station may be held accountable, even if it is an unrelated problem or failure. Finally, you may not be allowed to make certain equipment modifications because, for instance, F.C.C. authorization is required to make internal changes in a telephone [4]. Should modifications be necessary, work with a qualified and factory authorized technician--this can save you a lot of headaches in the long run.

There are additional items to consider: if the problem equipment is new, it may be defective and covered by a warranty. Perhaps a different model or brand of equipment may be less sensitive to interference and should be substituted.

With this background on various corrective measures, you can help provide information to the complainant either by telephone or by mail. You also may want to consider setting up an appointment for a house call, if the problems are severe, or if the simpler measures fail.

THE HOUSE CALL

Making a visit to the complainant's residence or business is not exactly going into enemy territory, but it is foreign. You will be dealing with unfamiliar equipment, surroundings and faces, and possibly unfamiliar problems. When you go, you may want to bring along a few hand tools, and perhaps a trap if the complaint indicated this might help.

After presenting your business card, ask the complainant to demonstrate the problems being experienced. Ask if it only happens in certain modes or on certain channels, along with any other pertinent questions. Check all grounds and look for improper connections and installations. If it is a component system, try to isolate the piece of equipment that is suspect--once found, try moving it around for a location less sensitive to R.F. pickup. Television problems are usually caused by an input overload--try the trap that you hopefully brought along. In severe cases, two traps may be required.

If you're lucky, the inspection and troubleshooting will be successful in solving the interference problem. If it were not, there may be one other option to consider, which is assistance from the manufacturer. Perhaps, the station may want to contact the manufacturer on behalf of the complainant, or at least be available to assist in explaining the problem. A service technician can then make the necessary modifications, but if the unit is out of warranty there may be a charge for these changes.

If no cost effective solution can be found, this is a problem that the complainant may have to live with, without incurring additional expenses for new and more R.F.I. suppressed equipment or costly internal modifications. Most complainants will not like to hear this, but at least an honest effort was made for them.

Be sure to close each complaint file by documenting what follow-up action was taken, by whom, where and when. Also, if the complaint was unresolved, always send out a letter explaining why the problem could not be corrected, and keep a copy of this letter on file along with the rest of the complaint information.

ADDITIONAL HELPS

Every station should have the current F.C.C. rules and the N.A.B. Engineering Handbook available. Do not overlook the F.C.C. "Interference Handbook." Be sure that you have at least one copy on file; and, if you have several, copies can be sent to complainants. The booklet is available from the Government Printing Office, and also from the Electronics Industries Association (E.I.A.). To receive copies of the handbook or copies of the E.I.A. pamphlet "Consumers Should Know Something About Interference," send a large S.A.S.E. along with your request to:

E.I.A./Consumer Electronics Group
P.O. Box 19100
Washington, D.C. 20036

The E.I.A. can also help in reaching manufacturers and provide other technical information and pamphlets.

Some vendors and manufacturers are more aware of potential interference problems now, and are addressing this matter. A major retail electronics chain now sells F.M. traps at its local outlets. A large telephone manufacturer markets plug-in R.F. filters at its phone stores. Just a few years ago, finding some of these filters and traps could be somewhat difficult.

For more information you may also want to contact your local F.C.C. Field Office. Other broadcast engineers in you area and your local Society of Broadcast Engineers chapter may also have some good ideas on correcting difficult R.F. interference problems.

Remember, there has been no deregulation of R.F. interference complaint regulations. But except for those blanketing interference problems that meet the criteria in the current F.C.C. rules, the licensee is probably not required to correct many of these complaints. The station should develop a policy for dealing with radio frequency interference complaints, especially when new F.M. construction is proposed or now occurring, as there is a definite obligation in this instance. Helping each complainant is a good public service--every complainant is a potential listener and a potential diary respondent.

REFERENCES

[1] F.C.C. Blanketing interference, 47 C.F.R. §73.88 (1987).

[2] F.C.C. F.M. blanketing interference, 47 C.F.R. §73.318 (1987).

[3] F.C.C. Retention of letters received from the public, 47 C.F.R. §73.1202 (1987).

[4] Federal Communications Commission, "Interference Handbook," 1986, p. 19.

APPENDIX

The following is the current F.C.C. rule for F.M. complaints:

§73.318 FM blanketing interference.

Areas adjacent to the transmitting antenna that receive a signal with a strength of 115 dBu (562 mV/m) or greater will be assumed to be blanketed. In determining the blanketing area, the 115 dBu contour is determined by calculating the inverse distance field using the effective radiated power of the maximum radiated lobe of the antenna without considering its vertical radiation pattern or height. For directional antennas, the effective radiated power in the pertinent bearing shall be used.

(a) The distance to the 115 dBu contour is determined using the following equation:

$$D \text{ (in kilometers)} = 0.394\ (P)^{1/2}$$
$$D \text{ (in miles)} = 0.245\ (P)^{1/2}$$

Where P is the maximum effective radiated power (ERP), measured in kilowatts, of the maximum radiated lobe.

(b) After January 1, 1985, permittees or licensees who either (1) commence program tests, or (2) replace their antennas, or (3) request facilities modifications and are issued a new construction permit must satisfy all complaints of blanketing interference which are received by the station during a one year period. The period begins with the commencement of program tests, or commencement of programming utilizing the new antenna. Resolution of complaints shall be at no cost to the complainant. These requirements specifically do not include interference complaints resulting from malfunctioning or mistuned receivers, improperly installed antenna systems, or the use of high gain antennas or antenna booster amplifiers. Mobile receivers and non-RF devices such as tape recorders or hi-fi amplifiers (phonographs) are also excluded.

(c) A permittee collocating with one or more existing stations and beginning program tests on or after January 1, 1985, must assume full financial responsibility for remedying new complaints of blanketing interference for a period of one year. Two or more permittees that concurrently collocate on or after January 1, 1985, shall assume shared responsibility for remedying blanketing interference complaints within the blanketing area unless an offending station can be readily determined and then that station shall assume full financial responsibility.

(d) Following the one year period of full financial obligation to satisfy blanketing complaints, licensees shall provide technical information or assistance to complainants on remedies for blanketing interference.

A DISTRIBUTED ARCHITECTURE FOR A RELIABLE SOLID-STATE VHF TELEVISION TRANSMITTER SERIES

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Abstract

To design an all solid state, high powered VHF television transmitter series requires fresh insights. A totally new transmitter has been designed around some basic tenets of broadcasters: on-air reliability, user friendly, and performance consistent with the need for a broad array of power levels. The foundation for this series is the concept of a Distributed Architecture.

The power amplifier, cooling, control, monitoring, mains distribution, RF combining, and power supplies are distributed across the entire transmitter in a hierarchically fashion. To achieve reliable on-air operation, parallelism of critical circuits were used.

VHF power transistors have become reliable and cost effective with the rising usage of the Field Effect Transistor (FET). Power FETs are inherently more rugged than their bipolar counterparts. Additionally, gain is high and growing usage of FET technology has driven costs down. Class B operated, wide band FET amplifiers were used to generate just over a kilowatt of linear power to form the building block module for the entire series of transmitters.

Reliable operation of these RF amplifier modules were enhanced by using conservative rated FETs and a unique cooling system. The FETs are mounted on novel heat sink with air directed squarely at the transistors. The RF modules are combined in a ring type isolating combiner which is immune to hot module changeout or module failure.

A Distributed Architecture was employed to control, monitor, and protect the transmitter. Each subassembly has its own monitor and protection system, each cabinet has a higher level of monitor/control/ reporting, and this is all communicated to and from the central system for operation of the transmitter.

Transistor Transmitters

All transistor transmitters have been in use for many years[1]. High power transmitters for television usage and simultaneously cost effective, have not been possible. Over ten years ago the bipolar transistor was the device of choice for VHF amplifier designs that required hundreds of watts of RF power. The bipolar transistor was limited in its capabilities and designers sought to work around these limitations. VHF bipolar power transistors are minority carrier semiconductors with charge storage problems, low gain (6-8 dB), thermal run away potential, and high saturation resistance.

Meanwhile, Field Effect Transistors (FETs) were gaining in their wide spread usage. FETs had been produced by only a few manufacturers for some unique applications. Several transistor manufacturers began turning resources toward the use of FETs for more and more applications. FETs are majority carrier devices and thus have no charge storage problems, exhibit higher gain, have low saturation resistance, and don't thermally run away. Many manufacturers turned their attention toward FETs and have come forth with a dazzling variety of new high performance devices suited for broadcast applications.

The popularity of the power RF FET has made high power VHF television transmitters practical. A review of the component literature will show that several VHF transistors can now be found to deliver over a hundred watts and are available from several manufacturers. This broad base of available FET transistors has set the stage for the all solid state VHF television transmitter. The task was to put these devices together with the other parts of the transmitter in the best manner possible[2]. This, we discovered, led to an Distributed Architecture based design.

Distributed Architecture

Application specific design is the method by which a single product is designed to meet a given set of needs. Historically, broadcast transmitter OEMs, including Harris, have approached designs in this manner. The difficulty with this approach is that only one product results from the entire design effort. Architecture is a method by which a family of products can be designed. By approaching the opportunity of designing a family of VHF products instead of a single product, led to an Architecture approach.

As the design of the family Architecture began to form, a second requirement of the design approach became evident. The Architecture needed to be an "Distributed and Open System". That means that the design should use common standards and common interfaces to permit future growth. By combining the family Architecture with an

Distributed/Open System approach, the basis for the design of the series of solid state VHF transmitters began.

Power Partitions

The transmitter power output (TPO) needs for VHF vary considerably. The FCC regulates the limitation of effective radiated power (ERP) of 100 kW for low band and 316 kW for high band. How each broadcaster meets the ERP requirements is dictated by the individual station circumstance. Antenna gain, transmission line efficiency, tower/building height, and derating factors. Harris undertook a comprehensive study of tower heights, antenna gain, and transmission line efficiencies. This study indicated that by incrementing the transmitter in about 15 kW cabinet blocks, the wide range of VHF broadcast needs could be met without excessive costs. This 15 kW cabinet building block was checked against practical design limits of combiners and power amplifiers.

To achieve 15 kW of RF power, more than one transistor is necessary. Thus the need for a second partition of power arose. Power VHF FETs can deliver up to several hundred watts of RF power. However, device lifetime and thermal management dictates that the FETs be used appropriately. It is one thing to design a power amplifier for say military use such as a bomb fuse which requires only seconds of lifetime versus the 15 to 25 years of life for a commercial VHF television transmitter! Transistor design theory says that we double the lifetime of a transistor for every ten degrees Celsius cooler we keep the junction of a transistor[3].

Our design tradeoffs were further complicated by the needs for aural power. Our data shows that a VHF transmitter uses a wide range of aural power. The FCC limits the aural power to a maximum of 20% of visual. Broadcasters use power primarily at either 10% or 20% with no clear trend for either. The growing popularity of BTSC stereo has encouraged higher aural powers while costs have prompted thoughts of lowered aural power. The design Architecture had to accommodate either 10% or 20% aural.

Against the backdrop of transistors and aural/visual power, a nominal 1 kW (peak of sync or CW) design limit for a module was selected. At the 1 kW module power, several transistors could be easily combined and thermally managed for long life.

The net result of the Architectural approach to the VHF transmitter series dictated that the transmitter be built from 1 kW modules (aural and visual), combined in cabinets to the 15 kW visual level, and capable of further combining of 15 kW cabinets to 60 kW.

RF Chain

The design Architecture resulted in a design of the RF chain as shown in Figure 1. As shown, a system of small and large building blocks are required to give the desired power outputs to match broadcast needs. But the design needed to be matched to the other needs of the broadcasters. We began a process of reviews of the design with people within the industry.

What we learned was that VHF television broadcasters are interested in:

* on-air reliability
* user friendly
* redundancy

Of course, many other important things were mentioned. But consistently broadcaster after broadcaster repeated these requirements. How to bring together the needs with the Architectural approach remained a question in need of a solution.

PA Power Supplies

The B+ PA supply for a VHF tube transmitter is a single unit usually at several thousand volts and a few amperes. A 30 kW solid state transmitter, however, using a single supply would require nearly 1000 amperes peak from a 50 VDC supply! Viewed differently, it is also possible to power each 1 kW RF module with its own supply. Although possible, neither the concept of a single power supply or individual RF module supplies didn't meet the requirements for a Distributed Architecture and the broadcaster.

Power supply costs per watt decrease with increasing power. Thus the fewer supplies, the lower the cost with one being the lowest cost. But with a single transmitter PA supply redundancy is lost. To strike a balance, two 50 VDC, 300 A, PA supplies per cabinet were used. This preserved the desired redundancy without excessive costs. By placing the supplies inside each cabinet, the original concept of an easily expanded family was maintained.

Cooling

Keeping the transmitter cool is vital to long life. Transistor temperature is an exponential function of junction temperature. All chemical activity proceeds more rapidly at elevated temperatures hence other components are likewise longer lived by keeping them cool. To achieve the required low operating temperatures, some obvious but not routinely employed techniques were used:

PA Cabinets

1. Parallel air flow across RF modules. This means

that air is only used once by a module.

2. Metered resistance control of air flow to each module. This mean that the same air flow is present to each module location whether or not the module is in its location.
3. Use of heat pipes to cool dummy loads in combiners.
4. Directing the air flow directly at the transistor thus increasing the effectiveness of the cooling air.
5. Use of a very low thermal resistance heat sink for the FETs to rapidly remove heat.

Special attention was also paid to the selection of fans for their audible noise properties.

Control and Monitor

Fundamentally, as good engineers, we would like to measure every thing inside every transmitter. But an all solid state 30 kW transmitter typically will have four cabinets, over 40 RF power modules, a few hundred power transistors, and associated combiner/splitters. The distributed concept of a transmitter control and monitor was born of a need for simplification.

Normally, an engineering designer begins by wishing to monitor and measure everything. Imagine, for a moment, that it is desired to measure key parameters of each pair of push-pull transistors used in the RF amplifiers. Measurement could include current, voltage, input RF power, reflected input power, out power, reflected output power, and so on. It would be easy to want to measure each and every parameter but this would lead to several thousand pieces of data necessary to characterize the performance of a transmitter. Obviously, something new had to be done.

Likewise, control could be extended to the ability to turn each module on/off, turn each cabinet on/off, and so forth. This approach is also unwieldy and in need of simplification.

The control and monitor approach adopted was one of each module, next higher assembly, and cabinet must "take care of itself". Each subassembly level would report only major summary forms of information. This summary data had to then be adequate to maintain and service the transmitter. All this philosophy fit the architectural hierachy of the design rather nicely. This is shown in Figure 2. The control of the entire transmitter is only by hardware while display of data was with software.

The type of information exchanged between hierarchy assemblies is best described by an example:

1. A single RF module overheats due to a local air passage blockage.

2. A sensor on the RF module detects the excessive module temperature and signals the module to shut down before any permanent damage can be done. This over temperature is one of several parameters monitored that will cause this action (over voltage, overdrive, high output VSWR, etc.).

3. The RF module begins flashing a light on the front of the module in a coded fashion identifying the specific nature of the problem for local usage.

4. A single signal is sent to the local cabinet control monitor module. This condition is not threatening to damage any other assemblies, thus no further action is necessary but to report the problem.

5. The cabinet that has the RF module problem signals the main transmitter monitor that alarm condition exists.

6. The main transmitter monitor system then alerts the local control with a flashing alarm light and a remote control contact is closed for remote sensing.

7. At the remote control point, the engineer is signaled by the remote control equipment of an alarm condition. At this time, he would not know what specifically the problem was. The transmitter parameters, however, can be monitored to see if other operational parameters are normal. If all is OK, the engineer schedules a trip to the transmitter site at a convenient time.

8. At the transmitter plant, the engineer will see the visual main control alarm signal and immediately see a flashing red light on the RF module. The coded nature of the flashing indicates an over temperature condition. The engineer then determines the cause of the overheating.

This sort of hierarchial logic prevailed in all the monitor functions of the transmitter.

The control function (all by hardware) was accomplished with a similar methodology. The main control function (On/Off, etc.) was directly tied to each cabinet which then commences the appropriate actions desired by the main transmitter control function.

RF Combining

The RF PA needs several levels of combining to each the 30 to 60 kW output power need by full power broadcasters. To reach this power, it is necessary to combine hundreds of transistors. To put the total problem into perspective, the combining was segmented into modules, cabinets of modules, and cabinets.

Transistor Combining

Class AB, push pull, transistor pairs is the lowest level of combining. Pairs are then combined within phase combining circuits (total of four transistors or what we call a quarter module). It is significant to note that a single transistor failure in a push pull pair is not a total failure. Power does drop and performance is reduced but operation continues.

These quarter modules are then summed in a four-way combiner to give over one kilowatt.

Module Combining

The 1 kW visual modules are combined in 16-way ring type combiners designed to permit operation with modules failed or removed. With this arrangement, servicing can be done on the air without interrupting air time.

Cabinet Combining

The output power of a aural PA is just above 15 kW. Two of these cabinets are readily summed with 3 dB hybrid type combiners to get 30 kW. This technique gives parallel redundancy similar to the familiar combined transmitters of today. If power out of one cabinet drops to zero, then the output power is 25% of the original power.

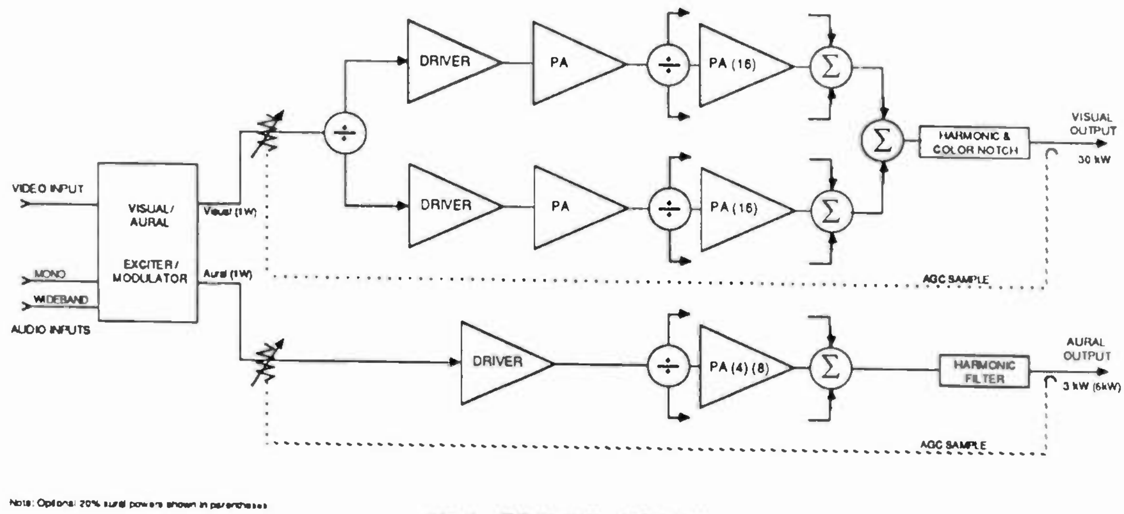
Significantly, the parallel redundancy is carried out from the lowest level to the highest level of RF combining.

Conclusion and Summary

The efforts that lead to this design have significant ramifications for transmitter designers. We have demonstrated that it is possible to form a complete transmitter series from one design effort. The approach is that of a Distributed Architecture. The benefit is a design that is highly repetitive and thus made from parts used again and again. This permits design attention on fewer parts and improved reliability. Broadcasters benefit from this approach because of the user friendliness, on air reliability, and ease of maintenance.

References

1. Cox, B. & Swanson, H., "World's First FCC Type Accepted 1 kW AM Transmitter; 1974.
2. Svet, F.A., "Factors Affecting On-Air Reliability Of Solid-State Transmitters," 1988 NAB Proc.
3. Black, Proc. IEEE, 57, 1587, 1969



Note: Optional 20% aural powers shown in parentheses

Fig 1. RF FLOW - HT-30LS

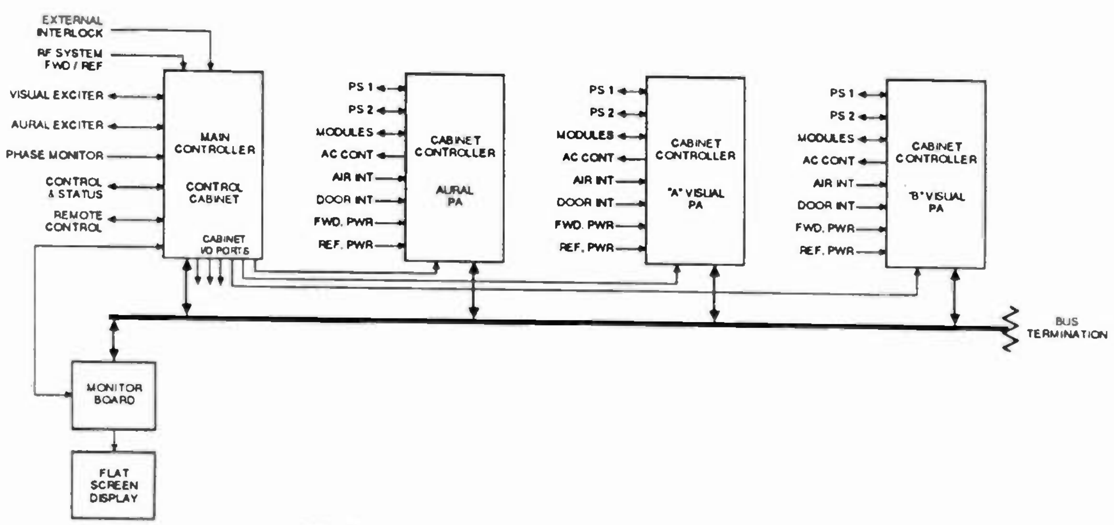


Fig 2. CONTROL SYSTEM - BLOCK DIAGRAM

ADVANCED RF MEASUREMENT TECHNIQUES

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ABSTRACT

Modern vector network analyzers with time domain capability provide the user with error-corrected, computer controlled RF system measurements and hard copy results. The use of error-correction yields accuracy never before obtainable. Microprocessor control makes measurement of complex impedance, insertion loss, and group delay, quick and simple.

This paper discusses these advanced measurement techniques and demonstrates tests on dual channel combiners, long transmission runs, and ACTV diplexers. It will familiarize broadcasters with how these tests can improve the efficiency and on air quality of their TV or FM plant, as well as evaluate RF system performance for ACTV/HDTV signals.

INTRODUCTION

Broadcast engineers have long known that low loss, low reflection RF systems maximize plant efficiency and quality. The reasons are obvious; lower loss means lower electricity bills for the same ERP. Low reflections (VSWR) means clearer video as well as increased efficiency. Low VSWR at the tube output assures optimum power transfer to the system. Removing antenna and transmission line reflections wards off the threat of "ghosts". Unfortunately, old loss and VSWR measurement equipment was often bulky, unstable, and produced inaccurate results. Higher electricity costs and increased transmitter efficiencies have pushed RF system performance requirements beyond the limits of this old equipment. New measurement techniques and equipment provide a quantum leap in the resolution and accuracy obtainable. A recently introduced network analyzer, the Hewlett Packard 8753A, utilizes these techniques. Both the RF system manufacturer and the engineer in the field benefit from it's use.

NETWORK ANALYZER BACKGROUND

To begin our discussion of these techniques we should start with a short explanation of the theory behind the network analyzer. A network analyzer determines the transmission and reflection characteristics (impedance and insertion loss) of any device connected between it's two ports. It does this by relating the devices input and output signals to a known reference signal. Once the device's magnitude and phase responses (vectors) have been obtained its complex impedance, insertion loss, and group delay can be determined. By finding the complex s-parameter vectors of any two-port network, it's performance can be completely defined.

The difference between vector and older scalar analyzers is that a vector analyzer measures the systems complex response rather than its magnitude only response. It is this vector capabilty which permits mathematical manipulation of the results and removal of measurement uncertainty. Error vectors that can be mathematically subtracted from the obtained response are found during a calibration routine. Error correction has made it possible to remove the impedance and loss characteristics of the test adapters from the device measurement. Now, by placing known values of impedance at the ends of the test cables the analyzer can reposition its reference ports to their ends. Figure 1 shows the effects of correction on the measurement of a band pass filter's VSWR using high grade lab cables. The figure shows that the uncorrected values are quite different and higher than those with the cable errors removed.

Modern vector network analyzers consist of four basic units (figure 2) combined into one frame. This produces a self-contained measurement system that is easily transported. The most important of the units is a synthesized sweeper source. A synthesizer is used rather than an old fashioned sweeper due to its inherent stability and frequency resolution. Its discrete measurement point capability (selectable from 201 to 1601) makes mathematical manipulation simple. Stopping at each frequency point, obtaining a response vector, storing, and displaying the data is all done almost instantaneously. The second of the analyzer units is the signal separator. A signal from the source is applied to the device under test and the separator extracts a sample of the forward signal as a reference. The reflected and transmitted signals of the device are also sampled by the separator.

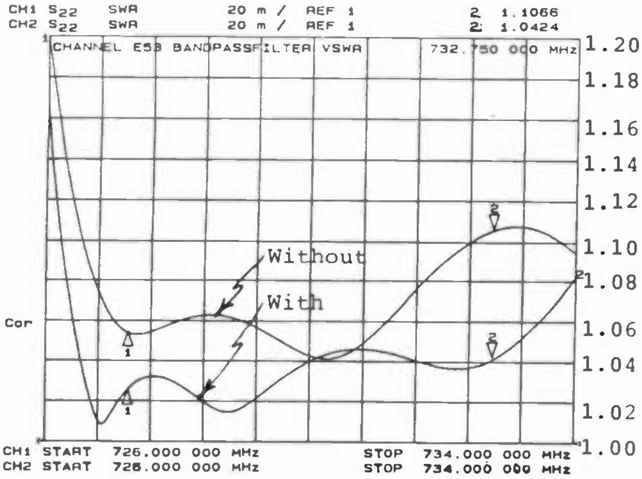

FIGURE 1
FILTER VSWR WITH AND
WITHOUT ERROR CORRECTION

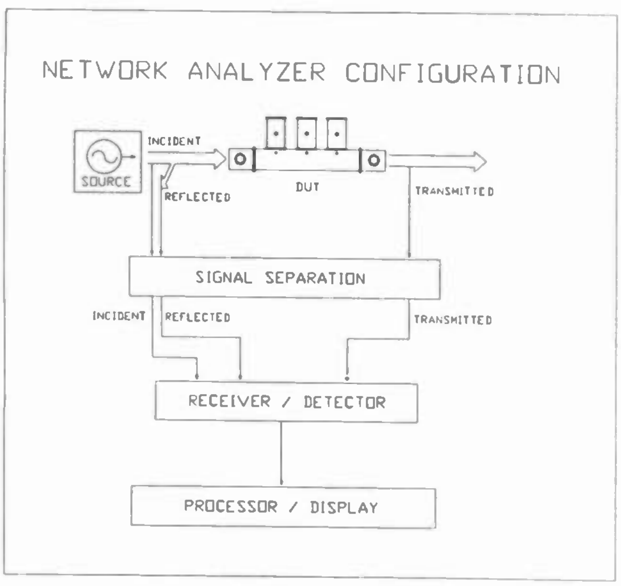

FIGURE 2 NETWORK ANALYZER BLOCK DIAGRAM

These signals; incident, reflected, and transmitted are then applied to the inputs of three phase-locked receivers. These convert the signals to an IF frequency and digitize them. The signals now in digital form go into the processor/display unit. Here they are modified for presentation on the CRT. The processor contains storage space for all of the data from each sweep of the device, any necessary math logic, as well as five instrument set-ups. A main micro-processor drives and coordinates all functions of the analyzer. The data can be stored in memory and displayed in various formats; linear, logarithmic, phase, group delay, smith chart, and polar. Changes in format are made without retaking the measurement. All measurement and display parameters are easily adjusted from the front panel.

The analyzer contains an IEEE-488 computer bus which allows remote operation of the analyzer. Completely automated testing can be done by joining a computer to the bus. The bus also drives a multi-pen plotter for hard-copy output. A built in graphics program directs the plotter to produce exact multi-color duplicates of the screen. This method differs from old x-y plotters by producing a fully labeled and graphed plot from a blank page. Special graph paper is not necessary.

PRECISION MANUFACTURING TOOL

The analyzer is used extensively for the tuning of RF systems and components in the factory. Large systems are assembled from a multitude of components, each separately checked with the analyzer. Some components require tuning to insure that the entire system VSWR is low. The accuracy of the analyzer is extremely important at this point since errors accumulate when combined in a system. To demonstrate, take four components and match them for unity VSWR with both old and new measurement systems. Then assemble the components as a system and compare the results. Both measurement systems have values of VSWR uncertainty which affect the displayed results. Uncertainty is a measure of the overall analyzer directivity. Poor directivity means more uncertainty. The Contributors to bad directivity consist of; limited coupler directivity, poor source match, varying test set frequency response, and high VSWR test cables. The new analyzer, however, has a calibration routine to remove these constant uncertainty terms. The old one can not remove their effects. Typical total directivity for the old system assuming type "N" couplers, adapters, and cables would be -34dB or a VSWR uncertainty of 1.041. For the new analyzer after correction, -44dB and 1.013 are typical values. If each component is perfectly matched on the respective systems we find that the old system yields actual VSWR values between 1.00 and 1.04. Those tuned with the new system range between 1.00 and 1.01. When the four components are assembled and measured as a system the accumulation of these uncertainties can be dramatic. An input VSWR of 1.174 may actually be present with the old system and 1.053 with the new analyzer. These numbers assume that all the mismatches are at the correct phase to sum together. Actual system values may or may not be lower. The more components assembled into a system the worse the problem becomes.

The first step in any component or system tuning is to obtain matched test transitions. For coax systems these transitions adapt the line size down to type "N". Coax transitions are matched using two identical adapters on the ends of a 20 foot length of line. Each adapter is adjusted for minimum VSWR looking from the other end. The minimization is repeated until unity VSWR is obtained at both ends. The analyzers data memory is used to store the results of one ends measurement. While adjusting the other end the memory data obtained is displayed and a direct comparison between the ends is made. This technique results in two transitions, tuned simultaneously with very little effort. Waveguide system testing utilizes waveguide to type "N" adapters. The adapter is tuned by bolting it to the end of a straight waveguide section containing a low VSWR waveguide sliding load. Match is obtained by making a tight impedance grouping which moves around the smith chart center as the load is moved.

VSWR optimization of most coaxial system components is unnecessary. If tuning is needed, it is done by placing correctly sized and located capacitive rings on the inner conductor. The rings are placed as closely to the mismatch as possible. They provide an impedance which is of the correct magnitude and opposite reactance phase to cancel the mismatch. The rings, known as slugs, are silver brazed in place on the inner conductor after positioning. Waveguide components are tuned, when necessary, using a similar technique. Small capacitive domes, known as buttons, are placed on the broad wall of the guide. Residual mismatches are cancelled by proper sizing and location of the buttons. Care must be used when tuning components, both coax and waveguide, to insure that the bandwidth of the device remains wide. Tuning methods can never replace the benefits of proper low VSWR component design.

System sub-assemblies are built and tested following component checkout and tuning. Diplexers, filters, switchless combiners, and hybrid combiners are some considered sub-assemblies. They are parts of an overall RF system. Each is tested and adjusted for the desired performance. Measurement of insertion loss, isolation, and group delay must be made in addition to impedance. Diplexer cavity tuning, a complicated procedure, is aided by the analyzers dual measurement capability. The simultaneous monitoring of two of the diplexer outputs with a single input shows the load port and visual port isolation together. Figure 3 shows the visual input insertion loss, impedance and aural input impedance for a typical diplexer.

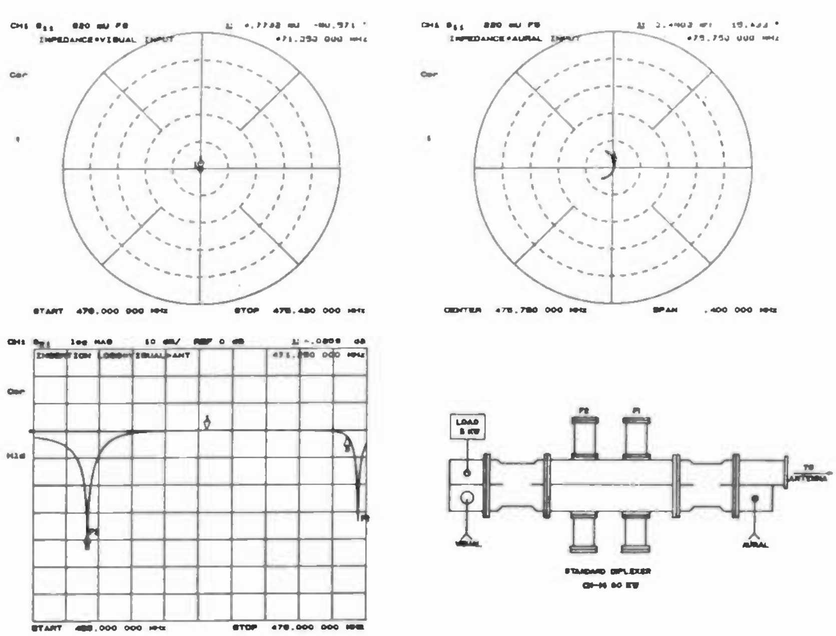

FIGURE 3
TYPICAL DIPLEXER IMPEDANCE AND VISUAL INSERTION LOSS

The final step in the production measurement cycle is to assemble and test the entire system as it will be installed at the station. Systems are shipped assembled as unitized systems or match marked and disassembled for shipping. The system check insures correct operation in all possible modes. A four plot per page output format reduces the number of pages in the final data report supplied with every system. The system performance and signal routing in all operating modes is included in the report.

Recent proposals for advanced concept television (ACTV) have indicated that stringent diplexer and RF system specifications are necessary. Group delay and VSWR variations may be extremely detrimental to video quality, defeating the purpose of enhanced definition. Study of RF system effects has already begun on several proposed methods. New schemes may be required to carry the signal from the tube and radiate it properly without distortion.

FIELD MEASUREMENTS

Recent increases in broadcast plant output power have increased the necessary number of output tubes. It is imperative that maximum power output be maintained under all fault conditions. Concurrently the RF system switching, combining, and routing requirements have become very complex. This increased complexity makes it difficult for the average station engineer to install, adjust and repair the RF system. It is often advantageous, both economically and technically, to leave RF system tasks to the manufacturer's field engineers. The economic advantages result from the field engineers dedication to the RF system only. Trained and experienced in the proper operation of the equipment, he can quickly and correctly locate problems and supervise installation. It is unnecessary for station personel to become RF system experts, a costly undertaking. Since the field engineer brings along his own test equipment and transistions no station equipment expenditures are needed. The technical advantages result from essentially the same things. The experience gained from the completion of many successful installations leads to a system which operates efficiently and requires little maintenance. A poorly installed and adjusted system costs money for maintenance. It may also continuously cost money by being mistuned and inefficient. (Table I)

Field measurements and services available to the broadcaster cover both inside and outside transmission systems. All measurements are made using the network analyzer and associated test transitions.

They consist of:

A) Check and tuning of inside system modes, VSWR, isolation, insertion loss, and group delay.

B) Check and tuning of single or multi-channel coax and waveguide transmission lines with or without the antenna using time domain analysis.

C) Tuning of antennas either at the input or through a transmission line using TDR.

D) Complete system evaluation from the tube output to and including the antenna.

E) Clean, repair, and refurbish any faulty components found in the system and check.

F) All types of field strength measurements.

	VSWR	LOST POWER DUE TO VSWR MISMATCH (kW)	INSERTION LOSS (dB)	LOST POWER DUE TO INSERTION LOSS (kW)	LOST POWER IN REJECT LOAD (kW)	TOTAL LOST POWER (kW)	COST PER YEAR @ 10¢/KWH
WELL TUNED SYSTEM	1.03:1	.056	.09	5.026	.076	5.158	$ 4,518
POORLY TUNED SYSTEM	1.13:1	.882	.19	10.732	2.400	14.014	$12,276

COST PER YEAR
OF TUNED VERSUS MISTUNED
240 kW RF SYSTEM

TABLE I

Transmission line testing and tuning benefit the most from analyzer technology advances in comparison to other system measurements. Two major problems have always made accurate tuning of long lengths of coax or waveguide difficult. First, the poor performance of the input transitions or test equipment often shadowed the performance of the line beyond. Second, the location of mismatches along the line could not be determined close enough to allow their removal. The introduction of error correction and a new technique of line pulsing, known as TDR, have cured these problems.

Error correction helps solve the first of the problems. TDR helps solve both of these problems. Using time domain reflectometry (TDR) a graph of the mismatches on the line versus distance is displayed. Line pulsing is not new for testing broadcast antennas[1] however, old techniques required a large amount of equipment and interpretation. Modern TDR uses the analyzers digital data format and built in computer to generate the response. An inverse fourier transform (FFT) is used to calculate the time domain response from standard frequency response information (VSWR, insertion loss). For narrowband devices the bandpass time domain response must be obtained. This is used when the devices response is correct only over a small frequency range. The mismatch values averaged over the frequency range selected are displayed versus time. The location of the mismatch is then easily determined. Distance resolution is determined by the frequency span of the measurement. Wider bandwidth yields higher distance resolution (Table II).

MEASUREMENT FREQ. SPAN (MHz)	DISTANCE RESOLUTION (ft in air)
6	41.0
20	12.3
50	4.9
100	2.46

TDR
SPAN VS RESOLUTION
TABLE II

Because the time scale is mathematically derived, any desired display increment can be selected. The line can be examined in detail or a top to bottom view can be displayed. By mathematically removing or gating out a portion of the line and converting back to the frequency domain, the VSWR or insertion loss without that portion is obtained. Gating is used to remove input and output transitions from the measurement. Solving the first problem mentioned above. By removing everything <u>but one</u> response we can measure a single device. An antenna can be measured from the base of the tower without disconnecting it from the transmission line.

The TDR is now used for all transmission line tuning. It has become an irreplaceable tool for locating line difficulties. Loose flanges, dents, and other assembly problems can be easily found. On a recent job it was even used to find a connection where, instead of one, two gaskets were installed. Single channel operation dictates that the line and antenna have very low VSWRs over a six MHz band. This results in a resolution of 41 feet, a distance too great to locate problems. Bandwidths of 20 or 50 MHz yield resolutions good enough to pinpoint problems on long runs. These bandwidths maintain the necessary narrowband restriction. Two TDR measurements are therefore made. One over six MHz to determine the <u>magnitudes</u> of the reflections in the desired channel. A second measurement over a 20 or 50 MHz band is used to locate the <u>positions</u> of the mismatches. The magnitudes of the wideband measurement are used for reference only. Figures 4 and 5 show six MHz and 50 MHz TDR data obtained on a recently installed 1000 foot waveguide run and side mounted antenna on Mt. Sutro in San Francisco, CA. This line included several sets of odd angle elbows, offsets and twists. From the 50 MHz data we can locate mismatch areas. Comparing the antenna response in figure 4 to that in figure 5 shows the effect of widening the bandwidth on the measured mismatch magnitude. The average antenna VSWR which appears at approximately 2.6 s, over six MHz is very low (<1.03). The antenna average over 50 MHz is much higher (1.40),

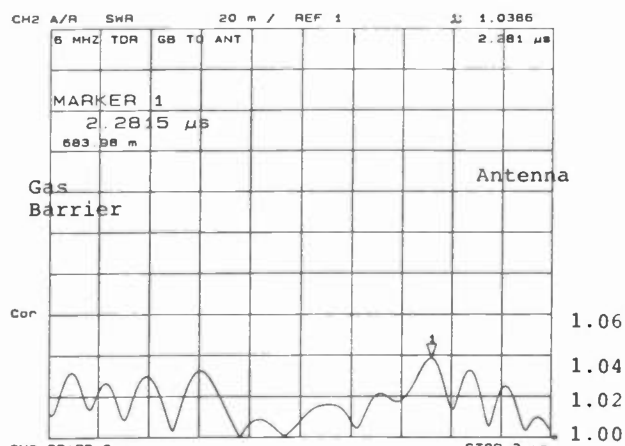

FIGURE 4
WAVEGUIDE AND ANTENNA VSWR
6 MHz TIME DOMAIN

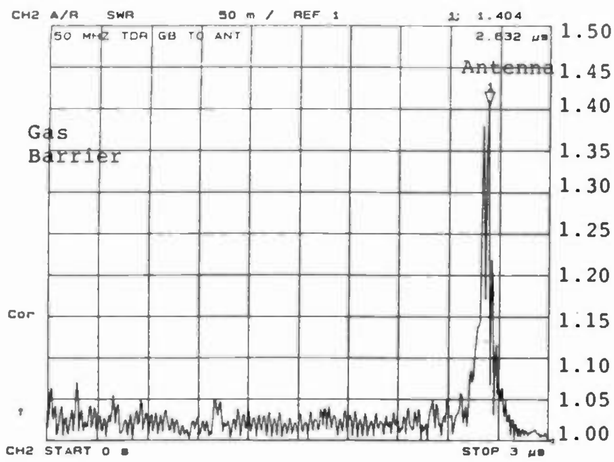

FIGURE 5
WAVEGUIDE AND ANTENNA VSWR
50 MHz TIME DOMAIN

showing how narrow it's tuning is. Figure 6 shows the six MHz data in the frequency domain format. The line and antenna have a VSWR less than 1.13:1 over the band. The 50 MHz frequency domain data in figure 7 shows the relatively narrow system VSWR performance which results.

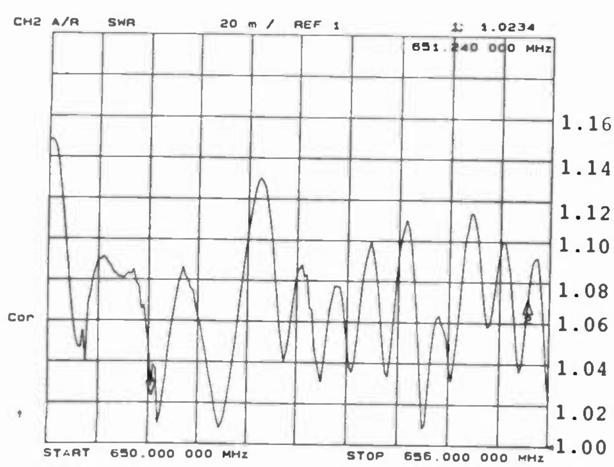

FIGURE 6
WAVEGUIDE AND ANTENNA VSWR
6 MHz FREQUENCY DOMAIN

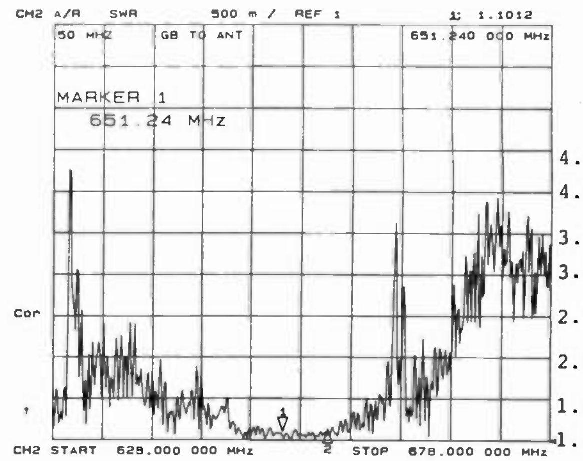

FIGURE 7
WAVEGUIDE AND ANTENNA VSWR
50 MHz FREQUENCY DOMAIN

The recent introduction of high power UHF dual channel installations[2] in the US and abroad has led to another new measurement technique. In the past the tuning of dual channel systems was done by sweeping the entire bandwidth necessary to cover both bands. Markers were placed on the carriers and these points were optimized. Since it is difficult to tune one channel without effecting the other, broadband sweeping was the only answer. The analyzer has made dual channel tuning simple. Two separate narrow measurement channels can be set up and displayed simultaneously. The effects of line tuning are shown on both channels together. Two TDR plots can also be displayed. This technique was used on a 240kW dual channel 1200 ft. dual waveguide system in Kuwait. The broadband return loss data obtained after tuning one of these lines is shown in figure 8. The line VSWR is less than 1.10 over approximately 30 MHz. Dual channel combining system alignment is also simplified with this technique. The response of the two input channels can be obtained by feeding the antenna port with a single cable.

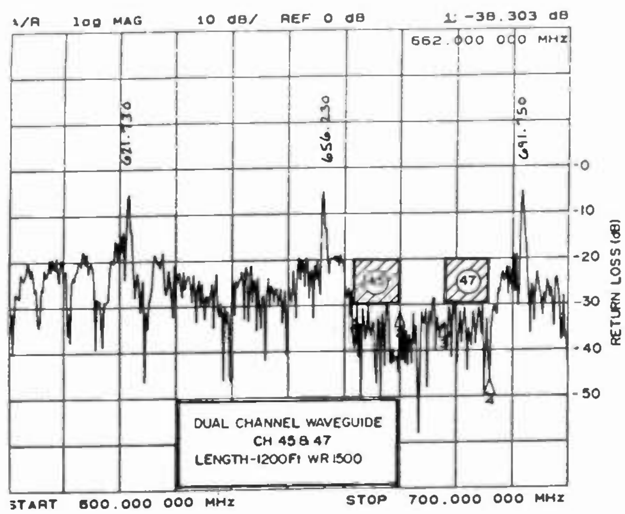

FIGURE 8

CONCLUSION

RF system measurements during manufacture and in the field have benefited from the introduction of vector network analyzers. Several techniques used for these measurements have been presented. The advantages to the station engineer of good RF system performance have also been demonstrated.

The analyzer accuracy discussed will be necessary for the next generation of ACTV/HDTV RF equipment.

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USING AMATEUR PACKET TECHNOLOGY FOR LOCAL EBS

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Abstract

The American Radio Relay League (ARRL) published Version 2.0 of the AX.25 Amateur Packet Protocol in October, 1984. Designed to work in the unpredictable and noisy Amateur radio RF environment, this protocol has appeal for non-amateur services where time sharing of available spectrum for data exchange is the goal. This paper covers an application, still under development, where Amateur packet technology is used to distribute data to participating local EBS stations in Los Angeles County from the Los Angeles County Sheriff's Radio Center. Readily available and relatively inexpensive amateur radio equipment is employed. Also covered is a plan being developed to use this technology to make local EBS information available to hearing impaired citizens of the County.

Basic Concept and History of LA County EBS

The Los Angeles County Operational Area's local EBS system depends on an existing radio system used daily by the County's Air Quality Management District to inform businesses and schools of the steps they must take when smog alerts are declared. This radio system consists of a number of bases and FM transmitters on 39.98 mHz.

Use of this channel for local EBS voice messaging began in 1982. A base station at the Los Angeles County Sheriff's Radio Center was equipped with a NIAC two-tone generator, and Sheriff's personnel were trained on system testing and activation. Many Los Angeles County broadcast stations who participate in the Plan have purchased special receiver/cassette systems with NIAC decoders for reception of tests and alerts. A package of scripts was prepared. One station even printed up Orange Local EBS Procedure cards for posting in participating stations.

Telephone response to the Los Angeles Sheriff's Department from participating stations after weekly tests shows a high level of participation and support. This support underscores the premise that, for Southern California, it is not a question of IF a major earthquake will happen, but WHEN.

The LA County Operational Area Plan

Originally drafted in 1982, the Los Angeles County EBS Operational Area Plan was designed only for voice messaging. To our knowledge, no other EBS Plan has begun to incorporate data transmission from the political entity in charge of the Operational Area in times of major disasters. The Los Angles County Sheriff is the designated person in charge in LA County. Over twenty cities are part of LA County, covering an area of thousands of square miles. The Plan is designed so all cities can alert the Sheriff if a condition exists that warrants local EBS activation.

LA County Operational Area Plan Objectives

The Los Angeles County Operational Area Plan has been designed to meet four key objectives to meet the unique needs of our region:

- Provide a direct, rapid flow of information to the public through broadcast media.

- Avoid relay of information from broadcast station to broadcast station.

- Not depend on certain stations remaining on the air.

- Not rely on the public switched network.

Stated Purpose of the Los Angeles EBS Plan

The Los Angeles EBS Operational Plan rewrite has not been reapproved by the County at this time. However, all advisory parties have "signed off" on a revised statement of purpose that outlines the EBS mission for our region:

The System will relay public emergency information and instructions to the Media concerning protection of life, limb, and property within the Los Angeles Operational Area, and adjacent Operational Areas.

The Need to Improve The System

EBS plans outline the flow of information from government to the media; participating radio and TV stations relay that information to the public. Since the original Los Angeles

Plan was drafted in 1982, there was a deepening understanding that the volume of information that must get to the public for a major earthquake is staggering, and new methods might be needed to deliver it to the Media. Given the volume of information, and the lack of certainty about which broadcasters will be on the air after a major earthquake, any method must be compatible with existing EBS.

Changes leading to the data transmission system portion of the Plan are a product of many Advisory Group meetings over a period of years. Work on this project began in 1986 when the Emergency Public Information Advisory Group (EPIAG) to the Plans and Programs Committee of the Los Angeles County Emergency Preparedness Commission looked at enhancing the local EBS voice message system to solve several problems that had been identified:

- Voice messages currently need to be transcribed by TV stations for the hearing impaired community.

- Staffing is not adequate 24 hours a day in stations to make sure this responsibility is covered. After the October 1, 1987 Whittier earthquake, research showed there was not adequate means to keep hearing impaired citizens in the County informed. Many hearing impaired citizens with teletypewriter capability were cut off when phone service failed.

- Many radio and TV newsrooms will not have enough staff to assign one person to listen to all EBS voice messages, or to review cassette tapes of messages previously recorded. If an EBS condition arises during an evening, weekend, or holiday, the problem will compound.

EPIAG members familiar with Amateur Packet technology suggested this might be an easy and cost-effective way to add data messaging to the existing voice system. Working closely with the Los Angeles County Facilities Management Department, who is in charge of County radio equipment, preliminary tests showed this idea has merit. Tests and adjust-adjustments now underway have the following goals:

- Improve data transmission reliability

- Test various types of Amateur packet equipment

- Test receivers that can be used with the system

- Demonstrate the system to stations considering implementing data reception

- Perfect the system to get EBS information to the hearing impaired community.

Packet Basics

The AX.25 Link Layer, unlike most link-layer protocols, does not assume that a primary device (usually called a DCE, or data circuit terminating equipment) is connected to a slave device a DTE (or data terminating equipment). Instead, AX.25 assumes both ends of the circuit are equal. Such terminals are called DXE's.

Transmissions using this protocol are sent in blocks of data called Frames. Fields within the frames take care of housekeeping information and actual data transmission.

Flag	Address	Control	FCS	Flag
01111110	112/560 bits	8 bits	16 bits	01111110

Fig. 1 - U and S Frame Structure

Flag	Address	Control	PID	Information	FCS	Flag
01111110	112/560 bits	8 bits	8 bits	n*8 bits	16 bits	0111111

Fig. 2 - Information Frame Structure

The Flag field is used as a marker between frames. Two adjacent fields may share the same Flag frame. The Address field identifies the source of the frame and its destination. The Control field identifies the type of frame being passed. The PID field (Protocol Identifier) appears only in information frames. For reasons beyond the scope of this paper, the PID field is not included as part of the count of the information field. The Information field can contain up to 256 8 bit "octets". A "zero bit" may be inserted in this field to prevent adjacent flags from appearing accidentally in this field.

The FCS field (Frame Checking Sequence) is a 16 bit number calculated by both ends of the transmission system. A Frame will be transmitted again if the receiving station's FCS calculation does not agree with the FCS calculation of the transmitting station. Since the Los Angeles EBS Packet system is "one way" only, this feature is not currently usable.

Invalid Frames

One of the mixed blessings of one way packet is what happens when the receiving station determines that an invalid frame has been received. The frame will be discarded. It is far more likely for this reason that whole phrases or sentences will be dropped, rather than single words within sentences. Since important messages will be repeated, and voice information will still be available on the channel, this risk is deemed to be acceptable at this time. This is far less risky than a data transmission mode where a single key modifier word like "NOT" can be dropped from the middle of a sentence. Sheriff's Department personnel are being instructed, nonetheless, to avoid negatives in messages to minimize possible confusion, and to repeat "life and death" phrases in messages.

The System

The heart of the packet radio system is a TNC (terminal node controller), sometimes called the "packet box". The TNC is a specialized intelligent modem adapted for two-way radio. It accepts serial data from a computer or terminal, adds the error detecting codes mentioned above, and converts it into audio that is fed into the microphone input of a two-way transmitter. At the receiver, an identical TNC extracts the data from the encoded audio, corrects some errors introduced by transmission, and provides serial ASCII data that can be routed to a computer, teletype machine, terminal, or directly to a serial interfaced printer.

The TNC at the receiving end can be set to respond to any packet transmission, or it may be set to decode only those transmissions designated for a unique address code. The system usually used for amateur VHF transmissions operates at 1200 baud, or roughly 100 characters per second. The input device to the TNC at the transmitting end, or the output device at the receiving end, can be set to other data speeds that may be faster or slower than 1200 baud. For instance, slower speed output will be necessary for real time video display character generators TV stations will use to display EBS information for the hearing impaired community.

Reliability

A few words about data transmission reliability are in order. While this transmission system utilizes packet concepts, it does not take advantage of error checking that is a key part of two-way packet links. Packet data in this system is sent out with no confirmation of reception from individual stations. This is a known risk in the current system.

Why Not Use Two Channels For True AX.25?

A second radio channel (or channels) for true, two-way packet is not available at this time. Public Safety and broadcast RPU channels are at a premium in the Los Angeles region. All parties felt that implementing one way transmission of information packets now is better than no data capability at all.

County communications and broadcast engineers are experimenting with methods to make the best of and improve the existing one-way system, since there are drawbacks to a true two-way packet system:

• There would be added cost to participating stations and to the County.

• Each participating station would need a unique call sign, not to mention a transmitter and the associated red tape of FCC licensing not needed with the current one-way system.

• More than one return channel might be necessary for such a system. A large number of packet stations each responding with a unique call sign on the same channel might slow the system down too much.

• Accidental or intentional jamming on the response channel could make the system unusable.

• Breaking the system into "nodes" using more than one response channel might be the answer, provided spectrum is available. However, this would complicate the system and add greatly to its cost.

Regions considering packet who are not experiencing spectrum shortages might consider full, two-way implementation of AX.25, but should keep in mind these drawbacks.

The PACLEN Command

AX.25 lets an information packet to contain up to 256 bytes. Most packet controllers default to a value of 128. We are now experimenting with shorter packet lengths to minimize data loss, although at some cost to the overall speed of data transmission. Many packet box manufacturers recommend lengths of 64 (or even 32 under difficult conditions). We are currently setting PACLEN to a value of 32 to minimize the amount of data that will be lost if a Frame is invalid.

The Amateur Packet community is aware of this problem. "One-way" packet is used for certain bulletin board applications that resemble the EBS application. Amateurs are currently working on a new terminal program to solve this problem. If their experiments are successful, we would merely change the terminal program at the Sheriff's Radio Center (SRC) to implement it.

Other Reliability Factors

There are a number of other factors that can impair data transmission using a packet protocol. All of them relate to the fidelity of the FM transmission system:

• The setting of audio levels throughout the system, especially through microwave relays in the LA County Radio network that feed this system.

• A poor signal path that does not provide sufficient signal strength at the receiver.

• A receiver that introduces distortion into the FM decoding process.

• A receiver that is experiencing multipath will decode phase distortion that will impair data transmission.

• The setting of audio levels from the receiver into the packet controller is by no means super critical, but must be within an acceptable tolerance range.

Transmitting Equipment

A packet controller supporting the protocol was installed at the Sheriff's Radio Center in mid-1988. A lap top PC compatible computer is used as a terminal to configure the packet operation mode, and compose, store, and play back messages over the system. The packet controller has a contact closure feature so it can key associated transmitting equipment.

The Equipment currently installed at the SRC is:

- AEA Model PK-232 Terminal Node Controller
- AC power adapter for above (1 A. DC)
- DataVue PC/XT compatible lap top with 640 K memory with:
  - 2 - 720K 3.5 inch floppy disk drives
  - Serial port
  - Parallel printer port
  - Real time clock
  - Back lit LCD 25 line x 80 column screen

Software is the AEA "PC Pakrat" program for PK-232 operation using a PC compatible computer. This program provides both the transmission function and a simple text editor in a single package that is easy to operate. Text can also be edited in a stand-alone word processor and stored on floppy disk, and transmitted over the system as complete files. This would eliminate the need to manually type in all messages.

Receiving Equipment

Participating EBS stations already have scanner or dedicated receivers tune to the 39.98 mHz. channel. A packet controller must be connected to the audio output of this receiver. A terminal with a screen or printer can be connected to the controller as an output device, or a properly configured printer can be connected directly to the packet controller once it is configured using a terminal. The A.E.A. controller, as do all of the controllers investigated, have non-volatile memory storage of critical setup parameters. With careful shopping, the total cost of a receiver, TNC, and printer should be $500 or less.

The least inexpensive equipment compliment would be a scanner receiver coupled to a Commodore C64 computer equipped with a plug-in module that converts the computer into a packet box. These modules are being built and distributed by several amateur radio clubs.

Many inexpensive packet controllers using the AX.25 protocol are available through amateur radio outlets. Only one is available now (so far as we are aware of) that implements a command that makes it possible to send messages in a mode that suppresses the station address (call sign) preceding each packet block of 256 "octets".

This is the Advanced Electronic Applications PAKRAT 232. It supports the MBX ALL mode of operation that suppresses the call sign of the addressee of the packet. This MBX ALL mode is exactly what is needed for EBS data sent from a central location to a number of reception points. Text broken up by repeated printing of the station call signs would be more difficult to read, and would present problems for the proposed system to serve the hearing impaired community (HIDAT).

Installing Receiving Equipment

The packet box (TNC) has a miniature phone jack that can be connected to the receiver's earphone or speaker jack. We are recommending stations install a transformer at the audio output of the receiver to prevent hum. The installation at KFWB would not work properly until a CALRAD 8 ohm to 600 ohm transformer was installed to break a ground loop.

TNC's come with detailed instructions on installing battery backup for memory, making up RS-232 cables, and assuring that the audio level from the receiver is set properly. Receiver audio should be derived from a point ahead of the front panel volume control so audio to the TNC cannot be turned down accidentally. Steps must be taken to prevent accidentally overloading the audio input of the TNC. Murphy's Law has special cases that apply to emergency systems. His laws must be respected at all times.

A front panel THRESHOLD control is set so the DCD light just goes out when unsquelched receiver noise is used as the audio source. An LED tuning indicator is built in so the TNC's response to the high and low tones that make up a packet signal can be monitored.

Receiver Types

Any receiver capable of tuning the 39.98 mHz. channel should work, but FM receivers that have appreciable phase distortion will present problems for proper packet data reception. Several scanner receivers have been tested so far that seem to work reliably, however.

Most participating stations in Los Angeles have purchased a fixed-tuned PLECTRON receiver that incorporates a NIAC two-tone decoder, and a simple cassette deck that activates when the decoder receives the proper EBS attention signal. The PLECTRON receivers are far from ideal. We are currently investigating installing separate receivers for the data transmission part of the system.

The Sound of Packet

Listening to packet transmissions is not very entertaining, and is quite obnoxious to many people. There is a calculated risk in mixing packet and voice on the same channel. The risk is that people working near the receiver will not want to listen to packet transmissions, and turn down the receiver audio, or completely disable the receiver. A separate receiver operating with carrier squelch would solve this problem as long as packet transmissions are not preceded with the NIAC attention signal that activates most of the receivers now installed.

Display Devices At The Receive Location

The TNC must be connected to some type of display device. A printer is recommended so stations will have a record of transmissions. The printer can be part of a computer or terminal attached to the TNC, or if it is a serial device, it can be connected directly to the TNC once the TNC is configured using a terminal or computer. A null modem cable that cross-connects RS-232 connector pins 2 and 3 and carries through pin 1 for ground is all that is necessary. When the packet box is properly configured, and set up parameters are safely stored in non-volatile TNC memory, merely turning on the TNC will initialize the printer and display a test message.

Having a computer or terminal available is a "must" for set up, is an aid to testing, and does make changing various packet parameters much easier. Disconnecting the computer or terminal after installation and successful testing is one of the goals of the system to keep the installation as simple as possible.

Protecting The System

Placing all equipment on an uninterruptible power supply, and installing transient protection are two recommendations we heartily endorse.

Testing

Although stations can request special tests by calling the Sheriff's Radio Center, the EPAIG is encouraging everyone to use a special cassette tape. This tape, recorded on a high quality cassette deck, has the two packet transmission tones recorded at reference level, and several minutes of packet data. Data transmission should approach 100% reliability if the system is functioning properly when this test tape is played back on most portable cassette decks found at broadcast stations.

The PLECTRON units are capable of recording packet data transmissions on cassette. The PLECTRON unit can play transmissions back and the TNC will decode the tape as if it were audio coming from a receiver. These PLECTRON cassette decks are not stable enough to play the system test tape, but do seem to work on tapes recorded and played back on the same unit.

The Hearing Impaired Data Transmission (HIDAT)

The Los Angeles County Operational Area Emergency Area will transmit EBS information to hearing impaired citizens of the County via participating television and cable entities using packet radio technology. Broadcasters are developing this mode of EBS data transmission in cooperation with Los Angeles County to provide hearing impaired citizens with visual information to be aired as slow speed "crawls" on TV screens. This visual information will provide enhanced emergency communication with TV viewers tuned to participating stations.

While no one can guarantee home TV receivers will have a source of power, or participating TV stations will be on the air after an emergency, such events may require EBS activation for days or weeks. As services are restored, hearing impaired citizens can receive information through this system.

Data transmissions will be in addition to voice announcements. Any broadcast entity, school, or business wishing to receive these data transmissions can connect a low-cost packet controller (TNC) to the 39.98 mHz. local EBS channel.

HIDAT Equipment

Now that packet data is being sent over the system for routine weekly tests, we are gathering information on low cost character generators that can be driven from the TNC's RS-232 output. The goal is to find an NTSC generator that will accept data, and produce a "crawl" across a TV screen at a reasonable reading rate.

Low cost and reliability are important considerations. We are trying to encourage as many local TV stations as possible to install this equipment at the studio and at their transmitter sites as well.

The Future of the HIDAT System

The hearing impaired community was encouraged to learn that we are going to provide a system that will help them deal with the isolation imposed by a major disaster. It is up to the Los Angeles broadcasting community to come through with a finished system. Once prototyped and tested, we believe other communities may find this system will solve their emergency communications needs to their hearing impaired citizens.

Information Input to Local EBS

While weekly tests will continue, and improvements will be made to the system, no one will know what will happen in the ultimate test, a major earthquake. Since any information system is only as good as the sources available to it, the EPIAG is also working to assure all players in the emergency drama can communicate with the Emergency Operations Center at the Sheriff's Radio Center. Amateur radio operators in the Disaster Communications Service (DCS) and the Amateur Radio Emergency Service (ARES) will play a key role in the drama. When phone lines are down, and many normal radio systems are inoperative from lack of power or other reasons, Amateur Radio may indeed provide much of the information that will be transmitted to broadcast stations from the Emergency Operations Center via the Amateur technology of packet.

Conclusion

The heart of the system has been installed. Packet tests are now conducted weekly. At KFWB's installation, test data reliability exceeds 95% at this time. Experiments are underway to improve the system to as close to 100% data integrity as possible. A close watch is being kept on developments within the Amateur community for improvements for data transmission on one-way systems. The system has been tested sufficiently at this point that we are encouraging participating stations in Los Angeles County to consider installing necessary packet receiving equipment.

This paper has been a preview of a system that is still in the early stages of development. We hope to provide updates on this project at broadcast engineering seminars in the future.

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Credits

The author gratefully acknowledges the body of work and experimentation performed by the Amateur Radio community, and especially theTucson Amateur Packet Radio Corporation.

Thanks and much credit are due to Mr. Stephen Smith, an Engineer with the Los Angeles County Facilities Management Department. Mr. Smith spent much of his own time and drew upon his personal resources to implement this system. Mr. Smith is also an amateur radio operator holding the call sign WA8LMF.

Mr. Burt Weiner(K6OQK) of KHJ Television, and Mr. Jack King, now with the Los Angeles County Facilities Management Department, must be credited with directing the EPIAG toward actions that resulted in this system.

EIGHT-CITY DS3 DIGITAL VIDEO TRIAL—PROGRESS AND NETWORKING FEATURES

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ABSTRACT

Progress continues toward a U.S. field trial of a terrestrial digital network for broadcast television distribution and collection between a network broadcaster and its affiliates or news bureaus. There are two primary challenges: first, to be sure that the picture quality, including the effects of errors and protection switching hits, will be acceptable; and second, to be sure that the networking capabilities and controls are acceptable. Both of these challenges are being faced and, with the continued good will of the numerous trial participants, the trial is moving forward. Papers presented at the 1987 and 1988 NAB Conferences describe this activity in detail. This paper describes the progress and changes since then.

TRIAL PARTICIPANTS MEETING

The second Trial Participants meeting was held in Minneapolis on September 29, 1988 hosted by Bellcore and U S West. Attendance included Broadcasters, Suppliers, Inter-exchange Carriers, Regional Companies and Bellcore.

Several important decisions were announced by Bellcore and by the various trial participants that will help to assure the continued momentum that is building toward beginning a landmark trial system connecting 8 cities across the US and a separate system connecting 5 cities in Canada.

The 8-city, 2-way, DS3 channel connection is being provided by CNI, CTI, Lightnet, Norlight, Southernnet and Wiltel on fiber optic and microwave transmission facilities. All seven Regional Companies are participating as well (see Figure 1). Participating broadcasters and their affiliates or news bureaus include ABC, CBS, NBC, PBS and Fox.

CONCERNING VIDEO TERMINALS

A major milestone had been reached, on June 21, 1988, when all the trial participants were invited to observe two potential video coding algorithms for use in the trial. This was accomplished with the considerable help of CBS, who provided the necessary high quality picture material and chroma keying to test the Telettra (discrete cosine transform component coder with NTSC interface) and NEC (differential PCM composite NTSC coder) standards proposals (Ref. 1 and 2). Both proposals were judged by the participants to be suitable for use in the trial.

At the September 29th meeting, Telettra announced that they would contribute eight complete Video Terminals, one for each of the eight US cities, to be used as a baseline by all the broadcasters/affiliates for the trial. This includes the necessary video codecs and the additional functionality (Ref. 3) required to be compatible with the remote surveillance and control and express order wire functions of the system plan. Other suppliers codecs or complete Video Terminals may be substituted between two or more cities by agreement with the broadcasters on their own initiative during their portion of the trial. ABL, NEC and Northern Telecom will participate on this basis. Other suppliers are invited to participate on this basis as well.

Bellcore supports the formal standardization of an ANSI approved video coding algorithm based upon industry supported testing of actual hardware as part of the ongoing 45 Mbps coding project in T1Y1. However, the standards activity is still ongoing and so the trial plan accommodates other proposals if they are reduced to hardware and if the broadcasters agree to include them.

The baseline trial will be very flexible. It will support a wide range of television signals including:

(1) the high quality needed for both distribution and collection (contribution) of program material;

(2) either composite (NTSC) video or digital studio quality component (luminance and two color difference signals) video in the one DS3 channel;

(3) noisy pictures for those times when picture quality may not be so good - for example, when receiving pictures from a marginal overseas satellite connection, such as, from the Phillippines to the US;

(4) teletext signals - these are digital data signals that a broadcaster inserts in the vertical blanking interval of the picture, providing stock quotations, or other material; and

(5) video pictures coming out of an ordinary VCR and without time-base correction because it is not sensitive to the variations from picture line-to-picture line caused by inaccurate motor drives in the VCR.

CONCERNING NETWORKING

Bellcore previously indicated that it was considering the possibility of replacing the Multipoint Unit with a Multicast Switch as a result of suggestions by the television broadcasters for even more networking flexibility even though it might be more costly and might take longer to develop. At the September meeting, Bellcore announced that the generic requirements for the Multicast Switch had been completed (Ref. 4) and now supersede the Multipoint Unit.

Telinq announced that it was well along in the preliminary development stage and has since agreed to provide eight Multicast Switches, one for each trial city, to be located at telephone company central offices, for the baseline trial. If other suppliers are interested, they may be substituted during the trial. Telinq also agreed to provide a rudimentary controller for the trial in order to demonstrate all the networking surveillance and control and express order wire features provided by the Multicast Switch.

The Multicast Switch will allow tree like, or more complex, networks with all possible switching combinations for the DS3 payload, express order wire and surveillance and control channel. Each appears on a separate electronic crossbar switching fabric permitting any input to be connected to any one or more outputs including a return path to create a loopback, or loopback/through. The network will operate with full DS3 frame synchronization in both directions. This allows the three subchannels to be switched or looped back independently and completely avoids DS3 reframing every time a switching operation takes place so that the control network always remains intact and to keep the speed of switching as fast as possible.

The surveillance and control channel and express orderwire channel are created from the so-called C-bit positions contained within the DS3 bit stream. These channels are expected to provide the flexibility needed to both emulate and exceed the capabilities of today's satellite networking arrangements for fixed locations.

Bellcore's new proposed generic requirements include having the network controller able to be connected either to a Video Terminal or directly to any Multicast Switch or to a DS3 Networking Test Set via a common interface.

OPERATIONAL FEATURES

The proposed generic requirements for the intelligent nodes (Multicast Switches and Video Terminals) incorporate a number of additional features to create a built-in operational capability. Much of this capability is fundamental to the two-way nature of the network allowing the customer to, from one location, probe the network and receive replies. Since the Multicast Switch is symmetrical, this can be done from any terminal location that is equipped with a controller.

The operational features include:

1. The Multicast Switch provides addressable, customer-controlled loopbacks of either the payload, orderwire, or control channel while still allowing the forward signal to go through in order not to disturb service.

2. The Video Terminal provides addressable, customer-controlled analog loopbacks of the video picture and stereophonic sound, and digital loopbacks for the DS3 payload and DS1 subchannel used for audio. These loopbacks are for both the line side and the station side of the terminal.

3. Overhead bits are included within the surveillance and control channel for rugged error checking to avoid false responses to commands.

4. Overhead bits are included within the payload to correct errors (forward error correction) so they will not affect the received picture.

5. Alarm indications are provided on all incoming ports to the Multicast Switch and Video Terminal, plus this will cause a far-end alarm signal to be generated on the corresponding outgoing circuit on the control channel so that a trouble in any one or both directions on any network link will be caught at both ends of the trouble so that it can be picked up in response to a poll by the customer.

6. A repetitive, self-addressed alarm signal will be autonomously generated by the Multicast Switch to inform all downstream terminals of both the existence and the location of an incoming failure.

7. Customer-controlled turn-on and turn-off of individual ports on the Multicast Switch will enable circuit additions, rearrrangements, or repairs to be made to the network without interfering with the rest of the network.

8. Master and Alternate Master identification numbers, settable by the customer, are stored in each intelligent node (Multicast Switch and Video Terminal) to avoid control chaos which could otherwise occur since the Multicast Switch can be controlled from anyone of its ports. Only the Master can control the network unless it relinquishes control to the designated (resettable) Alternate Master.

CONCERNING A DS3 NETWORKING TEST SET

The generic requirements for features needed to upgrade an ordinary DS3 Test Set to be able to monitor or insert/receive network commands/responses or to monitor error performance have been documented (Ref. 4). In addition to Tautron and Anritsu, previously announced, Telinq announced at the September meeting that they too would provide a test set for the trial.

A DS3 test set will permit Telephone Company craftspeople to remotely access a multipoint network and to perform tests or monitor performance of the service. An entire multipoint network can be tested from one location provided the broadcaster agrees to relinquish the network, or a portion of it, and informs the Telephone Company of the secure codes it is using to address and control individual Multicast Switches and Video Terminals.

NEW TRIAL SCHEDULE

In order to allow sufficient time to develop the more complex Multicast Switch, a new trial start date of October 1989 has been agreed to by all trial participants.

AUDIO TERMINALS

Participating Audio Terminal suppliers are: ABL, AEG Bayly, Coastcom, Comlux, Intraplex, NEC and RE Instruments. These terminals will provide 15 kHz, high quality multi-channel television sound (stereo and Separate Audio Program) plus six additional channels for voice or data transmission.

ORDER WIRE EQUIPMENT

Compatible order wire equipment will be used for each of the eight cities in the trial. This will permit 2-way, off-the-air oral communications between the network broadcaster and its affiliates - something not normally available in today's mostly one-way satellite networks. Selective calling may be used or all stations may communicate simultaneously.

CANADIAN TRIAL

CBC and Bell Canada will be conducting a separate 5-city trial using the same requirements as for the US trial.

CONCLUSION

The trial effort has come a long way. With the agreements reached at the Trial Participants meeting, and with the completion of the proposed generic requirements for the Video Terminal, Multicast Switch, and DS3 Test Set features, there is every reason to expect that this significant new industry-wide effort, joining the broadcast television and the telecommunications industries and their suppliers will succeed. Any company that has not yet agreed to participate in the trials and may wish to do so may still be accommodated.

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Bellcore
Document Registrar
Room 2J-125
435 South Street
Morristown, NJ 07960-1910

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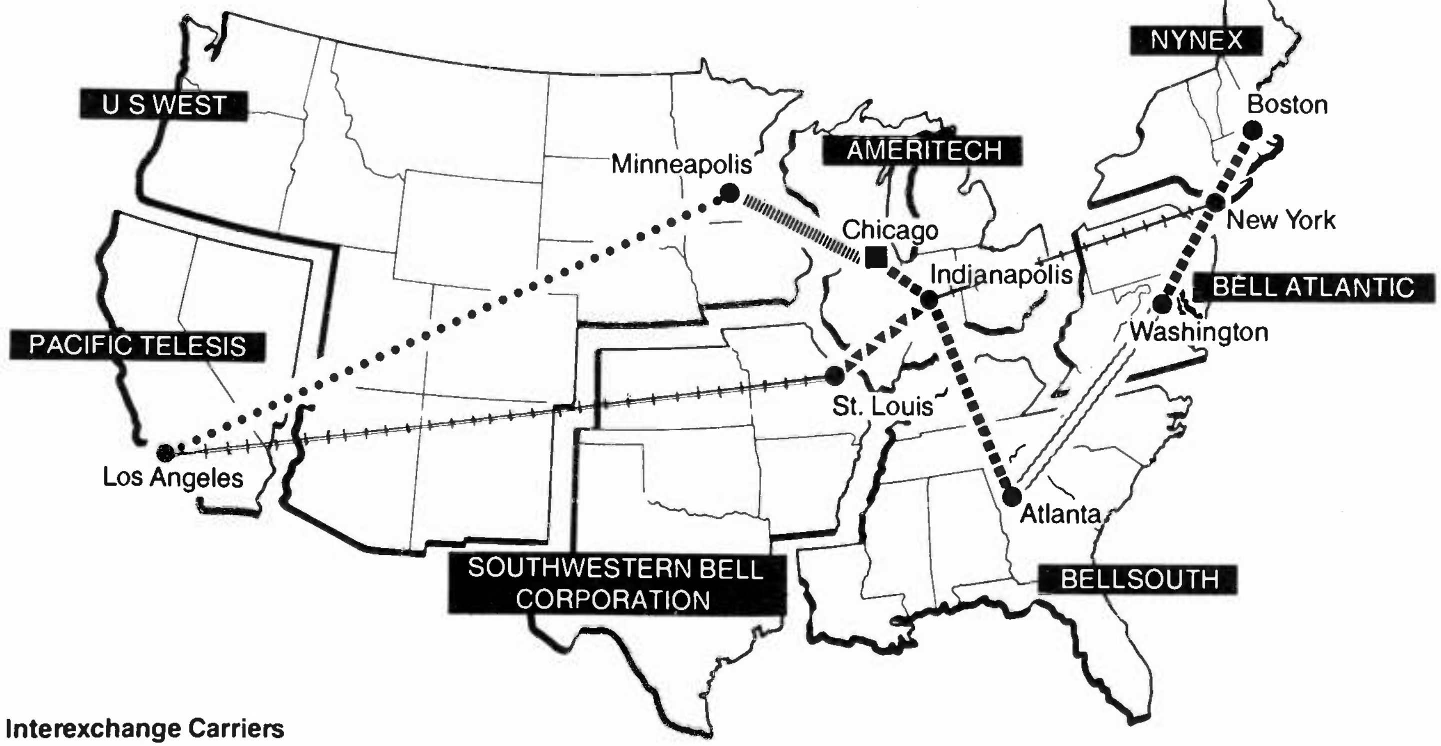
DS3 DIGITAL VIDEO TRIAL



Interexchange Carriers

■■■■ Lightnet	┼┼┼┼ CTI - Communication Transmission	
Southernnet	•••• Wiltel	■ Intermediate Point
Norlight	▲▼▲▼▲ CNI - Consolidated Network	● Trial Cities

Figure 1

BROADBAND ISDN ARCHITECTURE

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ABSTRACT

Market pull and technology push are the major forces driving future broadband ISDN networks. Firstly, this paper examines these two driving forces. Secondly, it describes a long-term broadband ISDN architecture. Finally, it describes possible evolutionary paths from early-availability architectures through the longer-term.

1. BROADBAND SERVICES MARKET PULL

New service opportunities are a major driving force for future Broadband Integrated Services Digital Networks (B-ISDN) [1] [2] [3]. Business customers have a growing need for information technology and increasingly sophisticated telecommunications. Local Area Networks (LAN) have increased the intrapremises data rates from kbit/s to Mbit/s over the past decade, but interconnection of these LANs is still a problem.

Residential customers are demanding more selectivity in video entertainment, evidenced by the increasing number of channels demanded on cable television systems and the increasing rentals of video cassettes. Future entertainment services may range from viewing normal broadcast video sources, to viewing special events such as first-run movies on a pay-per-view basis, to video-on-demand in which a user is connected directly to a remotely-located video source with complete control over that source.

This section discusses some of the existing and future services that a broadband ISDN should support.

1.1 Existing Services

A broadband ISDN should support existing voice telecommunications services. Other voice services, such as 7 kHz and 15 kHz speech communications, may also become important, especially for business customers.

A broadband network should support existing data communications services, such as those that provide features for X.25 and SNA applications. In addition, higher-speed data services, discussed in the next section, will become increasingly important.

A broadband network should support existing private line services, including the DS-0, DS-1 and DS-3 rates. New innovations in the signaling protocol for broadband networks may enhance these capabilities by allowing private virtual networks to be defined dynamically.

1.2 High-Speed Data Services

The widespread penetration of LANs has firmly established very high-speed networking (1-10 Mbit/s) in the local premises environment. The success of LANs can be attributed to the growing use of more powerful PCs and workstations for high performance applications such as remote file serving and distributed processing. The growing need for LAN interconnection and such applications as desktop publishing and CAD/CAM suggest an emerging market for multi-megabit communications services that extend beyond the local premises, across metropolitan and wide area environments. Switched Multi-megabit Data Service (SMDS) is a high-speed connectionless data service proposed for broadband networks to meet these future data communications needs [4].

1.3 Video and Image Services

The availability of high-speed communications can completely change the character of business interactions. In desktop teleconferencing applications, windows on the user's terminal can display full-motion images of the other parties in a conference. Viewgraphs, sketches, photographs, and text files can be exchanged and either displayed or printed. Other parties can be added to a conference as needed for consultation, and subconferences created for private conversations.

Electronic mail, commonly used today for transmitting text messages, can be stimulated by the availability of broadband networks. Integrating multiple media, including text, voice, image and full-motion video, can increase the effectiveness of these systems.

Switched access television service, the delivery of entertainment video to residential customers over a switched broadband network, can tap a large existing market and offer several advantages over current delivery alternatives. These advantages include access to a wider variety of programming, elimination of the need for scrambling equipment, opportunities for new services such as pay-per-view, and higher quality video such as advanced television (ATV).

Recent increases in the sale and rental of video recorders, players and cassettes show that users want to control their own programming. Video-on-demand service provides the user with ultimate control over video programming without having to purchase and maintain a video cassette player and without having to pick up and return video cassettes. Instead, a user calls a video vendor and selects a program from a list of available video sources. Video information from the vendor is switched to the user via the broadband network, and control signals from the user (play, stop, pause, rewind, etc.) are transmitted to the vendor over a data connection.

1.4 Other Future Services

Other future broadband services include access to reference information. Text, graphics and images can be viewed on a student's terminal, printed on a local printer, or stored in a disk file on the student's PC. Some people predict that broadband ISDN will completely change the way data is viewed. Just as the telephone system displaced the telegraph in the nineteenth century, *telesophy*[1] may displace today's simplistic data communications [5]. Telesophy advocates predict a new mass communications system that will explicitly support the sharing of knowledge within a global information community.

Video browsing may become popular for shopping. For example, by calling a vendor via a broadband network, a shopper can view a full-motion demonstration of a riding lawn mower illustrating its maneuverability around trees and its ease of operation. With interactive capability, the seller can tailor the presentation to the level of detail desired by the buyer. A complete home merchandising system can provide cost and availability information to the buyer, allow the buyer to order the product and specify shipping information, and allow the seller to update an inventory system automatically. Real estate sales is another possible application. A person transferring to a different city can access video information remotely about homes for sale, schools, churches, shopping centers or maps of the new community.

1. Telesophy is a word coined by Bruce Schatz of Bellcore. It comes from the words telephony and philosophy and means "wisdom at a distance". The "tele" part indicates that the access can be made independent of physical location in the communications network. The "sophy" part indicates that filtering can be done independent of physical data type.

2. BROADBAND TECHNOLOGY PUSH

Over the past few years, broadband technology has emerged as an attractive alternative for future telecommunications networks. It promises an integrated access structure with flexible allocation of bandwidth, capable of providing existing services and positioned to serve a broad range of future services. It promises a more integrated switching fabric, eliminating the need for a switching fabric for each service type and the associated complexity and high operations costs. Finally, it promises to simplify interoffice transmission and switching facilities.

Several parallel advances are responsible for the shift in economics of integrated broadband networks.

2.1 Optical Fiber

Optical fiber is used extensively in the interoffice plant. It is now starting to be used in the loop between the Central Office (CO) and subscriber carrier locations. The cost of single-mode fiber, which offers virtually unlimited bandwidth, dropped from 5 dollars/m in 1982 to about 23 cents/m in 1987. With continued progress some experts predict that the cost will drop to 4 cents/m by 1992.

2.2 Low-Cost Optical Devices

Until recently, single-mode fiber implied the need for laser transmitters. However, single-mode laser sources had inadequate reliability for use in the loop and were not compatible with near-term cost objectives. On the other hand, lower cost light-emitting diodes (LED) as used in low-bit-rate multi-mode systems were known to meet loop reliability requirements. It has now been demonstrated that light emitting diodes can be used with significant lengths of single-mode fiber at bit rates in the 600 Mbit/s range [6].

More recent work has been aimed at reducing the cost of laser sources through simple packaging and relaxed yield criteria [7]. As laser technology for loop applications matures and as production quantities reach levels characteristic of the loop market, laser costs and reliability should improve. Thus, both LED and laser sources now show the potential for use as low-cost optical sources for integrated broadband networks.

2.3 Low-Cost Broadband Circuit Switches

Much progress has been made in broadband circuit switching technology. Just a few years ago, video switches used discrete relays or *pin* diodes to implement an analog switching fabric and were both large and costly.

Several experimental devices, using both CMOS and ECL technologies, have been fabricated for space-division circuit switching of broadband channels. Using such VLSI technologies and new packaging methods, it is now possible to create a compact 64x64 switching subsystem with a 4x4 matrix of 16x16 CMOS crosspoint chips on a single card measuring four inches square [8]. In addition to the advantages of digital switching and smaller size, this board represents about a 10:1 reduction in the size and the cost per crosspoint over that of the analog switching fabric.

2.4 Synchronous Optical NETwork (SONET)

The advances in synchronous multiplexing technology, the widespread deployment of incompatible fiber systems, and the need for flexible management of the bandwidth of fiber transmission systems led to the formulation of the Synchronous Optical NETwork (SONET) concept [9]. SONET is a network hierarchy of octet-interleaved synchronous signals. This hierarchy forms a family of standard electrical synchronous transport signals (STS) that can be simply converted to a family of standard optical carriers (OC). Through these STS and OC interfaces, SONET provides a set of standard transmission interfaces for customer premise equipment, switches, distribution and interoffice transport systems.

The STS rates are integer multiples of 51.730 Mb/s. The interface structure contains separate overhead and information payload. As a result operations, administration and maintenance capabilities are an integral part of the transport systems. Currently, there is a significant interest in introducing SONET based transport in exchange carrier networks when the equipment becomes available over the next several years. Moreover, SONET is proposed as a basis for defining broadband user-network interfaces for ISDN.

2.5 Label Multiplexing

Position multiplexing, also called Synchronous Transfer Mode (STM), is used for multiplexing channels into a common transmission stream, such as DS-1 or DS-3. An STM stream has a fixed channel structure; each bit within the payload is associated with a specific channel, which has a specific information transfer rate.

With label multiplexing, also called Asynchronous Transfer Mode (ATM), each frame is divided into cells that are similar to small packets [10] [11]. Each cell contains a fixed-length header and a fixed-length payload. A label in each header explicitly identifies the channel. Therefore, multiplexing and demultiplexing hardware is simplified and the structure of the transmission stream can be changed dynamically. Furthermore, due to the fixed size of the cells, STM can be emulated by ATM. ATM has been proposed for use in the SONET payload of integrated broadband networks.

2.6 ATM Cell Mergers and Sorters

Cell mergers and sorters provide flexible handling of ATM cells and have a variety of uses [12]. For example, cell mergers in the Interface Module (IM) can multiplex information from many different users onto a single STS-3c signal for transmission to the CO; a cell sorter in the CO can provide the demultiplexing function. A cell sorter in the IM can separate video information from lower speed signals and route it to a circuit switch. A cell merger in the IM can combine a circuit-switched video stream with lower-speed voice and data streams for delivery over a single fiber to the customer.

2.7 ATM Cell Switching

The technology required for broadband cell switching is much different from that used in lower speed packet switching networks. One objective of conventional store-and-forward packet switching networks is improving the error performance of noisy facilities. Copies of packets are held at the transmitting side of a link until acknowledged by the receiving side of the link. Using optical fiber transmission, the error performance of a broadband network is expected to be much better than that of current networks, so retransmission is much less likely. The primary objectives for broadband cell switching are low delay for traffic such as speech, high throughput for traffic such as still frame video, low cell loss rate, and low internal buffer requirements for minimizing equipment costs.

One promising technology for broadband cell switching is the banyan routing network. A cell processor inserts a routing field, with the most significant bit first, into each cell delivered to the input of the banyan switch. The first stage of the banyan switch examines the first bit of the routing field and routes to the top or the bottom half of the second stage based on the value of this bit. The second stage examines the second bit of the routing field and routes to the corresponding fourth of the network. This process continues through each of the N stages. A switch with N stages has 2^N inputs and 2^N outputs and is composed of $N * 2^{N-1}$ switching nodes, with two inputs and two outputs each.

If a banyan routing network is preceded by a Batcher sorting network, the combination is non-blocking internally. A 32x32 Batcher sorting network and a 32x32 banyan routing network have been implementing in VLSI, the current chips operate at over 100 Mbit/s per port with the next version expected to operate at greater than 155 Mbit/s [13].

2.8 Broadband System Prototypes

Several broadband system experiments are proceeding, including The Prelude switch in France [14], the Elastic Basket switch in Japan [15], the Knockout switch at Bell Laboratories [16], the Experimental Research Prototype at Bellcore [17] [18] [12], and the PARIS switch at IBM [19]. Each of these experiments focuses on different aspects of broadband cell switching - the interface structure, the services, the switching fabric, the protocols and the addressing - and each experiment represents a different solution to broadband cell switching.

3. A LONG-TERM BROADBAND ARCHITECTURE

Figure 1 shows a view of a long-term target architecture for a broadband network. An interworking unit (IWU) at the customer's premises provides the conversion between various customer interfaces and an integrated user-network interface. This user-network interface uses SONET transmission at the STS-3c basic rate of 155.52 Mbit/s or at a higher rate such as 622.08 Mbit/s, depending on the service needs of the customer. ATM cells within the SONET payload provide flexible multiplexing in order to support a wide range of existing and future services.

In the distribution and drop plant between the Remote Electronics (RE) and the customer, single-mode fiber is dedicated to the customer, forming a star topology[2]. Two fibers can be provided to each customer, one for upstream and one for downstream transmission, or both directions of transmission can share the same fiber using WDM techniques. The cost difference between these two alternatives is slight, with short loops favoring two fibers and longer loops favoring one fiber [20]. Operations costs, while not quantified, may favor the use of a single fiber.

Single-mode fiber is also used in the feeder plant between the RE and the CO, but operates at a higher rate near 2.4 Gbit/s. Although the RE-to-CO distances are longer than the customer-to-RE distances, the feeder costs are offset by the concentration provided by the RE and by the higher transmission rate in the feeder and represent less than 5% of the total cost [20].

Customers near the CO will be served directly from an IM in the CO. Customers further from the CO may be more economically served from an IM located remotely [20], trading off the additional cost of placing the IM in a remote location against the cost of longer lengths of dedicated fiber.

Several topologies are possible for interconnecting the RE locations to the CO. In a star topology, dedicated fiber connects each RE to the CO. In a ring topology, a fiber ring starts at the CO and passes through each RE using add/drop multiplexers to drop and insert channels. A combination star/ring is also possible for the feeder topology in which switched channels are interconnected in a star topology and broadcast channels in a bus topology. The latter alternative may offer some savings, but the effect on the overall cost is slight since the feeder cable represents only a small portion of the total cost.

The physical implementation of a Next Generation Switch (NGS) [21] may incorporate different switch fabrics for different traffic types. For example, ATM cells carrying video information may be separated from cells carrying lower-speed information; video cells may routed to a circuit switching fabric while cells of other traffic are concentrated by ATM mergers in the IM and routed to a self-routing Batcher-banyan switching fabric. Although the physical implementation may incorporate multiple switching fabrics, the fabric appears to be integrated from the point of view of switch control and from the customer's perspective.

User-network and network-network signaling are integrated, with the same signaling protocol being used for both circuit and packet communications. Multi-media and multi-party communications are an integral part of these signaling procedures, facilitating the development of future interactive applications.

Hub offices provide the interconnections between CO locations. Like the feeder plant the interoffice plant connecting the CO locations to the Hub office may be connected in various topologies, such as star, ring or bus. If the community of interest is high enough, two COs may also have direct trunk groups in addition to connections to the Hub office.

Narrowband networks will continue to exist for many years. Interworking units (IWU) in central offices and hub offices provide the necessary conversions for interconnecting to other networks. In addition, fiber loops may be installed in exchanges with existing narrowband offices. An IWU in these offices provides conversion between the SONET interface on the fiber and an Integrated Digital Loop Carrier (IDLC) interface to the existing switch.

Integrated network operations and maintenance systems provide continuous surveillance of the network, reconfiguring the network during failure conditions to provide continuous service, isolating the cause of the failure, and dispatching personnel to repair the failed component. Integrated administration systems provide rapid service provisioning and accurate billing.

If coherent optics or dense WDM systems become practical and low-cost, Fig. 1 may change as follows. The active electronics in the RE may be moved back to the CO and multiple wavelengths used on the RE-to-CO fibers. This provides dedicated channels from the CO to the RE without increasing the number of fibers in the feeder plant. From a first cost standpoint, this alternative trades off the added cost of WDM equipment on a per-line basis against the cost

2. A recent T1S1.1 contribution proposed another level, called a Remote Multiplexer (RM), between the RE and the customer in order to increase sharing of the distribution plant. The fiber between the RE and the RM is called the sub-feeder plant and can use either a star, bus, or ring topology.

of 2.4 Gbit/s multiplexers on a per-feeder basis. Also, if only passive components remain in the RE, the operations costs may be lower. In either case, the physical architecture of the installed fiber is identical independently of whether the RE contains active electronics.

4. EVOLUTIONARY PATHS TOWARD A BROADBAND NETWORK ARCHITECTURE

The evolution toward the broadband network architecture described in the previous section will be constrained by several factors. First, standards for broadband are still being discussed, although much progress has been made over the past year, particularly in standardizing SONET. Second, even after broadband standards are complete, the development of a next generation switch (NGS) is likely to require several years. During this time, some components of the NGS will become available earlier than others, resulting in network architectures using combinations of old and new technology. Third, initial broadband systems will be more expensive since they are at the beginning of the cost learning curve. Finally, even after the cost of broadband systems reach the lower part of the learning curve, capital constraints will limit the rate at which broadband systems can be installed.

In light of these factors, an evolutionary strategy for introducing broadband network elements is needed that satisfies several conditions. First, it should allow some broadband services that are already in demand, such as high-speed data services, to be offered using early availability technology. Second, it should allow fiber to be installed for growth in the loop plant instead of copper, at a cost competitive with copper, in order to be prepared for the provision of future broadband services. Finally, the evolutionary path should allow cost effect upgrades as NGS systems become available.

The next two sections outline a possible evolutionary strategy for residential and business network architectures.

4.1 Evolution of Network Architectures for Business Services

The desire to provide early availability of high speed data services coupled with current development of Metropolitan Area Networks (MAN) equipment, leads to the possible architecture implementation shown in Fig. 2. A gateway, G, provides the conversion from a LAN protocol to the Switched Multimegabit Data Service (SMDS) protocol carried in a DS-3 rate transmission stream. The DS-3 electrical signal is carried to the CO by an optical transmission system with a proprietary interface. In the CO, the DS-3 signal is cross connected to a multiplexer and carried to a hub office, again using a transmission system with a proprietary interface. At the hub office the DS-3 signals are connected to a MAN switching system (MSS), where each SMDS packet is segmented into cells and routed to the destination port.

As SONET transmission systems become available, the architecture of Fig. 2 allows for evolution of the network to include SONET transmission systems with standard interfaces. Furthermore, when ATM mergers and sorters become available, the architecture shown in Fig. 2 evolves to that shown in Fig. 3. Standard SONET transmission is used on the optical fibers with ATM cells carried in the SONET payload. Segmentation and reassembly of SMDS packets into ATM cells is done at the customer's premises. ATM mergers and sorters in the COs reduce the interoffice transmission requirements. Finally, the MSS can be based on an ATM merger/sorter pair operating in a MAN emulation mode with static translation tables.

As the demand for SMDS service increases, the throughput capacity of the ATM merger in the hub office is reached. At that point, a higher capacity ATM cell switch can be placed between the ATM sorter and merger, as shown in Fig. 4. A cell switch with N ports can route cells from any of the N mergers at the input ports to any of the N sorters at the output ports, increasing the capacity of the system by a factor of N. As the data traffic further increases, cell switches are placed in the COs to handle intra-CO traffic, reducing the load on the interoffice links. Cell switches in the COs also allow for direct trunks between COs with a high community of interest, further reducing the load on the interoffice links.

Voice services can also be provided over the same fibers in each of the three architectures shown above. In Fig. 2 the proprietary transmission can provide multiple DS-3 channels over the fiber. For example, one DS-3 channel may carry voice traffic to the existing voice switch via an Integrated Digital Loop Carrier (IDLC) interface. In Fig. 3 the voice information is carried in ATM cells on the user-network interface, split off from the data traffic in the CO using ATM sorters, and routed to the existing voice switch. When ATM cell switches are installed in a CO, the voice traffic may be carried by the cell switch, with the interworking unit moved to the trunk side of the office. This evolution is discussed further in the following section.

4.2 Evolution of Network Architectures for Residential Services

For growth of voice services in the loop, optical fiber is becoming a cost-effective alternative to copper. In addition, it positions the loop for providing future broadband services.

Several suppliers are now developing equipment for providing POTS service over fiber using proprietary interfaces. After completion of the broadband standards for SONET and ATM, standard equipment will become available for providing POTS service using the architecture shown in Fig. 5. The IWU provides the BORSCHT functions and places speech samples into ATM cells. At the RE, the cells from several users are merged together into a common stream, reducing the transmission requirements between the RE and the CO. At the CO, the aggregated SONET/ATM interface is converted to an IDLC interface compatible with the existing voice switch.

As the amount of fiber in an exchange increases, the existing voice switch may be replaced with an ATM cell switch capable of switching both voice and data services. The architecture shown in Fig. 5 is easily evolvable to that shown in Fig. 6, since the electronics in the RE are unchanged. When the cell switch is installed in the CO, an IWU on the trunk side of the office replaces the IWU on the line side.

A loop based on optical fiber has the capability to support video distribution services in addition to voice and data services. The architecture shown in Fig. 7 is one alternative for early availability video distribution. This architecture is similar to that shown in Figs. 5 and 6, except a second wavelength is multiplexed onto the fiber using WDM techniques. This second wavelength may carry several analog video signals using techniques such as frequency-modulated (FM) microwave subcarriers [22]. The incremental cost of adding video service in this architecture is the cost of the optical source and the WDM equipment in the RE, and the cost of the tuner at the customer's premises.

A second alternative for providing video distribution service is shown in Fig. 8. A channel-change message from the user results in a crosspoint being closed in the circuit switch. Thus, the selected channel is switched to the input of the ATM merger. This alternative uses ATM for multiplexing the voice, video, and data signals into a single transmission stream carried by a single wavelength. The incremental cost for providing video service with this architecture is the cost of the merger and the allocated cost of the circuit switch in the RE, and the cost of an ATM cell sorter and a codec at the customer's premises.

The architectures shown in Figs. 7 and 8 provide different service capabilities. The first architecture provides broadcast access to a limited number of video signals, but does not constrain the number of channels simultaneously accessible from that limited set. The second architecture provides switched access to an unlimited number of video sources, but limits the number of sources available simultaneously. The incremental cost of accessing an additional channel with the first architecture is the cost of an additional tuner. The incremental cost of accessing an additional channel with the second architecture is the allocated cost of an additional port on the circuit switch and the cost of an additional codec. Also, depending on the video coding algorithm, the line rate requirements may change. For example, a 155.52 Mbit/s signal is expected to carry one ATV signal or three NTSC signals, while a 622.08 Mbit/s signal carries up to four ATV signals.

Other early-availability architectures are also possible for carrying video signals to residential customers, but are not discussed here.

5. CONCLUSIONS

This paper described possible services providing a market pull for Broadband ISDN and recent technological advances providing a technology push. A long term Broadband ISDN architecture was described based on a double star topology, optical transmission, and high-speed circuit and packet switching. Finally, evolutionary paths for broadband ISDN architectures were described, which gradually introduce broadband network components and capabilities as they become available in order to provide early-availability of some broadband services.

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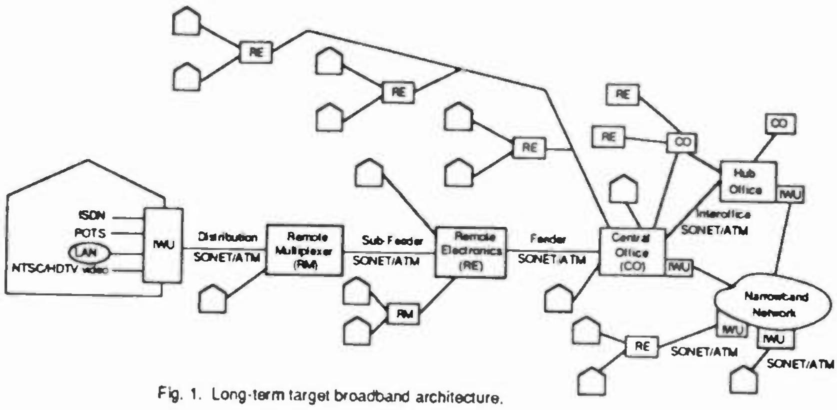

Fig. 1. Long-term target broadband architecture.

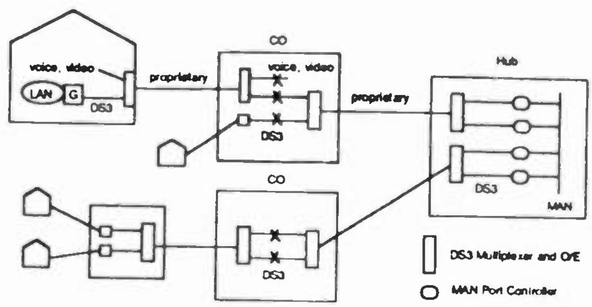

Fig. 2. Early availability architecture for providing SMDS service using MAN technology.

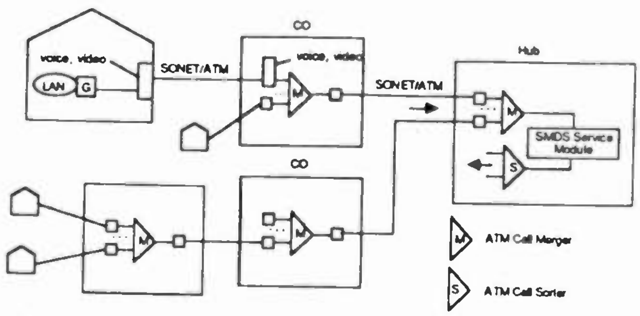

Fig. 3. Architecture for providing SMDS service using ATM technology.

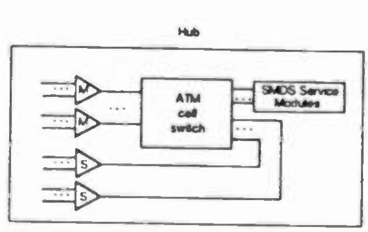

Fig. 4. Expanding the capacity of a switch for providing SMDS service.

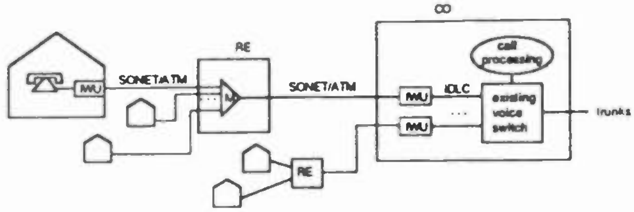

Fig. 5. Architecture for POTS-on-fiber using ATM and existing voice switches.

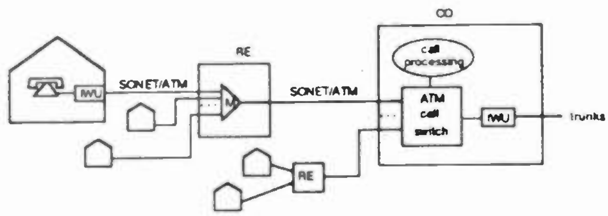

Fig. 6. Architecture for POTS-on-fiber using ATM cell switch.

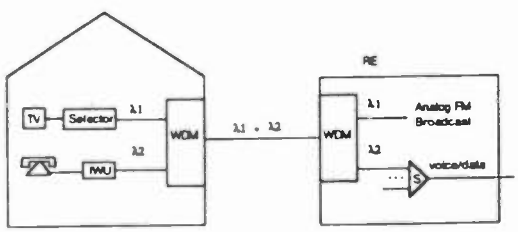

Fig. 7. Architecture for adding broadcast analog video.

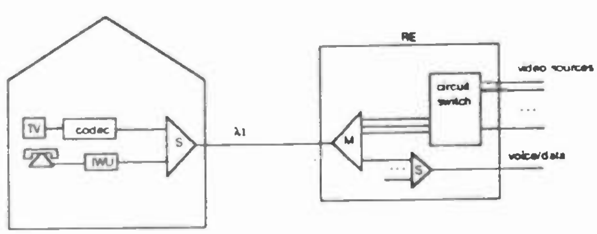

Fig. 8. Architecture for adding switched digital video.

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