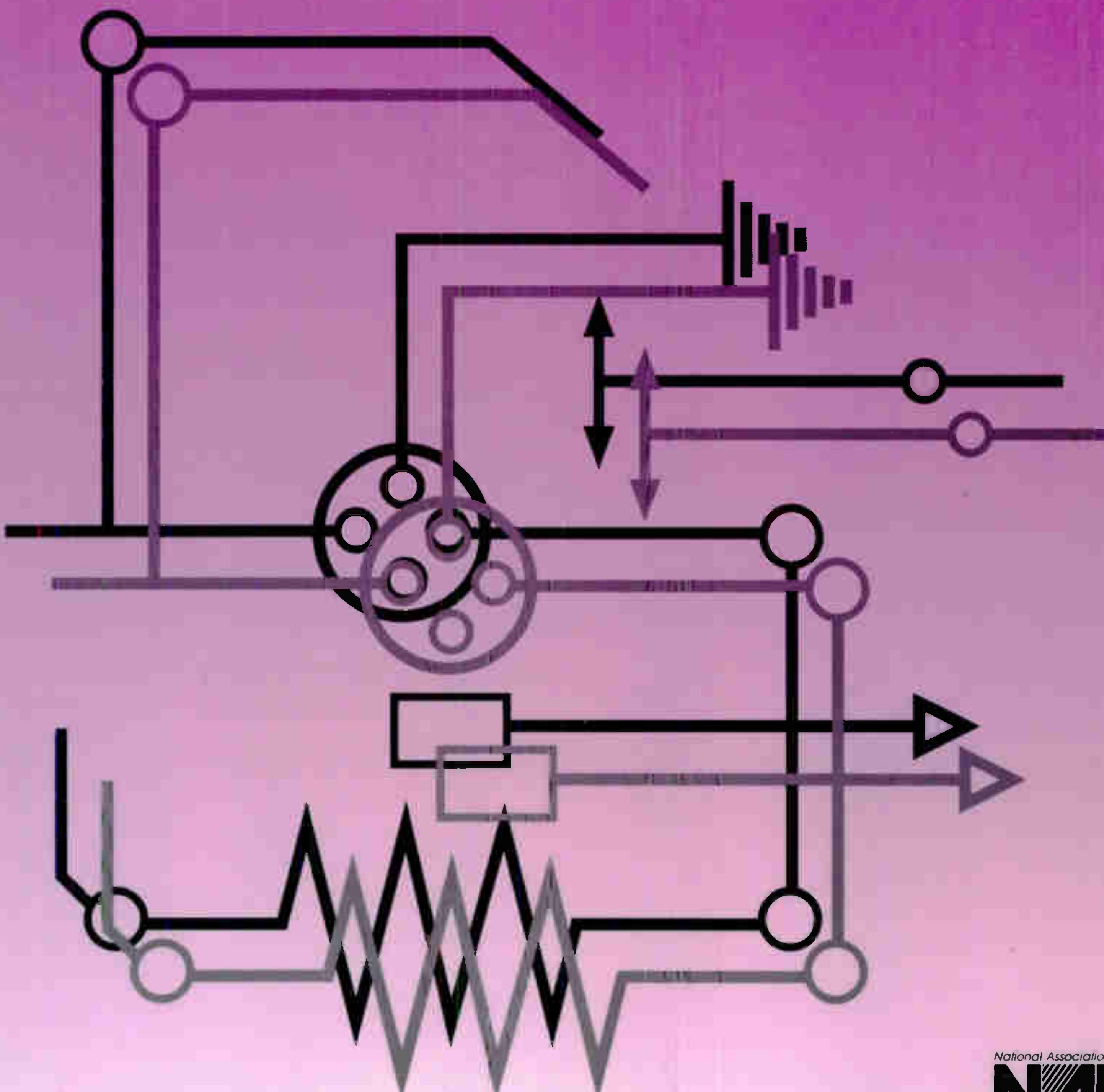
1992 Proceedings

46th Annual Broadcast Engineering Conference Proceedings



National Association of Broadcasters
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Bob Moore
4-92

1992 PROCEEDINGS

46th Annual
Broadcast Engineering
Conference Proceedings

Las Vegas, Nevada

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These proceedings contain technical papers presented at the NAB Broadcast Engineering Conference April 12-16, 1992.

Published by the NAB Office of Science and Technology

ISBN 0-89324-139-3

Additional copies of the 1992 Proceedings are available through NAB Services, 1771 N Street, N.W. Washington, DC 20036-2891. For price information, please write or call 1-(800) 368-5644.

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March 1992

Dear Industry Engineer:

On behalf of NAB's department of Science & Technology, I am pleased to present the *1992 NAB Engineering Conference Proceedings*. The immense diversity of this year's papers reflects the broad scope of concerns every engineer addresses every day.

In radio, the emphasis in Digital Audio Broadcasting (DAB) is shifting from theoretical systems to hard data and hard choices. Even while we consider entirely new transmission systems, we continue to explore ways to maximize the coverage of existing signals and improve the quality of station audio with digital techniques. In television, we consider exciting new technologies to improve the quality of broadcast video, the impact of digital techniques on facility planning, innovative options for expanding video production, and the opportunities in interactive video. Perhaps most interesting is NHK's concept for an entirely new kind of broadcasting service, called Integrated Services Digital Broadcasting, or ISDB.

I'm equally excited by our NAB '92 authors and presenters. I am especially pleased to welcome the participation of the Society of Broadcast Engineers (SBE) and the continuing increase of international presentations on broadcast engineering.

Bringing together a diverse group of professionals to share technical understanding across international and association borders is the essence of communications, perhaps also the essence of broadcasting. Continued exchange is essential as we grow to meet the challenges of broadcasting in the 1990s.

Best regards,

Michael Rau

Michael C. Rau
Senior Vice President
Science and Technology

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Donald Wilkinson
Chairman, NAB Engineering Conference
and Advisory Committee
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*Paper not available at the time of publication.

LOW COST DIGITAL AUDIO STORAGE UTILIZING 3.5 INCH FLOPPY DISKS

William Franklin
Fidelipac Corporation
Moorestown, New Jersey

ABSTRACT

This paper deals with a new digital audio system which is designed specifically for the recording and reproduction of audio in the broadcast format, and operates identically to the endless loop cartridge machine. By using digital compression and variable sampling rates, it is possible to store up to 5 minutes of stereo audio on the same inexpensive 3.5 inch floppy disk which is used in your personal computer.

BACKGROUND

The NAB cartridge provides broadcasters a simple, high quality recording media unique to their requirements for production and on-air presentation. It has continued to evolve with technology and manufacturer's ongoing research and development to become a very dependable recording format. From its origins as a mono, commercial-only sound source, to its current position as a music length around-the-clock mainstay of a majority of stations, the NAB cart offers broadcasters an economical system for recording and playback of audio.

The key features of the NAB cart are:

1. Instant access.
2. Simplicity of operation.
3. Durability
4. Interchangeability.
5. Low cost of both media and hardware.

The NAB cartridge has its technical limitations. The rigors of continuous insertion and play eventually cause the tape and cartridge to wear out. Engineers must clean and replace pinch rollers and heads, not to mention making lengthy electronic and mechanical alignments which the cartridge system requires. And finally, while the original claims of interchangeability between cartridges and machines have always been stated, in actuality, the best one can hope for is an in-house standard of one brand of cartridge and splitting the difference in stereo phase consistency. But despite its limitations, after 30 years the NAB cartridge remains the standard due to convenience, simplicity, reliability and cost effectiveness.

We now find ourselves in the digital audio age. Numerous recording formats offer excellent sound capability and enhanced control of our recordings. It is a given that at some point, all audio will be represented by digital bits rather than the analog method that has supported us for the last 50 years.

Digital Audio Criteria

As we see it, the prerequisites for a digital cartridge machine replacement are as follows:

1. It must be as simple to operate as a cart machine.
2. It must have CD quality audio.
3. It must cost no more than a high quality cart machine.
4. It must be durable and easy to maintain.
5. Media cost must be in line with existing cartridges.

There are many new digital audio storage formats available, all of which offer digital audio quality. Operational complexity ranges from simple reel-to-reel emulations to multi-tasking, GUI based audio control centers. As one discovers the ravenous amount of memory storage digital audio requires, the media and storage space become expensive and quite large. With it comes an uneasiness in having all archives stored in one location with no provision for interchangeability or recovery in the event of a mass storage system failure.

All new technologies have a cost. Unique digital audio storage architectures are very costly due to the low quantity of units produced, the development costs and the sourcing of the appropriate components. The development of a low cost digital cartridge machine replacement must take advantage of existing low cost components and media, and offer a sophisticated level of control while retaining the ease of operation of the cartridge machine.

DCR1000

Transport

The 3.5 inch floppy drive has made its way into almost 40 million PC's in the United States. It has proven itself as a rugged, reliable storage system. The low cost of the drives is tied to the tremendous volume of units being manufactured. The design is compact and self-contained, so that the drive is really a disposable item should it ever fail. Replacing a floppy drive involves only four mounting screws and a ribbon cable and takes just a few minutes.

The 3.5 inch floppy disk is a sturdy, jacketed media which comes in storage densities of 1, 2, 4, and 10MB. There is even talk of a standard for 20MB disks within the next year. The advantages of using this consumer based commodity is that the large demand drives pricing down and broadens distribution. In addition, trade organizations exist which establish standards for manufacturing and technology improvements. This offers many sources of disk drives that are interchangeable and upgradable.

It becomes clear that the 3.5 inch floppy drive system is an excellent choice on which to base a digital audio storage system. The next problem is how to fit enough digital audio onto a 3.5 inch floppy.

Digital Compression

Current professional digital audio systems use 16 bit samples. The rate at which the audio is sampled has an effect on the total amount of memory necessary to store a length of music. For example, 60 seconds of audio using 16 bit samples with a sampling rate of 44.1 kHz will require 5.1MB of memory. By reducing the sampling rate to 32 kHz, we can fit the same 60 seconds in 3.7MB. Reducing the sampling rate further continues to reduce the necessary storage space of the audio. However, by reducing the sampling rate, one also begins to reduce the audio bandwidth of the recording.

apt-X™100

By using an extensively researched and proven audio data compression system, the apt-X™100, we can compress the 16 bit PCM data to 4 bits. The system works on a coding process based on ADPCM (adaptive differential pulse code modulation). This coding process works without audible sound degradation and minimal time delay between the encode and decode reconstruction. Typically, the delay is 2.7ms at 44.1 kHz and 3.8ms at 32 kHz.

The three key parts of the apt-X™100 system, which allow reduction of data by 4 to 1, are sub-band coding, adaptive quantization and linear prediction.

The apt-X™100 system divides the audio spectrum into four sub-bands. Bands with higher energy levels receive more precise coding. The four band sub-coding maximizes the perceived quality according to the spectral response of the human ear.

In addition to band splitting, the system adapts the quantizing step size according to the energy of the input signal. The system continually analyzes the previous sample and permits the decoder to function without new gain information. The old sample data is constantly compared against the accuracy of the current sample and gain correction is applied in the form of an adaption multiplier. This process offers excellent tracking over a wide dynamic range and also maintains a constant signal-to-noise ratio.

The third part of the apt-X™100 process is the use of linear prediction. The sinusoidal nature of the majority of music and vocal sounds allows for a great deal of predictability. Here again, by analyzing recent data, an accurate prediction can be made of the next value and reduced coding can be used for signal representation.

The high speed and processing power of today's DSP chips make possible utilization the apt-X™100 compression algorithm in real time. By reducing the 16 bits of digital audio to 4 bits, we can now fit 60 seconds of audio into 1.27MB and .9MB at 44.1 kHz and 32 kHz sampling rates respectively, and without degradation to the sound quality.

Selectable Sampling Rates

By combining high density media and digital compression, we find a means of storing useful recording lengths on 3.5 inch floppies. As mentioned earlier, by reducing the sampling rate, storage time can be increased. Users can select from four sampling rates to increase the recording time of disk depending on their final application. The four sampling rates available are: 44.1 kHz, 32 kHz, 24 kHz, and 22 kHz. The user also selects recording in mono or stereo. By selecting the mono recording mode, the available time on a disk is again doubled. Figure 1 shows the available recording times with various sampling rates and media.

Master Player

The DCR1000 series consists of three models: a Master Player, Record Module, and Sub-Player. All units are 1/3 rack wide and 3 rack units high.

The Master Player contains the disk drive, a microprocessor board, a D to A board and power supply. The front panel of the player contains a 2 x 24 character LCD display which gives instructions, status, title information and a countdown clock. The front panel switches are the usual PLAY and STOP buttons with the addition of a CUE switch for accessing multiple cuts and re-cuing the disk. The cuing of a floppy disk is virtually instantaneous. Cuts can be previewed and then re-cued in one second. You no longer need to wait for a cart to fast forward.

The rear panel includes the power entry assembly, XLR analog outputs, an AES/EBU digital output, power supply interface connector, the DCR bus connector, RS232/422 port and the usual cart machine remote control interfacing using open collector outputs and contact closures to ground for input control.

Record Module

The Record Module is a separate unit which is connected to the Master Player via the DCR bus and receives power from it. The recorder contains an A to D board, LED audio metering, and sampling select buttons and status lights. The front panel function switches include RECORD, SEC, and TER. A START ON AUDIO switch allows recording to commence on the detection of audio. A variable audio threshold is provided to set the start on audio level.

A standard AT computer keyboard plugs into the Recorder Module for titling the disks and editing cue points. The keyboard also allows additional features and functions. The LCD display can show song title and artist, or the outcue of the cut being played. The display also contains an accurate countdown

Stereo Audio Storage Time

	Sampling Rate	22 kHz	24 kHz	32 kHz	44.1 kHz
	Audio Bandwidth	10 kHz	12 kHz	15 kHz	20 kHz
Media					
2MB		1:14	1:08	0:51	0:37
10MB		7:30	6:53	5:10	3:45
20MB		16:34	15:13	11:25	8:17

Mono Audio Storage Time

	Sampling Rate	22 kHz	24 kHz	32 kHz	44.1 kHz
	Audio Bandwidth	10 kHz	12 kHz	15 kHz	20 kHz
Media					
2MB		2:28	2:16	1:42	1:14
10MB		15:00	13:44	10:20	7:30
20MB		31:08	28:36	21:26	15:34

Figure 1

clock for each cut as it is played. The keyboard is not required for operation of the DCR. All functions and instructions are available through front panel switches and the LCD display.

The rear panel of the DCR Record Module contains XLR analog inputs and an AES/EBU digital input. The recorder is connected to the master player through the DCR bus and power supply cables. A keyboard port and a printer output allow labeling and titling the disks.

Formatting

Before it is first recorded, a blank disk must be formatted and labeled with the DCR header and format information. This format is different from the automatically by pressing the format switches on the machine's front panel. The format function can format a new disk or erase a previously recorded disk.

Sub-Player

The third unit in the DCR1000 set is a Sub-Player which contains a disk drive, an LCD display and front panel switches but shares the D to A board and power supply with the Master Player. This operation is similar a triple deck cart machine. This creates a lower cost unit but does not allow audio overlap between two decks. One Master Player can control up to three Sub-Players.

Production Capabilities

The DCR1000 emulates the operation of the NAB cart machine but goes a few steps further. The principle of inserting one-cart/one-song is maintained. Short commercial spot announcements can be recorded on the 2MB disks; music length material can be stored on the 10MB disks. The DCR permits a total of 16 cuts per disk. These cuts can be accessed manually by repeatedly pressing the CUE button. Additionally, the DCR can be configured to "remember" the last cut played in a rotation just like the NAB cart.

The header information placed on the disk during the recording operation contains all the data necessary for automatic playback. These instructions are read every time a disk is inserted. Kill dates may be stored on commercials to prevent playing them after expiration. A provision to manually override this feature is included.

Eight tertiary cues are allowed per cut and secondary cues are used as a true EOM (end of message). Cue placement and duration can be edited from the keyboard. The end preview function plays the last 5 seconds of the cut.

The looping function of carts for continuous background beds also can be accomplished by the DCR. A disk may be permanently labeled as a looping disk or manually set to loop when required. The editing feature makes looping a very precise way of splicing the beginning and ending of the loop on the disk in real time. This eliminates razor blades or assembly editing on a reel-to-reel before dubbing to cart.

Setup Options

By use of the keyboard port or front panel switches, a station can set up its own in-house DCR standards of operation which lock out button pushers and operator errors. For example, users may opt for the consistency of recording all disks in stereo at 32 kHz sampling by default. However, if a disk is intentionally recorded differently, the DCR automatically recognizes this from the header information and selects the appropriate playback mode.

Specifications

The advantages of digital audio are evident in the electronic measurements of the system. Figure 2 is a comparison of an analog cart machine versus the DCR.

	Cart Machine	DCR1000
S/N Ratio	-60dB	-90dB
THD	0.5%	.05%
Wow & Flutter	.12%	Unmeasurable
Crosstalk	-50dB	Unmeasurable

Figure 2

The digital unit offers significant improvement in the sound quality. Moreover, reliability is enhanced dramatically by the disk drive which provides a MTBF of greater than 30,000 hours and with disk life at greater than 3 million passes per track. The total system performance clearly takes the single play cart machine to a new level of quality.

CONCLUSION

The advantages of a digital audio system are quite clear. Current digital audio devices are designed as application specific units. They address a particular recording storage need and focus on the end user's requirements.

The DCR1000 addresses broadcasters' need for a single and multiple play on-the-air presentation. It maintains the operational simplicity of the analog tape cartridge machine. In addition to standard cart machine characteristics, new and improved features like editing, and instant end preview can be incorporated into the new digital design without complicating operation.

The size, shape and feel of a machine has an effect on how the user interfaces with it. By designing hardware in familiar packaging and following existing operational characteristics, the integration of new technology is an easier pill for operators to swallow. From the engineering perspective, the DCR offers lower maintenance, higher quality sound, interchangeability, and last but not least, lower cost of operation.

RF DESIGN CONSIDERATIONS IN THE DEVELOPMENT OF A HIGH-SPECTRAL EFFICIENT, MULTI-CHANNEL, ALL-DIGITAL STL

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ABSTRACT

An all digital approach to the design of an aural studio-transmitter-link (STL) for the 944-952 MHz band imposes several unique requirements on the radio frequency (RF) design of the transmitter and receiver portions of the system. The FM radio designs that have served the broadcasters' STL needs so well in the past are no longer adequate. Where high performance and spectrum efficiency are the primary needs only new approaches in transmitter and receiver RF designs will suffice for the digital-studio-transmitter- link (DSTL™).

This paper describes the RF technologies behind the design of the radio-frequency portion of the DSTL. These include the use of gallium-arsenic microwave monolithic integrated circuits (MMIC), power devices, dielectric and surface-acoustic wave (SAW) filters, multiple p-i-n diode arrays, and high dielectric-constant stripline circuits.

INTRODUCTION

The heart of the aural 950 MHz STL link for many years has been the tried and true analog FM radio. This classical approach to the broadcasters' needs, until recently, has served the industry well. Now with spectrum congestion in all the major markets there is talk of Category A antennas[1], compatible sharing of spectrum and segment-allocation schemes[2]. At the same time the radio listening audience has come to realize the benefits of digital audio performance and are now demanding the benefits of this technology in their radio entertainment.

One totally different approach to the congestion/performance problem is a spectral efficient radio. A spectral efficient radio could mean lower power for the same or higher system fade margin, lower occupied bandwidth or a combination of these two specifications. A DSTL radio with a 250 kHz occupied bandwidth could double the capacity of the current STL band while also improving both the signal-to-noise performance (figure 1) and the system fade margin (table1) by a considerable amount.

There are a variety of analog ways to build a spectrum-efficient radio with analog technology; single-side band (SSB), narrow-band frequency modulation (NBFM), etc. However, none of these approaches provides the necessary audio fidelity required by the marketplace. Then there are the digital modulation techniques such as pulse code modulation (PCM) that provide the high-fidelity performance, but are inefficient users of the available spectrum.

Recently digital audio coding technology[3,4] has provided a new tool for developing spectral-efficient modulation techniques. A family of digital audio encoders and decoders that allows high quality stereo

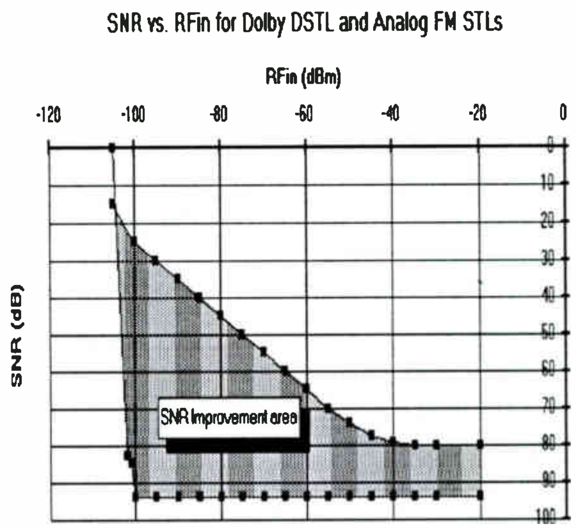

Fig. 1 DSTL SNR Improvement area.

audio to be transmitted efficiently through existing digital data channels over terrestrial, wired or wireless, or satellite links are available today.

When one of these digital audio coding technologies is combined with a judicious choice of digital modulation a new performance standard for STL service is created. Hence the DSTL. The benefits of the DSTL are:

- Wide audio bandwidth
- High signal-to-noise ratio
- No Crosstalk
- Degradation-free multiple hops
- Constant audio SNR during substantial fades
- Higher system gain (greater fade margin)
- Lack of background chatter
- No phase distortion
- Encryption against pirates

To successfully implement the DSTL concept into a workable approach it is first necessary to develop the desired carrier signal, perform the modulation and amplify the result to a usable signal level. The first two functions are performed in the DSTL's exciter module and the latter in the power amplifier module.

The key to maintaining the spectral-efficiency of the audio coding technology and modulation format is processing those signals through highly linear stages. If during any of the RF modulation, up-conversion or amplification processes the signal experiences any form of odd-order non-linearity it will corrupt the occupied bandwidth.

Figure 2 illustrates the three technologies that are integrated into the DSTL system.

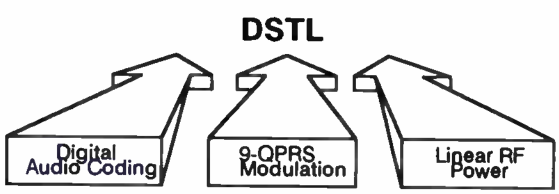

Fig. 2 DSTL Technology Base

RF DESIGN CONSIDERATIONS

Transmitter Design

The DSTL transmitter, as shown in Figure 3, is comprised of an A/D, DSP, Digital Modulator, Exciter, Power Amplifier and Power Supply modules. The A/D module provides the multi-channel analog-to-digital conversion. The DSP board contains the Dolby AC-2 data compression technology. The Modulator board performs the digital 9-QPRS encoding. The Exciter board performs the RF frequency synthesis and RF modulation functions. The Power Amplifier performs the amplification and power control functions. Special consideration is given to the linear power amplification required in an all-digital radio system.

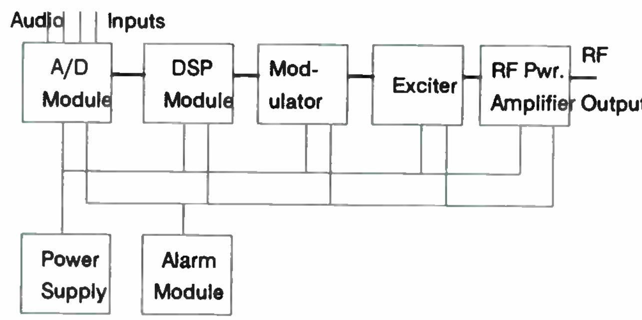

Fig.3 DSTL transmitter Block diagram.

In addition, the transmitter has an Alarm Module which monitors the status of all of the other modules and the power supply.

Spectrum Efficiency and Output Power Capability

Proposed spectrum mask requirements would need a 250 kHz occupied bandwidth. For this requirement Dolby chose to utilize AC-2 encode technology combined with 9-QPRS modulation which provides a spectral efficiency of 2 bits/s/Hz.

Estimated receiver performance and path length calculations determined that, for most applications, only a 1 Watt output would be necessary to provide adequate fade margins.

In order to maintain the spectral efficiency of the modulation it was further determined that the power output stage would have to have a third-order intermodulation intercept point of +60dBm[5]! Linear microwave power takes on new meaning in the DSTL radio.

Exciter Design

The various blocks in the exciter module are shown in figure 4.

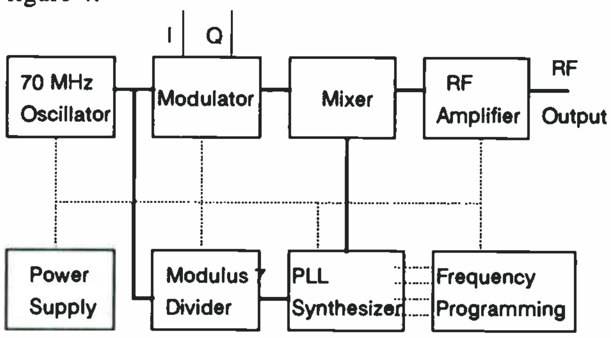

Fig.4 DSTL exciter module block diagram.

Master Oscillator
A temperature stabilized 70 MHz oscillator is used as the source for the reference signal for the frequency synthesizer and the carrier for the modulation. The reference signal is processed by a fixed modulus-7 divider to provide a 10MHz signal for the phase-locked-loop (PLL) integrated circuit (IC). The other 70 MHz signal is modulated by the I and Q channels from the modem and then up converted to the 944-952 MHz band in a later part of the exciter.

Field-programmable Frequency Synthesizer
If this radio is to be used in an area that plans to take advantage of its two-for-one spectral-efficient capability it must have the flexibility of being frequency agile. A field-programmable frequency synthesizer approach gives the DSTL transmitter and receiver frequency agility in order to manage circumstantial frequency relocation.

The CMOS PLL IC has an internal reference divider that is programmed to divide the 10 MHz input signal, from the modulus-7 divider by another factor of 400. This division, when used with a dual-modulus prescaler, allows the DSTL to be frequency programmed in 25 kHz steps anywhere in the 944-952 MHz U.S. STL band. User programming is via switches located on the front panel of the frequency synthesizer.

The emitter-coupled logic (ECL) dual-modulus, 128/129 prescaler, translates the 1014-1022 MHz voltage controlled oscillator (VCO) signal into a lower frequency signal that the CMOS PLL integrated circuit can handle.

Audio Source Coding Technology
Dolby AC-2 coder technology provides the two high-quality 15 kHz channels, a 7 kHz auxiliary and a 3 kHz voice/modem channel to the DSP module. In this module 16-bit, 44.1 ksamples/sec bit-rate reduction is achieved. By using a low time delay implementation of Dolby AC-2 data compression, less than 250 kHz of STL bandwidth is required with approximately 8 msec. time delay in the main audio channels.

9-QPRS modulation
Spectrum efficiency is achieved by the use of QPRS signaling, at 70 MHz, in conjunction with a system cosine filter that results in partial response signaling as disclosed by Todd[6]

Up-conversion
The 9-QPRS modulated 70 MHz signal is up-converted by a high-side injection, passive, double balanced mixer. This mixer and its drive level were chosen to produce the lowest possible third-order intermodulation distortion. Broadband resistive terminations are used on all ports of the mixer to properly terminate the image and spurious frequencies.

Dielectric Filters
In order to filter out the undesired frequency products from the up-converter, a ceramic-block dielectric filter is used. The filters operate, in theory, similar to microwave quarter-wavelength interdigital-line comb filters constructed from round rods. These, however, are constructed out of a material that has a very high dielectric constant (Er range from 20-100) and low loss tangent. It is not uncommon for these structures to have unloaded Q's greater than 5,000.

Unlike the machined metal construction of older microwave filters the dielectric is machined and then plated. This results in a very cost-effective, physically small (7 mm x 9 mm x 27 mm) filter with less than 2 dB insertion loss.

Two and four-pole versions of these filters are used in both the DSTL exciter and the receiver designs.

RF Power Amplifier Design
The heart of the power amplifier is a stripline sub-assembly. This sub-assembly exhibits over 50 dB of linear power gain. Since amplifier load mis-matches can impact the IM performance of the last stage an

isolator is an integral part of the sub-assembly. Following the isolator is a six-port directional coupler. Part of this structure is used as a 30 dB monitor port for the transmitter. Another part of the six-port coupler is used to allow for customer setting of the output power and to maintain the power level constant over temperature and aging. An ALC loop has been designed into the power amplifier. An output filter keeps the harmonic content of the transmitter greater than 70 dB below the nominal 1 Watt output signal.

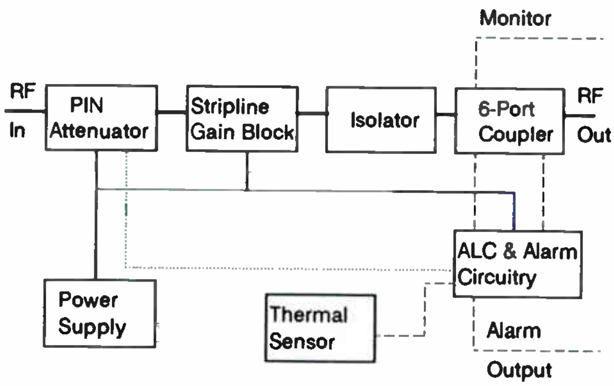

Fig. 5 DSTL power amplifier module block diagram.

Third-order Intercept Requirements

As mentioned earlier, the 9-QPRS modulation format requires a hyper-linear power amplifier to preserve the spectrum-efficiency. To ensure that the amplifier intermodulation products are greater than -60 dBc the third order intercept performance must be close to +60 dBm.

There were three different amplifier design approaches that were investigated: "Pre-distortion", "Feed-forward" and "Back-off". Pre-distortion did not lend itself to the need for long term stability over time and temperature. Although significant improvements in distortion cancellation have been reported and achieved using feed-forward techniques, a design using this approach also exhibited time and temperature effects that were difficult to control. In addition the feed-forward technique requires an additional side-chain amplifier and had high component and labor costs.

Upon first investigation the back-off approach would appear to be too costly because of the amount of raw input power and thermal dissipation required. However, further investigation indicates that if hyper-linear devices are used in the amplification stages the input power and heat problems are manageable.

GaAs MESFET Design

Due to their many non-linear mechanisms[7] bi-polar silicon devices clearly do not have the linearity required for DSTL applications. Prior to the availability of microwave Gallium-Arsenic (GaAs) metal-gate field-effect (MESFET) power transistors, high intercept amplifiers had to be constructed using feed-forward or pre-distortion techniques. As mentioned, both of these techniques have gain and phase stability problems that result in complex and costly support circuitry.

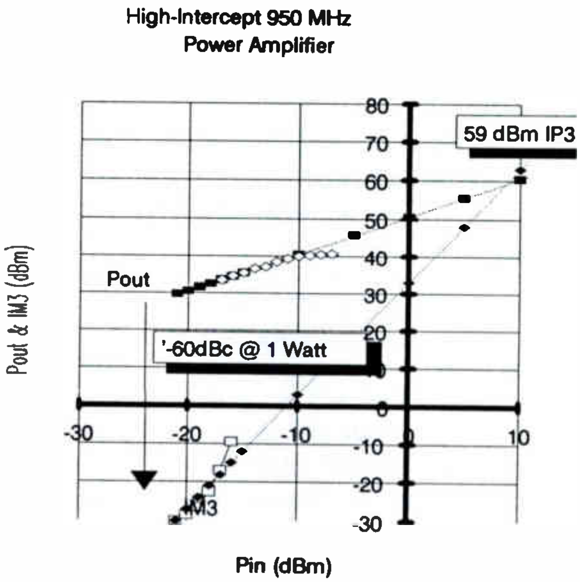

Fig. 6 Measured DSTL Third-order Intercept

Today, linear Class-A GaAs FETs are available with 20-Watt performance up and into the C-band frequency region.

Because of their inherent higher gain compared to available bi-polar transistors only three stages of power amplification are required. Figure 7 illustrates the gain distribution that is realized by each device. The gain numbers reflect the intrinsic forward gain of the device and the gain realized by providing the optimum input, interstage and output impedance match.

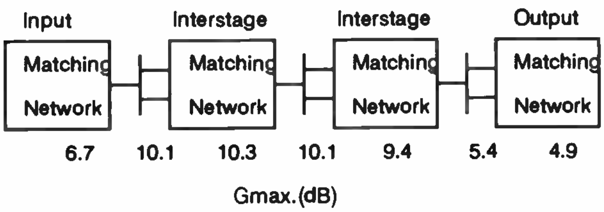

Fig. 7 DSTL power amplifier gain distribution.

Impedance Matching Structures
To provide the device impedance transformation, DC block function and some degree of filtering a unique tri-plate, broadside-coupled quarter-wavelength stripline resonator structure was developed. These structures were optimized for bandwidth and manufacturing yield by the use of Touchstone software.[8]

Ceramic Loaded Teflon Substrate
Conventional microstripline design techniques were avoided because of their large physical size and spurious radiation problems. The final power amplifier design was realized by using totally enclosed three-layer stripline. The stripline structures were realized using high-dielectric constant, e.g. 10.5, microwave, ceramic loaded Teflon, material.

The above mentioned material and stripline structure combine to provide the advantages of being physically small, wideband and capable of handling high power. This amplifier is constructed without the need for expensive discrete blocking capacitors or spring wound coils. The structure is thermally stable and provides its own EMI shielding.

Monitor, Filter and Control Circuitry
The power amplifier is activated by an OPERATE signal from the transmitter front panel. This operate signal is processed by a logic circuit which monitors all of the module's operating voltages. This circuitry prevents the power amplifier module from being activated in the event of the loss of any power source which might damage the devices in the power amplifier sub-assembly.

Part of this control circuitry also monitors the power amplifier sub-assembly temperature. The circuitry is designed to disable the power amplifier in the event that there is an over temperature condition.

Samples of the forward and reverse output signals are coupled via part of the six-port directional coupler to biased Schottky-barrier diode detectors. Signal processing circuitry monitors the detected forward power signal and adjusts the input p-i-n attenuator to keep the output forward power level constant.

The detected reverse power signal is compared to the forward power signal to determine if high VSWR conditions exist. When the VSWR exceeds 3:1 the circuitry initiates an alarm signal which in turn activates the alarm LED on the module and a summary alarm LED on the transmitter's front panel.

Performance
The spectrum mask of the completed amplifier at the 1 Watt output level is shown in the figure below.

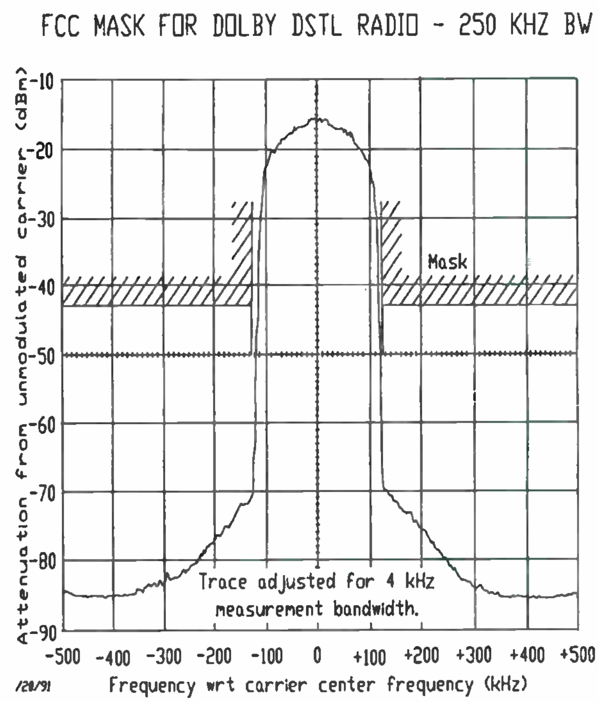

Fig. 8 Occupied spectrum mask of DSTL

All IM products are well below the unmodulated carrier by 60dB. Since the power amplifier is so linear, harmonics of the unit are greater than -70 dBc. Very little output filtering is required for normal amplification.

Receiver Design
Description - The receiver is comprised of a

Receiver/Synthesizer, Modem, DSP, D/A, Alarm and Power Supply Modules. A digitally synthesized FM Stereo Baseband Generator module is also available.

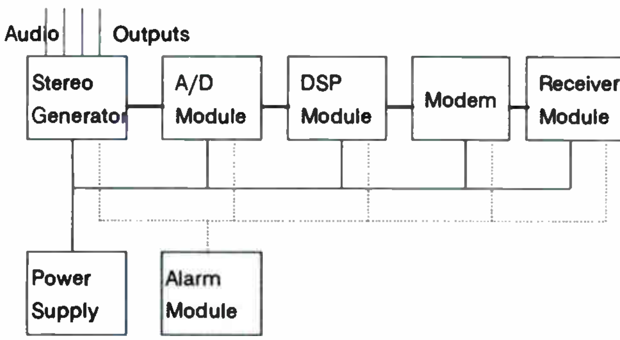

Fig. 9 Receiver block diagram.

Front-end Design

Preselector considerations and design

In many metropolitan areas the most common STL receiver locations are fraught with a variety of undesired high level signals. In addition to multiple STL signals, there are quite often pocket pager, mobile and other services just below and above of the 944-952 MHz STL band. Often, these signals are too close to the STL band or too high in amplitude for the receiver front-end to handle.

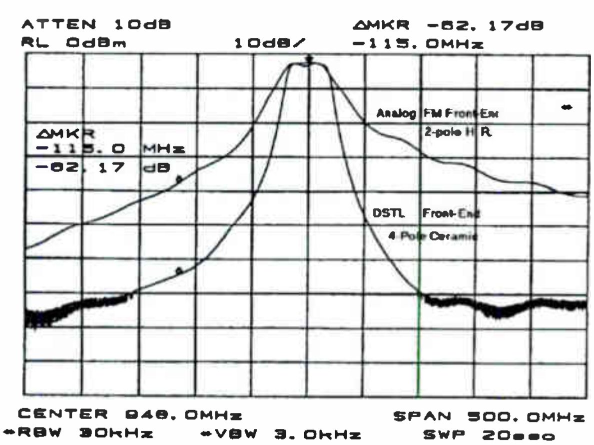

Fig. 10 Receiver Selectivity Curves

The above figure demonstrates the receiver front-end selectivity of a typical analog FM STL receiver and the DSTL receiver. As can be seen from the curves even a high amount of selectivity will still allow some out-of-band energy to get through to the RF pre-amplifier stage. The only way to prevent this stage from producing IM is to select amplifier and mixer stages with moderate gain and high third-order intercept capability. Even then it will be necessary to balance the gain distribution with the right amount of AGC action.

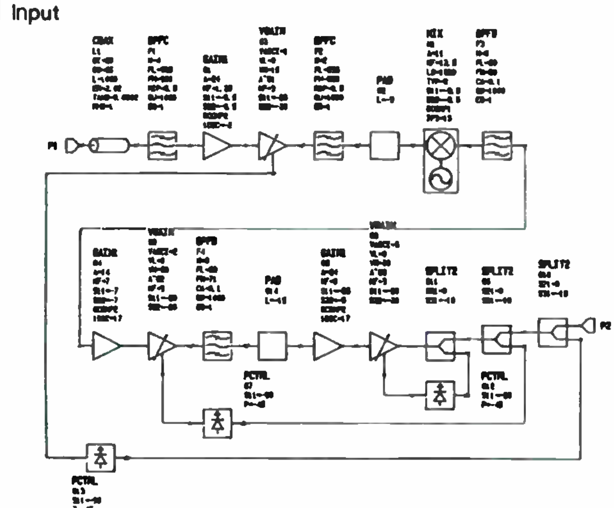

Fig. 11 DSTL receiver AGC block diagram

Third-Order Intercept Requirements

In the DSTL receiver the RF pre-amplifier device is a GaAs MMIC which was chosen to have a good noise figure and high third-order intercept to prevent the creation of IM products. The majority of the front-end selectivity follows this stage where the filter's insertion loss has lower impact on the noise figure. Since no active mixers could be located with high third-order intercepts, i.e. greater than 18 dBm, a passive mixer was chosen to perform the frequency conversion function.

IF Design

Third-order intercept requirements

Like the transmitter chain, the receiver must be highly linear. Critical to over all performance of the DSTL system is low IM distortion. Each stage of the 90 dB gain IF chain has been designed to maximize it's third-order IM intercept-point and control the amount of gain.

Surface-Acoustic-Wave Filtering

IF selectivity is provided by a surface-acoustic-wave filter. This filter has a 3dB bandwidth of 1MHz and exhibits 60dB of alternate channel attenuation. Its small physical size and good temperature stability make it an ideal choice for this application.

AGC Considerations and Implementation
Another key part of maintaining the IM distortion in the receiver chain is to provide the correct amount of interstage AGC action. Improper front-end AGC action can result in either front-end IM or degradation in the receivers noise figure. Improper AGC action in the IF section can result in undesirable clipping resulting in a high bit-error-rate.

The figure below demonstrates the DSTL receiver's AGC characteristic over a -120 to -20 dBm range of input signals. The figure indicates that the RF signal level in each stage is controlled to keep the generation of IM as low as possible.

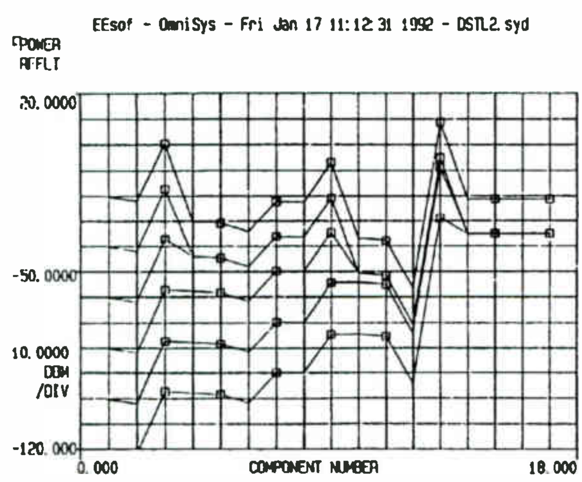

Fig. 12 AGC budget analysis for various input levels.

MECHANICAL DESIGN CONSIDERATIONS

Unique Requirements

Need for Modularity
Rapid technological changes can make a product obsolete if it has no ability to be up-graded. With the advances in audio, digital signal processing power, direct digital frequency synthesis it was felt that a modular approach would provide the user with flexibility over his years of ownership. In addition, the modular approach provides service and exchange benefits should they ever be needed.

Thermal Design Aspects

From the RF electronic packaging point of view the most difficult problem is thermal management. The power amplifier module dissipates close to 30-Watts of heat in its normal mode of operation. Most RF amplifier design engineers are used to junction temperature values of 200°C in bi-polar devices. In order to meet the DSTL's desired MTBF performance the channel temperature of the GaAs power FETs must be keep well below their 175°C maximum operating point. Data provided by the device manufacturer indicated that the typical thermal resistance is about 70% of the given maximum values. From these device ratings and the amplifier's operating efficiency the heat sink and thermal interface requirements were determined.

The resulting heat sink design runs the full length of the power amplifier module and results in a device MTBF of over 750,000 hours at +70°C ambient.

ELECTROMAGNETIC-MAGNETIC INTERFERENCE CONSIDERATIONS

A product design that takes into account EMI considerations from conception will have fewer of those problems in the product launch cycle and over its operating life. Interference and susceptibility from either the RF, digital circuitry or external environmental fields have the potential of reducing a product's performance. To preclude this possibility, all of the modules developed for the DSTL contain EMI suppression. All entry and exit lines of the RF modules contain RF filtering and are shielded.

SUMMARY

It was once thought that all digital radios required more bandwidth than analog radios. As shown, when the optimum audio coding technology is combined with the proper choice of digital modulation and RF technology this is no longer true. The multi-channel DSTL radios will now start to replace their less spectrum efficient analog predecessors.

These new technologies will also have an impact on other future high-spectral efficient digital radios.

ACKNOWLEDGMENTS

The author wishes to acknowledge the efforts of Edmond Chu in the fabrication and testing of the RF circuitry.

Table 1

Sample Path Length and Fade Margin Comparisons					
		DSTL	Analog FM		Notes:
LOSS					
	Path	-122	-122	dB	20 mi. / 32.2 km.@ 950MHz
	Transmission Line	-5.6	-5.6	dB	400 ft. / 122m. (7/8" foam)
	Connectors	-4	-4	dB	Total
	Others			dB	
TOTAL SYSTEM LOSSES		-131.6	-131.6	dB	
GAIN					
	Transmitter Power	30	35	dBm	DSTL 1W. / FM 7 W.
	Transmit Antenna	15	15	dB	6 ft. grid parabolic
	Receive Antenna	15	15	dB	6 ft. grid parabolic
	Others			dB	
TOTAL SYSTEM GAINS		60	65	dBm	
TOTAL SYSTEM LOSSES		-131.6	-131.6	dB	
TOTAL SYSTEM GAINS		60	65	dBm	
Received Signal Strength		-71.6	-66.6	dBm	
Desired signal Level		-95	-66.9	dBm	
Fade Margin		23.4	0.3	dB	DSTL @10^-4 BER. FM 100 uV for 70 dB SNR

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DIGITAL AUDIO INTERFACE STANDARDS

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ABSTRACT:

In recent years there has been an accelerating transition from analog broadcast audio products to a new generation labeled as "digital." Already many of these products-- including audio storage, audio editors, audio processing, stereo generators, RF modulators, CD players/recorders, DAT equipment, STLs, and more-- have been developed. Many already are proving themselves in the field.

Broadcasters most frequently cite cost savings, improved as well as consistent quality, and greater flexibility as reasons for relatively quick acceptance and adoption of these digital products. In fact, the broadcast marketplace itself is rapidly resolving engineering arguments about whether analog or digital technology is better.

Nevertheless, today's digital products share a common characteristic. Virtually all of them have analog (not digital) input and/or output. This has been necessary since the vast majority of broadcast hardware has been analog. However, the rapid proliferation of digital replacement products has made the need for a digital interface audio standard apparent.

To this end, a group of volunteers representing equipment manufacturers met in San Francisco during Radio '91, the fall show of the National Association of Broadcasters to begin developing a digital audio interface standard. This paper will report on group's activities and progress to date.

GROUP GUIDELINES

Before beginning work on a standard, the group adopted the following guidelines:

First, its purpose is to develop a digital interface standard (or more than one) which meets the needs of the broadcast community.

Second, the standard(s) will be voluntary, not mandatory.

Third, no company or organization will directly benefit financially from the standard(s).

Fourth, the group agreed to:

--seek a compromise in finding standard(s), with the understanding that no perfect solution likely exists.

--recognize that speed of decision-making will minimize problems in the future.

--do nothing if nothing is needed.

MAJOR CONSIDERATIONS

Area of Interface To Be Addressed: When developing a digital (or any) product for the broadcast environment, equipment manufacturers must consider two levels of interface: Those interfaces internal to the equipment and those external to the equipment.

Traditionally, internal interfaces have fallen within the domain of each equipment manufacturer. In digital product development, internal interfaces involve such decisions as whether to use linear or compressed data; serial or parallel processing, or many other unique schemes which will define a particular product. Ultimately these choices contribute beneficially to a selection of differentiated products being offered to the market. For this reason, the group decided to limit its focus to external interfaces.

External interfaces are the inputs and outputs (I/Os) which allow a product to interface with other equipment. Because current digital "black boxes" have had to interface with analog equipment which has dominated the broadcast facility, they have used analog input(s) and output(s). Thus, the incoming analog signal is converted to the black box's internal digital format at the input, and then re-converted to an analog signal at the output. Each conversion from analog to digital (A to D) and digital to analog (D to A) results in a degeneration of signal caused by round-off errors; quantization errors, or other factors. By keeping data in a digital format which is passed digitally from equipment to equipment, these errors can be virtually eliminated.

Quality: Clearly, any digital interface standard must preserve signal quality-- but what is quality? Information in a broadcast facility can have a 10:1 range of bandwidth depending on service type (see Table 1). Quality requirements will vary and also depend on service type. Existing broadcast facilities have differing levels of quality which typically are quantified by signal to noise; distortion; frequency response, and channel separation.

Generally, two levels of thought prevail concerning the ultimate level of sound reproduction or quality. The first level recognizes that since the last part of the audio chain is the human ear, limitations of human hearing create a reasonable boundary for technical specifications. The second level believes that such a parameter is far too loose or too tight, depending upon the application. For example, AM radio performance in an automobile may measure a 50 dB signal to noise; FM tuner performance may exceed 80 dB signal to noise, and the human ear exhibits over a 100 dB dynamic range.

An understanding of the ability to hear audio defects is a separate, but nevertheless significant aspect of performance limit setting. The group recognizes that in-depth research into the audibility of defects is on-going and may re-write previously accepted limits.

Compression Algorithms: Because the goal of an external digital interface standard must be to preserve signal quality, the group has carefully considered use of compression algorithms.

In the analog and digital domains, compression algorithms have demonstrated their ability to discard certain audio information while retaining data essential to listening. Various algorithms have been based on psychoacoustic models of human hearing. Listening tests by experts have confirmed the quality of these algorithms is quite high. Digital data compression of 4:1, 6:1 and 8:1 is routinely used.

What these algorithms can do is demonstrated by assuming an audio signal to 20 kHz linearly sampled at 44.1 kHz to 16 bit accuracy. The data rate is:

44.1 kHz x 16 bits = 705.6 kbits/s.

Using an 8:1 compression algorithm, information is used at 88.2 kbits/s instead of at 705.6 kbits/s. This decreases requirements for computer disc space for audio files; reduces RF bandwidth requirements, and permits simpler, slower hardware. Digitally compressed audio is them expanded to the full range of the original signal, missing only the audio to which the human ear is least sensitive.

However, these algorithms are not without limitations. Multiple digital compressions and expansions reportedly result in audible defects. As a result, caution must be exercised in repeated use of digital data compression and expansion in the on-air chain. Certain equipment manufacturers already have opted to use internal linear digital techniques with no digital compression in their handling of audio to minimize risk of audible defects.

At this writing, it appears that repeated use of compression algorithms in an external digital interface standard will be avoided as well. Either information will be passed linearally, or it will be passed through the chain in a compressed state to avoid audible defects from repeated compression and expansion.

COMMON INFORMATION TYPES USED IN BROADCASTING

Before deciding what external standard(s) would be needed, the group identified information types currently used for most common broadcast purposes (see Table 1). Spectra has been defined by FCC requirements as well as practical considerations of today's equipment:

Table 1: Broadcast Information Types

Description	Information	Frequency
Cue Tones	Start/stop	media-dependent
Data	Headers, timing, etc.	relatively slow
AM Mono	Single channel audio	30 Hz-10 kHz
AM Stereo	Two channels audio	2 x 30 Hz-10
AM Composite	Not used	NA
FM Stereo	Two channels audio	2 x 30 Hz-15
FM SCA	Single channel audio	50 Hz-5 kHz (typical)
FM SCA	Data	5000 baud (typical)
FM Composite	Stereo and SCAs	30 Hz-100 kHz

Based on frequency requirements, examination of broadcast information types indicates that two digital audio interface standards most likely will be required. While FM composite baseband information is popular and practical, it requires up to 100 kHz which is generally not encompassed by today's digital audio standards.

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At this point, the group is focusing on standards for each of the two types of information handling. One will address studio (primarily audio, encompassing information described as cue tones, data, AM mono, AM stereo, FM stereo and FM SCA) and the second will focus on STL/transmitter (audio/FM composite baseband) requirements. Conceivably the studio (audio) standard and the STL/transmitter (audio/composite baseband) standard would be transformed by a link.

Further discussion of studio and STL/transmitter requirements follows:

STUDIO (AUDIO) INTERFACE

Studio information which is primarily audio. It includes monoaural used by many AM stations; stereo audio used by virtually all FM stations as well as a growing number of AM stations, and FM SCA. It also encompasses other data useful in audio processing:

- o Header information including a description of the digital file. This may be quite detailed.
- o Timing information.
- o Compression/expansion information.
- o Sample rate.
- o Error correction.

One of the top contenders for a studio interface standard is the AES (Audio Engineering Society) format for digital audio interface. This standard is based on an audio sample rate of 48 kHz $\pm$12.5% which includes 44.1 kHz. Data is transported serially, and the transmission format is packet based high level data link control communications protocol. Provisions also are included for other ancillary data which may or may not be time-related to the audio signal.

STL/TRANSMITTER (COMPOSITE BASEBAND) INTERFACE

The STL/Transmitter interface standard will focus on studio audio which is sent to the transmitter site as composite FM. Indeed, this has become most popular given the physical location of today's common integrated audio processor/stereo generator at the studio.

The stereo generator accepts the L/R inputs and creates the stereo multiplex signal. This stereo multiplex signal then is combined with subcarriers (called SCAs) to form the FM composite baseband signal. The composite baseband signal covers from 30 Hz to 100 kHz. This 100 kHz bandwidth is far wider than the 15 kHz normally considered adequate for FM audio. Since FM composite baseband is not just an audio signal, techniques such as compression no longer are applicable.

The stereo generator may be located at the studio, connected to the STL transmitter, and the SCA(s) summed at the transmitter site, or the SCA(s) may be summed before the STL transmitter and sent as composite. The FM composite baseband is also processed in clipper/limiters to enhance loudness and prevent overshoot.

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Many stations still operate with the stereo generator at the transmitter site. In this case, L/R is transported to the transmitter site, coupled into the stereo generator, and summed with the SCA(s).

At last, the FM composite baseband is used to frequency-modulate (FM) the FM exciter. Many different variations of this scheme of products are used to accomplish the same function. Because of the importance of FM composite baseband techniques, digital standards need to be addressed.

CONCLUSION

There has been a recent proliferation and acceptance of digital audio equipment in broadcast facilities. That equipment will best serve the broadcast industry if it can be connected in a simple electrical and mechanical manner. Setting a voluntary digital audio interface standard for broadcast equipment manufacturers will permit them to design compatible products that meet broadcasters' expectations.

DIGITAL AUDIO PRODUCTION IN THE CBC: PAST, PRESENT, AND FUTURE

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Abstract: Digital audio technology was applied piecemeal to program sound production in the CBC several years ago. This resulted in the construction of several fully digital production suites. A review of these facilities, and the "hands on" experience gained by our producers during daily operation indicates very satisfactory results, but also raises many questions about their integration with an increasingly digital world. These questions will be examined in the light of recent technology developments.

INTRODUCTION

The CBC decided two years ago to build several digital audio suites. It was obvious that digital techniques would be applied to sound production, and that there was a lot to learn about how to implement and apply the technology. The studio installations have answered many of the original questions, but have raised others.

Studio H in Toronto was the first digital Radio studio. It was built essentially from separate components, and provided some challenging system integration and synchronization problems. The TV sound post production suite (Montreal, Studio 76) and the most recent radio drama studio (Montreal, Studio 14) have both used more completely integrated systems to avoid this.

All three studios are built around a hard disk recorder and a digital console. The random access and non destructive editing capabilities of the recorder have changed production techniques enormously, and for the better. These new techniques can be applied to both radio and TV sound production. Any significant differences will be mentioned in the applicable section.

Production is only the first step in radio programming. Finished programs have to be distributed across the network and transmitted to the listener. The combination of digital audio rate reduction encoder/decoders (codecs) and the common carriers' digital network facilities promise to lower program distribution costs significantly, as well as improving the signal quality. Digital transmission (or to use the internationally recognized term, emission) facilities that are being developed and tested now will eliminate virtually all of the existing signal degradations, and support many new listener services.

THE PRODUCTION IMPLICATIONS

Sequential media

The tools: It is worth reviewing tape based production techniques to appreciate the enormity of the changes that random access storage has brought to sound program production.

Tape (even if it is digital tape) is sequential. Because the playback head has no choice other than to follow the length of the tape, edits made on 1/4 inch tape are destructive. The source material on the "unwanted" side of the edit point is no longer accessible once the desired portion has been spliced into, or transferred to the final piece of tape. If for some reason, an edit has to be redone, there is a practical physical limit to the amount of material that can be reclaimed or trimmed. Electronic (assembly) splicing permits previewing the edit, and provides lots of resolution in selecting the edit point, but cannot overcome the destructive characteristics of edits based on a sequential storage medium. This really means that once the edit has been made, it is impractical to change it. Even if the original material is still available, changing one edit point almost always requires changing at least one other edit, and often several others.

This is true of multitrack tape as well. Once the material has been transferred to the multitrack tape, the timing relationship among the segments is fixed; shifting the out point of one segment will shift the in points of all the related segments. Figure 1 shows an edit point that has been changed to "stretch" the original segment. This clearly is only possible if there is enough material available, and requires that the "in" point of the corresponding track be moved. If the "in" point of this track can't change, the only solution is to re-lay the original segment to the multitrack tape. All this can be done, but it is slow enough and requires enough effort to discourage producers from "trying it this way" to see if the result is any better.

Production Techniques: Tape based production of a documentary, for example, begins in a "production office" with stacks of pre-recorded material and the program plan. The desired material is located, and if there are no transitions or level adjustments necessary, segments are edited and spliced on the spot. If some processing is required, marked segments are assembled into different reels with the appropriate processing for each cue noted. For more elaborate productions that require music and effects elements, the segments are selected, cut roughly to length, and prepared for transfer to multitrack tape.

Production then moves to the studio where the transitions are tried, the timings adjusted as required, and all the elements balanced for the final mix. The whole process is essentially linear, with each segment completed and recorded on the final master tape before the next is started. The tape

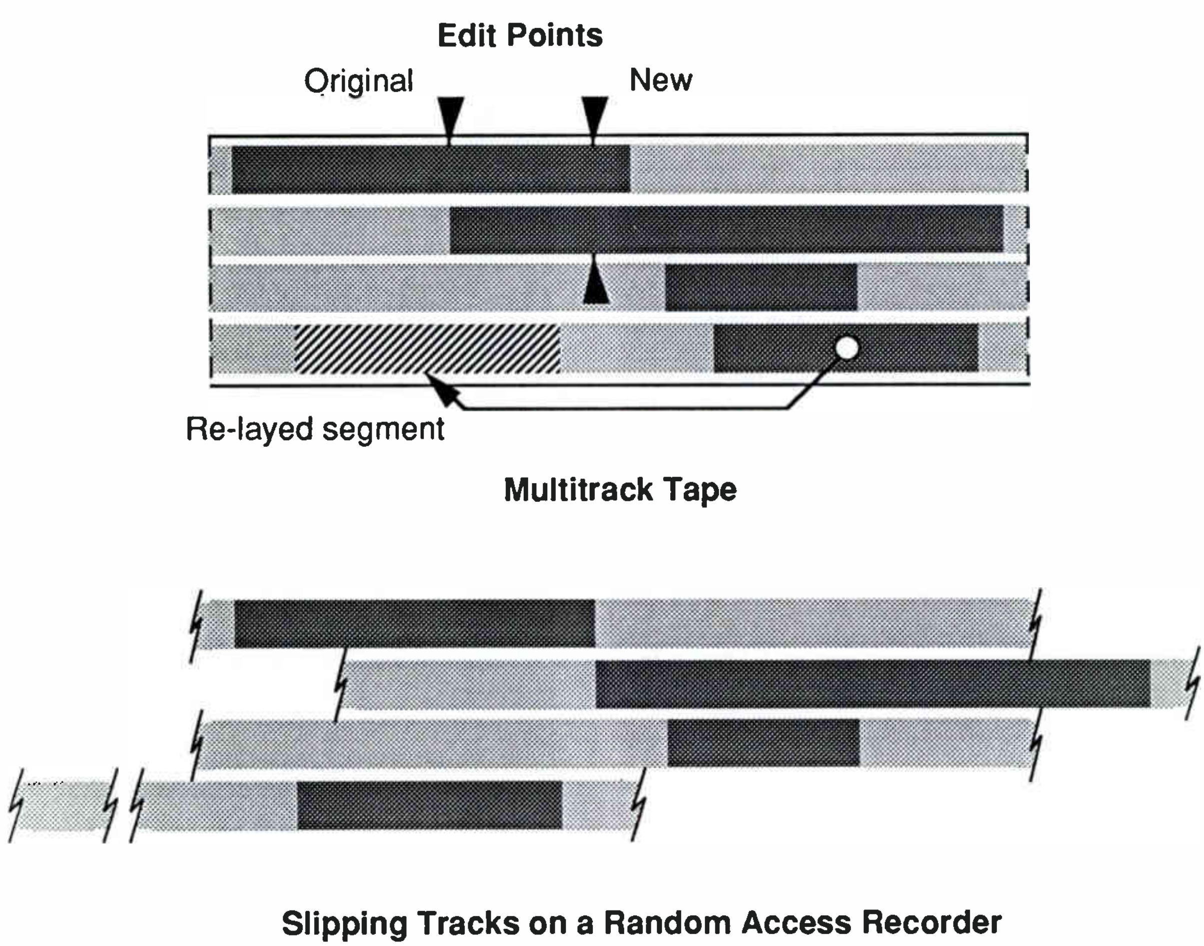

Figure 1: Changing the edit point, segment duration and segment location on multitrack tape contrasted with the same operations on a disk recorder

medium pretty much demands this method of operation, because the relative timing of each element is determined by its physical position on the tape.

Random Access media

The Tools: Digitized sound is a sequence of numerical data, the same as any other data file, so can be recorded on a hard disk or any other random access device having sufficient capacity. The structure created by the sampling process allows precise and repeatable retrieval of the sound. Editing thus becomes a matter of retrieving the desired portion of the sound file at the desired time, rather than discarding an unwanted section of tape. Edit points do nothing more than define a set of data locations within a sound file, so changing them does not destroy any of the original sound information.

The combination of random access storage and non destructive edits makes it possible to "slip" tracks freely in time, and to change edits at will. This forms a very flexible and fast production tool, particularly when automated. Fortunately, digital recording and signal processing tools lend themselves particularly well to automation, so all three tools are evolving rapidly.

The basic technical problem with "what if" editing and digital signal processing is that the audio samples have to be identified, and all the signals in the system must be coincident in time. This requires some method of labeling individual samples and of synchronizing the signal sources to a common clock.

The requirement for sample and edit point identification is obvious. Even though the 48 kHz sampling rate provides very high edit point resolution, and the AES/EBU data format carries a sample identification code in the "channel status" ... stream, none of the equipment in early ...llations provided the facilities to use this ...nation.

...of the equipment addressed the problem of ...g the output signal to a common clock. The ...nits that did provided an input for some sort of reference signal, to which they locked their internal clock. One supplier solved the problem by providing enough buffering at the signal inputs to realign the signals before any internal signal processing was done. It proved possible to modify some of the remaining equipment in Studio H to accept an external reference signal. Outputs from equipment that couldn't be synchronized to the reference had to be run through a sample rate converter or an analog to digital converter that was locked to the reference clock. The word clock signal (a 48 kHz square wave) turned out to be the most convenient reference signal in the Studio H installation.

The problem was still not solved because of various processing delays and the lack of a common relationship between the reference word clock phase and the phase of the output signals. In order to align (or in TV jargon, to phase) all the output signals, a word clock splitter with phase controls for the individual feeds to each device had to be built. The whole system had to be re-timed whenever the signal path was changed. This is clearly not the way to run a studio.

The situation is improving, but slowly, and apparently in stages. Many of the new so-called professional digital audio devices appearing use the video "colour black" signal as the common reference signal. This makes it possible to build digital audio studios around a common "sync" signal, the same way video studios have been built for years. It is, however, an incomplete solution that could cause a false sense of security. The relationship between the reference signal phase and the the output signal phase still changes from manufacturer to manufacturer. The relationships between the various video clock rates and the three common audio sampling rates are complex. Unless some common derivation scheme is adopted, small differences in sampling rates that will cause occasional sample slips between signal sources that are locked to the same reference may be built into the system.

The use of a video reference makes it easy to use video time code to label the audio edit points. This provides adequate (sub-frame) resolution for "lip-syncing" audio to video, but without some as-yet-unagreed-upon extensions, SMPTE or EBU time code cannot provide audio sample level resolution. Considering that audio and video (post) production are essentially separate processes, and the difficulties of using a video reference with conventional time code for digital audio, it seems logical to adopt a more appropriate audio reference signal and method of identifying audio samples. There would, of course, be a well established relationship to all the video standards.

Quite a bit of progress toward this goal has been made within the Audio Engineering Society. They have written AES11-1991, a standard for the "Synchronization of Digital Audio Equipment in Studio Operations", and have several proposals under consideration that will solve many of the problems that CBC has experienced in digital audio production. The question now seems to be to make the equipment suppliers and system designers aware of the problems, and of the progress toward solving them.

Production Techniques: The advantages of random access recorders can be illustrated by looking at the example of the documentary production again. The process starts in the production office as usual, but this time no attempt is made to locate edit points precisely, or even to do any of the simple edits. The only task is to locate material for transfer to the disk recorder. The material has to be trimmed, but only approximately, so that not too much disk capacity will be wasted on material that won't be used. This simplifies the "pre-production" work enormously, reducing the time required by at least two thirds.

The bulk of the work is done in the production suite, where the selected segments are labeled during transfer to the disk recorder. The labels automatically stay with the (segments of) material during the whole process, so reference to them is more intuitive and relevant to the production.

The recorder's random access and non destructive edit capability make it possible to preview and change edit points and transitions between segments virtually in real time. An automation system integrated with both the recorder and console can store the details of all the trials, then invoke the selected versions during final program assembly. A really effective automation system makes it possible to work on the individual elements of the program, then sit back and listen to the overall effect of the program as it is being assembled by the automation system. Many producers have welcomed this opportunity to "get out of the trees, and appreciate the woods".

We have found that many of the producers take a while to get used to using the new facilities. At first, productions more or less follow traditional methods of building up programs sequentially, with all the details put in place as production proceeds. Once they become accustomed to the ability to test and change segments freely and rapidly, producers start building the key segments pretty much independently of the other portions of the program. Sound production has thus developed a parallel with word processing, in that individual elements or complete modules of sound can be "cut and pasted" easily and rapidly within the program. There seems to be a great deal of comfort in knowing that it its possible to build a complex montage of sounds that can be shifted in time or changed to match other segments better, without destroying the structure of the entire program. Once the complex or crucial segments are in place, the supporting segments and more mundane details like program duration can be dealt with.

The time spent building a program in a digital production suite increases, compared to that spent in a conventional studio, because the detailed editing work is transferred there from the production office. The total time required, however, drops by about 30% to 50% for most programs. In some cases, the production time has not decreased very much, but the quality of the product has improved because of the greater flexibility available.

Many of our operators have complained that some of the earlier control systems were obviously designed by computer programmers who had never had any production experience. They found that procedures that might not be intrusive when using a computer, such as long key stroke command sequences, got in the way of the creative flow of program production. There is no doubt that computer assisted production is here to stay, but neither is there any doubt that the assistance should be unintrusive. "Keep the emphasis on the sound, not the computer", in the words of one operator.

CBC's early experiences with digital systems produced the impression that the most sophisticated productions would benefit most from the new tools. This is true in a broad sense, but even very straightforward forms of programming benefit.

Music: Music production, at first glance, seems not terribly well suited to random access media. It is sequential, and doesn't have the complex montages of many independent elements that are characteristic of drama productions. It does, however, require very precise edits whose in and out points have to be determined. Our operators have found that the waveform visualizations offered by many digital editors very useful for locating and eliminating coughs, and other noises that detract from a performance. They have also found the precise editing and rapid preview facilities save a lot of time whenever a small segment of music has to be replaced. These features, along with the multiple generation capability of digital techniques tend to improve the overall quality of the musical product.

Drive: Another style of program that would seem not to benefit from digital techniques is the morning and afternoon "Drive" show. These are done live, use very few effects, and depend more on content than on quality. CBC has not used the digital facilities to produce this type of program yet, but there are some promising applications for random access storage devices. The shows are made up of a lot of short segments, many of which do not occur at predetermined times, or necessarily in any specific order. CBC is investigating the use of disk recorders equipped with some sort of control panel that will give an operator very rapid, intuitive access to a large number of segments. These systems would replace the cart machines in use now, and eliminate the duplication, handling and "traffic" problems associated with the carts themselves. Many other functions can be envisioned, but they depend on the implementation of future interfaces, rather than the basic technology, so are outside the scope of this discussion.

News and Current Affairs: Experiments that apply digital technology to news and current affairs programming are underway. The first of these promises to cure an annoying problem that CBC has faced essentially since its own beginnings; that is the difficulty of getting reasonably good quality remote material into the production system quickly. In many cases, there is not enough time for a reporter to bring a taped interview back to the studio, so news has to rely on the telephone system (either land line or cellular). Good as it is, the phone was never designed to provide great on-air intelligibility, no matter how good our equipment and operators are.

The solution to this problem seems to have appeared in the form of low data rate digital audio encoder decoders (codecs) and the Integrated Services Digital Network (ISDN) that the common carriers are currently putting in place. Several different codecs that can provide 7.5 kHz voice facilities at 56 or 64 kilobits per second data rates are available, with the so-called G.722 codecs being the most common. These have, in some cases, been packaged with all the line interface, dialing and microphone and line audio input and output facilities required to provide dial up, two way circuits by merely connecting a pair of units to an ISDN facility. The great advantage of these developments is that, in addition to providing surprisingly good voice quality, the broadcaster can simply "dial up" a circuit whenever and wherever required (this is not universally true now, but soon will be), and is only charged for the time that the circuit is "off hook".

CBC is currently using ISDN circuits for many of its overseas news feeds. Radio Canada International is also using them for some program feeds because even though some of the available codecs degrade musical programs more than they affect spoken material, the only alternative is to ship tapes, or to use telephone circuits.

Digital audio rate compression is the basis of another project. The Radio News Audio Text System (RNATS), or Desk Top Radio project combines local area network and file server (computer) technology with compressed audio files and databases. This permits low cost workstations have access to and exchange a wide range of material while producing sound and text material for news. The workstations will combine the features of the INFO text processing system now used by radio news with (some of) the sound editing capabilities demonstrated by the equipment in our developmental studios. It should be possible to create links between the sound and text files to create searchable pointers into the recorded items. It will also permit simultaneous access to the original "raw" sound files so that multiple edited versions can be produced as required for different broadcasts. The finished items will be assembled into program lists, and made available at a simplified "on-air" control panel in the control room so that the operator will not have to rewind, reload and re-cue several several tape machines during a newscast. In the future, it may be possible to add other features, such as full random access to the Radio Program Archives, or cart machine emulation that would let the RNATS system serve other programming needs. As this is being written, the system specification is out for bid.

Variety: CBC does a lot of variety programming. The format is not as complex as drama, but requires the same tools. The major difference is that variety relies heavily on "live" contributions from studios across the country. These can be interviews with an artist in a different city, or in some cases, multiway hookups between several studios, all coordinated by the "host" studio. The contributed material is recorded along with the host's portion, then combined with the other program material in a post production and packaging stage.

DIGITAL NETWORKS

DAB is poised to change program emission, but digital technology will probably be felt sooner in the program contribution and distribution networks. The common carriers would like to dispose of the analog networks that they maintain for broadcasters, and treat programs as just another data stream. The current hope is that the perceptual codecs developed for DAB will be able to lower the required bit rates enough to provide economical circuit costs, while improving on the signal quality of existing circuits.

The perceptual codecs are based on the assumption that a relatively loud signal in a "critical" band of frequencies will conceal or mask a lower level sound in that same band. (There are additional secondary effects that can be ignored for the purposes of this discussion). The codecs reduce the total data rate by allowing the noise floor to rise to some point below the "masking threshold" in each of the critical bands. The recovered signal thus has a dynamic, program dependent noise associated with it that is (ideally) concealed from the listener by the program signal itself.

Perceptual codecs obviously have to do a great deal of signal processing to achieve the low data rates being discussed. This cannot be done instantaneously; the lower the data rate, the more signal processing is required, and the longer the insertion delay of the codec.

CBC and others have done basic audio quality tests on several different codecs and found that some program material (generally referred to as critical material) produces artifacts in the recovered signal. In general, more material becomes critical as the the data rate is lowered, and the artifacts become more severe, as intuitively expected.

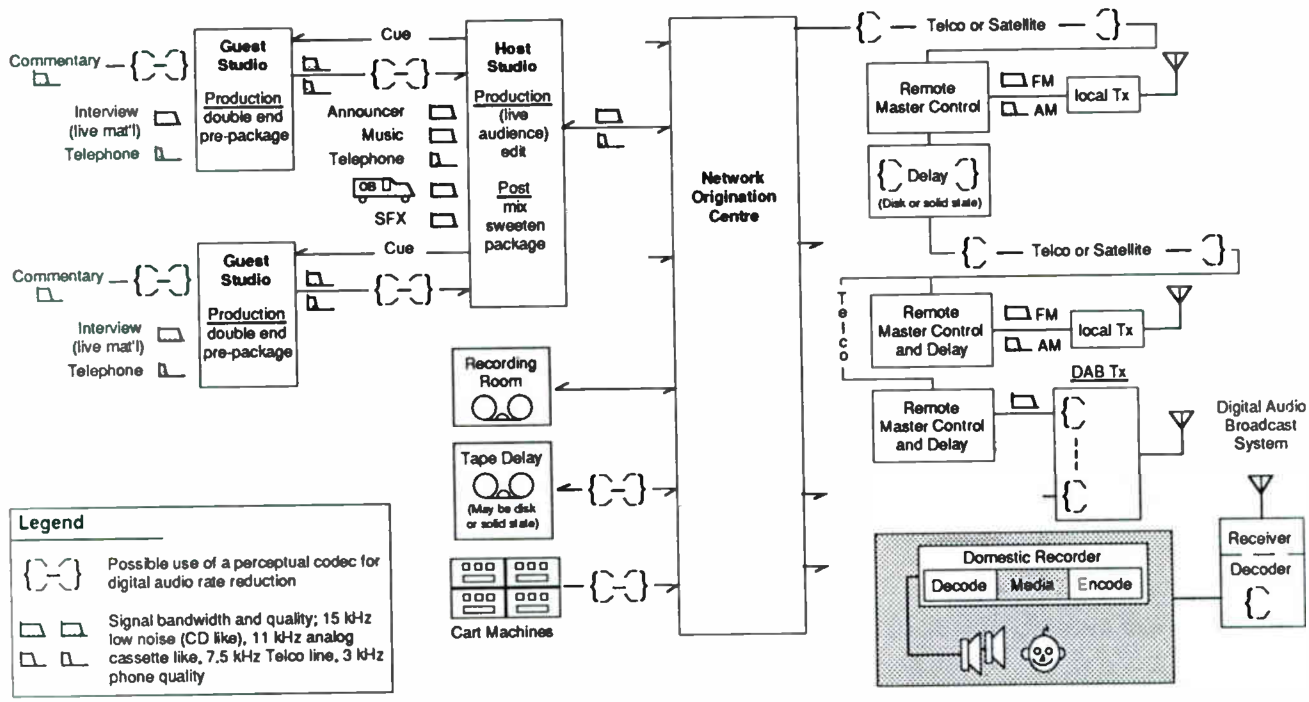

Figure 2: Potential opportunities for rate reduction codecs in the Contribution, Distribution and Emission portions of a digitized Radio network

Program Contribution:

The program contribution network is generally confined to the left hand side of the network origination (radio master control) block shown in Figure 2. It includes the high quality links between the host and guest (contributing) studios, and the link between the producing (host) studio and the network origination centre. The diagram shows where rate reduction codecs might be used in the links. There can be up to five tandemed codecs in a contribution circuit, if all the links are included.

It has always been assumed that circuits used for program contribution should preserve as much of the original sound quality as possible. Any impairments occurring at this point, after all, will become a permanent part of the program.

Sound quality turns out to be only one of several important parameters. As mentioned earlier, the CBC does a lot of "two-way" hook ups between host and guest studios. These are impossible if there is an annoying amount of delay between studios. The "annoying amount" varies with the program material. An interviewer could tolerate more delay than several musicians trying to play together, particularly if the topic was very important. The cue circuits shown in Figure 2 may or may not contribute to the delay. High quality sound (via a codec, therefore with some delay) is more important for music programming than for an interview, for instance.

There is a another untested factor that may affect the operation of program contribution networks. The CBC is concerned that the signal processing normally done during the production of programs using material contributed via rate reduced networks may "unmask" portions of the program dependent noise or other artifacts. Early tests of codecs operating at 128 kilobits/s per mono channel have shown some problems. More extensive "post processing" tests with codecs operating at 192 kilobits/s per mono channel are planned for early spring of 1992, as part of the CCIR's efforts to select codecs for future radio systems.

In short, perceptual codecs are not, at this stage of their development, an unmixed blessing. The trade-off between quality, bit rate and delay may have to be pushed towards lower delay and higher quality at the cost of bit rate to make the "two-way" hookups now in common use practical. The topology of future contribution networks will almost certainly have to be planned to minimize the number of tandemed codecs, and the total delay.

Program Distribution:

The program distribution network carries program material from the network origination centre to the transmitter inputs. The right hand side of Figure 2 provides a fairly good idea of what might happen in a section of this network. It is evident that the most important problem in digitizing the distribution network will be dealing with the number of tandemed codecs. Current CCIR tests (Jan. / Feb. '92) are evaluating the basic audio quality of three tandemed codecs operating at 128 kilobits/s, and will base their distribution application recommendations on these tests.

Figure 2 shows two codecs in the "Telco or Satellite" links and a third in the Delay recorder that is used to compensate for the time zones across the country. Three codecs is a conservative estimate of the total number of codecs in some distribution links, especially in the early stages of the conversion to a rate reduced network.

CBC (and other broadcasters) now lease links with analog terminals from the carriers, so have no control over the number or type of codec that may be part of the link, as long as the link meets contract specifications. (Testing and specifying codec performance is beyond the scope of this paper, but is a major problem). Additional codecs may find their way into the system within the "Remote Master Control" areas. None of these currently have the facilities to switch encoded digital signals, so codecs may be added just to provide program routing.

As the digitization of the distribution network proceeds, the number of embedded codecs will drop. Current assumptions are that the carriers will provide data channels with digital terminals, leaving the broadcasters to provide (and control) the codecs. Broadcast plants will gradually convert to digital routing switchers, so will pass the network signal without having to re-encode it. Ideally, there might be only one encoder at the network presentation center, and a decoder in the listener's receiver. Reality will probably eventually validate the choice of three tandemed distribution codecs.

One major advantage of digital signal distribution is the opportunity to multiplex network cue and control information into the program data stream. It may, for example, be necessary to distribute "network time", which won't match local time because of codec delays, along with the control information to manage the network cleanly. There have also been some proposals to include non program related data (RDS information) with the program data. Because there is only 8 kilobits/s reserved for all the auxiliary data, it seems shortsighted to limit a potentially lucrative service to something less than this.

Emission:

The emission link may be a bit more complex than Figure 2 implies. Some form of encoding will have to be done at the transmitter, even if the distribution system uses the same source coding algorithm as the emission link, because of the different channel coding requirements. The network cue and control information referred to above may be replaced by other data more pertinent to the emission system. Several feeds from the distribution network may be combined to provide multichannel sound services, as the advanced television systems now being developed will do.

Implications:

The emission codec may not be the final link in the system. If rate reduced domestic recorders proliferate, it may be part of a tandem, rather than the last link to the listener. The broadcaster clearly cannot control the situation downstream from the receiver decoder, but must control the upstream system. Because the sheer quantity of receiver decoders make it the only immutable component of the system, it has a crucial role in defining the rest of the system.

There is a great deal of codec development work currently under way. This will not stop when, for instance, the CCIR recommends which of the existing codecs are most suitable for the contribution, distribution and emission "applications". The reality is that there will be a constantly evolving mixture of rate reduction algorithms and codecs upstream of the receiver decoder.

The broadcasters' long term task is to ensure that these upstream devices are compatible with the receiver decoders. Their short term task is to ensure that the selected emission algorithm does not compromise future development of other codecs, and hence of the complete system.

CONCLUSIONS

CBC has built several digital audio studios to explore new techniques in radio and television sound production. Producers have welcomed the freedom to try out different sound montages without wasting time, and the "cut and paste" editing that random access recorders provide. The relative ease of automating digital processes has (when correctly implemented) removed many of the procedural impediments to production process. Program quality has improved, and the time required to produce complex programs has decreased by about a third, when compared to tape based techniques. There are additional innovative benefits to less complex forms of programming, exemplified by the Desk Top Radio experiments being conducted for news and current affairs.

Experiments in sound production naturally extended to questions about program distribution technology, and its influence on future systems. The perceptual codecs originally conceived to reduce data rates for DAB emission have migrated upstream into network service areas in conjunction with the common carriers' move to digitize their networks. These codecs promise to improve sound quality and reduce network costs. Broadcasters will, however, have to insure that the receivers selected for a DAB service are compatible with, and do not compromise the future development of these codecs.

TELEVISION AND NEW TECHNOLOGY

Sunday, April 12, 1992

Moderator:
Tony Uyttendaele, Cap Cities/ABC, New York, New York

*Paper not available at the time of publication.

TELEVISION DATA SYSTEM FOR PROGRAM IDENTIFICATION

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Abstract- Congress has passed the Television Decoder Circuitry Act of 1990 and the FCC has issued rules requiring 13-inch and larger television receivers manufactured after July 1, 1993 to include decoder circuitry to display closed captioning data[1]. Manufacturers are busy preparing products for introductions in compliance with this new requirement. The EIA's Television Data Systems Subcommittee has been working at the request of the FCC to develop standards for extended data services on line 21 of field two, which will make use of the same decoding hardware.

Because of the certainty of the required decoding circuitry, an opportunity was created that guarantees widespread availability of a new data communication channel from the broadcaster to the TV receiver. The resulting new features will make TV receivers more user friendly and create new opportunities for broadcasters.

WHO IS DEVELOPING THIS SPECIFCATION?

In September of 1990, a group of TV receiver manufacturers and caption providers* formed a Task Force under the Electronic Industries Association (R-4 TV Systems Committee) to formulate the TV receiver display standards for Closed Captioning. This group worked assiduously to deliver a document to the FCC by December of the same year. With minor modifications that document was incorporated into the FCC Report and Order of April 15, 1991 which defined the requirements for closed caption decoders that must be part of every TV set manufactured after July 1, 1993.

The same group became the EIA R4.3 Subcommittee: "TV Data Systems" and has since been working at the request of the FCC[2] to define *Extended Data Services* for line 21 of field 2.

WHAT ARE THE NEW DATA SERVICES?

Extended Captioning Services

In addition to adding the capability to provide a second, independent caption service with all the same features as the original, several new features will be added.

The second (field-2) service will be able to provide complete captions in a second language, with all the necessary special characters without being restricted by the presence of the original captions. Additional character symbols will be included to allow support for many languages not now supported. New options will be added to allow caption providers to select a variety of background colors.

Extended Text Service

The new service will add the capacity to provide two additional TEXT services in the same format as the current service.

Program Identification Service

A completely new service will be added to allow program providers and local broadcasters to transmit data which can be used to describe numerous aspects of the current and future programs. This new service will also be capable of sending automatic station identification, clock setting data, aspect ratio information, and National Weather Service text messages.

What are the Benefits for TV Viewers? There will finally be an end to the blinking 12:00 on VCRs and TV screens. Automatic clock setting, with self calibration, will permanently end this problem.

"Channel Maps" that list which station is on what channel will no longer get misplaced or clutter the coffee table. No matter how often the local CATV company shuffles the line-up, viewers will always be able to find their favorite stations by on-screen station identification and channel selection.

Channel browsers will be able to identify the name of programs quickly by on-screen program titles that are generated by the TV. These titles can even appear during a commercial break. Figure 1. on the next page, illustrates an example of how a browsing screen might appear

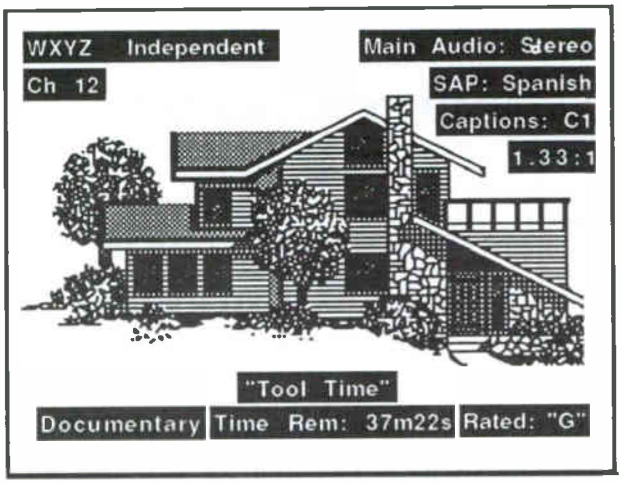

Fig.1 An example of a "Browsing Screen"

In addition to information about the current program, stations will also be able to send their schedule for upcoming programs. The receivers can be made to seek and store this information during non-use and compile it into an on-screen program schedule. This information can be used in on-screen displays to make program selection quite easy. Figure 2 shows how this might be displayed.

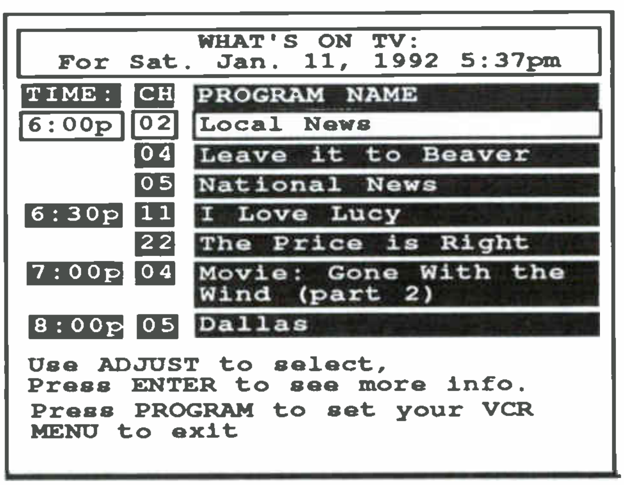

Fig.2 An example of an On-Screen Guide display.

A second screen could be used to view more detailed information about the current or scheduled programs. Figure 3 shows how this might look.

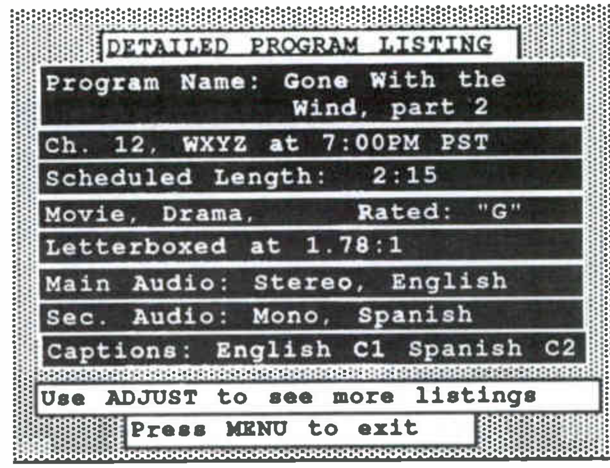

Fig.3 An example of a Detailed Program Listing.

Information about the program's aspect ratio is also included so new wide-screen TVs can automatically zoom in to fill the screen on letterboxed programs or commercials.

The system also provides VCR users with a number of features. Since each program is uniquely identified, VCRs can be designed to record desired programs completely, even if they run overtime or are delayed. Program identifiers for future programs can be transmitted during previews so VCRs can be instantly programmed by impulse response.

The system will also have the means to retransmit the National Weather Service's Weather Radio Specific Area Message Encoder (WRSAME) messages. This capability would allow TVs to use on-screen displays to show weather warnings for their specific area as quickly as they are issued.

What are the Benefits for Broadcasters? At last the broadcaster will have the means to identify his station by call letters (and network affiliation) directly on his viewers TV screen whenever channels are changed. CATV operators will no longer be able to confuse consumers by re-mapping channels.

Program providers will be able to fully utilize the wide-screen modes of new wide-screen TVs by controlling when the TV switches into the various modes.

Channel browsers will be captured by advertisers when they see the name of the program they are interested in viewing. Ratings can be improved because it will be easier for viewers to find the programs they want to watch or record.

HOW AND WHERE WILL IT BE USED?

These new data signals will be transmitted exclusively on line 21, field 2 in the same format as closed captioning signals. The new signals may coexist with the new closed captioning services & text services also planned for this line.

Until now, this line has been reserved but unused. Currently there is a FCC authorization[3] to place a 9 bit pseudo random framing code signal here. This signal is supposed to occupy about 1/2 of this line when Closed Captioning signals are transmitted. It should be made clear that this signal has never been used by any captioning equipment.

The EIA Subcommittee will be making a recommendation to the FCC for authorization and protection of these new services on line 21, field 2.

WHY IS THIS SYSTEM THE ANSWER?

There have been many similar and competing proposals for program identification schemes in recent months. Many of the other proposals offer higher data transfer rates and more capabilities. However, there are several advantages that are only available on the EIA system.

What are the Hardware Advantages?

The receiver hardware comes for free. All the hardware requirements for this system will be included in every TV manufactured after July 1, 1993 in order to comply with the closed captioning requirements. Receiver manufacturers have championed this cause to provide a low-risk solution. Only some additional control software is required beyond the minimum caption decoding and display hardware that will already be included.

The encoding hardware is almost free. Since the encoding equipment needed is exactly the same as the equipment used for encoding the closed captioning signals, many stations may already have this equipment. Because of the increased exposure and more widespread use of captioning, new encoder suppliers will have more equipment available at lower costs. Again the only difference to this equipment is some additional software to automate the transmission of the codes.

What is the Format Advantage?

Because this system is defined as part of the extended services for the line 21 closed captioning system, it is expected that it will gain the same FCC protection as the current captioning system. This protection ensures that the data is not removed or destroyed by down-stream processes. There will only be one type of signal on line 21.

Why not Other Proposed Systems?

It would be unlikely for receiver manufacturers to embrace a different system that involves additional hardware costs. Why should one build a new data decoder for a system that may never materialize or get widespread support?

Program providers or local stations have no incentive to transmit data to an audience they are uncertain will ever exist. This creates a question of which comes first, the data or the receivers? The EIA proposal faces no such uncertainty, all the hardware will be in place.

WHEN WILL THIS HAPPEN?

As it now stands, caption decoding hardware will be in every TV (13" or larger) after July 1, 1993. The goal of the EIA subcommittee is to make the submission to the FCC by the end of the second quarter of 1992. Allowing 6 months for FCC action and finalization of the EIA specification, manufacturers could then begin adding the necessary software updates to TVs.

It would then be possible for some Program ID capable TVs and VCRs to appear by late 1993 or early 1994.

SUMMARY

TV manufacturers, caption providers and other industry representatives have joined forces in an EIA subcommittee to establish a single standard for transmitting program and station identification data on an unused portion of the vertical blanking interval. This proposed system will allow many advanced, new features to both TVs and VCRs at little cost to the consumer. Broadcasters can benefit by providing the data which allows viewers to more easily find their favorite programs. This system is very likely to succeed because it will require no new hardware at either the transmitter or receiver ends. Broadcasters are encouraged to recognize the new business opportunities that will be offered and participate in the process that can make the current schedule for implementation of these program identification services a reality by the end of 1993.

ACKNOWLEDGEMENTS

I would like to express my appreciation for the dedicated work and encouragement of the following individuals for their part in the Task Force and on the Subcommittee: (1) Mr. Julius Szakolczay, Engineering & Development Manager for Mitsubishi Electronics America Inc., and Chairman of the Task Force and Subcommittee; (2) Mr. Tom Mock, Director of Engineering EIA CEG and all those on the Subcommittee who have contributed their valuable ideas and efforts to this new system.

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1. Code of Federal Regulations, Title 47, part 15.119 as amended [GEN. Docket No. 91-1; FCC 91-119]
2. Report & Order, FCC 91-119 April 15, 1991
3. Code of Federal Regulations, Title 47, Part 73.682 (22)(i)(C) and Part 73.699, figure 17 C

* **Participants in EIA R4.3**

TV Receiver Manufacturers:
Mitsubishi, J.V.C., Matsushita (Panasonic/Quasar), Philips (Magnavox), Samsung, Sanyo, Sony, Thomson (RCA/GE) and Zenith.

Caption Equipment Suppliers:
EEG and NCI.

Caption Providers:
Caption America, The Caption Center (WGBH), Real-Time Captioning

Others Represented:
Texas Instruments, Gemstar Development Corp, Insight Telecast, E.I.A., N.A.B., and NCTA.

DBS FOR LOCAL BROADCASTERS

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Abstract- A satellite system whereby every existing television broadcast station in the U.S. can transmit program(s) to a 200 or 300 mile diameter coverage area is described. Four (or more) NTSC channels or two HDTV channels are transmitted in each satellite transponder, free of the impairments and limitations imposed by traditional terrestrial broadcasting.

The satellite to provide this service is described in detail showing how the narrow beams are formed and how the signals are processed. Frequencies are re-used through geographic isolation allowing a large number of reliable low power transponders on board. Therefore a large number of users can be accommodated.

A companion EIRP budget is given which shows that performance margins are equal to or better than National DBS networks with 230 watt transmitters on board.

Digital source coding and transmission are planned for both NTSC and HDTV channels. Software features will include conditional access, blackout of restricted areas within the beam to avoid duplication of programs, and a variety of data and personal messages to selective receivers.

INTRODUCTION

Did you know that the FCC has set aside 1000 Megahertz for television broadcast? Yes, one whole Gigahertz for broadcasting of television directly to homes. If this speaker was a broadcaster, I would be plenty angry if someone other than a broadcaster were to use that spectrum. The spectrum has been allocated to a satellite service. Internationally it is called the Broadcast Satellite Service (BSS). In the U.S. it is commonly called DBS for Direct Broadcast Satellite, or more properly Direct Broadcast Satellite Service as in Part 100 of the FCC rules. No matter what you call it, it is a Broadcast Service, and it would be tragic if broadcasters as we know them did not use it.

DBS has been perceived as a national kind of service without the localism of broadcasting. Up to now all planning by applicants, licensees, and programmers interested in using this resource has been directed toward serving the entire U.S. with TV programs. Now, the convergence of several technologies has made it possible for localism in satellite broadcasting.

TECHNICAL PARAMETERS OF DBSS

Technical aspects of the Direct Broadcast Satellite Service for the International Telecommunications Union (ITU) Region II, which includes the U.S., are given in the Final Acts of the 1983 World Administrative Radio Council (WARC). The FCC has, with very few caveats, adopted those standards but they have not yet been codified into the rule and regulations.

TABLE 1
SOME TECHNICAL FEATURES OF DBS
IN THE UNITED STATES

PARAMETER	ALLOCATION/STANDARD
1. Orbital Slots (West Longitude)	61.5°, 101°, 110°, 119°, 148°, 157°, & 175°
	This spacing allows extremely small receive dishes from an interference standpoint. Dishes of less than 12 inches in diameter can be

	used if the power from the satellite is adequate.
2. Operating Frequency Band	Feeder (UP) Link: 17.3 to 17.8 GHz Downlink: 12.2 to 12.7 GHz
3. Polarization	Circular; Both left and right hand polarization are allowed from an orbital slot.
4. Channelization	16, frequencies on each polarization, for a total of 32 from an orbital slot. Center frequencies are about 30 MHz apart.
5. Downlink EIRP	Left to discretion of operator, but limited at the border of the country.

ANTENNA BEAMS

At geostationary orbit, the earth subtends a solid angle of about 19°. If one chose to cover the entire globe with a usable signal from a satellite in that orbit, a flared horn with 19° half power beamwidth could be used at any frequency allocated to satellites. Indeed, the Intelsat satellites utilize such a horn antenna for global coverage. To cover a land mass such as the Continental United States (CONUS) a beamwidth of about 3° (latitude coverage) and 8° (longitude coverage) could be used. Figure 1 depicts global and CONUS coverage graphically.

At 12.5 GHz, a four foot diameter parabolic antenna will have a half power beamwidth of about 1.5°, a 8 foot dish about 0.7°, and a 12 foot dish 0.5°. From Synchronous altitude these beams would produce half power coverage of 600 miles, 300 miles, and 200 miles respectively. Figure 2 shows this land coverage centered on Puget Sound. The 12 foot dish is a convenient size although it is a little bit larger than most satellite dishes. It fits very nicely on a modern satellite launch vehicle such as Arienne, Titan, Atlas, Latest Delta, or cargo space on Shuttle.

Figure 3 shows two adjacent beams in Washington State. There are many ways to produce a multiplicity of beams without having separate antennas. The simplest way is shown in Figure 4. If a feed horn of a parabolic reflector antenna is displaced from the parabolic axis of revolution, a beam is produced that is squinted from the boresight (axis of revolution). If a single reflector with a multiplicity of feeds is used, a multiplicity of beams are formed. Figure 5 shows how a single reflector with multiple feeds can produce a "shaped beam" to fit coverage of a particular land mass; in this case Mexico. In the shaped beam case, the feed horns are connected by a power splitting network to produce the desired result. In the Local DBS case, a separate feed horn is used for each beam.

FREQUENCY REUSE

Since each beam produced by the satellite antenna is isolated from every other beam, a geographical isolation is created. With this isolation, the frequencies can be reused without fear of interference. In the planned local DBS system, alternate beams can use the same frequencies. A frequency reuse factor of 16 is planned for a fully loaded system of 60 beams. All that is required for reuse on alternate beams is that the sidelobe energy be low enough to preclude harmful interference.

It must be emphasized here that use of the term "frequency" denotes the band of frequencies allocated to a transponder. In other words, in this context, a frequency is synonymous with a transponder. By using video compression techniques a transponder can handle more than one television signal. Local DBS currently plans transmission of a minimum of 4 NTSC signals, or 2 HDTV signals in each transponder.

An average of 4 frequencies or transponders will be connected to each beam. The number assigned to a beam will depend on market conditions. Urban areas will have more and rural areas will have less. Furthermore, it is likely that lightly populated areas will have wider beams of about 300 miles in diameter.

SIGNAL PROCESSING

The Local DBS is similar to every other U.S. domestic satellite currently flying in the way it processes signals. It is a type of repeater known as a "Single Conversion Heterodyne Repeater." Sometimes it is called a bent pipe in the sky. What makes this satellite unique is that it has many more transponders on board. However, the transponders utilize low power amplifiers in the 7 to 15 watt output range as opposed to the National DBS which utilize 120 to 230 watt output amplifiers. The weight and power capacity of the national and local satellites is about the same, and they can share a common bus. Instead of a single common input from the receiving antenna and a single common output to the transmitting antenna, the Local DBS has input and output from individual feed horns. Figure 6 is a partial black diagram of the local DBS.

EIRP COMPARISON NATIONAL AND LOCAL DBS

TABLE 2 - EIRP COMPARISON

PARAMETER	NATIONAL DBS	LOCAL DBS
1. Power Amplifier Output - dBW	+23.6 (230 Watts)	8.8 (7.5 Watts)
2. Output Multiplexer and Feed Line Losses - dB	1.5	1.0
3. Antenna Beamwidth - Degrees	3° x 8°	0.5°
4. Peak Antenna Gain - dB	31 (55% eff)	49 (40% eff)
5. Peak EIRP - dBW	53.1	56.8
6. Geographic Loss - dB	1.0	3.0
7. Edge of Coverage EIRP - dBW	52.1	53.8

The EIRP from the Local DBS is expected to be at least 1.5 dB better than the national service everywhere in the coverage area of both services. It should be noticed that the difference in transmitter power is more than made up by antenna gain. The loss between power amplifier and antenna is less in the Local DBS case since a power division network to a multiplicity of feed horns is not required.

Reliability of the Local DBS should be better than the National Service. Five to 10 watt power amplifiers in satellites have a proven track record of phenomenal reliability. Use of satellite amplifiers of 120 watt and higher has been limited, but the results have been somewhat disappointing. Recent experience has been good however. Use of the low power amplifiers represents very low technical risk.

DIGITAL SOURCE CODING AND TRANSMISSION

The Local DBS system plans to use digital compression for both video and sound because of its spectrum efficiency. Transmission will be digital utilizing a power and spectral efficient technique. Spectrum efficiency is not as important in satellite transmission as is power efficiency. Digital compression in the source coding will allow more television channels in a transponder than might otherwise be possible without compression. Use of QPSK modulation with modest error correction will allow use of extremely small ground receiving antennas.

Even the purest of the purists in the broadcast industry concede that today's technology of compression using 6 to 8 Mb/s on NTSC video produces an acceptable result for broadcast. Proponent HDTV systems are utilizing basic data rates in the range 15 to 20 Mb/s. Almost everyone in the industry believes that a digital system will be selected by the FCC for terrestrial broadcast. Even if the FCC doesn't standardize on a digital system, the satellite industry will adopt one.

Audio

TV associated audio will be sent via some digital compression technology. The most likely candidate is Musicam where monaural audio can be transmitted with true CD quality at 128 kb/s, and left and right audio can be transmitted in 192 kb/s. Musicam is very close to becoming an international standard by a joint ISO and MPEG committee.

NTSC

At least 4 channels of compressed NTSC video will be transmitted in each transponder. The method used will be determined by whatever technology that proves to be best at the time the Local DBS is launched, probably in 1996.

HDTV

Each transponder will be capable of handling 2 HDTV channels. Whatever proponent system is ultimately selected as the standard by the FCC will be used in the Local DBS.

BLACKOUT FEATURES

Consider the case shown in Figure 7. At least 3 TV markets are covered by a single 200 mile spot beam. This will be the case in many urban areas in the U.S. Market integrity will be maintained through software by way of a blackout feature. At least 32 blackout combinations can be accommodated in each channel. Blackout regions can be defined by Postal Zip Codes and/or geographic coordinates. If a satellite station obtains program exclusivity for the entire beam, the coverage area and potential viewers can extend over twice the area of the average B Contour of a full service station. In any case, the satellite signal is not subject to the blockage and multipath so common in terrestrial broadcast. The so-called white areas of fringe reception in some markets will obtain excellent reception. The need for troublesome translators used in many markets will be eliminated.

A farsighted broadcaster might establish a second or third channel for a variety of reasons. Use your imagination. If the program is non-duplicating, the entire 200 or 300 mile area could be served. Selective data services and personal messages can be sent via the conditional access system used.

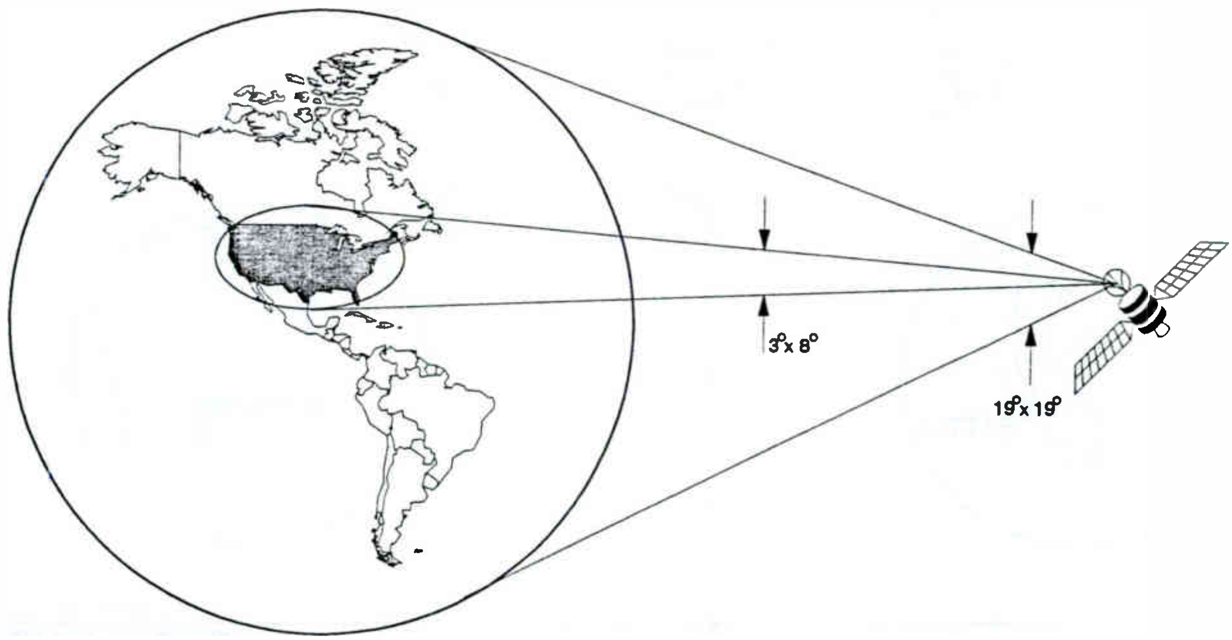
GROUND RECEIVERS

Ground Receivers will be capable of receiving either local or national DBS since they both will use the same technologies of compression and modulation. The upscale models will probably have antennas that can readily be pointed at any orbital slot in its field of view, either mechanically or preferably electronically. Figure 8 shows a block diagram of a DBS receiver, with the desirable features. Current planning calls for HDTV and NTSC digital signals to have about the same symbol rate in a transponder. The NTSC stream to contain a minimum of four TV channels and the HDTV stream to have two TV channels. This will allow commonality in the Demodulator, Demultiplex, and Forward Error Correction circuits as well as the decryption/authorization protocols. The NTSC decompressor can be in either the TV set or in the satellite receiver. For HDTV, the decompressor will be in the TV set.

ACKNOWLEDGEMENTS

The author wishes to thank the NAB and the staff of the Science and Technology Department for the invitation and the opportunity to present this paper. Ed Taylor and Selman Kremer are hereby acknowledged for their entrepreneurial spirit to move forward on this concept. Space Systems/Loral is also thanked for its contributions to this unique satellite system.

FIGURE 1 - GLOBAL AND CONUS COVERAGE FROM SYNCHRONOUS SATELLITES



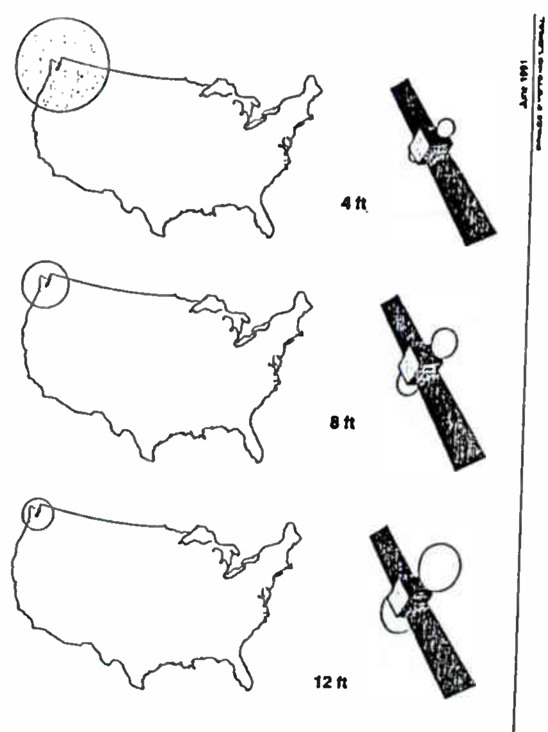


Figure 2 - Land Coverage for Different Satellite Dish Sizes

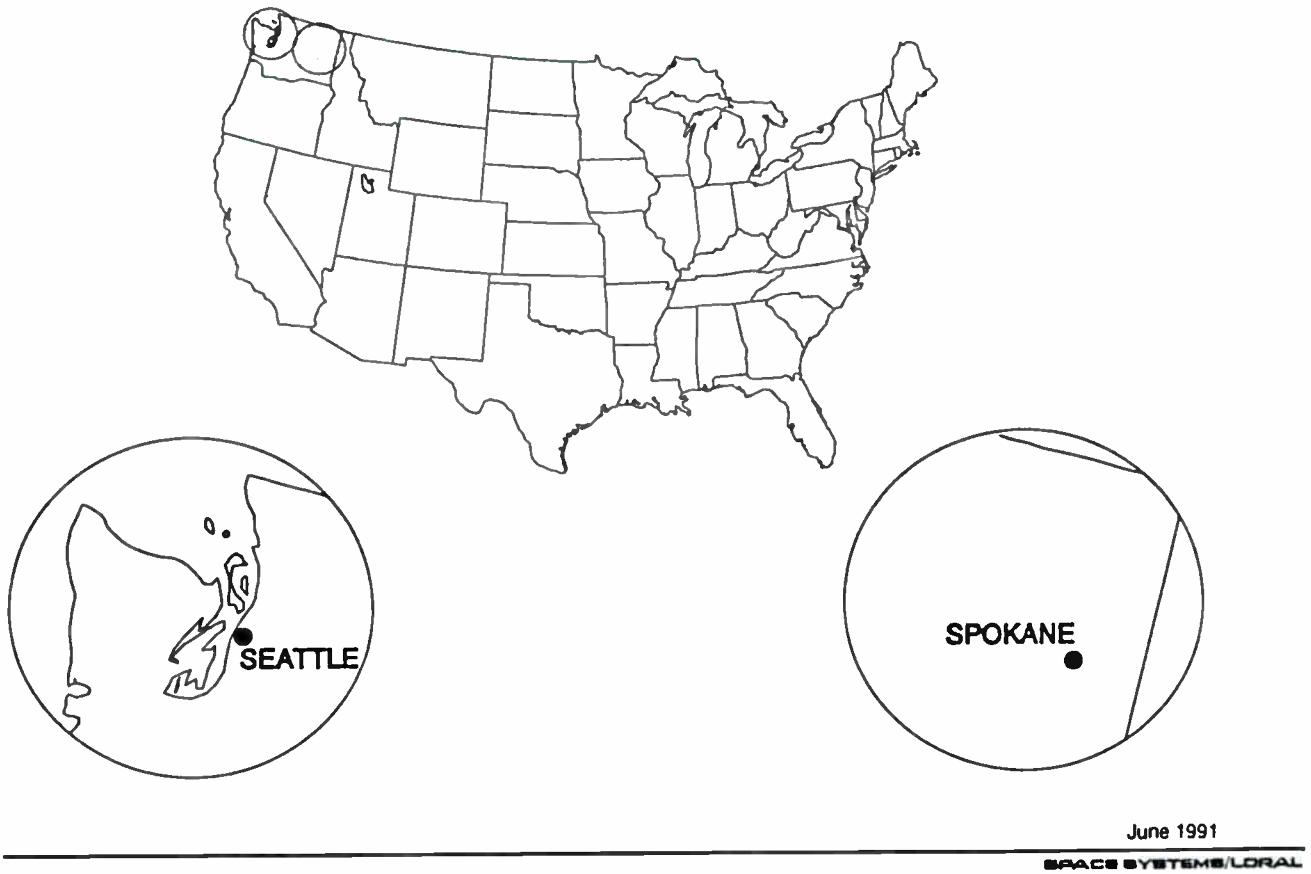


Figure 3 - Adjacent 200 Mile Beams

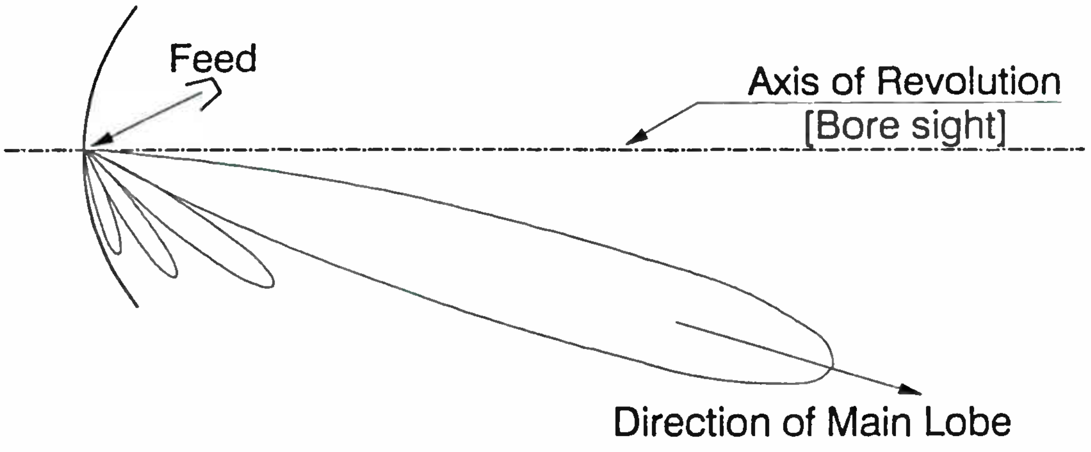
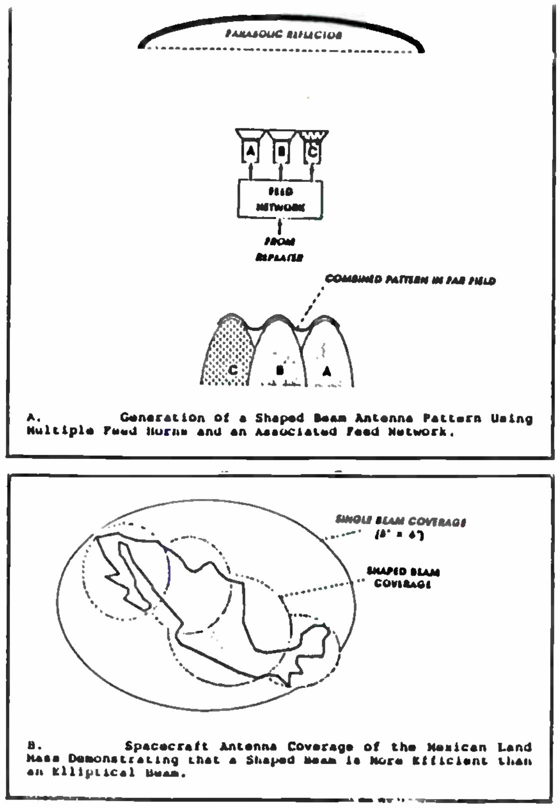


FIGURE 4 - ANTENNA PATTERN WITH FEED OFFSET FROM FOCUS



A. Generation of a Shaped Beam Antenna Pattern Using Multiple Feed Horns and an Associated Feed Network.

B. Spacecraft Antenna Coverage of the Mexican Land Mass Demonstrating that a Shaped Beam is More Efficient than an Elliptical Beam.

Figure 5 - Formation and Application of Shaped Beams

FIGURE 6 - SATELLITE PARTIAL BLOCK DIAGRAM

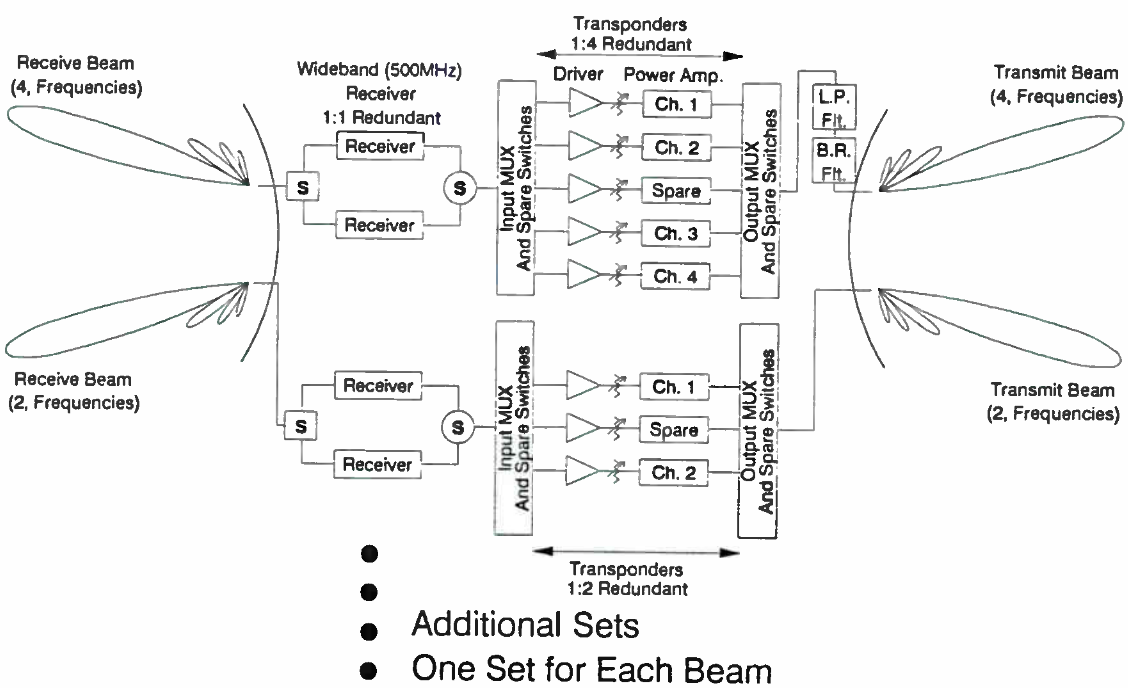


Figure 7

- ELECTRONIC BLANKING with the BEAM
- Washington Local Stations cannot be received in Richmond or Baltimore.

June 1991

SPACE SYSTEMS/LORAL

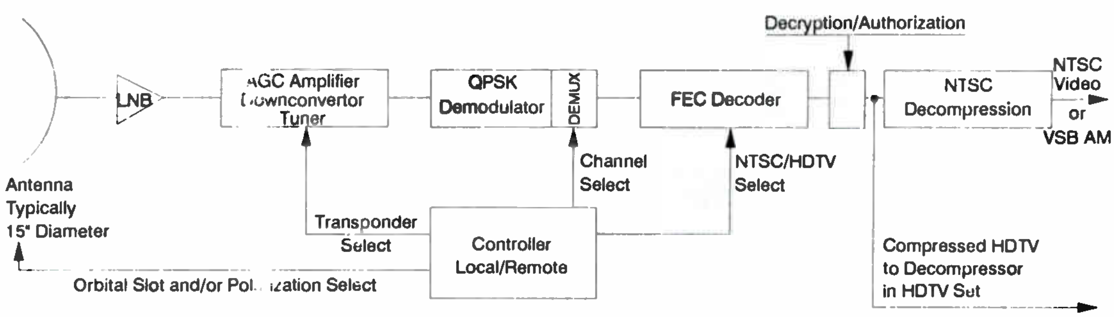


FIGURE 8 - DBS GROUND RECEIVER, BLOCK DIAGRAM

ISDB TRANSMISSION SYSTEM IN THE 12 GHZ BAND DIGITAL SATELLITE BROADCASTING

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ABSTRACT

ISDB (Integrated Services Digital Broadcasting) is a total digital broadcasting system that can not only integrate and transmit a large variety of services but also provide multi-media services using a number of different media. This paper describes the concept of ISDB and discusses the flexible signal transmission system using one satellite broadcasting channel in the 12GHz band. And Digital TV, supposed a main service in ISDB, is also investigated. Digital TV is expected to provide high-quality services nearly all of the time when compared with FM-TV. In one satellite channel, it may become feasible to transmit more TV channels than FM-TV system, and various data services by means of lately developed efficient coding methods.

INTRODUCTION

Digital technology has been intensively introduced in broadcasting studio equipment. But dedicated digital channels has not been used for digital broadcasting systems. The reasons why realization of full scale digital broadcasting has been delayed in the past are the lack of frequencies for transmission, high cost of high-speed LSI's and immature encoding technology.

Recently, however, high-efficiency encoding technology for pictures and sounds has made rapid progress, and the digital signal transmission has been expected effective in frequency utilization. The evolution of the information society has increased the needs for various data services and the development of these encoding technology and systems has matured, leading to standardization in various fields.

Aside from these developments, research of ISDB using satellite broadcasting has been conducted as a system to digitally integrate information and to flexibly provide services.

Under these circumstances, the environment to implement ISDB in which source signals including motion pictures are encoded in digital, multiplexed and transmitted has been established. As a new broadcasting infrastructure, ISDB has various possibilities, such as efficient utilization of frequencies, participation by a wide variety of service broadcasters and supply of new services suiting needs. Much is expected from ISDB as broadcasting to meet the diverse wishes of the viewers.

This paper describes the concept of ISDB using 12 GHz-band satellite broadcasting and studies of the encoding and transmission systems for its realization.

CONCEPT OF ISDB

Digitalization of Broadcasting

ISDB can be implemented based on the overall advantages of digitalization in broadcast production, signal transmission and reception.

(1) In broadcast production, news gathering equipment and studio facilities will be implemented with digital technology and digital lines in and outside of stations will be consolidated, so that the production systems will be unified and a diverse, efficient production environment will be built.

(2) More channels can be offered and diverse services can be integrated more easily by digital signal transmission. Information will be able to be sent to homes without quality deterioration.

(3) Signals of this broadcasting can be received by one integrated receiver for all services. By integrating circuits in LSI's, receivers can easily be produced in small size, at a low cost. Furthermore, using a computer technology, new receiving functions with a good human interface can be offered.

Digitalization toward the 21st Century, as mentioned above, will enable broadcasting to be a new means of providing information to homes. The following items will be demanded in the future as broadcasting functions.

(1) More Television channels will be offered by effective frequency utilization, to provide diverse programs and wide information which the viewers need.

(2) Presentation functions of the television will be enhanced through the introduction of various digital picture effect functions and through providing with additional information by the use of media suiting the programs, such as sound, printing and software.

(3) The program identification and supply of user guide information will make automatic recording, reception and selection of information easy, to improve the reception environment of each viewer.

(4) Various data and new broadcasting services will be handled uniformly for flexible organization of broadcasting programs.

To accomplish them, a broadcasting system can be considered to transmit integrated services by providing broadband digital broadcasting channels on one transmission channel and by multiplexing picture, sound, data, etc. with user guide information in this transmission channel. This is called Integrated Services Digital Broadcasting.

Transmission Channel of ISDB

Satellite channels, terrestrial channels, optical fiber cable and other means can be used as ISDB transmission channels. Among them, 12 GHz-band satellite broadcasting channels have more possibilities as transmission channels that will actualize ISDB in the near future. When a broadcasting signal is transmitted by digital modulation using one satellite broadcasting channel, the interference protection ratio used in WARC-BS planning (no interference disturbance must be detected in the same channel as FM-TV: 30dB, adjacent: 14dB) must be satisfied.

There are possibilities of using a frequency band near 22GHz for broad-band ISDB, including high definition TV, however, attenuation of this frequency band by rain and water vapor is large, so radio wave propagation characteristics of this band have many technical problems.

CONFIGURATION OF TRANSMISSION SIGNAL

Compression Encoding and Transmission

The compression encoding technology of television pictures is making rapid advances as shown in Figure 1.

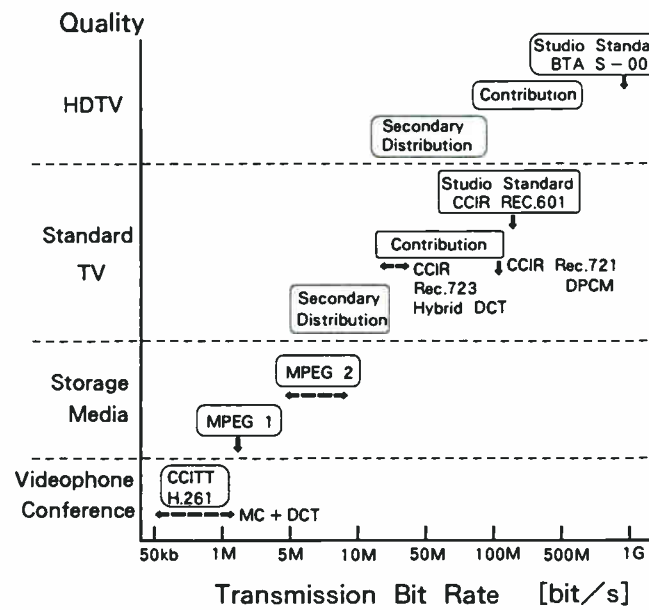


Fig.1 Transmission Bit Rate of Picture Coding Standards

Using this picture encoding technology, some television channels of broadcasting quality can be transmitted in the bandwidth for one satellite broadcasting channel. Furthermore, using compression-encoding of television signals, not only the television signals but also a large volume of data can be transmitted in satellite broadcasting channels. For example, using a transmission channel of 1Mbps, approximately 8 sound programs can be sent with the 128kbps compression-encoding of sound signals. Six newspaper pages per second can be sent if characters are sent by encoding. Or one to ten A4 pages per second can be sent by facsimile.

Figure 2 shows the ISDB transmission system which uses the bandwidth of one satellite broadcasting channel entirely for digital signal transmission. This transmission system requires flexible transmission of some television signals and various data signals. A transmission system has been studied to send various digital signals commonly and efficiently through the same transmission channel, as follows.

Signal Multiplexing System

There are two multiplexing systems, the structure and packet systems. The former allocates services in fixed time multiplexing positions by control codes. The latter identifies services by the header added to the data and

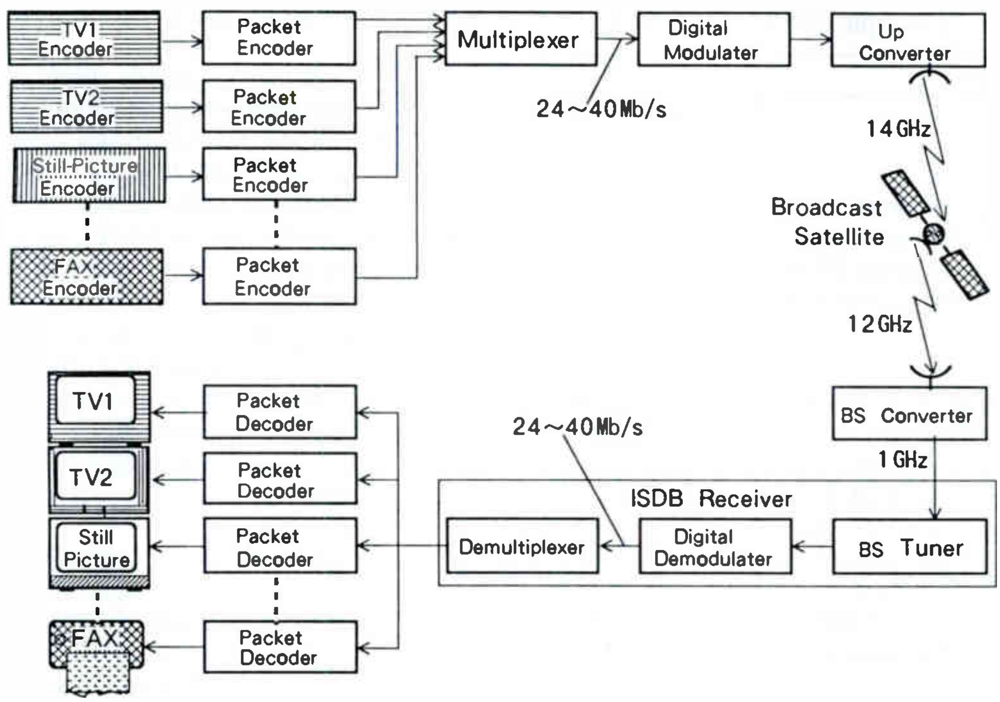


Fig.2 Block Diagram of ISDB Transmission and Reception System

does not fix multiplexing positions. Table 1 compares the two systems. Services sent in ISDB are numerous in types and many of them have variable transmission rates and different transmission characteristics. Under these conditions, the packet system, which excels in transmission efficiency, flexibility and expansibility, is suitable.

Packet Configuration

The items shown in Table 2 are contained in the header to be added to packets in broadcasting, and approximately forty bits are necessary. The packet length suiting television picture transmission, if this header is used, has been evaluated.

Table 1 Comparison of Signal Multiplexing Method

Comparison Item	Structure System	Packet System
System Requirement	Allocation configuration of all services must be decided by mode control.	Configuration by packet format can be decided independent of individual services.
Expansibility	Transmission parameter changes and new service additions are limited.	Transmission speed and other items can be changed freely and new services can be added relatively freely.
Operability	Program organization is free inside allocation configuration.	Transmission capacity has to be adjusted.
Reliability	Stable by acquisition through synchronization.	Header reliability depends on error correction.

Table 2 Example of Items in Header

Header Item	Bits
Operator ID	12 to
Program ID	12 to
Service ID	8 to
Continuity Index	2 to
Scramble Control	2
Data Transmission Flag	2

Once an encoding error occurs in television picture signals compressed by variable-length encoding, subsequent code strings cannot be decided correctly and are aborted. For this reason, picture field unit and stripe unit (e.g., horizontal picture block) are resynchronized as a data group to prevent propagation of error. Therefore, the mean data amount of data groups will be 117kbits in the field and 7.8kbits in the stripe if the television picture data transmission capacity is 7Mbps.

When an optimal packet length [P] is examined for such a data length [D], the header loss [a=H/P] by the header length [H] and mismatching loss [b=(P/2)/D: proportion of invalid data part of packets not filled by data to data length] are evaluated and the transmission efficiency (c=a+b) combining both is used as the decision criterion. Figure 3 shows the transmission efficiencies of packets of different lengths, regarding the data length. The diagram shows that approximately 2,000 bits are adequate as the packet length, with a high transmission efficiency for both the field and the stripe.

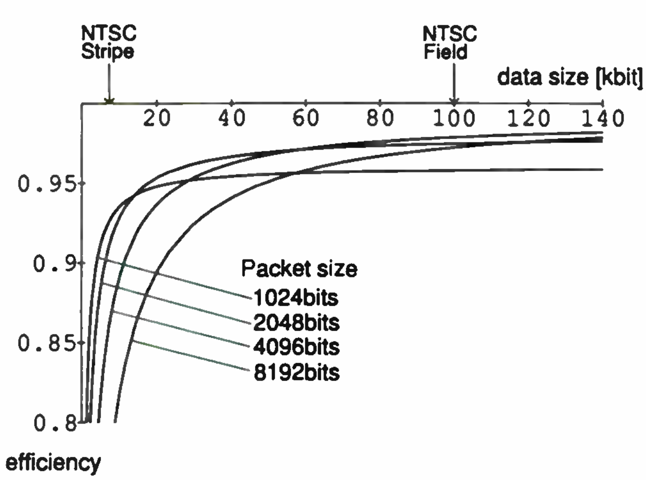


Fig.3 Efficiency of Various Packet Size as a Function of Data Size

Basic Transmission Format

The ISDB transmission signals should preferably meet the following transmission conditions:
1. Suitability to the hierarchies with international communication networks (e.g., 2.048Mbps)
2. Packets with a good transmission efficiency can be configured easily. Packet lengths of approximately 2000 digital bits are suitable lengths as mentioned above.
3. Expansibility to narrow and broad band channels other than 12GHz-band satellite broadcasting channels.
4. Packets, structures and various other multiplex signal configurations can be contained.
5. Ease of receiving small-capacity data services as well as large-capacity services such as digital TV.

Figure 4 shows the basic configuration of the ISDB transmission signal that incorporates these conditions into its system. One frame of ISDB is configured by structuring a 2048bits stream as the basis, in accordance with conditions 1. and 2. as the sector shown in Figure (a) and by interleave-multiplexing N number of sectors

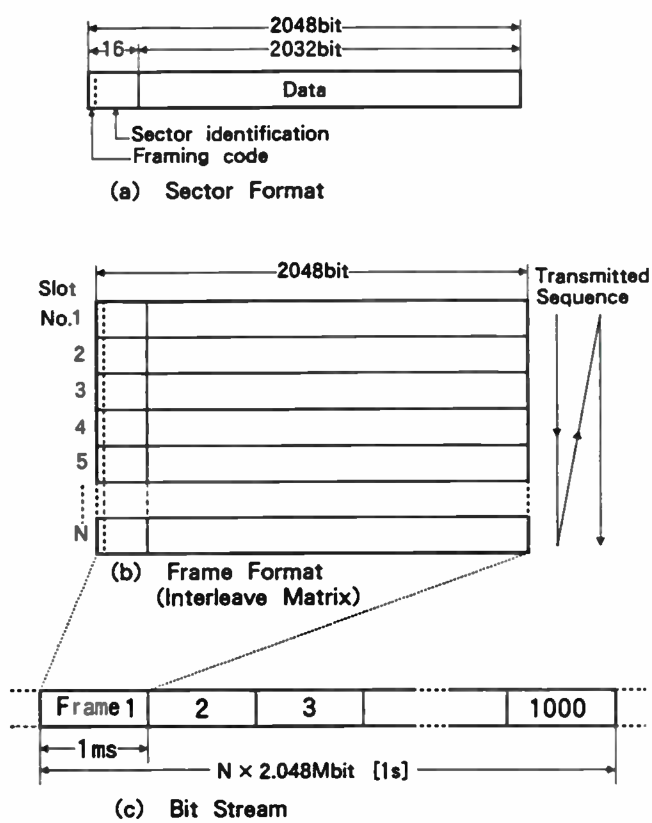


Fig.4 Basic Configuration of Signal Transmission Format

as shown in Figure (b). As shown in Figure (c), periods of 1ms are repeated in one frame and the transmission bit rate will become N · 2.048Mbps in accordance with the number N. By suitably setting N as explained above, ISDB transmission signals of a different capacity, shown in 3., can be configured. The first 16 bits of sectors become common parts as frame sync and sector ID codes. The remaining 2032 bits can be set freely. Therefore, structure and packet multiplex signals unique to ISDB can be set to meet the condition 4..

The concept of slots is used when sending a service by these transmission signals. One slot is a multiplexing position in a basic transmission unit of 2.048Mbps in the ISDB transmission signal. Each service is transmitted by designating a specified slot position in slots of number N. Various data services are small in capacity and are transmitted by designating one slot. Digital TV services with a large transmission capacity use some slots. Many services are mixed in one slot, so various data services are transmitted in vacant capacities of slots designated by digital TV services . If services are multiplexed in ISDB in the above mentioned large-capacity transmissions only in the concept of packets, all the packet headers to be transmitted must be identified at a high speed. However, by using the concept of slots, the speed is changed to a low speed sufficient for signal processing of each service after extracting slots needed for deinterleaving. Thus, services of a small capacity as mentioned in condition 5. can be received easily.

ISDB TRANSMISSION PARAMETERS

Satellite Link Parameters

Table 3 shows down link parameters when broadcasting signals are received by an antenna of 45cm in diameter and assuming the satellite output to be 120W, as in the BS-3 now in orbit in Japan. The table shows that receiving C/N during normal operation (referred to as "noise bandwidth 27MHz" in the following) would be 15.7dB. An appropriate transmission margin must be secured from this to design the ISDB system. For this reason, the receiving C/N values as boundaries of various services must be set by taking the time factor during effects of rain attenuation into account.

Table 3 Satellite Broadcasting Link Parameters

Parameters	Value
e. i. r. p.	59. 0 dB(120W)
Free Space Loss	-205. 6 dB
Rain Margin	- 2. 0 dB
Receiver G/T	10. 0 dB/K
Boltzman Constant	-228. 6 dBW/Hz·K
C/N (27MHz)	15. 7 dB

Transmission Bit Rate

Multi-channel PCM audio broadcasting is one example of broadcasting service development to digitally modulate carriers directly in 12GHz-band satellite broadcasting in Japan. In this case, transmission bit rates of 24 and 32Mbps were studied to equalize the service boundaries during low C/N as those of satellite TV sound.[1] 40Mbps is also possible if the range is satisfies the interference protection ratio conditions specified by WARC-BS.[2] Three transmission bit rates, namely, 24, 32 and 40Mbps, would be studied as ISDB systems. The use of easily produces QPSK and MSK for home-use receivers with a good efficiency will be studied as the digital modulation system.

Transmission Quality by Error Correction

In the digital modulation of satellite transmission described above, transmission quality deteriorates greatly during low C/N as it is, and a sufficient transmission margin cannot be secured. For this reason, error correction codes must be used to improve transmission quality. The following is required as error correction codes used in ISDB:

•Coding rate that sufficiently assures the service transmission capacity.

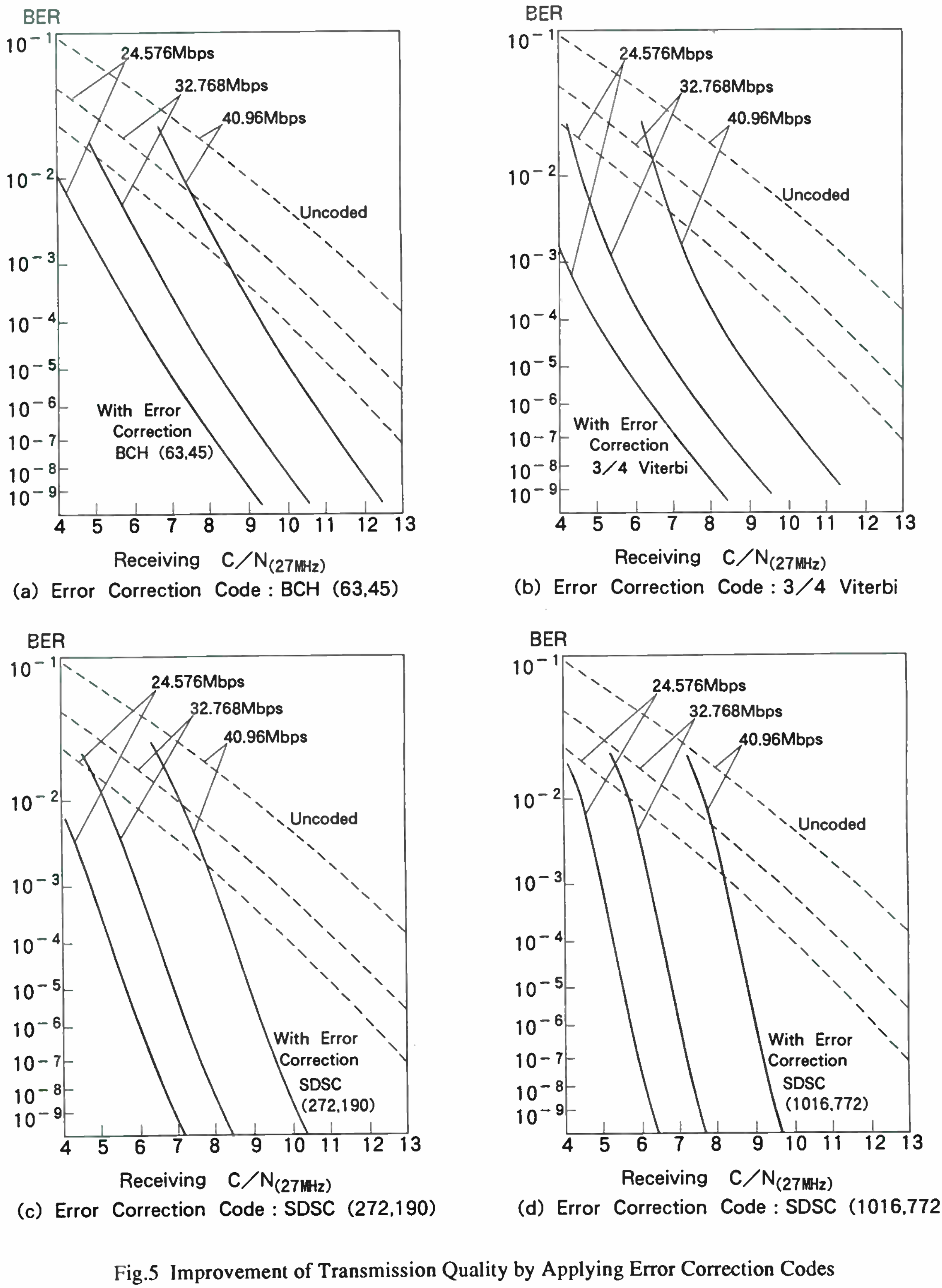
•Decoding circuit scale that can be obtained at low cost, suitable for a home receiver.

•Sufficient encoding gain that allows quality improvements.

The following codes were studied as error correction codes:

1. BCH (63, 45): BCH code with an encoding rate of 0.7.
2. 3/4 Viterbi: Convolutional code with an encoding rate of 0.75 is Viterbi soft-decision decoded.[3]
3. SDSC (272, 190): Shortened difference set cyclic code with an encoding rate of 0.7.[4]
4. SDSC (1016, 772): Difference set cyclic code with an encoding rate of 0.76 making shorter the (1057, 813) code by 41bits.[4]

Figure 5 relates the bit-error rate and receiving C/N when these four types of correction codes are applied to



(a) Error Correction Code : BCH (63,45)

(b) Error Correction Code : 3/4 Viterbi

(c) Error Correction Code : SDSC (272,190)

(d) Error Correction Code : SDSC (1016,772)

Fig.5 Improvement of Transmission Quality by Applying Error Correction Codes

12GHz-band satellite broadcasting transmission. The use of MSK was assumed in the diagram if the transmission bit rate was 24 or 32Mbps. One characteristic of MSK is that characteristic deterioration is small to the non-linearity of transmission paths. Based on past experimental results,[1] 2dB was included as deterioration from the theoretical value for home receivers which were mass produced. In the case of 40Mbps, the specification of interference protection ratio is forecasted as difficult to satisfy by MSK so the use of QPSK was assumed. Deterioration of 3dB from the theoretical value was included, in that deterioration increases by the non-linearity of the transmission path and the high demodulation clock rate.[2]

Digital TV Service Quality

At present, the picture encoding method for broadcasting has not been defined yet and the relationship between the bit-error rate and picture quality is not decided. Assuming the use of adaptive DCT as the encoding system, the relationship was estimated tentatively as shown in Table 4.[5] Based on the figure 5 and Table 4, the quality of digital TV is derived to be as shown in Figure 6 if receiving C/N lowers. The diagram also shows the quality of the existing FM-TV. These criteria are quality deterioration evaluation values in five stages. Compared with FM-TV, digital TV can maintain picture quality perfectly, even if the receiving C/N lowers, however the quality rapidly deteriorates as a cut-off characteristic once C/N lowers below a certain level.

Table 4 Assumed Video Quality Deterioration of Digital Television as to Bit-error Rate

Quality Deterioration Evaluation	Error Frequency per sec	Bit-error Rate
Detection Limit	1/30 times	3×10^{-9}
Allowable Limit	1 time	1×10^{-7}
Patience Limit	30 times	3×10^{-6}
Reception Limit	900 times	9×10^{-5}

Table 5 shows the annual time-rate when picture quality lowers to below the patience limits (evaluation 2.5) and annual interruptions that continue longer than one minute in the satellite transmission.[6] Using SDSC (1016, 772) for error correction, Table 5 assumes reception in high-rain areas in Japan. All three transmission bit rates have very low annual time-rates. The annual interruptions would become h/24 if actual viewing is h hours in a day, so that interruptions would

Table 5 Presumed Annual Time-Rate and Number of Interruption under Patience Limit

System BitRate	C/N	Time-rate	Number of Interruption
24Mbps	5.7dB	0.01%	13times
32Mbps	6.9dB	0.015%	18times
40Mbps	8.9dB	0.025%	30times

be confined to a few times a year. Thus, cases of actually being unable to view television would be rare, and digital TV is expected to provide high-quality service nearly all of the time when compared with FM-TV.

As mentioned earlier, an important item in service design of digital TV is whether or not service can be provided sufficiently, beginning with the time-rate which shows low C/N. A consideration not to deviate the digital TV service boundaries greatly from FM-TV must be made, referring to Figure 6. However, if an effort is made to maintain service to very low C/N, the transmission bit rate is lowered unnecessarily, making it difficult to meet the requirement of securing a large number of TV channels.

Number of Digital TV Transmission Channels

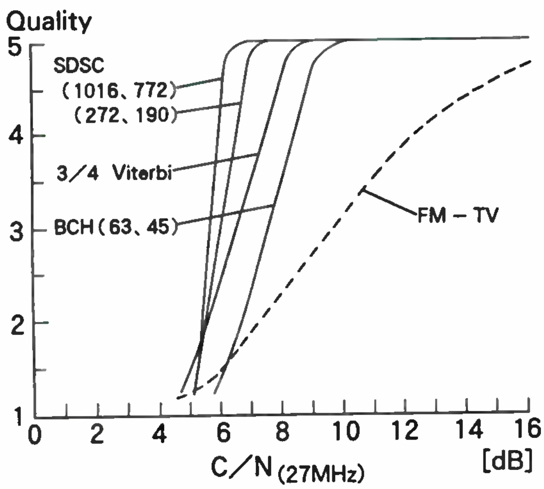
The number of TV channels that can send signals in one satellite channel varies in accordance with the transmission bit rate. Figure 7 shows the relationship between the picture encoding rate and the number of digital TV channels using the transmission bit rate as a parameter. Four sounds (each sound 128kbps) added to one TV channel and sync and control codes needed for transmission signals are included in the transmission bit rate, as shown in the following equation;

Transmission bit rate =

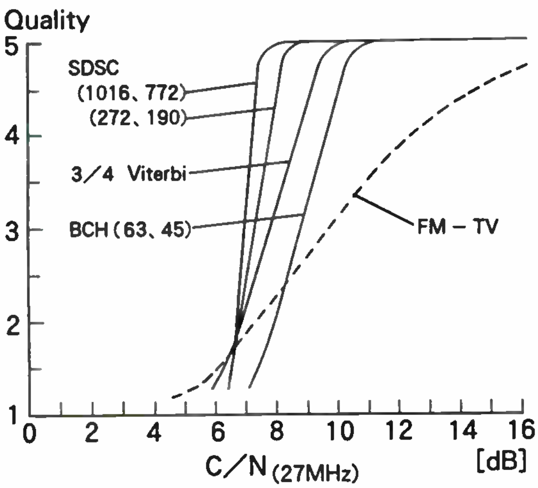
{(picture+sound×4)×TVch}/coding rate+sync+control

The picture encoding rate is calculated using SDSC (1016, 772) with an encoding rate of 0.76 as the error correction code.

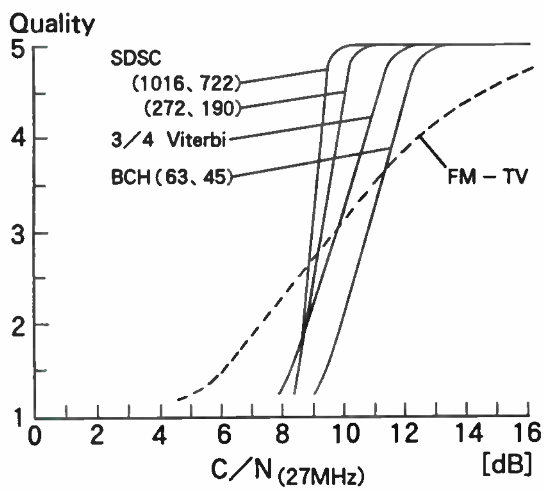
The diagram shows that the picture encoding rate will be approximately 7Mbps on an average when three TV channels are broadcast at the transmission bit rate of 32Mbps. Similarly, the picture encoding rate must be approximately 6.5Mbps on an average if four TV channels are to be broadcast at the transmission bit rate of 40Mbps. A picture encoding rate that satisfies broadcasting picture quality has not been determined yet. The number of TV transmission channels will vary in accordance with the development of picture encoding.



(a) Transmission Bit－rate : 24Mbps



(b) Transmission Bit-rate : 32Mbps



(c) Transmission Bit-rate : 40Mbps

Fig.6 Presumed Video Quality as a Function of Receiving C/N

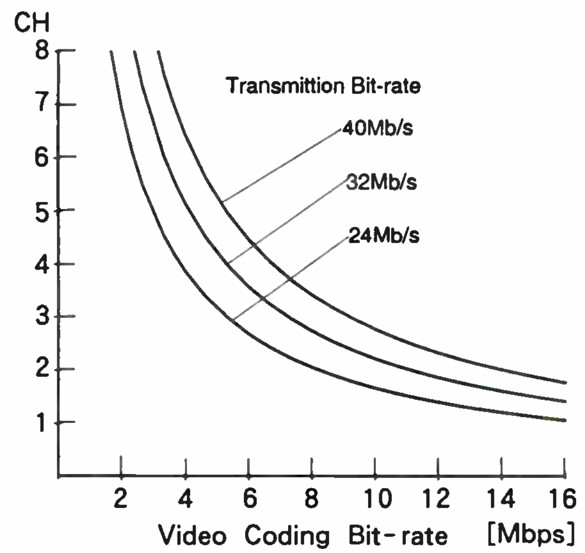


Fig.7 Number of TV Channels as a Function of Video Coding Rate

Transmission Quality of Other Services

In addition to digital TV, ISDB also transmits various digital services. These services are not only provided independently, but are also provided as additions to TV or as multi-media. Care must be exercised so that quality of individual services does not differ greatly when C/N is low. Table 6 shows the bit-error rate estimated to be the deterioration detection limits of various services and the receiving C/Ns for them. The table is an example of when the transmission bit rate is 32Mbps and SDSC (1016, 772) code is used as the error correction code. The table shows that the receiving C/N corresponding to the detection limits of the various services is within a narrow range of 6.8 to 7.4dB and that there are no large differences in the service boundaries.

Table 6 Required Bit-error Rate as to Detection Limit for Various Digital Services

Services	Bit-error Rate	C/N
Teletext	1×10-8	7. 3dB
Facsimile	1×10-5	6. 8dB
Still － Picture	1×10-6	7. 0dB
PCM Sound	4×10-6	6. 9dB
Digital TV	3×10-9	7. 4dB

SUMMARY

ISDB is expected to be the infrastructure of new broadcasting that can supply high and diverse programs to homes at low cost. It has a flexible transmission structure with expansibility and is promising as a future medium that can integrate various digital broadcasting services.

In ISDB of 12GHz-band digital satellite broadcasting, various digital services can be transmitted efficiently, achieving effective utilization of radio wave resources. In particular, broadcasting digital TV can offer more channels than FM-TV. The number of digital TV channels is decided by the picture encoding rate, which concerns picture quality, and by the transmission bit rate . The transmission bit rate must be decided taking account of the service boundaries when the C/N is low and the time-rates during this time.
The basic configuration of the ISDB transmission signal for expansion into various transmission channels has been shown. The configuration of multiplexing signals inside sectors suiting digital TV and various data services will be studied in the future.

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RESULTS OF FIELD TESTS OF GHOST CANCELING SYSTEMS FOR NTSC TELEVISION BROADCASTING

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ABSTRACT

In the past decade, significant progress has been achieved toward developing technologies to reduce the visual effects of multipath propagation and eliminating ghosts. This effort has culminated in the design and development of five different ghost canceling systems for consideration for a single voluntary standard for the U.S. television industry. This document reports on a recently completed field test measurement program in the Washington D.C. urban area designed to evaluate these five competing ghost canceling systems.

All five systems were effective in reducing and/or eliminating ghosts. The performance, however, varied significantly from system to system and depended to some extent on the transmitting frequency (VHF or UHF), the type and complexity of the ghosting condition, and the received signal level. One system, the Philips system, consistently exhibited superior performance relative to the other four systems.

INTRODUCTION

Multipath distortion, or ghosting, has been a pernicious problem with television transmission since the beginning of television broadcast service. While many improvements to the television broadcasting system have been implemented over the years, the degraded images associated with multipath ghosting have not diminished and ghosting continues to reign as the most annoying impairment of over-the-air television transmission.

A standardization program is underway within the television industry to evaluate competing ghost canceling systems and develop a single voluntary standard for the broadcast, cable and consumer electronics industries. The field tests of proponent systems reported here are the most critical element in that evaluation. Only in the field can the effectiveness of ghost canceling systems be evaluated under the actual conditions that will exist in the marketplace. As shown by the results of these tests, the technology to successfully implement ghost canceling in the current television broadcasting service clearly exists. It is hoped that this data will aid in reaching the important goal of achieving consensus within the broadcast, consumer electronics and cable industries on a preferred ghost canceling system.

BACKGROUND

Theoretical descriptions of ghost canceling systems have existed for many years. However, until the present program was undertaken, the Broadcast Technology Association (BTA) of Japan ghost canceling system, which was adopted as a standard in Japan in 1987, represented the only functioning system that had been publicly demonstrated. The systems that participated in the field test program were submitted in response to a Request for Proposal that was released by NAB in July, 1990. Field testing is the most appropriate decisional criteria for selecting a preferred system, since the plethora of ghosting conditions experienced in the field cannot be easily duplicated in the laboratory. By assessing the performance of the ghost canceling systems at a large number of diverse locations in the field, the ghost canceling systems were subjected to a wide range of ghosting impairment situations.

Five proponent organizations submitted functional systems in September, 1991 for participation in the NAB ghost canceling field tests as follows:

1. AT&T/Zenith Electronics Corporation
2. Broadcast Technology Association (BTA) of Japan
3. David Sarnoff Research Center/Thomson Consumer Electronics
4. Philips Laboratories
5. Samsung Electronics

Implementing each of these systems requires the insertion of a Ghost Canceling Reference (GCR) signal in the vertical blanking interval of the broadcast television transmission. Only the GCR signal is under consideration for standardization. Receiver manufacturers then could choose to process the signal in any desired way, encouraging products that balance manufacturing costs with customer requirements. The GCR signals of the proponents are shown in Figures 1 through 5. The waveforms shown for each proponent GCR are presented in a four or eight field sequence. Complete descriptions of the proponent systems can be found in References 2-7.

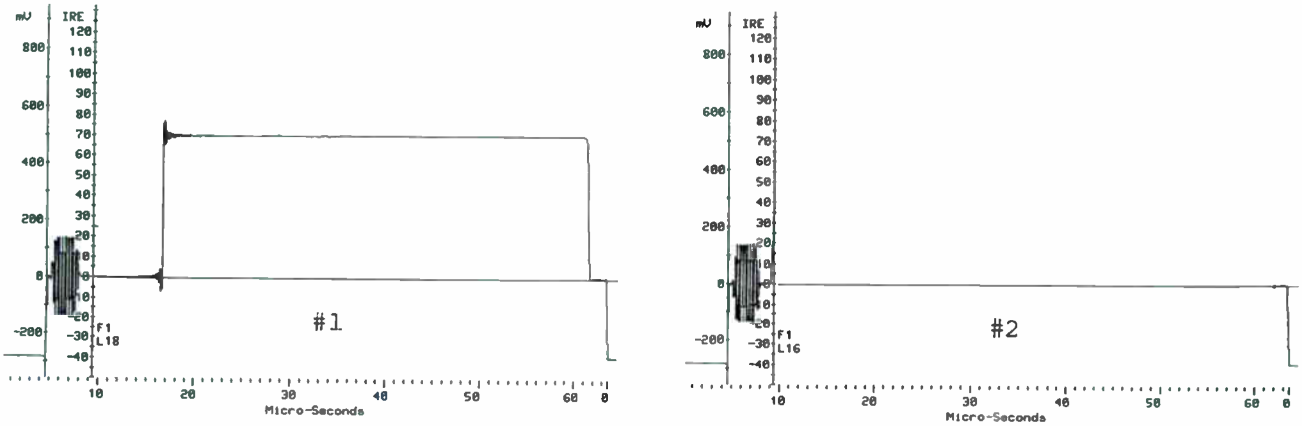

Figure 1
BTA GCR Waveforms

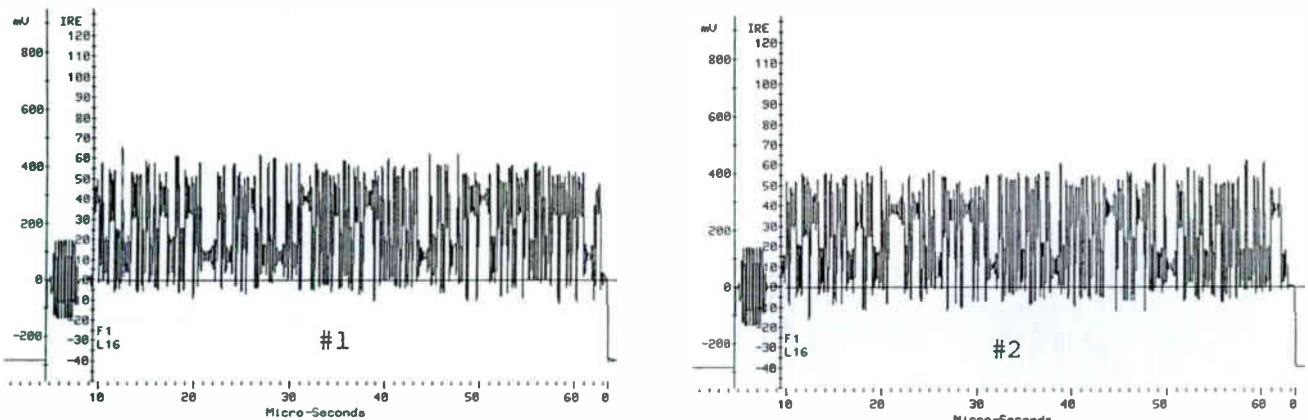

Figure 2
AT&T/Zenith GCR Waveforms

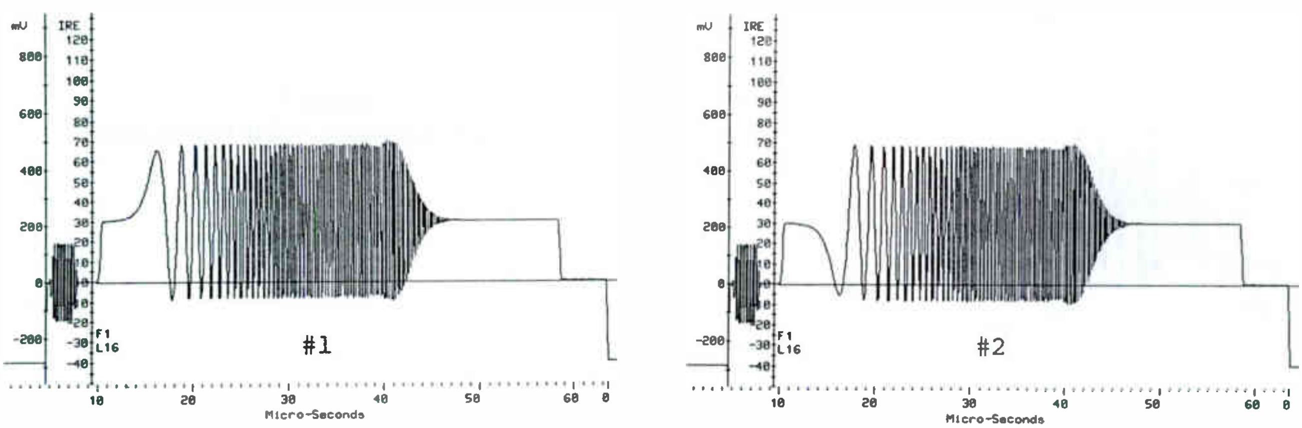

Figure 3
Philips Laboratories GCR Waveforms

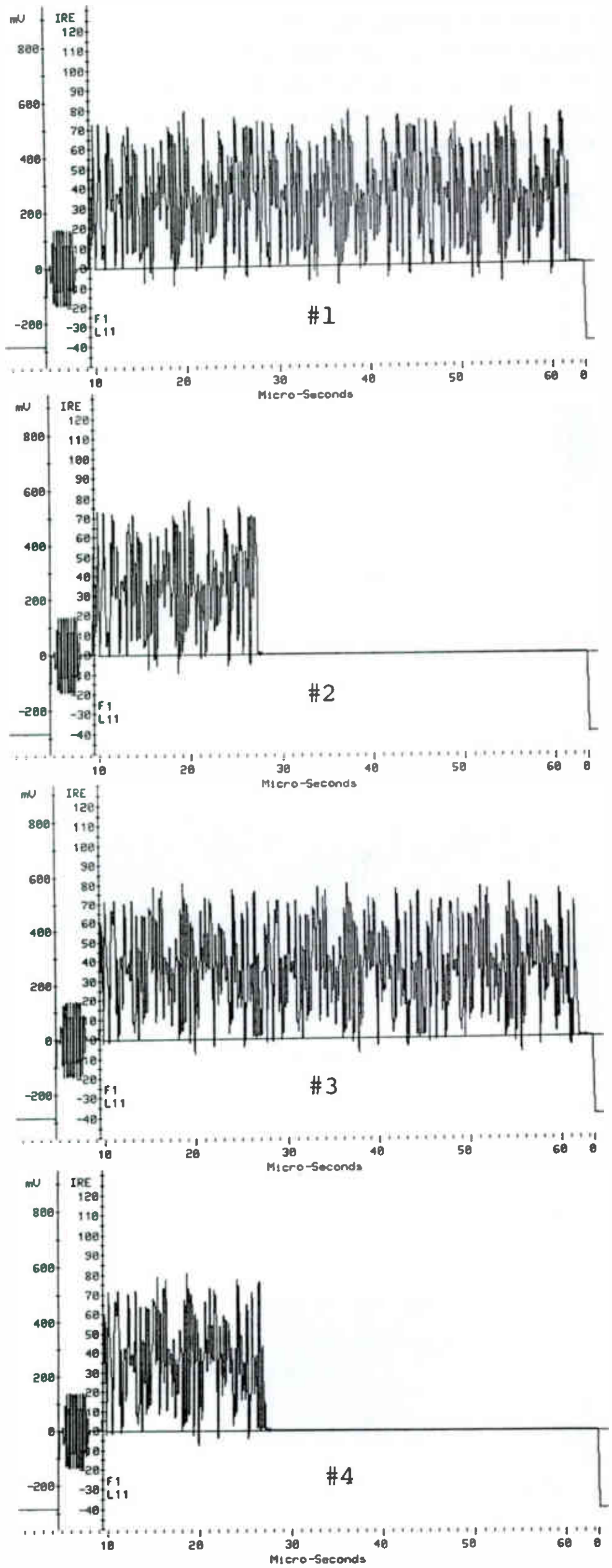

Figure 4
Sarnoff/Thomson GCR Waveforms

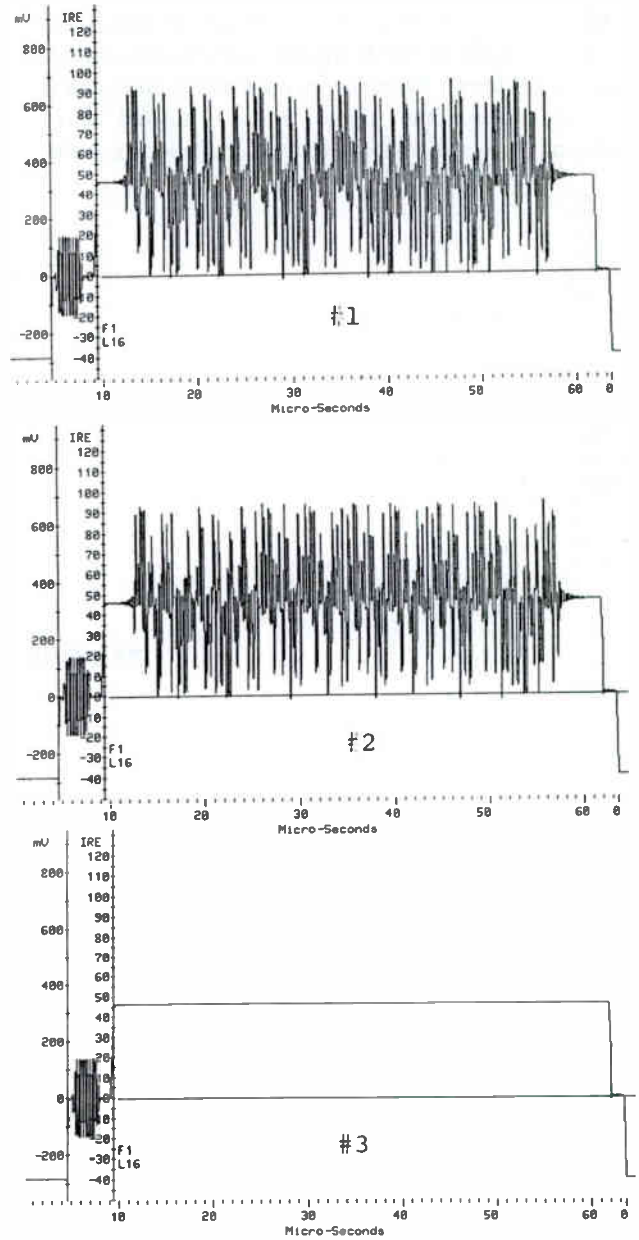

Figure 5
Samsung Electronics GCR Waveforms

THE FIELD TEST PLAN

The test plan was developed by technical representatives from Capital Cities/ABC, ATTC, CableLabs, CBS, EIA, MSTV, NAB and NBC.

The field tests took place in the Washington D.C. area between September 27 and November 8, 1991. Three local stations participated in the field tests; one VHF station (WRC-TV, channel 4) one low UHF station (WDCA-TV, channel 20) and one high UHF station (WFTY, channel 50). Due to the mild Washington climate, almost all of the summer tree foliage remained in place for the duration of the test period.

NAB served as project manager for the field test effort. The Electronic Industries Association (EIA), the Association for Maximum Service Television (MSTV) and Cable Television Laboratories (CableLabs) contributed to the direct funding of the project and the Public Broadcasting Service (PBS) contributed equipment and technical support. The Carl T. Jones Corporation was contracted to conduct the field tests using their field van. This contractor had previously conducted ghost canceling field tests on the BTA ghost canceling system for NAB in Atlanta in 1990 (see Ref. 1) and was familiar with the complexities of such an effort. The analysis of the data was undertaken jointly by NAB and MSTV.

Site Selection

One hundred and six locations were selected for measurement sites. Sites were specifically chosen in areas where a variety of ghosting conditions might be found. The locations were chosen to represent two general conditions: strong signal areas (less than 15 miles from the transmitters) and weak signal areas. Approximately 70% of the selected sites were in strong signal areas and 30% fell in the weak signal category. For the strong signal areas, fourteen areas were identified where a variety of ghosting conditions might be found including highly urbanized and built-up areas, areas with varying terrain and locations likely to be affected by clusters of tall buildings. For each area, or "cluster," a central site and four other sites on equally spaced 1/2 mile radials from the central point were identified, subject to accessibility of those sites in the field van. The close-in cluster sites are shown in Figure 6. In the weak signal category, six radials from the transmitter sites were identified with terrain profiles that suggested multipath conditions might be present. Five measurement sites were identified along each radial, starting at about the 35 mile point from the transmitters, with the other measurements being taken every mile or so closer to the transmitters. If a viewable picture did not exist at the 35 mile point, the van moved closer to the transmitters along the radial until passable viewing conditions were found. The radial site

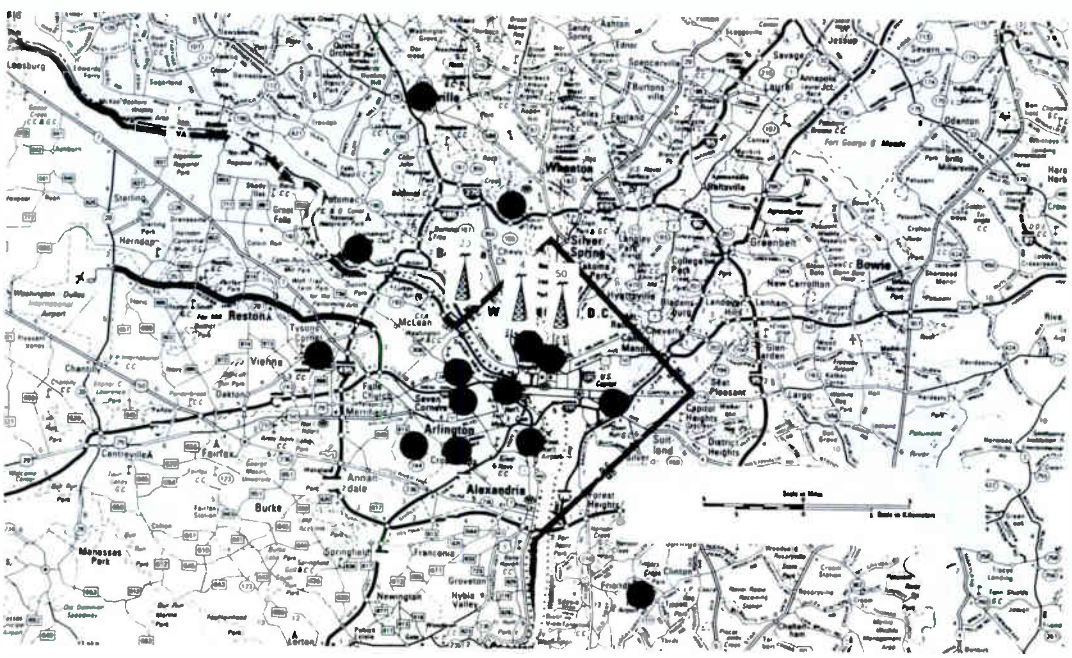

Figure 6

Close-In "Cluster" Measurement Sites

locations are shown in Figure 7. Figure 8 shows a photograph of the field van at a typical site location.

It is important to note that the measurement sites were chosen based on the likelihood of encountering a multitude of ghosting conditions. The data from the present study is not indicative of the level of ghosting that would be encountered on a statistical basis throughout the urban area.

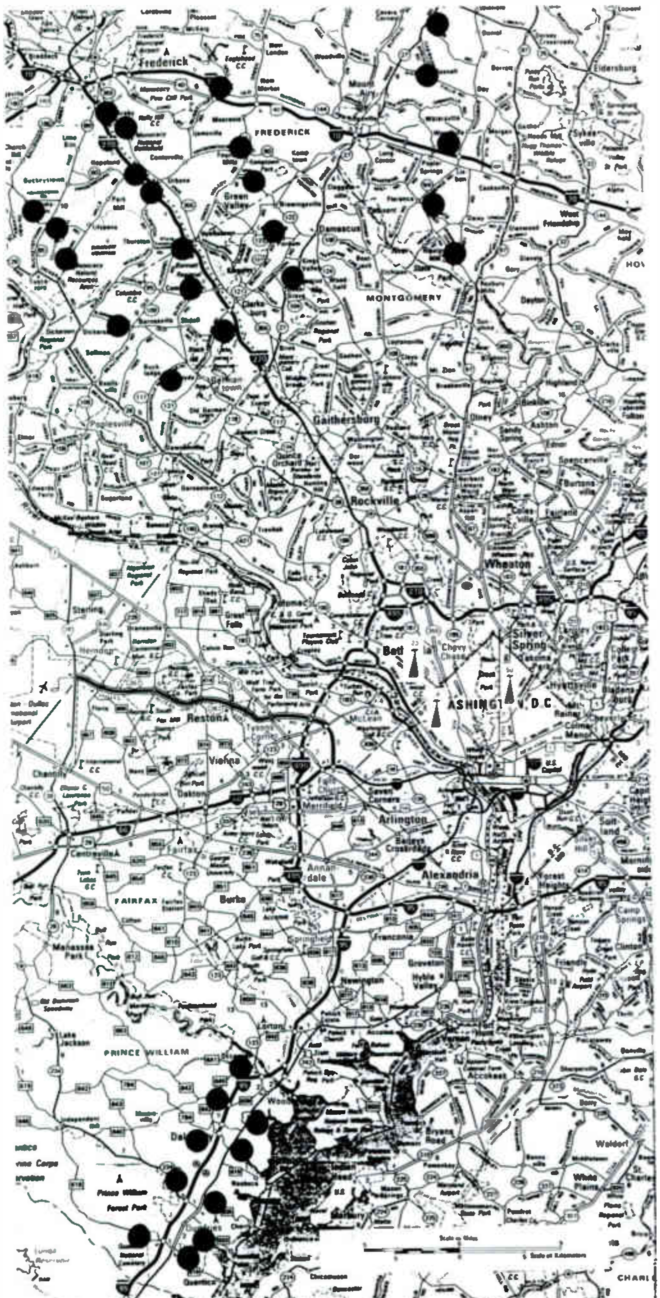

Figure 7

Far-out Radial Measurement Sites

Figure 8

Field Van at Typical Site

Hardware

Field Van Equipment. The equipment complement included an AC power generator, 30-foot mast, field strength meter, a Radio Shack VU-75 antenna, two Tektronix model 1450 television demodulators (one configured with a VHF module and one with a UHF module), the five proponent ghost canceling systems, a video switcher, a Tektronix model 1480-R video waveform monitor, a Tektronix VM700 video measurement system, Hewlett Packard Laserjet printer, a Toshiba
S-VHS video cassette recorder, two Samsung TC 9845S 20" video monitors, a microphone and other miscellaneous equipment. Figure 9 shows a block diagram of the measurement system. The ghost canceling decoders were all prototype equipment, with the exception of the BTA unit which was a laboratory unit developed by NHK and NEC. The Tektronix 1450 was used as a common demodulator for all the ghosting decoders. Each decoder accepted a baseband video input. In addition, the Sarnoff unit required both the I (in-phase) and Q (quadrature phase) outputs of the demodulator.

GCR Insertion Equipment. Three Tektronix 1910 generators were used at each of the three stations. The Tektronix 1910 is a digital signal generator that has the capability to add new test signals via the insertion of appropriately configured PROMs. (The BTA GCR is currently available as an option on the Tektronix 1910). Tektronix, in coordination with proponents, fabricated PROM sets containing all five GCR signals for installation in the three Tektronix 1910 generators which were then installed at the stations.

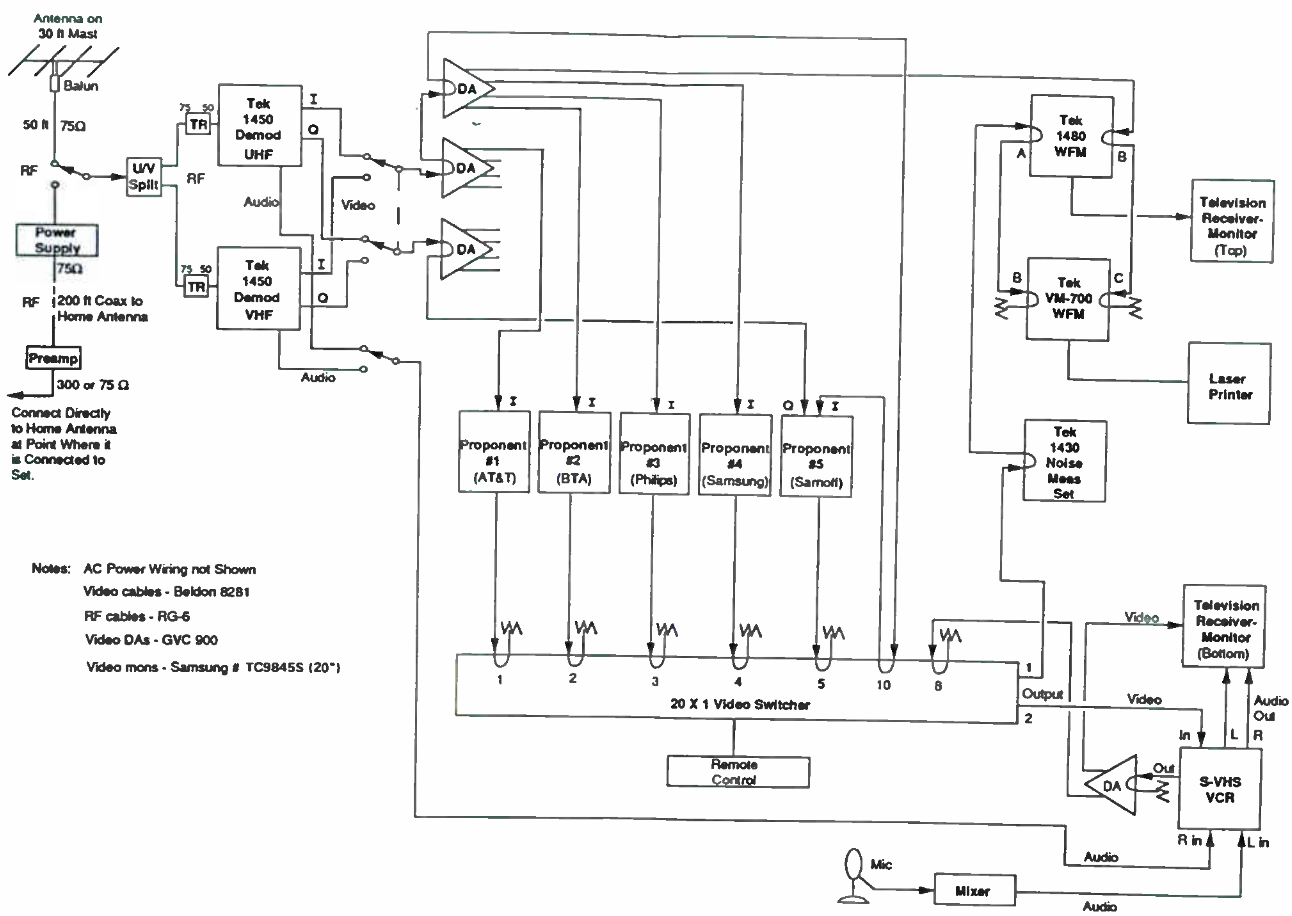

Figure 9

Field Van Measurement System

Proponent GCR signals were placed in the VBI on the following line numbers (as requested by proponents):

System	VBI line
AT&T/Zenith	16
BTA	18
Philips	16
Samsung	16
Sarnoff/Thomson	11

Activation of the proponent GCRs was facilitated through the use of a small laptop computer installed in the field van and interfaced to a cellular telephone. At the transmitter sites, a phone line and an autoanswer modem were installed and interfaced to the remote control serial port of the 1910 generator. Using standard off-the-shelf communications software, script files were written that allowed the field team to activate the desired GCR via a menu.

Methodology and Procedure

The five proponent ghost canceling decoders were installed in the field van. The majority of measurements were made using a mast-mounted antenna attached to the vehicle. Data from a number of indoor locations was also gathered. In this case, home antennas were connected to the field van equipment via a front end preamp and coaxial cable.

The field van visited a given measurement site only once. While at a particular site, the performance of all five systems on all three television stations was assessed. This was desirable so that, for each channel, propagation conditions were similar for all systems. Also, since each system was tested sequentially in time, the program material on a given channel was often very similar. While the objectionability of a given condition of ghosting may be somewhat dependent on the source material being viewed, every effort was made to insure that the evaluation for all five systems was made during the same program or with similar scene content.

At each outdoor measurement site the following data was collected: RF signal level at the input to the television receiver (calibrated to obtain field strength readings at the antenna); photographs of the general vicinity of the observation location showing reception conditions; waveform monitor plots of a standard multiburst test signal before and after correction; subjective evaluation of quality level and degree of ghosting impairment before and after correction; and video tapes of the observed signals for off-line analysis. The video tapes were subsequently used to validate the judgments of the two expert observers.

Evaluation consisted principally of visual evaluation of the received television picture by two expert observers rating the overall picture quality and level of ghost impairment before and after insertion of the ghost cancelers. Both observers had extensive experience in critical technical assessment of video picture quality. The same expert observers were used throughout the tests. The visual evaluation of each received picture was based on the subjective quality and impairment scales specified by the International Radio Consultative Committee (CCIR) (Ref. 8). The five-grade CCIR scales are as follows:

CCIR Quality Scale	CCIR Impairment Scale
5 Excellent	5 Imperceptible
4 Good	4 Perceptible, but Not Annoying
3 Fair	3 Slightly Annoying
2 Poor	2 Annoying
1 Bad	1 Very Annoying

To rate the signal quality and impairment level, the two expert observers each evaluated the image and arrived at a rating by consensus. Differences of opinion were discussed, and if no agreement was reached, the average of the two opinions was recorded. Both quality and impairment assessments were made to the nearest half step on the five point scale, an acceptable practice in CCIR studies. This was done to reveal instances where improvements were evident but did not result in a full grade of improvement on the CCIR scales.

At each site, specific characteristics of the ghosting impairments were identified. The delay ranges in which ghosts were discernible were noted for the following categories (microseconds of delay of ghosted signal relative to main signal): -9 to -2, -2 to 0, 0 to 2, 2 to 10, 10 to 19, 19 to 28, 28 to 37, greater than 37. The number of simultaneous ghosts was also noted as follows: 1, 2, 3, 4, 5-8, greater than 8. Ghosts were also catalogued as to whether they were distinct images, exhibited a diffuse appearance or both. In addition, ghosts were categorized as to whether they exhibited a time variable nature. This was determined by careful examination of the impaired image in conjunction with observing the demodulated GCR signal on a waveform monitor for a period of one to two minutes. If the GCR waveform exhibited visible multipath-induced changes within two seconds, the ghosting condition was identified as being time variable.

At each site the antenna was raised to a height of 30 feet above ground and was then rotated toward each station in turn and the signal was observed on the three channels. If no ghosts were observed, the antenna was rotated to ascertain if ghosts could be produced. If no ghosts were evident, the van was moved to a new location.

With the desired channel being received, subjective judgments of quality and impairment conditions were rendered and recorded. The first proponent ghost canceler was then reset and an appropriate time period was allowed for the equipment to perform the ghost reduction process. The VCR was then used to record a segment of the received signal. An S-VHS VCR was used to accurately record the full bandwidth of the video signal and preserved the nature of even very subtle ghosting effects. During the recording the canceler was turned off and on, or periodically bypassed, to more readily show the effect of the ghost canceler. The subjective quality and impairment assessments were then made. This process was then repeated for the other systems. After all systems were tested, the sequence was repeated for the second and third channels.

Test Management Policy

Interface with proponents before and during the course of testing was handled in a structured manner. Prior to the start of testing, the GCR signals, as generated by the Tektronix 1910 unit, were demonstrated for proponents who then verified that the GCR signal generation was accurate. Proponents were also invited (but not required) to participate in installing the ghost canceling decoders in the field van. All proponents (with the exception of BTA, whose unit required no special alignment) were active in installing their own equipment. After the GCR generators were installed at the television stations, several days were made available for proponents to adjust and align their systems using the field van equipment configuration and the broadcast transmissions with the embedded GCR signals.

During testing, proponents were allowed to visit the field van and observe the performance of the ghost cancelers and the testing process. Permission for such visits was based on adequate advanced notice and was subject to revocation if the schedule became compromised or if the presence of proponents proved to be disruptive to the measurement team. In practice, there were no serious problems.

When any of the ghost cancelers failed, either partially or completely, or when a unit's performance suddenly degraded, evaluation of that unit was suspended and the proponent was immediately notified. Access to the field van for maintenance or repair of proponent equipment was made available at night and on weekends. Testing was not halted for proponent equipment failures. Several incidents of this nature occurred and proponents generally responded rapidly to remedy any hardware problems.

Modifications or improvements to proponent ghost decoding equipment was permitted, subject to the constraint of requiring the work to be done at night or on weekends in order to avoid schedule delays. Sarnoff, AT&T and Samsung instituted hardware and/or software changes to their decoders during the testing period. Changes to GCR signals were not permitted.

DATA ANALYSIS

Overall Measurements

As previously mentioned, a total of 106 measurement locations were selected for this measurement program. Data was collected from all five systems on three television stations (one VHF and two UHF) for a total of approximately 1500 field strength measurements and around the same number of subjective observation sequences. An observation sequence encompassed both a quality and impairment evaluation of the received picture before and after the activation of the ghost canceler, as well as a judgment of specific characteristics of ghosting impairments.

Approximately 70% of the measured locations were close-in measurements (less than 15 miles from the transmitters) where strong or moderate signal conditions are normally encountered. The close-in measurements, referred to herein as "cluster" measurements, were collected to analyze and evaluate specific characteristics of ghosting impairments such as the performance of the canceler under a range of simultaneous ghosts, different time delay ranges, and whether the ghosting condition was static or time varying and determine the general performance of all five systems under similar testing conditions. The remaining 30% of the measurement sites were selected along six different far-out radials, referred to herein as "radial" measurements, for the purpose of assessing the performance of these systems under weak signal conditions. In addition, a number of measurements were gathered at several homes using indoor antennas. While this data is useful and interesting, not enough indoor sites were examined to be able to generalize about the overall performance of the systems in indoor situations.

To assess the overall quality of the received pictures under investigation, subjective evaluation of picture quality (the experts were told to ignore any ghosting impairments in the quality assessment) before and after the activation of the ghost canceler was performed. This exercise, while not directly used in the analysis, was useful during the measurement phase of the project in preparing the observers for the impairment evaluation. Prior to the activation of the ghost canceler, approximately two-thirds of the observations were rated as "good" or "excellent" in quality; the remaining one-third were rated as either "fair", "poor" or "bad" for all three channels. The percentage did not significantly improve after the activation of the canceler.

All 106 measurement locations experienced some level of ghosting impairment. The type and complexity of the ghosting impairment, however, varied significantly from location to location. Ghosts were visible on all three channels and at all measured locations. Table 1 presents the impairment statistics for all three channels prior to the activation of the ghost canceler. The combined cluster and radial measurements were included in the tabulation of the impairment statistics.

Note that none of the systems tested exhibited a processing improvement on all of the ghosting conditions encountered in the field. In fact, all five systems encountered at least some ghosting conditions where degradation of the impaired image was observed when the ghost canceler was activated. Also, it should be noted that the above statistics do not include data relating to the number of times the ghost canceler failed to operate or produced a totally incoherent image at the ghost canceler's output. Both the AT&T and Samsung systems exhibited failure conditions of this type on all three channels for approximately 5% of the observations.

To assess the general performance of the five systems under test, the difference in the CCIR impairment scale before and after the activation of the ghost canceler was tabulated for all the cluster and radial measurement locations. For each channel, an Average Improvement Index (AII) for each system was determined by adding the difference in the CCIR impairment scale of all the observations made on that channel and dividing it by the total number of locations. In addition, the average CCIR impairment value before the activation of the ghost canceler for each channel was also determined. This value, along with the information in Table 1, is useful in determining the relative degree of impairment prior to activation of the ghost canceler between the three channels. Table 3 presents the Average Improvement Index for each system for all three television channels along with the average impairment value for each channel.

CCIR Impairment Scale	Ch. 4 (%)	Ch. 20 (%)	Ch. 50 (%)
5 Imperceptible	0	0	0
4 Perceptible, but Not Annoying	14	11	5
3 Slightly Annoying	45	36	44
2 Annoying	28	34	43
1 Very Annoying	13	19	8

TABLE 1
Overall Impairment Assessment of Measurement Sites
(All sites)

All five systems were effective in reducing and/or eliminating ghosts. The degree of effectiveness, however, varied from system to system and depended to some extent on the transmitting frequency (VHF or UHF), the type and complexity of the ghosting condition, and the received signal level. Table 2 presents the percentage of observations where the ghost canceler degraded, did not change or improved the impaired image for all five systems. Here again, both cluster and radial measurements are included in the tabulation.

A review of the statistics in Table 3 reveals that the Average Improvement Index is highest for the Philips system, followed by BTA, Sarnoff, Samsung and AT&T.

System	Ch. 4 %	Ch. 20 %	Ch. 50 %
BTA			
Degrade	5	0	1
No change	14	14	14
Improve	81	86	85
Samsung			
Degrade	7	4	9
No change	59	60	60
Improve	34	36	31
AT&T/Zenith			
Degrade	35	27	41
No change	37	28	36
Improve	28	46	23
Philips			
Degrade	0	1	1
No change	5	10	3
Improve	95	89	96
Sarnoff/Thomson			
Degrade	1	3	4
No change	12	19	17
Improve	87	78	79

TABLE 2
Overall Effect of Ghost Canceling Systems on Observed Impairments
(All sites)

	Ch. 4	Ch. 20	Ch. 50
Average impairment before activation of ghost canceler	2.7	2.6	2.5
System	Average Improvement Index		
BTA	1.20	1.21	1.00
Samsung	.23	.34	.13
AT&T/Zenith	-.06	.32	-.20
Philips	1.54	1.49	1.52
Sarnoff/Thomson	.91	.85	.88

TABLE 3
Average Impairment and Average Improvement Index
(All sites)

The data also reveals that the Average Improvement Index was nearly the same on all three channels for the Philips and Sarnoff systems. However, BTA, Samsung and AT&T showed a slight decrease in the AII value for Channel 50.

Cluster Measurements

As noted earlier, the cluster measurements were primarily collected to evaluate the performance of the five systems for different ghosting conditions. Specifically, the performance of the ghost canceler for each system was evaluated with respect to the following categories of ghosting conditions: 1) static (non-varying) or time-varying ghosting conditions; 2) ghosts that are distinct replicas of the original signal, a smeared or diffused version of the original or both distinct and diffuse appearances; 3) the number of simultaneous ghosts in the impaired image; and 4) different delay ranges of the ghosted signal relative to the main signal.

Table 4 presents statistics relating to the effectiveness of the ghost canceler in reducing and/or eliminating ghosts for the cluster measurements; i.e., strong and moderate signal conditions. The table shows the percentage of observations that were not changed, degraded or improved after activation of the ghost canceler for all the systems tested.

Here again, none of the systems tested exhibited a processing improvement on all the ghosting conditions encountered in the field. Philips exhibited the lowest image degradation percentage -- less than 1% on all three channels -- followed by BTA, Sarnoff, Samsung, and AT&T.

Table 5 presents the Average Improvement Index for each system for the cluster measurements, along with average impairment value for each channel before activation of the ghost canceler. The data again reveals that the Philips system has the highest Average Improvement Index

System	Ch. 4 %	Ch. 20 %	Ch. 50 %
BTA			
Degrade	5	0	2
No change	8	10	8
Improve	87	90	90
Samsung			
Degrade	5	2	10
No change	50	54	56
Improve	45	44	34
AT&T/Zenith			
Degrade	25	20	38
No change	40	27	34
Improve	35	53	28
Philips			
Degrade	0	1	1
No change	3	10	3
Improve	97	89	96
Sarnoff/Thomson			
Degrade	2	4	6
No change	10	16	14
Improve	88	80	80

TABLE 4
Overall Effect of Ghost Canceling Systems on Observed Impairments
(Cluster sites only)

followed by BTA, Sarnoff, Samsung and AT&T.

Table 6 presents the percentage of observations which exhibited static (non-varying) ghosting conditions along with changing or time-varying ghosting conditions, categorized by channel. As expected, the data shows that the percentage of non-varying ghosts is much greater for VHF than UHF frequencies. Furthermore, the data supports the supposition that the higher the frequency, the greater the likelihood of encountering time-varying or changing ghosts.

Table 7 shows the Average Improvement Index for each system on all three channels for both static and time-varying ghosts.

A review of the data in Table 7 reveals that the Philips system has the highest Average Improvement Index for both the time-varying and static ghosting conditions, followed by BTA, Sarnoff, Samsung and AT&T. The data also reveals that the Philips and Sarnoff systems have a higher Average Improvement Index for time-varying ghosts than for static ghosts. The data supports comments recorded by the expert observers during the measurement phase which indicated that both systems adequately tracked and corrected time-varying ghosts.

Table 8 shows the percentage of observations for each channel that exhibited ghosts that were distinct replicas of the original image, smeared or diffused versions of the original image or a combination of both distinct and diffuse appearances. The data shows that approximately one-third of the observations exhibited distinct appearances, while more than half of the observations contained both distinct and smeared ghosts. The data further reveals that the type and appearance of this ghosting condition was about the same for VHF and UHF and is therefore a frequency independent effect.

	Ch. 4	Ch. 20	Ch. 50
Average impairment before activation of ghost canceler	2.5	2.5	2.5
System	Average Improvement Index		
BTA	1.44	1.33	1.15
Samsung	.33	.39	.17
AT&T/Zenith	.13	.43	-.15
Philips	1.80	1.52	1.62
Sarnoff/Thomson	.99	.91	.85

TABLE 5
Average Impairment and Average Improvement Index
(Cluster sites only)

	Ch.4 (%)	Ch. 20 (%)	Ch. 50 (%)
Static or non-varying ghosts	81	37	14
Changing or time-varying ghosts	19	63	86

TABLE 6
Relative Presence of Static Vs. Time-Varying Ghosts
(Cluster sites only)

System/State	Average Improvement Index		
	Ch. 4	Ch. 20	Ch. 50
BTA Non-varying Time varying	 1.61 1.08	 1.48 1.23	 1.40 1.06
Samsung Non-varying Time varying	 .33 .17	 .39 .39	 .06 .17
AT&T/Zenith Non-varying Time varying	 .10 .18	 .27 .52	 -.11 -.15
Philips Non-varying Time varying	 1.76 2.00	 1.40 1.65	 1.44 1.65
Sarnoff/Thomson Non-varying Time varying	 .86 1.08	 .80 .98	 .83 .92

TABLE 7
System Performance for Static and Time-Varying Ghosting Conditions
(Cluster sites only)

Ghost Type	Ch.4 (%)	Ch. 20 (%)	Ch. 50 (%)
Distinct	34	36	36
Diffuse	15	5	11
Both	51	59	53

TABLE 8
Relative Presence of Distinct and Diffuse Ghosts
(Cluster sites only)

Table 9 presents statistics relating to the Average Improvement Index for each system on all three television channels for distinct, diffuse and the combined diffused and distinct ghosting conditions.

A review of the data on Table 9 shows that the Philips system has the highest Average Improvement Index for all three conditions (distinct, diffuse and both), followed by BTA, Sarnoff, Samsung and AT&T. The data also reveals that all five systems were more effective in reducing and/or eliminating the ghosting impairment at locations where the ghosts were distinct rather than smeared.

Table 10 presents the percentage of observations that contained 1, 2, 3, 4, 5 to 8 and greater than 8 simultaneous ghosts in a picture, categorized by channel. Here again, the data shows that the distribution of multiple ghosts was roughly the same for VHF and UHF.

System/Type	Average Improvement Index		
	Ch. 4	Ch. 20	Ch. 50
BTA			
Distinct	1.76	1.30	1.17
Diffuse	1.06	0.80	0.55
Both	1.31	1.49	1.33
Samsung			
Distinct	.44	0.32	0.56
Diffuse	0.0	0.63	0.0
Both	0.31	0.43	0.23
AT&T/Zenith			
Distinct	0.38	0.33	-0.12
Diffuse	-0.22	0.10	-0.25
Both	0.07	0.56	-0.05
Philips			
Distinct	2.11	1.78	1.61
Diffuse	1.28	0.70	1.22
Both	1.58	1.65	1.75
Sarnoff/Thomson			
Distinct	0.98	1.07	1.07
Diffuse	0.72	0.80	0.50
Both	1.00	0.86	0.77

TABLE 9
System Performance for Distinct, Diffuse and Combination Ghosting Conditions
(Cluster sites only)

Number of Simultaneous Ghosts	Ch. 4 (%)	Ch. 20 (%)	Ch. 50 (%)
1	8	14	9
2	18	24	20
3	35	22	41
4	20	18	14
5-8	16	22	16
>8	3	0	0

TABLE 10
Relative Presence of Multiple Ghosts
(Cluster sites only)

Tables 11 (a), (b) and (c) presents statistics of the Average Improvement Index for all five systems on all three television channels respectively for 1, 2, 3, 4 and 5 to 8 simultaneous ghosts in an picture. The data shows that the Philips systems has the highest Average Improvement Index on all three channels for single and multiple ghosts, followed by BTA and Sarnoff, Samsung and AT&T. Philips did extremely well for pictures with

Number of Ghosts	Average Improvement Index				
	BTA	Samsung	AT&T	Philips	Sarnoff
1	0.80	0.0	-0.13	1.00	0.60
2	1.67	0.21	0.0	1.96	1.33
3	1.30	0.26	0.34	1.57	0.86
4	1.58	0.29	0.39	2.15	1.08
5-8	1.55	0.50	0.14	2.23	1.27

TABLE 11 (a)
Ch. 4 System Performance for Multiple Ghost Conditions
(Cluster sites only)

Number of Ghosts	Average Improvement Index				
	BTA	Samsung	AT&T	Philips	Sarnoff
1	1.50	0.55	0.50	1.65	0.90
2	0.83	0.22	0.03	1.28	0.84
3	1.86	0.45	0.46	1.29	0.75
4	1.75	0.50	0.15	2.40	1.35
5-8	1.29	0.25	0.71	1.69	1.04

TABLE 11 (b)
Ch. 20 System Performance for Multiple Ghost Conditions
(Cluster sites only)

Number of Ghosts	Average Improvement Index				
	BTA	Samsung	AT&T	Philips	Sarnoff
1	1.25	0.25	0.75	1.58	1.42
2	0.77	0.08	-0.58	1.32	0.73
3	1.77	0.13	-0.04	1.69	0.89
4	1.22	0.19	-0.35	1.56	1.39
5-8	1.39	0.17	-0.17	1.94	0.17

TABLE 11 (c)
Ch. 50 System Performance for Multiple Ghost Conditions
(Cluster sites only)

4 or more simultaneous ghosts, and all five systems exhibited a relatively lower AII score for a single ghost at VHF and for two simultaneous ghosts at UHF.

Finally, the performance of the ghost cancelers were also evaluated with respect to the delay ranges of the ghosts. Specifically, the picture was evaluated based on whether the ghosting condition was advanced (leading ghosts) and/or delayed (lagging ghosts) in time relative to the original signal. Tables 12 (a), (b) and (c) presents statistics on the Average Improvement Index for all five systems and all three channels respectively for delay ranges of -2 to 0, 0 to 2, 2 to 10, 10 to 19, 19 to 28 and 28 to 37 microseconds relative to the main signal. It was noted that approximately one-third of all observations made on all three channels contained leading ghosts.

Delay Range	Average Improvement Index				
	BTA	Samsung	AT&T	Philips	Sarnoff
-2 to 0	1.35	0.13	-0.01	1.70	1.08
0 to 2	1.20	0.0	0.0	1.20	0.50
2 to 10	1.54	0.23	0.11	1.82	1.20
10 to 19	1.70	0.27	0.06	2.00	0.73
19 to 28	0.94	0.22	0.50	2.00	1.00
28 to 37	1.63	0.50	0.25	1.88	1.13

TABLE 12 (a)
Ch. 4 System Performance Relative to Ghost Delay Range
(Cluster sites only)

Delay Range	Average Improvement Index				
	BTA	Samsung	AT&T	Philips	Sarnoff
-2 to 0	1.13	0.22	0.27	1.55	1.03
0 to 2	1.00	0.31	0.40	1.38	1.10
2 to 10	1.28	0.45	0.20	1.55	0.85
10 to 19	1.63	0.37	0.52	1.67	0.91
19 to 28	1.29	0.21	0.38	1.50	1.40
28 to 37	1.10	0.50	0.80	1.50	1.40

TABLE 12 (b)
Ch. 20 System Performance Relative to Ghost Delay Range
(Cluster sites only)

Delay Range	Average Improvement Index				
	BTA	Samsung	AT&T	Philips	Sarnoff
-2 to 0	1.09	-0.07	-0.27	1.43	0.80
0 to 2	0.80	0.20	-0.30	1.10	0.80
2 to 10	1.02	0.12	-0.02	1.20	0.76
10 to 19	1.19	0.21	-0.26	1.76	0.69
19 to 28	1.55	0.50	0.05	2.05	0.45
28 to 37	--	--	--	--	--

TABLE 12 (c)
Ch. 50 System Performance Relative to Ghost Delay Range
(Cluster sites only)

A review of the data on Tables 12 (a), (b) and (c) reveals that the Philips systems has the highest Average Index for all delay ranges on three channels, followed by BTA, Sarnoff, Samsung and AT&T. All five systems have a relatively low Average Improvement Index for delay ranges between 0 and 2 microseconds, i.e., close-in ghosts.

Radial Measurements

As noted earlier, the radial measurements were primarily collected to assess the general performance of the five systems under moderate and weak signal conditions. This information is useful in determining the processing gain of all systems in the field.

Table 13 presents statistics relating to the effectiveness of the ghost canceler in reducing and/or eliminating ghosts for the radial measurements. The table shows the percentage of observations that were not changed, degraded or improved after the activation of the ghost canceler for all the five systems tested.

None of the systems tested exhibited an improvement on all the ghosting conditions encountered in the field. The Philips and Sarnoff systems did not exhibit image degradation in any of the observation sequences. Also, a comparison between the data in Table 4 (cluster measurements) and the data presented above showed a marked decrease in the percentage of observations that indicate improvement in the picture for the BTA, Samsung and AT&T systems when compared against the Philips and Sarnoff systems. This information suggests a processing gain advantage for Philips and Sarnoff relative to the other three systems.

Table 14 shows the Average Improvement Index for all five systems for the radial measurements, along with the average impairment value for each channel before activation of the ghost canceler. Here again, the data reveals that the Philips system has the highest Average Improvement Index, followed by Sarnoff and BTA. Samsung and AT&T showed little or no average improvement.

A comparison of the data in Tables 5 (cluster measurements) and 14 (radial measurements) generally shows a slight to moderate decrease in the Average Improvement Index on all five systems and nearly all three channels. This data suggests that the performance of all five ghost cancelers is affected by moderate and low signal conditions, some more than others. This assessment is certainly applicable to the BTA, Samsung and AT&T systems.

System	Ch. 4 %	Ch. 20 %	Ch. 50 %
BTA			
Degrade	4	0	0
No change	29	25	29
Improve	67	75	71
Samsung			
Degrade	12	0	7
No change	80	78	68
Improve	8	22	25
AT&T/Zenith			
Degrade	58	40	52
No change	29	26	39
Improve	13	34	9
Philips			
Degrade	0	0	0
No change	10	10	4
Improve	90	90	96
Sarnoff/Thomson			
Degrade	0	0	0
No change	17	28	25
Improve	83	72	75

TABLE 13
Overall Effect of Ghost Canceling Systems on Observed Impairments
(Radial sites only)

	Ch. 4	Ch. 20	Ch. 50
Average impairment before activation of ghost canceler	3.2	2.7	2.6
System	Average Improvement Index		
BTA	0.64	0.93	0.65
Samsung	0.0	0.22	0.04
AT&T/Zenith	-0.48	0.04	-0.32
Philips	0.93	1.41	1.28
Sarnoff/Thomson	0.71	0.69	0.93

TABLE 14
Average Impairment and Average Improvement Index
(Radial sites only)

CONCLUSIONS

A field measurement program was conducted to evaluate and document the performance of five competing ghost canceling systems under consideration for a single voluntary standard for the broadcast, cable and consumer electronic industries. The information described above encompasses details of the measurement program and presents findings and observations along with general statistics on the overall performance of all five systems. It is hoped that this information will aid in the selection of a single voluntary standard for the United States.

Based on the information collected, one can conclude that all five systems were effective in reducing and/or eliminating ghosts. The overall performance, however, varied significantly from system to system and depended to some extent on the transmitting frequency (VHF or UHF), the type and complexity of the ghosting condition, and the received signal level. The Philips system consistently outperformed the other four systems.

ACKNOWLEDGMENTS

We would like to thank the following individuals and companies for their efforts in this project: Lori Gildow-Rebert of PBS for software development and serving as one of the expert observers; CBS, Capital Cities/ABC, CableLabs, PBS and EIA for providing equipment pieces; Ron Marquez, Jim Capps and Steve Kronschnabel of Tektronix for the substantial effort to develop and fabricate the GCR PROMs for the Tektronix 1910 generators; John Hidle of the Carl T. Jones Corporation for technical support and serving as the second expert observer and driver of the field van; Ed Williams of PBS for technical support; Curtis Garris of WFTY, Pedro Perez of WDCA-TV and Jim Powell of WRC-TV for technical assistance and their stations' gracious participation in the test program.

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COMPUTER AND LABORATORY EVALUATION OF VIDEO GHOST CANCELING REFERENCE SIGNALS

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Abstract - Five different video Ghost Cancelling Reference signals have been proposed to improve television reception in North America. One of them must be selected as a standard.

This paper presents the computer simulations and laboratory tests completed to evaluate and compare the effectiveness of each video Ghost Cancelling Reference (GCR) Signal proposed.

INTRODUCTION

Broadcasters have been looking for a long time for a device or a system which could reduce or eliminate video ghosts. These ghosts are created when television signals are reflected one or more times from obstacles such as hills or buildings. A television picture received under these conditions will be degraded and annoying to watch. Viewers are particularly sensitive to this kind of degradation since they have access to ghost free sources of video such as video cassette recordings.

Video ghosts were accepted as a fact of life until 1989 when the Broadcasting Technology Association (BTA) in Japan developed a video ghost cancelling system. This system was evaluated in the Atlanta area in the United States [1] and generally found effective in nearly all the test locations. However, some weaknesses were identified and several organizations made proposals for an improved system. A formal request for proposals was issued by the NAB in July 1990. Five different proposals were received as shown in Table 1. A more complete description can be found in the bibliography [2-6]. Each of them requires a different Ghost Cancelling Reference (GCR) signal to be included in the Vertical Blanking Interval (VBI) for television transmission. For obvious practical reasons a single system has to be selected as a standard.

To do so, it was planned to evaluate the proposals in three different ways, namely through computer simulations, laboratory tests and some field tests. This paper will present the results of some of the computer simulations and laboratory tests which were performed at the Communications Research Centre (CRC) in Ottawa, Canada. These tests were done in collaboration with the proponents and were part of the evaluation process of the Advanced Television Systems Committee's (ATSC) Specialist Group on Ghost Cancelling (T3-S5).

COMPUTER SIMULATION

Computer simulations were carried out to evaluate the channel characterization performance of each proposed GCR signal. With computer simulations, it is possible to isolate the channel characterization potential of each GCR signal from the performance of the complete Ghost Cancellation System. In particular they give an opportunity to evaluate the performance of GCR signals without the limitations of any hardware implementation.

To complete the computer simulation, each proponent with the exception of BTA provided CRC with a computer file of their respective GCR signals. Each file was 8 TV lines long in order to cover the entire length of some of the GCR which operate in a cycle of 8 fields. A channel simulation software was created by the David Sarnoff Research Centre and was found appropriate by the other proponents. It was used by CRC to impair each of the signals with ten different combinations of noise and ghosts known only by CRC.

The ghost combinations were representative of an average over the air reception (A in Table 3), of a reception under more severe conditions (B in Table 3), of microreflections (C in Table 3) and of a very long delay ghost (D in Table 3).

The above simulated channels contained only static ghosts. In the last simulations, one of three ghosts was varying over time. This combination would be representative of dynamic or moving ghosts sometimes observed in the field. It could give indications about the potential of a GCR signal to track a dynamic ghost.

Each proponent received the ten impaired combinations of his GCR signal and processed them to extract the 10 channel impulse responses. The responses from 3 proponents, Philips, Samsung and Sarnoff, were received by CRC and compared with the reference channel responses of each combination. These comparisons showed that all the GCR evaluated performed very well. For example, Fig. 1 illustrates how the responses from the 3 proponents and the reference kept at CRC were similar for impairment combination 8, one of the more severe ones. More detailed analysis is presently underway. However, it is already clear that simulations of more severe conditions would be required to really evaluate the performance limits of each GCR signal.

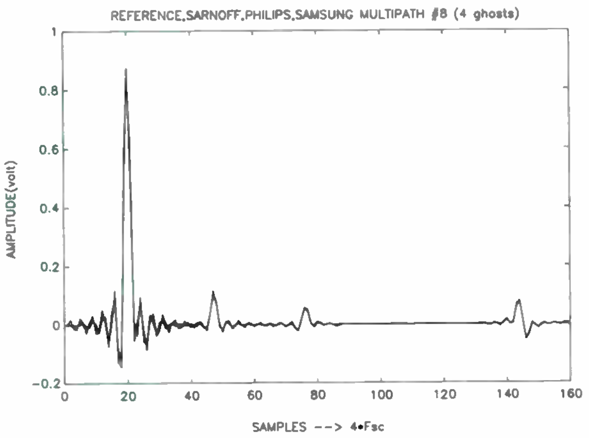

Fig. 1: Comparison of the reference impulse response #8 with the responses calculated by Philips, Samsung and Sarnoff/Thompson.

LABORATORY EVALUATION

The field tests and the computer simulations were complemented with laboratory experiments performed on a prototype from each proponent. These laboratory tests were completed at CRC early in 1992.

Equipment Set-Up

The equipment set-up for these tests is illustrated in Figure 2. Video sources included video test signals as well as real television pictures. Seven video sequences particularly sensitive to multipath were selected from a CCIR tape originally prepared to evaluate digital video codecs. The 5 GCR signals were added to the video program, each one occupying one line in the vertical blanking interval.

The video signal was then impaired by a ghost signal generator which can provide 7 ghosts with relative delay times of up to plus and minus 64 microseconds, amplitude range between 0 and -55 dB and phase between 0 and 359 degrees.

The intermediate frequency produced by the ghost generator was then up-converted to channel 11 which was free of interference from the Ottawa area transmitters. White noise was added to the modulated signal.

The impaired signal was demodulated by a Tektronix 1450 synchronous demodulator. The in-phase (I) output was distributed to the ghost canceller prototypes of 4 of the 5 proponents. The Sarnoff Laboratory prototype, was also fed with the quadrature (Q) output of the demodulator as it was operating on a complex signal. The ghost canceller from the Broadcast Technology Association (BTA) had its own demodulator and was fed directly with the RF signal.

The corrected output of each ghost canceller prototype was then monitored and compared with the reference signal on waveform and video monitors. Finally some pictures were also recorded for future reference.

The evaluation of each signal was done subjectively and objectively. The subjective evaluation was done by at least 5 expert observers using the CCIR impairment scales shown in Table 2. The observers were proponents' representatives or members of T3-S5. The observers were asked to make efforts to restrict their rating to ghost elimination and ignore artifacts created by hardware problems or the presence of noise.

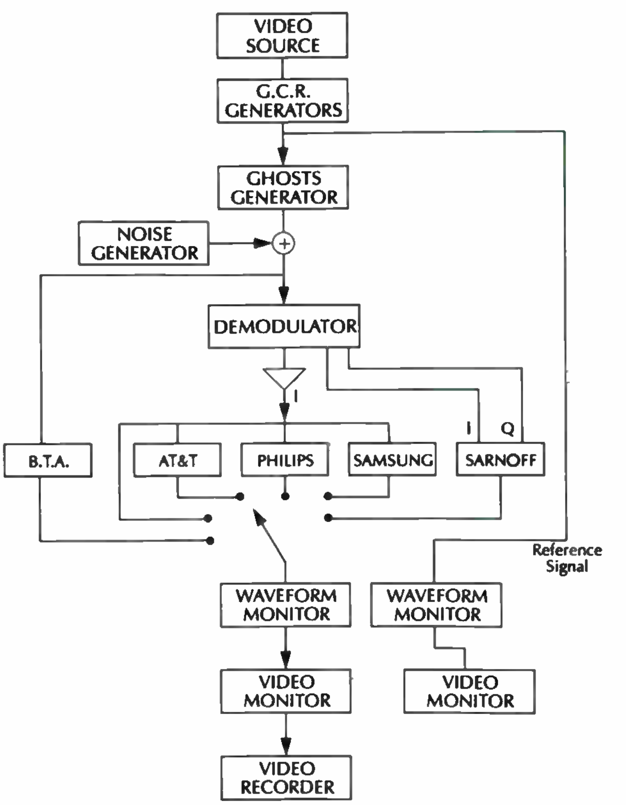

Fig. 2: Equipment set-up for the laboratory tests.

The objective measurements included the unweighted signal-to-noise ratio, pulse-to-bar ratio, the 2T K factor, the luminance non-linearity as well as the group delay and the gain of the Sin x/x. Bit error rate of a teletext sequence was also measured using equipment provided by PBS.

Description of the Prototypes

The results presented were obtained using the ghost cancellers provided by each proponent. These ghost cancellers were at different levels of development.

The synchronization circuits of the AT&T/Zenith prototype was built using low cost analog components. This was intended to demonstrate the ability of their GCR signal and algorithm to cancel ghosts with a low cost implementation. In the tests, their synchronization and phase-lock loop circuits were not able to always perform correctly. A proprietary algorithm was then used to correct for those defects as much as possible. However, even for the no ghost condition, their prototype degraded the picture quality more than the others. Its cancellation range was also limited by filter hardware to -1.8 to 29 microseconds. AT&T/Zenith system could not be tested in all the laboratory tests due to the delay in the shipping of their equipment to CRC.

The NEC ghost canceller, GCT-900, was used to evaluate the BTA GCR signal. It is a fully developed product already marketed in Japan. Its cancellation range was specified to be between -1.5 and 40 microseconds.

The prototype from Philips had been damaged during shipping. Some of the tests were done before it was repaired and while its synchronization circuit was still making frequent errors. After being repaired the defects were not visible anymore.

The Sarnoff/Thomson's prototype did not have any serious problems but it sometimes displayed white flashing lines near the top of the picture. This may be due to improper timing of the loading of the digital filter coefficients.

Finally Samsung's prototype was at an earlier stage of development than the other ones. It was still controlled by a personal computer. The time required to complete ghosts cancellation was relatively long and a software problem limited cancellation range to 35 microseconds. An improper dc shift and gain adjustment also occurred sometimes making the picture unviewable.

Before using the laboratory results to evaluate a GCR signal it has to be realized that some deficiencies of the tested prototypes are not related to the proposed GCR signal. It is therefore very important to remember that the results presented here show the capabilities of the presently available prototype. **In most of the cases these results will not show the ultimate capabilities of a particular GCR signal.**

Laboratory Test Results

The results of 4 of the most interesting tests are now presented: comparative tests, delay range tests, ghost amplitudes and convergence time tests. Other subjective test results as well as objective test results will be published as soon as possible.

The comparative tests were completed to see how each prototype was performing for the 6 typical combinations of ghosts listed in Table 3. Table 4 presents the impairment subjective rating given to the pictures before and after ghost cancellation.

The first tests were done with no ghost to observe how each prototype performed when no cancellation was needed. The results have shown that no significant degradation was observed except for the AT&T prototype which, as explained before, used low cost analog parts.

The results of the tests done with the different combinations of ghosts showed that the picture quality was significantly improved by ghost cancellation. The improvement was particularly significant when the original pictures had a low impairment rating such as in condition B, D, E and F. It is also interesting to mention that the rating obtained after correction of noisy pictures was also excellent. The reason for this, is that noise tended to hide the residual ghosts which may have been noticeable if there had been no noise.

The performances of some of the prototypes were affected however by hardware problems.

Due to shipping damage, the synchronization circuit of the Philips' prototype was more sensitive to noise for tests A, B, C and D which were completed before it could have been repaired.

The prototypes from AT&T/Zenith, BTA and Samsung did not perform as well as the two others for combinations D, E and F.

The AT&T prototype did improve the picture quality for these tests by cancelling the short ghosts. Due to hardware limitations, it was however unable to cancel the long ghost in tests D and E. This prototype was unable to operate for any of the 3 combinations when noise was added.

The BTA prototype cancelled some of the ghosts of combination D and E but could not eliminate the pre-ghosts which were outside its 1 microsecond range.

Samsung's prototype did not operate correctly with these ghost combinations because of the limitations described before. It was operating correctly however when the longest delay was reduced to 35 microseconds.

Table 5 presents the results of the tests performed to estimate the performance of each prototype for long delay ghosts. The tests were completed by setting the longest ghost of combination B to values of delay between 35 and 50 microseconds. The prototype from AT&T/Zenith was not available for these tests. Prototypes from both Samoff and Philips could correct ghosts delayed up to 40 microseconds. The ones from Samsung and BTA could eliminate ghosts with delays up to 35 microseconds.

Table 6 presents the results on the subjective performance of the prototypes for high amplitude ghosts. The tests were done with all the ghosts of combination B set to the attenuation indicated in Table 6.

The prototype from AT&T/Zenith was not available for these tests. The one from Samsung could not synchronize when the amplitude of the ghosts was too high.

Finally Table 7 presents the time required by each prototype to complete cancellation of ghosts. The time required by each prototype to cancel 2 different combinations of ghosts was measured with a signal-to-noise ratio of -48 and -25 dB.

Some ghost cancellers were correcting in 2 steps: first cancelling close ghosts, then longer ghosts in a second step. The values in the table are the ones measured for this second step. These values also include the time required for the prototypes to synchronize after the ghost combination has been switched in.

In conclusion the laboratory tests showed that all the systems proposed can eliminate ghosts in a very efficient manner. More advanced hardware will however be needed to demonstrate the real potential of each proposed GCR signal. It may also be necessary to develop different versions of each system prototype to evaluate the limits of each GCR signal for such parameters as ghost delay range, ghost cancellation time and ghost amplitude.

CONCLUSION

The results from the computer simulations and the laboratory tests of video ghost cancelling systems have been presented. They show that very good results can be obtained with each of the proposed GCR signals. It appears, however, that more tests are required before the best GCR signal can be selected as a standard.

Summary

The evolution of digital signal processing technology offers the broadcasters an opportunity to improve the quality of reception of their signals by eliminating annoying ghosts. Five different ghost cancelling reference (GCR) signals have been proposed for introduction as a ghost cancellation system in North America. The results from computer simulations and laboratory tests have demonstrated that a ghost cancellation system can significantly improve television service offered to the viewers.

ACKNOWLEDGEMENTS

The author would like first to sincerely thank the proponents without which this evaluation could not have been performed. Thanks are also expressed to the members of T3-S5 and to their chairman, A. Uyttendaele for their advice, guidance and assistance.

Thanks are also due to Philips Laboratory, Tektronix Canada and PBS for providing CRC with equipment essential to the laboratory tests; William Sawchuk, Director-General, Broadcasting Technologies Research, and Metin Akgun, Director, Broadcast Information Technologies for their support and encouragement all along this project.

Finally, very special thanks goes to the following CRC employees: Tom Green who set up the equipment and worked on the laboratory tests, Benoit Ledoux who did the first part of the computer simulation and helped for the laboratory tests, Orlando Rivera who also worked on the computer simulations, Alain Bergeron who worked on the laboratory tests, Tom Kahwa who helped to set the ghost generator and Karen A. Kennedy who typed this paper.

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TABLE 1: Proposed Ghost Cancelling Reference Signal

PROPONENTS	GCR TYPE
AT&T/Zenith	Pseudo-Random Binary Sequence (PRBS)
BTA (Japan)	Integrated Pulse
Philips Laboratories	Deterministic Sequence
Samsung	Complementary PRBS
Sarnoff/Thomson	Pseudo-Random Binary Sequence (PRBS)

TABLE 2: CCIR Impairment Scale

IMPAIRMENT SCALE
5 Imperceptible
4 Perceptible but not annoying
3 Slightly annoying
2 Annoying
1 Very annoying

TABLE 3: Ghost Combinations Used for the Laboratory Tests

DESIGNATION	NUMBER OF GHOSTS	DELAY (μsec)	ATTENUATION (dB)
A: Typical	2	.45 & 2.3	-19 & -24
B: Severe	4	.2 to 8.2	-14 to -24
C: Microreflections	5	-.7 to .4	-26 to -31
D: Long Delay	3	-1.8 to 39	-14 to -23
E: Very Severe	7	-4 to 40	-15 to -30
F: Strong Microreflections	5	.1 to 1.1	-9 to -17

TABLE 4: Subjective Impairment Rating of the 5 Prototypes for the 6 Ghost Combinations

TEST CONDITIONS (Unweighted SNR)	NO CORRECTION	CORRECTED BY:				
		AT&T/ZENITH(+)	BTA-NEC	PHILIPS (*)	SAMSUNG	SARNOFF
No Ghost						
49 dB	5	4.1	4.9	4.9	4.8	4.9
25 dB	5	4.3	5	5	4.9	5
Ghosts A						
51 dB	4	N.A.	4.8	4.5	4.2	4.6
25 dB	4.5	N.A.	5	4.5	5	5
Ghosts B						
48 dB	2.5	4.0	4.5	4.3	4	4.2
25 dB	3.5	3.9	4.9	4	4	5
Ghosts C						
46 dB	4	N.A.	4.3	4.5	4	4.6
25 dB	4.5	N.A.	4.8	4.5	4.8	4.9
Ghosts D						
49 dB	2	2.6(++)	3(++)	4.1	1(++)	4.4
25 dB	2.5	No operation	4(++)	4.5	3(++)	5
Ghosts E						
43 dB	1	2.6(++)	1.7(++)	3.7	1(++)	3.3
25 dB	2	No operation	2(++)	4.6	1(++)	4.8
Ghosts F						
43 dB	2.4	4	4.4	4.2	2.6	4
25 dB	2	No operation	3.4	4.4	3.8	4.4

(+ These tests were completed with the prototype unable to lock its internal clock to the received signal.)
(* These tests were completed before the Philips ghost canceller could be repaired after being damaged in the shipping.)
(++A ghost in this combination was out of the delay cancellation range of the prototype.)

TABLE 5: Subjective Impairment Rating for a Long Delay Ghost in Combination B

GHOST DELAY (µsec)	NO CORRECTION	CORRECTED BY:				
		AT&T/ZENITH	BTA-NEC	PHILIPS	SAMSUNG	SARNOFF
50	2.8	N.A.	3	3	3	3
45	2.8	N.A.	3	3	3	3
40	2.9	N.A.	3	4.7	1	4.7
35	2.7	N.A.	4.8	4.8	4.3	4.7

TABLE 6: Subjective Impairment Rating for Ghosts with High Amplitude (Combination B)

GHOST AMPLITUDE	NO CORRECTION	CORRECTED BY:				
		AT&T/ZENITH	BTA-NEC	PHILIPS	SAMSUNG	SARNOFF
-6 dB	1	N.A.	2	2.7	No Synch.	1.6
-8 dB	1.2	N.A.	2.6	3.2	No Synch.	1.8
-10 dB	1.5	N.A.	3.3	3.7	No Synch.	2.2
-12 dB	1.6	N.A.	3.5	3.9	3.1	3.0
-14 dB	2	N.A.	4.1	4.2	4.5	4.0

TABLE 7: Time (seconds) Required for Cancellation of Ghosts

TEST CONDITIONS (Unweighted SNR)	CORRECTED BY:				
	AT&T/ZENITH	BTA-NEC	PHILIPS	SAMSUNG	SARNOFF
Ghost B					
48 dB	4	46	6.5	31	12
25 dB	9	46	6	28	11
Ghost D					
49 dB	2	45	5	Out of range	19
25 dB	No operation	45	5	Out of range	15

AN OVERVIEW OF GHOST CANCELLATION REFERENCE SIGNALS

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Abstract: Removal of ghost images improves greatly the quality of received NTSC images. Implementation of cancellation at the receiver requires that the broadcaster transmit a standard Ghost Cancellation Reference (GCR) signal. The Advanced Television System Committee (ATSC) is currently in the process of selecting the best of five proposed GCR signals. This paper outlines the properties of a good GCR signal. It then describes how the five signals meet these criteria.

INTRODUCTION

If a television signal undergoes several reflections before reaching the receiver, then several copies of the same signal may be received at the same time. The receiver will lock onto the strongest signal. All the others are called multipath echoes or "ghosts". It has long been recognized that ghost cancellation is one of the few remaining possibilities for achieving significant new improvements in received image quality. By transmitting the required GCR signal, the broadcasting station needs to make but a very small financial investment and still make a large impact in improving his service to the community. Once a standard GCR signal is selected, receiver manufacturers will begin to sell home receivers that can cancel ghosts. The key for the initiation of these improvements is the selection of the standard GCR signal.

In the United States, the task of choosing the GCR signal has been assigned to the Advanced Television Systems Committee (ATSC). The ATSC Specialist Group T3/S5 has been accumulating data which the parent body can use to make their final selection. Five organizations have submitted proposals for a GCR signal. These are:

- The Japanese Broadcasting Technology Association (BTA).
- Philips
- Sarnoff/Thomson
- AT&T/Zenith
- Samsung

It has been decided that the choice of the GCR standard will be based on experience gained from:

- Field tests held by the National Association of Broadcasters (NAB)
- Field and laboratory tests conducted by CableLabs
- Laboratory tests and computer simulations run by the Canadian Research Centre (CRC)
- Evaluation of the claimed theoretical and practical advantages of each GCR, as presented by the proponents.

This paper will outline the ghost cancellation process and the requisite properties of good GCR signals. It will then present the salient features of each proposed GCR candidate and will discuss their relative merits.

THE PROPERTIES OF A GOOD GCR SIGNAL

Ghost images in television are due to the superposition of several copies of the main transmitted image, delayed with respect to each other. This multipath distortion can be represented mathematically by a linear, impulse response, or its Fourier Transform, the channel frequency response. Cancelling the echoes is a two step process: (1)characterize the frequency response of the echoing channel and (2)use adaptive filters to equalize the channel frequency response. The characterization of the channel frequency response is done via the GCR signal. The second step, cancellation of the echoes can be done with a variety of methods using adaptive digital filters. This paper deals with the selection of a GCR with is mainly pertinent to the first step.

The GCR signal is used to characterize the multipath channel. Since this is a linear channel, the characterization process is equivalent to finding the frequency response or impulse response of a linear "black box". Such responses are usually found by applying to the black box some suitable test signal and observing the response. The GCR plays the role of such a test signal. From linear system theory, one can tabulate the requirements such test signals should have. Namely:

- High energy to overcome noise that contaminates the measurement process. In the case of ghost cancellation, high energy is more crucial since moving ghost patterns represent a time-varying linear system that can only be characterized fast enough if the test signal energy is sufficiently high.
- The GCR should have a flat frequency spectrum (or equivalently an impulsive autocorrelation function) over the entire video band, so that it probes the unknown channel with equal weight at all frequencies.
- The group delay of the signal must be as linear as possible.
- The GCR signal should not limit the range of delays over which the ghosts can be cancelled.
- The GCR, which is transmitted over a VBI line, should allow complete channel characterization after the reception of a single such VBI line. This assures that there is no fundamental limit on echo cancellation speed.

THE GCR CANDIDATES

A brief description of each of the five GCR candidates follows. The relative merits of each will be discussed in the following sections.

- BTA: The BTA's GCR generation begins with a $\sin x/x$ type of impulse with a flat spectrum over the video band. This is integrated to generate a step, which is broadcast. Upon reception, the signal is differentiated to eliminate any DC drifts. The result is the impulse response of the channel.
- Philips: The Philips GCR resembles a linear-FM, chirp-like, frequency sweep signal. Unlike the classical chirp, this signal was synthesized from inception to meet all the criteria of the ghost cancellation task, including impulsive autocorrelation function, flat frequency amplitude response and linear group delay. The spectral flatness is independent of sampling rate and independent of the signal length..
- AT&T/Zenith: These proponents use a noise-like pseudo-random sequence. It is the maximal-length shift register sequence or "M" sequence that was borrowed from other, earlier applications. If repeated periodically and processed carefully and in a restrictive way, this signal also appears to have an impulsive autocorrelation function.
- Sarnoff/Thomson: Here two separate M-sequences are transmitted. A short one is transmitted with three periodic repetitions and a single short one is transmitted over a different field. The idea is that the single short one is processed in a manner that yields a possibly noisy solution. On the other hand, the three repetitions provide a clean solution with some time-delay ambiguities. By combining the results of processing both waveforms, the complete channel characterization can usually be found.
- Samsung: The AT&T and Sarnoff/Thomson versions of the pseudo-noise sequence have the potential problem in that the spectra are only flat if the sampling rate is not changed from the specified four times color subcarrier (or some integral multiple of this rate) **and** if the GCR is repeated periodically. Samsung uses a complementary set of two distinct pseudo-random sequences. When added together, the two sequences produce a flat spectrum at all sampling rates and without the need for periodic repetitions.

A FEW WORDS ABOUT ALGORITHMS

To translate a received, ghosted GCR signal to a channel model, it has to be processed by a suitable *algorithm.* With the BTA signal the algorithm may be an averaging of many successive received GCR samples to compensate for the low energy of the BTA signal. This is then followed by differentiation.

The AT&T/Zenith pseudo-random M-sequence has the mathematical property that low-noise, good quality channel characterization is only possible if the received GCR is repeated periodically several times in the receiver. However, the use of this algorithm carries the penalty that the sum of the total maximum (positive and negative) ghost delays must not exceed the length of the GCR itself. Echoes that have longer time delays will produce ambiguous results. The alternative is to avoid the time delay ambiguity by the use of an algorithm that produces noisy channel characterization coefficients.

Sarnoff/Thomson uses two GCRs of the same type as proposed by AT&T/Zenith. They process one with the noisy algorithm that does not have a time-delay limitation. They process the second with the low-noise method that does have a time-delay limitation. By comparing the two answers, they usually achieve a acceptable channel characterization. Equipment being used by Sarnoff/Thomson uses dual processing, calculating both in-phase and quadrature signals. This is not necessary for the other proponents.

Samsung's GCR can be processed without the penalties mentioned above. However, two or more successive waveforms must be averaged together to achieve acceptable channel characterization..

The Philips waveform can be processed by a variety of algorithms without any restrictions. Even one single received GCR has been shown to be sufficient to provide complete channel characterization.

THE EVOLUTION OF GCR SIGNALS

BTA

The first GCR signal to be implemented was selected by the BTA to be used in Japan. As the ATSC Specialist Group T3/S5 began its process of seeking a GCR to be recommended for the United States, it became apparent that the BTA signal suffers from having too low energy. This makes the BTA signal difficult or sometimes impossible to use under the conditions of:

- high noise
- rapidly moving echoes
- moving antenna structures (even antennas moving in the wind, in the case of short-wavelength UHF channels)

In addition the BTA signal has an other disadvantage. It requires that the GCR be differentiated after reception. The differentiation process accentuates the noise at the high-frequency end of the band, near the color subcarrier. This results in inferior color performance. Furthermore, the BTA specifies that the maximum ghost delay for which their GCR is used, should not exceed 1/2 horizontal line time. ["Ghost Canceller Requirements", letter to the ATSC T3/S5 Specialist Group, Dec. 23, 1991 from S. Matsuura for the BTA] That violates the restriction that the GCR should not impose a restriction on the ghost cancellation delay range.

AT&T/Zenith

In response to the low-energy limitation of the BTA signal, AT&T/Zenith decided to recommend the pseudo-random M sequence which has been used in earlier applications. It had the advantage of having about 20 dB more energy than the BTA GCR.

All the other GCRs all have much higher energy than the BTA. This makes any of them more robust under difficult conditions. AT&T/Zenith uses a pseudo-noise M-sequence for a GCR. The received, ghosted GCR waveform is correlated against a stored reference to achieve channel characterization. The problem with this approach is that the M-sequence only works in this context if it is repeated cyclically. (It is sufficient to repeat it in the receiver, only one copy need be transmitted per field.) The length of this signal is 53.4 μs. Due to the required processing the sum **of maximum positive and negative echo delays** can not exceed 53.4 μs. To achieve greater cancellation delay ranges, one must change to other processing algorithms which would produce a test signal spectrum that does not appear to be flat. It would therefore be subject to noise perturbations leading to inaccurate channel characterizations.

Sarnoff/Thomson

Sarnoff/Thomson also chose to use M-sequences. However, they tried to overcome the tradeoff handicap faced by the AT&T/Zenith signal. Therefore, they use a sequence of two waveforms. One is a short M-sequence, the second is three repetitions of the short M-sequence. The idea is that the short sequence will find short-delay ghosts very quickly and with higher signal-to-noise gain, but has an even more limited delay range than the AT&T/Zenith GCR. On the other hand the long sequence can be used to find longer-delay ghosts with in longer processing time, with no limitation on delay times and with less processing gain. This gives a noisy channel model. Combining the two answers is supposed to reduce the shortcomings of each. The approach is useful. However the suggestion that two separate characterization calculations be carried out simultaneously is an unnecessary complication. Furthermore, Sarnoff/Thomson not only requires the use of two separate GCR signals, it also uses for each of the GCRs, two separate calculations, one for the in-phase signal, the other for the quadrature signal. This so greatly increases the required hardware complexity that the method has questionable economic feasibility.

Samsung

Samsung realized that M-sequence type of pseudo random signal that is used by both AT&T/Zenith and Sarnoff/Thomson has the problems outlined above, due primarily to the fact that the Fourier spectrum of the M-sequence is not flat. However, Samsung wanted still to use a pseudo-noise type of signal. Therefore, Samsung recommended a GCR consisting of two full-length pseudo-noise "complementary sequences". When added together at the receiver, they provide a flat spectrum independent of sampling rate. They also do not have any echo-delay restrictions. However, since they can only be used in sets of two, the lower limit on cancellation time is two field times.

Philips

Philips realized that the pseudo-random, noise-like signal is straddled with compromises. Therefore, they decided to first compile a complete list of desirable GCR attributes. Then they synthesized the GCR that meets all these requirements simultaneously, without any of the limitations of the other suggestions. The Philips GCR looks like a linear-frequency swept FM signal. In fact this signal is not a standard chirp. It has been carefully optimized on the computer to have the following properties:

- The highest energy in the shortest time interval. This gives fast channel characterization with minimum use of the VBI line. The ability to characterize a noisy channel in one field time has been demonstrated.
- Flat frequency spectrum and linear group delay for accuracy in channel characterization. Spectral flatness is due to a truly flat Fourier spectrum, and is not subject to restrictive sampling rates, as are the AT&T/Zenith and Sarnoff/Thomson signals.
- A variety of algorithms have been proven in field tests. There is no need for dual GCRs or wasteful in-phase-quadrature processing as is done by Sarnoff/Thomson.
- Has been found by workers in the field to be useful as a test signal to give a visual indication of echo presence and channel frequency response.

CONCLUDING COMMENTS

The ATSC in the United States is scheduled to choose a national standard GCR signal during the Spring-Summer 1992 period. Five GCR candidates have been extensively compared in the field, in the laboratory, on the computer and in theoretical analyses. All but one of these signals have been found to have one or more shortcomings. The Philips signal is the only one that has exhibited simultaneously all of the desirable properties of a good GCR signal with none of the problems associated with the others.

Furthermore, in the field tests, only Philips and the BTA used commercial-grade, manufacturable cancellers. The other there proponents used prototype hardware and software that is far from economic implementations. For example, some use A/D converters with resolutions of 10 bits or greater (vs 8 bits for Philips). Some use dual RF demodulators for I and Q signals. Some even included a personal computer in their setup. When comparing the Philips GCR with the BTA version, only Philips was shown to have the high energy required to compensate for moving antennas and moving ghosts.

DIGITAL AUDIO PROCESSING

Sunday, April 12, 1992

Moderator:
Dennis Ciapura, Noble Broadcast Group, San Diego, California

***DEVELOPMENTS, STANDARDS, AND IMPLEMENTATION OF AUDIO TEST STANDARDS FOR COMPRESSION**
John P. Stautner
AWARE, Inc.
Cambridge, Massachusetts

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Frank Foti
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Cleveland, Ohio

DIGITAL AUDIO PROCESSING FOR FM: SYSTEM CONSIDERATIONS
Robert Orban
Orban, a division of AKG Acoustics, Inc.
San Leandro, California

***BROADCASTING ON THE ISDN (INDIVIDUAL SUBSCRIBER DIGITAL NETWORK)**
Steve Smyth, Hamish Eassie and Michael Smyth
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AC-2: HIGH-QUALITY DIGITAL AUDIO CODING FOR BROADCASTING AND STORAGE
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***THE ROAD FROM MASCAM VIA MUSICAM TO ISO/MPEG/AUDIO LAYER II: AUDIO CODING FOR THE 90'S AND BEYOND**
Gerhard Stoll
IRT
Munchen, Germany

*Paper not available at the time of publication.

DIGITAL AUDIO PROCESSING: KNEE DEEP IN THE HOOPLA!

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ABSTRACT

With the introduction of the latest broadcast transmission system audio processors, digital processing technology is being touted as the "latest and greatest". With digital technology embracing the broadcast industry by storm, the area of signal processing has been slow in keeping up. Discussion will reveal that digital signal processing will make a difference, in the future. At a time when the advantages of digital technology will be used to advance the technique and implementation of all audio processing functions. Instead of the analog "clone" artists using "digital" as a marketing gimmick.

INTRODUCTION

Where is processing today?

If ever there was a loaded question to open a technical paper, this may be it. For this writer, the phrases "pushing too far" and "threshold of pain" seem to come to mind. Due to the competitive requirements in the radio industry, audio processing has transformed from the augmentation of a broadcast signal to a strange artform.

It has been the quest of many to find that magic formula to the "louder than you'll ever be" sound. Many operations today employ enough processing equipment to provide maximum use of the modulated signal, but at the expense of sonic integrity. Can the digital processors now available make the difference?

Where can it go from here? The answer actually lies within the objective goals of the broadcaster. This may sound like pounding on a dead horse, but only two directions actually exist, towards quality and transparency, or in the competitive aggressive direction where "loudness is king". Is there a way to achieve both?

If you were to accept as truth the marketing campaigns of the manufactures, the answer would be a definite YES! From the objective, yet technical viewpoint put forth here, the answer is a compromise at best. The laws of physics, as we know them at this time, will not allow it to happen.

To date, there has not been a device created that will provide a qualitative/transparent, yet perceptibly more dominant signal to be transmitted within the limits of the law. Realizing of course that all opinions, including those of this writer, must stand on subjective reasoning, but are based upon actual real world, and in the field experiences.

Digital: Back to the future, or the new frontier?

Even with the dour view put forth, there are advances to be made with signal processing for broadcasting. While the industry looks at digital technology for the answers, it will be the designer whose research will provide the data that will unlock the eventual benefits digital technology brings to the world of signal processing.

As might be the case with most products that claim to be "leading edge", the market is attracted to those that provide the latest gimmick, promote benefits gained, or value added from improved technology. The Compact Disc player and the Personal Computer are a few well known examples. Through effective marketing campaigns these products became successful, once the users were able to achieve the stated benefits.

In broadcast engineering, the same holds true. The key top-of-mind-awareness technology today is digital. Every news story, topic, and/or product claims it. A quick look at any publication will reveal the detailed coverage Digital Audio Broadcasting (DAB) is receiving. Digital storage mediums continue to proliferate the industry at a fast pace. Just about every product advertisement promotes some advantage due to digital.

Is digital really any better? Or is the industry looking to this technology in hope that it will be better? These are the questions you should ask yourself. In many technical instances digital technology has proved overwhelmingly that it does provide increased benefits. But discussion and arguments still persist that the future of digital in the audio domain, and in this case audio processing, may still be that, in the future.

With audio processors there is no exception, digital is the key buzzword that must front any new product in order for it to gain stature and acceptance. What must now transpire is whether the performance of digital audio processors will be able to provide the earlier stated objectives needed from a transmission processing system.

Analog processors: a quick review

Before discussing digital processing, a quick review of analog processing technology is in order. For those of you new to the broadcast industry, the analog audio processor, or processors for those of you who must have a rack full of these devices, are sophisticated Automatic Gain Control (AGC) units that manipulate and prepare a broadcast signal for transmission. Without presenting too much detail, these functions include dynamic compression, limiting, and clipping, just to name a few.

Some of the drawbacks associated with this technology thus far, can be directly related to the sonic performance and ease of adjustment or modification. Since all analog processors rely upon some analog gain controlling device, additional sonic artifacts from the operation of the this device will detract from the sonic integrity of the processed signal.

For example, look at a typical analog processor made today. Most utilize a Voltage Controlled Amplifier/Attenuator (VCA) as the controllable gain cell. As processing action is implemented by the gain cell, additional sonic "impurities" are added to the signal by the gain cell during processing. This can usually be attributed to added Intermodulation Distortion (I.M.) within the signal. Most users will usually write off perceived I.M. as a result of the processing action, and much of it is, but there is also a quotient of distortion that is related to the gain cell as well.

Some benefits provided by manufactures, in recent past, have been the improvement of the gain cell structures so that it is more "transparent" to the signal passed through. Although still not totally transparent, just better in performance.

Another drawback is the ability to create changes in the processing system, repeatability of changes or settings, and the upgrade ability of the equipment in the field. How many times has the situation occurred when changes needed to be made, and then once performed, the decision was to go back to the previous operation? Returning to the previous settings can be a difficult task, especially when conventional analog methods do not provide for a manner in which repeatability can be easily accomplished.

Also there is difficulty when trying to add, upgrade, or modify equipment in the field. The most common manner is when multiple processing devices are "chained" together to create one complete system The task at hand is the ability to achieve performance where all the individual units will work together and produce the sonic objective. This usually requires effort on the part of station personnel to get a full understanding of each device in order for the individual parts to be assembled into a complete system. Then the arduous task of making all the adjustments necessary to achieve desired results.

Because analog products are "hardware" based, any changes to be made requires the altering of the component structure. This would involve the physical replacement or altering of the circuitry used. This can be a hardship in itself.

While the drawbacks of analog based processors have been focused upon, the benefits must be considered too. Analog products are time tested, and proven through their performance. Generally, they are less expensive in cost to their digital counterparts. Field maintenance of these units is easier to accomplish. And, another point to acknowledge, is that during the early stages of the digital invasion, the analog product does provide a "comfort" factor to the user, because of familiarity.

Here is an important point that must be considered. All audio processing algorithms today, which are the basis of the digital processors, were born out of the analog environment!

Digital: Is it the saving grace?

For audio processing, the advantages digital technology brings to the table are vast and many. Already at this early stage, advances have transpired that were only ideas of hope in the analog world. But, there are still advances yet to be made in the digital realm, that have long been accomplished in the analog domain. While digital provides a multitude of benefits to broadcast processors, the drawbacks of the current generation must be considered as well.

To accent the positive, digital processors provide benefits not before achievable with their analog counterparts. Through the use of software, these units can provide flexibility, programmability, along with being easily updated.

Current promoted features include operational settings set in software that can be easily changed, modified, stored, and recalled. Flexibility that will allow the device to transform from one type of processing system to another, all at the touch of a button. (In the analog domain this would require the physical addition or subtraction of equipment) Since software is portable, digital processors become easily updated with new software releases.

Another important advance is in the gain control function. Since the audio signal has been "digitized", both the control function, and audio path no longer must overcome the weakness of analog hardware design. This permits transparency of the signal with respect to the function of gain change. Further eliminating the earlier stated I.M. distortion from gain cell artifacts that may be added to the signal.

An analogy can be drawn where the digital audio processor, is a form of audio computer. Both are software based, programmable, flexible, and somewhat portable. Like the computer in the early going, the digital audio processor has some obstacles yet to overcome.

While the introduction of the Personal Computer brought a higher awareness to the computer industry, there were still many operational shortcomings that had to be overcome. Early generation units were very expensive. The digital power of the processors employed were slow and short on memory. Software, while sophisticated at the time, was actually very primitive and even cumbersome to use. The hardware utilized, became obsolete in short order time, and has since been replaced with much faster and more powerful products, two and three times over.

Ask yourself, how many early generation Personal Computers do you know of still in use performing a viable function? Most have probably been replaced by newer faster models that rendered them out to pasture.

The same will hold true for digital audio processors. Digital Signal Processing (DSP) is still somewhat in its infancy. It has been commercially available to the electronics industry for about five years. Suffice it to say that DSP is where the IBM-XT computer was at the beginning.

It has taken until now to obtain DSP hardware that can approach the data processing needed to perform the needed functions utilized within a transmission processing system. Because of this, present DSP hardware must take advantage of most or all the data processing power in order to perform it's function. This may lead to obsolete hardware once newer and improved hardware becomes available, and it will! This will happen because newer families of DSP hardware is in development that will provide higher processing power, and at lower cost.

Because of the above mentioned hardware considerations, software borders on that of being somewhat primitive. All DSP audio processors basically emulate or "clone" an analog counterpart. Evaluate the operational diagrams of any DSP unit available, and it will be seen that the DSP device is "clone" of an analog predecessor. Sort of re-packaging old goods and selling it under a new name.

Designers and developers are still very much in the "learning curve" of this very powerful and eventual useful tool. Where DSP will definitely come into play, is once this technology can be used to explore, create, and further the advances of signal processing. Using DSP to create and implement algorithms unachievable before in analog, or as of yet to be discovered.

A simple example of this would be the implementation of a "feed-forward" AGC circuit. In analog this circuit is achievable, but at the risk of unstable circuitry, and at a higher cost to produce. For this type of AGC to operate properly, it must rely on the stability of the analog hardware to produce the exact mathematical functions needed. This can utilize a fair amount of low tolerance components to perform this operation, in order for it to operate properly and remain stable.

In digital, this type of signal processor, is simple to create, stable in operation, and necessitates only the cost of development. Once operating it will remain consistent in performance throughout its existence.

Some questions!

With digital processing now a reality in the broadcast medium, here are some some questions to ponder when considering the evaluation or purchase of one of these devices.

1) Time Delay? How audible is the actual delay of the audio through the system. Some people have the ability to sense as little as a 5ms delay in an audio signal. If this is a problem, then off-air monitoring can be difficult for announcers. Another point to consider, with an influx of digital audio products utilizing data compression, where time delay is involved, additional delay from an audio processor will cause the overall delay to be cumulative. If this should exceed 10ms, off-air monitoring may prove to be almost impossible.

2) Final limiting technique? The true test of any broadcast transmission processor is the manner in which final limiting, or clipping is performed. The desired objective is to create a final limiter/clipper that provides exact peak control, while maintaining good sonic audio performance, and protects spectrum space. In the analog domain, creative methods have been developed to suppress, cancel, or control audio distortion due to the clipping process. This permits increased use of hard limiting, which will produce high average modulation that results in perceived loudness of a transmitted signal.

To date, this has been a design difficulty in the digital domain. Primarily because the harmonic energy created by a clipper scheme, when performed digitally, can alias the digital system and cause it to malfunction. What transpires is an additional distortion component added by the digital system that further degrades audio quality.

Some early digital final limiting methods include, *delay line limiters* that actually delay the audio, decide where the peak level should be, reset the gain, then pass the digitized signal. While this practice works well in theory, it does provide added I.M. to program material of music origin. Subjectively this can be more irritating, to listen to, than the harmonic distortion generated from an analog non-linear clipper.

Another method makes use of the *Hilbert Transform*, which is a mathematical model of the old RF clipper scheme. This process which transposes into digital fairly easily, still generates high order I.M. product through the nature of its operation, again degrading sonic integrity of musical program material.

Once progress is made in the digital hard limiting process, improved average modulation will result, and a digitally based transmission system will be Sonically more viable.

3) Does DSP provide "transparent" operation to established analog algorithms? In the present state of broadcasting, there are analog products that perform very well. If an exact model of an analog processor is developed, theoretically it should sound at least as good as the analog model. The whole process of digitizing the audio signal, perform all operations, and then undoing the digital signal should have no adverse affect to the sonic quality of the signal.

Any abnormalities due to the digital process, will depreciate the integrity of the processed signal. Although it was earlier stated that the digital gain control function was transparent to the signal, which is a benefit. It was also questioned that the final limit process may detract from the established analog technique, and that would be a drawback.

4) What new and/or improved techniques, structures, and functions are DSP units providing at this time? As presented earlier, DSP is definitely the future. The power to develop and program with this technology is limitless. But presently all DSP units "clone" analog functions. Where are the major sonic improvements over the analog units?

5) Cloning? There seems to be a "barnstorming" by manufactures professing their digital gear as the "latest and greatest". Marketing campaigns focusing on words such as "Quantum Leap" and "Re-optimize" stand out in reference to their products. Upon examination, the present generation of DSP products are digital models of analog designs. Now unless the operation of these products can out-perform their analog predecessors, where is the quantum leap?

6) Will Standards for digital signals be established? With the broadcast operation of tomorrow happening in the total digital domain, a common thread must exist if all these different digital devices are to "talk" to one another. At this time there has been some initial discussion within the industry to consider something of this nature. But nothing firm has come of it yet.

For example,while the AES/EBU standard is already in place, there currently is not a standard for a FM composite signal if one was to be digitally connected from a processor to an exciter. This has to be addressed. What about digital STL systems?

These questions need to be answered if DSP technology is to provide increased benefits to transmission system processing. With future broadcast methods in consideration, DAB, DSB to name a few, some of the above questions may appear moot. But to the present operation of the FM medium these questions are very relevant as they affect current operation.

As with any new technology, each of these will be addressed, along with new questions yet to be answered, as this technology develops.

Final thoughts

There is no question that digital processing will make a difference. As DSP technology proceeds along the learning curve, many creative, new, and improved signal processing techniques will enter this industry. But it is the viewpoint stressed here that the difference is still yet to come. But in the meantime, ask yourself, does it really make a difference in the end? If no, then re-evaluate the situation at hand, and wait for the future to become reality.

DIGITAL AUDIO PROCESSING FOR FM: SYSTEM CONSIDERATIONS

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Abstract

While digital signal processing for FM broadcast transmission provides numerous advantages, including programmability, stability, repeatability, and high audio quality, it must still interface gracefully with the rest of the broadcast plant. This paper discusses means of analog and digital input and output interfacing, remote control standards, and the advantages and disadvantages of digital versus analog stereo encoders. It presents measured performance, including baseband spectrum, of a system using digital signal processing and an analog stereo encoder. It also discusses how the broadcaster might upgrade the plant with minimum disruption and cost to include progressively more digital elements.

Introduction

Clearly, radio broadcasting will eventually originate from an all-digital plant. However, there will be a long transition period in which digital and analog equipment coexists. If the transition cannot be smoothly managed, the cost of the transition will rise and this, in turn, will delay upgrading the plant. It is therefore important to carefully design the input and output interfaces of digital broadcast equipment so that such equipment can be used in the widest variety of systems, and so that other equipment in the plant can be upgraded with minimum disruption.

This paper focuses on the FM transmission audio processor, which is responsible for controlling modulation and for processing and coloring (or not coloring) the sound to meet the goals of station management. Such processing must interface with studio equipment, studio-to-transmitter links (STLs), and transmitters.

Transmission System Design

Traditionally, transmission systems have had one of three basic configurations. In the first, the modulation controller is co-located with the FM transmitter, and the left and right stereo audio is passed to the modulation controller through two audio-bandwidth links, which may be land lines, analog microwave links, digital microwave links, or digital land lines. In the second, the modulation controller is located at the studio (for ease of adjustment), and the composite baseband signal is passed to the transmitter through a wideband microwave link or digital land lines. A third system modulates the carrier once at the composite STL transmitter (at the studio), and then frequency-shifts the received microwave signal directly to the final FM carrier frequency at the transmitter without decoding the RF signal to baseband.

One can only achieve most competitive on-air loudness if the composite link between the output of the stereo encoder and the input to the FM exciter is free from overshoot and bounce, which add peak level to the composite signal without increasing the average level. While the best composite links are acceptably free from these distortions (which can occur in the microwave transmitter or receiver), many are not. Co-locating the modulation controller and stereo generator with the transmitter and interconnecting these elements with short pieces of wire has always achieved the best results; correctly installed wire has no bounce or overshoot. The third system has a similar error budget, as there is only one modulator (although it is located at the studio, not the transmitter).

In the past, the convenience of having the processing readily accessible at the studio, plus the economy of being able to pass the whole composite signal on a single microwave link, has often dictated use of the second system despite its potential drawbacks. With the advent of remote-controllable digital transmission processors, there is no longer any need to have the audio processor directly at hand for adjustment. A processor equipped with an RS-232 computer serial interface can be connected to a conventional switched voice-grade public telephone line through a modem. It can then be remote-controlled at the studio (or from anywhere else with hard-wired or cellular telephone service) by a personal computer equipped with appropriate software, supplied by the manufacturer of the audio processor. Since this is a two-way link, the processor can supply information (such as gain reduction meter readings or current setup control settings) to the remote computer for display, and the operator can adjust the processor as if it were local.

If a digital STL is available, it is desirable to keep the signal in the digital domain as much as possible to avoid redundant, potentially quality-degrading analog-to-digital (A/D) and digital to analog (D/A) conversions. There exists in the professional audio community a strong preference for the AES/EBU digital interface[1], which uses XLR-type connectors and 110 ohm cable, and which passes both left and right audio channels (plus various auxiliary data) at 32kHz, 44.1kHz, or 48kHz sample rates. A recent draft[2] proposes a revision of the original specification to wrap up some loose ends and to improve the inter-operability of various pieces of equipment having the interface.

One issue that is presently unresolved is the sample rate. Two pieces of equipment having AES/EBU interfaces will interconnect correctly only if the driven equipment can receive and decode the sample rate sent by the driving equipment. Particularly in Europe, 32kHz is the standard for long-haul digital transmission of broadcast audio. However, all proposed Digital Audio Broadcast (DAB) systems have 20kHz audio bandwidth and a 44.1kHz (CD) sample rate, and forward-thinking broadcasters may wish to be able to pass 20kHz audio to the transmitter for application to a future DAB transmitter. Of the two digital audio microwave links manufactured at this writing by U.S. manufacturers, one uses a 32kHz rate and one uses a 44.1kHz rate. It is clear that future digital audio systems will contain an extensive number of sample rate converters to permit transparent inter-operability. Such converters are still expensive and costly, although at least one semiconductor manufacturer is promising a low-cost integrated arbitrary sample-rate converter by early 1993. Until the sample rate issue is solved, digital interconnection of digital audio devices in the broadcast studio will be non-trivial.

Another issue is the digitization of the composite stereo baseband (including SCA and RDS subcarriers, if used). The AES/EBU standard has insufficient bandwidth to pass the full baseband, which is ordinarily 99kHz, but which can be considerably higher if composite clipping is applied to the signal. At this writing, there is an ad-hoc open industry committee studying the issue, and several proposals are on the table. It is probable that, for ease of installation, the signal will be serial and will be transmitted on one coaxial or fiber optic cable. Such a connection is only useful if the FM exciter, stereo encoder, and/or digital composite STL have digital inputs and outputs.

This raises the question of whether the stereo encoder should be digital or analog. For the integrated digital audio processor and stereo encoder recently developed by the author's company, we decided that an analog stereo encoder was more appropriate. Such an encoder requires analog left and right inputs, which in turn require two audio-frequency-range D/A converters at the output of the audio processing prior to the stereo encoder. This automatically makes analog left and right outputs available to other equipment (such as a standby stereo encoder or transmitter) without adding cost.

At the current state of the art it is possible to make an analog *or* digital stereo encoder so good that the performance of either is literally several orders-of-magnitude better than required for audible perfection. However, the digital encoder requires a very fast (and somewhat expensive) 16-bit A/D if it is to provide a high-quality analog output. It also requires a very competent reconstruction (anti-imaging) filter to prevent modulation of the exciter by the composite sampling frequency and its sidebands. From 30-

[1] AES3-1985; ANSI S4.40-1985; IEC 958

[2] AES3-199X

53,000Hz such a filter must simultaneously have a deviation from linear phase of less than ±0.03 degrees and amplitude flatness within ±0.003dB to achieve 70dB separation.

If one instead requires a *digital* composite output, the problem becomes accommodating subcarriers, which must be digitized and added to the data stream prior to its application to the exciter's digital input. Digitization of the outputs of existing analog subcarrier generators may add considerable cost to the system.

If the broadcaster chooses to use composite clipping (which we discuss further below), then an analog clipper will provide better results than any digital composite clipper we have evaluated to date because the analog unit controls peaks precisely and does not add aliasing distortion to the expected clipper-induced harmonic and intermodulation distortion. An analog clipper requires an analog baseband input, which implies that the stereo encoder (regardless of technology) must provide an analog output.

The Link From Studio to Audio Processor

If the studio is digital, the appropriate link to a digital audio processor is clearly AES/EBU. (The sample rate is still an open question.) Because the audio processor ordinarily reduces the dynamic range, its input should be at least a 20-bit word, whose 120dB dynamic range will accommodate reasonable operator gain-riding errors without clipping or excessive noise, even from CD or other wide dynamic range digital sources.

The weakest link in any digital transmission audio processor is the input A/D, because this element must have a very wide dynamic range to accommodate the unprocessed audio. Current state-of-the art in audio-range converters is slightly less than 18 bits (about 103dB), which is usable, though not ideal. Operators may have to set levels a bit more carefully than they are used to doing with analog processing systems. Alternatively, one can install an analog AGC prior to the A/D to replace the digital AGC function in the transmission processor. This expedient can greatly ease the A/D's dynamic range requirements because the A/D is now being fed by level-controlled audio. An analog AGC is also appropriate for protecting a dual STL with limited dynamic range (such as a dual microwave analog STL) in systems where the transmission processor is located at the transmitter.

Modulation Control versus Spectral Cleanliness

Avoiding program-related "splatter" in the 57–99kHz range of the FM baseband is crucial to achieving clean SCA operation. To provide satisfactory SCA service, broadcasters must prevent imperfect operation of the program channel's audio processing and stereo encoder from causing such interference.

There have always been conflicts between preventing splatter and achieving the highest average modulation without exceeding legal peak deviation limits. If the broadcaster uses a high-quality stereo encoder with competently-designed linear (non-distorting) low-pass filters to band-limit the left and right audio channels to less than 19kHz, there is no problem: aliasing-related main-channel to subchannel crosstalk is negligible and the baseband above 57kHz is clean. However, few broadcasters use such stereo encoders because sharp-cutoff linear filters overshoot and ring, and they will do so even if they are perfectly phase-linear (that is, if they have constant delay at all frequencies in their passband). Therefore, all major stereo encoder manufacturers in the United States have developed proprietary non-overshooting analog filters that use various non-linear means to suppress overshoot and ringing. The performance of these filters is exceedingly level-dependent, and they must be accurately matched to the 100% modulation level of the FM channel to perform correctly. If their output level is set too high they will not suppress overshoot, and if their output level is too low they will not produce full modulation without also generating substantial distortion (because the broadcaster will over-drive their nonlinear elements in an attempt to achieve full legal modulation). Additionally, because all such filters are approximations, they never suppress overshoot absolutely, but only reduce it by comparison to equally selective linear filters.

As we will see below, digital signal processing (DSP) technology allows us to build non-linear filters with substantially fewer compromises than are required for analog filters, enabling the broadcaster to maximize loudness while simultaneously transmitting completely clean SCA subcarriers.

Controlling Composite Modulation by Peak-Limiting the Audio Channels

Controlling the modulation of the composite baseband by peak limiting the individual left and right audio inputs to the stereo encoder works because of the "interleaving" property of the stereo baseband: To a first approximation, the peak level of the composite is equal to the peak level produced by the larger of the left or right audio inputs.

There are several caveats. If the peak level of the 38kHz stereophonic subcarrier waveform does not coincide in time with the peak of the audio, the peak modulation is lower than it would be if the two peaks were precisely coincident. Although an inaccuracy in the approximation, this causes no overmodulation and is thus not a problem.

A more subtle inaccuracy is caused by the 19kHz pilot tone. Because it is correlated to the 38kHz suppressed subcarrier, its peak level does not add arithmetically to the peak level of the rest of the composite signal. Of course, if the modulation is pure monophonic (L+R), the pilot will add arithmetically because there is no stereo subcarrier — 91% L+R modulation will add to 9% pilot to produce 100% composite modulation. However, if the modulation is pure left or right (corresponding to equal levels of L+R and L-R), the same conditions will only result in approximately 97.2% composite modulation because the pilot has partially interleaved with the remaining composite signal. This means that as the stereophonic content of the program increases, peak modulation decreases slightly, and the transmission is not as loud as it could be if the full peak modulation capability of the channel were used.

Further (and usually more significant) inaccuracies can be caused by composite microwave STLs and exciters with poor low-frequency transient accuracy. Low frequency program energy can caused such devices to ring and "bounce," causing unnecessarily high peak modulation.

Controlling Composite Modulation by Composite Clipping

Except for exciter bounce, all of these problems can be "solved" by clipping the composite signal, which ensures that the peak level of the composite is always precisely constrained to the desired level. Early composite clippers acted on the entire composite signal (including the pilot tone), and were able to achieve significantly higher loudness at the expense of producing harmonic and intermodulation distortion throughout the baseband. The FCC halted this practice by stating that such processing could not cause any FCC Rule to be violated. Since clipping the entire baseband often brought the instantaneous pilot injection below the 8% modulation limit stipulated by the FCC Rules, it became clear that the pilot could not be clipped. A new generation of composite clippers that removed the pilot prior to clipping and re-injected it afterwards then appeared on the market. These new clippers still produced distortion throughout the baseband (including the 57–99kHz region occupied by SCAs), and had the additional disadvantage of being "unaware" of the interleaving between the pilot and remaining components in the composite baseband. However, since such interleaving only reduces modulation by about 0.25dB (an inaudible amount), it is of little practical concern.

Composite clipping has an advantage over conventional audio-domain processing only when the program has very large amounts of L–R energy. Assuming that the clipping is symmetrical, it will produce mainly odd-order harmonic and intermodulation distortion. All of the harmonic distortion (and some of the sum-frequency IM distortion) caused by clipping the 23–53kHz stereo subchannel will fall above 53kHz. If the receiver has a "Walsh-function" stereo decoder (which is insensitive to energy at harmonics of 38kHz), it will demodulate such above-53kHz energy so that it appears above 15kHz, where it is audibly suppressed by a combination of receiver de-emphasis, pilot tone filtering, and the natural insensitivity of the human ear to frequencies above 15–20kHz. Thus some of the distortion caused by clipping the stereo subchannel is completely inaudible.

Unfortunately, the amount of distortion that *will* be heard by the listener is unpredictable. It is highly dependent on the ratio of L+R to L–R energy in the stereo audio, and it depends on the technology used in the receiver's stereo decoder. Modern Walsh-function decoders are only sensitive to frequencies in the 0-57kHz range of the baseband. However, older-technology "square-wave switching" decoders demodulate any energy within 15kHz of 38kHz and all odd harmonics of 38kHz (114, 190, 266...kHz), and thus may sound noticeably more distorted than Walsh-function decoders when reproducing compositely-clipped audio, whose energy falls off quite slowly at very high frequencies.

If the program material is monophonic (such as center-channel voice), then the audible effect of composite clipping is similar to audio clipping, except that the composite clipper will produce distortion products that extend up into the stereo subchannel region (higher than 19kHz). The receiver assumes that the distortion above 19kHz is L–R and decodes it by frequency-shifting it down by 38kHz and making it appear out-of-phase in the left and right channels. The decoded distortion is thus not harmonically or spatially related to the undistorted program material being processed. It is well-known that such non-harmonic distortion (which is similar to aliasing in a digital system) is very offensive to the ear, and the fact that it decodes in a different spatial location than the original energy that was clipped to generate it may make it even more obvious.

Of course, if the station wants to broadcast acceptably-clean subcarriers in the 57–99kHz region, composite clipping is impractical except in the smallest amounts (meaning less than 0.5dB clipping of low-duty cycle overshoots in the transmission system), because it otherwise introduces too much spurious spectrum in the 57–99kHz region. From our previous discussion, it is also clear that the audible distortion produced at the receiver is highly dependent on the nature of the program material and the design of the receiver. We must therefore consider alternatives.

Digital Non-Linear Filtering for Modulation and Spectrum Control

Previous analog audio processors designed by the author have used the frequency-contoured sidechain (FCS) overshoot corrector[3] to provide a "band-limited safety clipper" function in the audio (left/right) domain. The FCS circuit operates by detecting any overshoots at its inputs with a center-clipper circuit (whose output consists only of the parts of the input waveform exceeding 100% modulation), low-pass filtering these overshoots with an analog filter having unity gain at low frequencies and a rising response just before its 15kHz cutoff frequency, and subtracting the output of this filter from a delayed version of the un-clipped input signal. (The delay approximately matches the group delay of the low-pass filter.) The resulting peak-controlled signal is then passed through a group delay corrector so that the delay through the entire system is approximately constant with frequency.

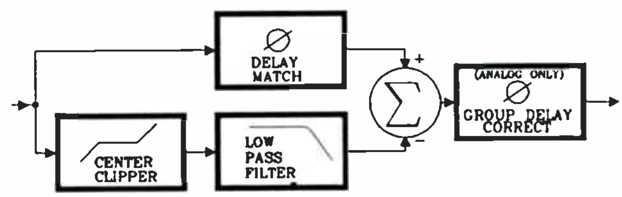

Figure 1: FCS Overshoot Corrector

Because of approximations within the system, a safety clipper is required after the system to catch residual overshoots caused by the approximations. When the system is operated "aggressively," the safety clipper generates spurious spectrum about 50dB below 100% modulation in the 57–99kHz region. While this is about 15dB better than the spurious spectrum caused by a comparably-operated composite clipper, experience has shown that it can still generate audible interference in SCA subcarriers. For this reason, the author's company developed an accessory filter card, consisting of two additional cascaded overshoot correctors per audio channel, to replace the safety clipper. This processing typically reduces the spurious spectrum above 62kHz to –80dB while simultaneously reducing residual overshoot to about 0.3dB. Field experience shows that the card does its job; improving this performance would provide no further audible benefit. However, the card is complex and costly.

[3] U.S. Patent #4,460,871

The principal performance limitation of the analog FCS system is that it cannot be perfectly phase-linear. Residual, uncorrected group delay errors cause imperfect overshoot suppression. If this system is instead realized in the digital domain, we can realize the frequency-contoured sidechain lowpass filter as a constant group-delay finite impulse response (FIR) filter to achieve more accurate overshoot suppression. By making the filter roll off more sharply above 15kHz than its analog counterpart, we can suppress undesired spectral energy above 57kHz more effectively. We have successfully done this in a new digital audio processor. Residual overshoot is reduced to about 0.2dB, while spurious spectrum is typically –85dB in the 57–99kHz region. Therefore, the performance of one stage of optimized digital overshoot correction is slightly better than three stages of analog correction.

Figures 2, 3, and 4 respectively show the 0–100kHz baseband spectrum of a single-stage analog overshoot corrector with safety clipper, a three-stage analog overshoot corrector (without safety clipper), and a digital overshoot corrector. All were taken with an 801-line FFT analyzer[4] using its "maximum peak hold" mode over a five minute observation with pulsed stereophonic USASI noise per NRSC-1 specifications[5]. This test signal was developed by the National Radio Systems Committee AM Subgroup to vigorously exercise the clipping, bandwidth-limiting, and overshoot correction systems in audio processors. The spectrum at the processor's output with this excitation is assumed to be closed to "worst-case."

One can see that the single-stage analog circuit introduces considerable interference in the 57–99kHz part of the baseband. The three-stage analog circuit protects the region from 62–99kHz by 80dB or better. However, the digital circuit is more selective, protecting the entire 57–99kHz region better than 80dB, and protecting the pilot tone far better than the analog circuit. The digital circuit will also provide greatly superior protection to a 57kHz RDS subcarrier.

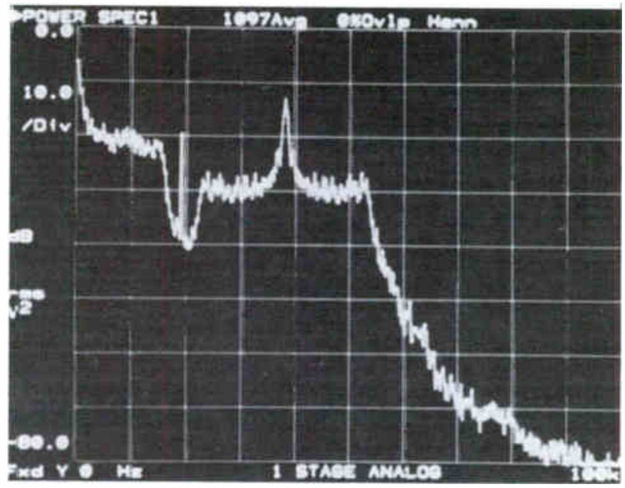

Figure 2: One stage of analog overshoot correction with safety clipper.

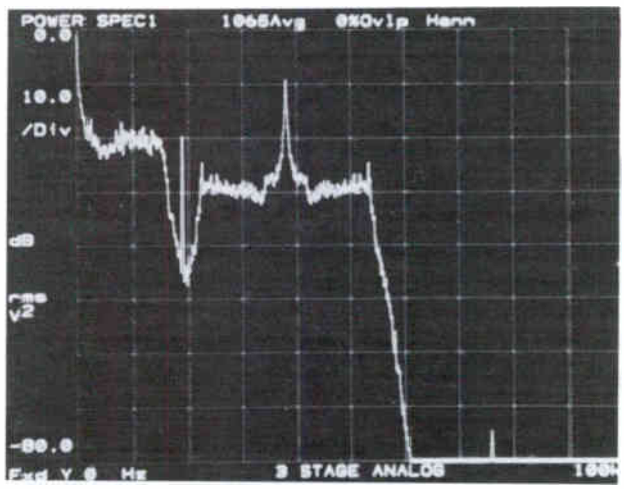

Figure 3: Three stages of analog overshoot correction with no safety clipper.

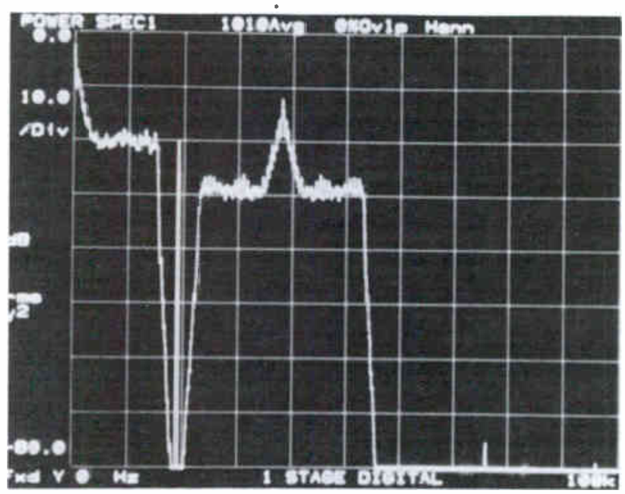

Figure 4: One stage of digital overshoot correction with no safety clipper.

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4 Hewlett-Packard 3562A

5 ANSI/EIA-549-1988, NRSC-1 AM Preemphasis/Deemphasis and Broadcast Transmission Bandwidth Specifications.

Conclusion

Incorporating digital signal processing in the FM plant increases on-air quality but provides many challenges to the broadcaster. Most plants will be hybrid, containing both analog and digital elements, for many years to come. Broadcasters are well-advised to use established industry standards for digital interconnection (most importantly AES/EBU), although further standardization efforts are necessary to determine the number of bits and the sample rates. Eventually, all professional equipment will probably incorporate sample rate converters to operate at least at 32, 44.1, and 48kHz to ensure easy inter-operability. Digital equipment should be available with (possibly optional) analog inputs and outputs for the foreseeable future, to facilitate interconnection with analog equipment.

The availability of wide dynamic range digital STLs using data compression, and of readily remote-controllable digital transmission audio processors will make co-location of audio processor, stereo encoder, and transmitter more attractive than in the past. The processor will be readily remote-controllable from a personal computer via modem, and the wide dynamic range of the STL will reduce or eliminate problems of noise build-up caused by audio compression in the processor.

Successful operation of interference-free SCA subcarrier services precludes using composite clipping (whether in the analog or digital domain) as a prime means of overshoot control, despite its simplicity. Provided that the composite studio-to-transmitter link (if used) and the FM RF exciter are free from overshoot and bounce, non-linear filters prior to the stereo encoder control peak modulation control almost as well as composite clipping, and typically yield a 57–99kHz baseband several orders-of-magnitude cleaner. However, a fully-effective (i.e., subjectively unimprovable) analog overshoot control system is complicated and expensive. Exploiting the intrinsic phase-linearity of properly-designed FIR filters, we have created a digital overshoot correction system with fewer approximations than its analog counterpart, achieving better performance with one stage than with three stages of analog overshoot correction. Further advantages of the digital circuit include far better stability over time and temperature and virtually no unit-to-unit variation.

Appendix: Effect of Pilot Interleaving on Peak Modulation

In this appendix we provide a mathematical model for the pilot interleaving effects discussed above. We describe the stereo baseband as follows:

$$B(t) = 0.45(L-R)\sin(\omega_{sc}t) + 0.1\sin\left(\frac{\omega_{sc}t}{2}\right) + 0.45(L+R) \quad (1)$$

where $\omega_{sc} = 2\pi 38\text{kHz}$

and the maximum permitted peak modulation level (normally ±75kHz deviation) is normalized to unity. We see that the three components are the stereo subchannel (generated by multiplying the L–R signal by a 38kHz sine wave), the 19kHz pilot tone, and the L+R main channel.

To compute the worst-case maximum peak level, we differentiate the expression for the baseband with respect to time and set the resulting expression equal to 0:

$$\frac{dB}{dt} = (9L-9R)\cos(\omega_{sc}t) + \cos\left(\frac{\omega_{sc}t}{2}\right) = 0 \quad (2)$$

To determine the time at which the maximum absolute value occurs we solve Eq. 2 for t. We then substitute the result into Eq. 1 and evaluate Eq. 1 to get the actual normalized peak level. Because Eq. 2 is periodic it has an infinite number of solutions, any of which will give the same result for the peak level when substituted into Eq. 1. For convenience, we ordinarily solve for the solution closest to $t = 0$.

To simplify the mathematics, we note that the highest peak modulation occurs when the peak level of the audio coincides with the peak level of the 38kHz subcarrier. We can therefore represent the left and right audio signals as fixed DC voltages having peak values equal to the maximum peak values of the left and right signals. To further simplify the mathematics, we normalize ω_{sc} to unity. Eqs. 1 and 2 then simplify as follows:

$$B(t)_{pk} = 0.45(L-R)\sin t + 0.1\sin\left(\frac{t}{2}\right) + 0.45(L+R) \quad (3)$$

and

$$\frac{dB_{pk}}{dt} = (9L - 9R)\cos t + \cos\left(\frac{t}{2}\right) = 0 \qquad (4)$$

Eq. 4 has no analytical solution for *t;* we must solve it numerically for each value of L and R that we wish to analyze, and then substitute the result into Eq. 3 to get the normalized peak level. Eq. 4 is well-behaved and any one of a number of standard techniques (like the bisection, false position, or secant methods)[6] can be used to solve it numerically.

The salient result is as follows: If we first set L = R = 1 (the pure monophonic case), then the normalized peak level of the composite, including pilot tone at 10% injection, is 1. If we now set L = 1 and R = 0, then the normalized peak level of the composite decreases to 0.9720, or –0.25dB. So the effect is negligible for almost all practical purposes.

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[6] *see, for example:* W.H. Press, B.P. Flannery, S.A. Teukolsky, & W.T. Vetterling: *Numerical Recipes — The Art of Scientific Computing,* Chapter 9. Cambridge, Cambridge University Press, 1986.

AC-2: HIGH-QUALITY DIGITAL AUDIO CODING FOR BROADCASTING AND STORAGE

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ABSTRACT

A family of wideband digital audio transform coders suitable for use in broadcast applications is presented. The coders operate at a variety of bit-rates ranging from 192 to 64 kb/sec per channel, and offer various tradeoffs in bit-rate, coding delay, and audio bandwidth. In particular, a next-generation audio coder is introduced in which both frequency-domain and temporal masking effects of human hearing are exploited during bit-rate reduction. The coder employs a new filterbank which dynamically adjusts frequency and time resolution on a signal-dependent basis. Current and future applications of the coding family are described.

INTRODUCTION

The need for a significant reduction in bit-rate for wideband digital audio signal transmission has recently led to the development of new psycho-acoustically-based compression techniques (see, for example [1,2,3]). In the basic approach, the audio frequency range is subdivided into a multiplicity of bands which approximate human auditory critical bandwidths. The frequency division is accomplished using a filterbank implemented as a frequency transform or sub-band filter. Frequency-domain masking properties of the human auditory system, ideally analyzed in such a framework, are exploited to maximize perceived fidelity of the signal transmitted at a given bit-rate. Temporal masking effects can equally be used to advantage in audio coding, although this notion has largely remained unexplored in the literature.

In this paper, the AC-2 family of audio transform coders operating at 192, 128, 96 and 64 kbit/sec per monophonic channel will be presented. The various coders within the family are related by the frequency-domain psychoacoustic models employed, and frequency subdivision according to critical bands. AC-2 operates at various sampling rates, including 48, 44.1, 44.0559, and 16 kHz. Encoding and decoding time delay can be traded for bit-rate within the family.

In addition to discussing the coders presented previously in [2], this paper introduces a next-generation design employing a time-varying frequency division scheme. This design incorporates models of both frequency and temporal masking in bit-rate reduction. Denoted AC-2A, the new coder dynami-cally selects an appropriate filterbank for each 10 ms analysis interval of the input signal. A characteristic feature of AC-2A is that an optimal window and frequency transform are selected for each analysis interval depending upon temporal characteristics of the signal contained within. This allows the usual tradeoff between filterbank time and frequency resolution to occur on a signal-dependent basis, as opposed to earlier approaches wherein filterbank resolution is selected a priori and then fixed in time. In this paper, the designation AC-2 generically refers to the family of coders, and AC-2A specifically refers to the next-generation version.

A primary design feature of AC-2 is low decoder implementation cost. Digital signal processing techniques and VLSI hardware have recently advanced to the point where AC-2 is a cost-effective solution in many different applications. Current-generation fixed-point DSPs are capable of implementing two channels of AC-2 encoding or decoding in a single chip. For example, a stereo decoder running at 128 kbits/sec with a 48 kHz sampling rate has been implemented in one 20 MHz Motorola DSP5600x processor. A stereo decoder has also been implemented on a 16-bit processor, Texas Instruments' 16-bit TMS320C50, in a single chip with no external RAM [4]. For the most cost-sensitive and high volume applications, custom VLSI offers an attractive solution. Our studies suggest that a stereo decoder can be implemented on a dedicated integrated circuit at a chip cost under $10 in quantity including all RAM and ROM.

APPLICATIONS

The AC-2 family of coders has been designed and optimized to fulfill design requirements commonly encountered in broadcasting. There are a number of current applications which exemplify this. For example, AC-2 has been selected for use in a new multichannel NTSC compression system. This system transmits multiple channels of compressed video and audio in a single satellite transponder for delivery of entertainment-quality programming to cable head ends and backyard dish owners. AC-2 has also been selected for use in High-Definition Television (HDTV) system prototypes submitted to the Advanced Television Test Center (ATTC) by two proponents in the United States. These two systems, employing advanced digital video and audio coding and modulation techniques, are undergoing objective and subjective performance evaluation in 1992 at tests conducted at the ATTC, under the auspices of the Federal Communications Commission. A low-delay (7.5 ms) AC-2 coder is employed in a new 950 MHz Digital Studio to Transmitter Link (DSTL) [5]. In this system, efficient audio coding and digital modulation techniques are combined to yield a 2:1 improvement in occupied bandwidth compared to analog STLs. AC-2 is also employed in the distribution of high-quality compressed audio, including two-channel surround-encoded movie soundtracks, over T1 telecom and satellite links, in disk-based storage and digital cart machines, and radio network contribution and distribution. Other applications include ISDN, digital audio broadcasting, and backhauling with SCPC or bandedge techniques.

GENERALIZED AC-2 CODING ALGORITHM

In this section, we present the generalized audio coding framework embodied in the AC-2A audio compression algorithm. AC-2A employs many of the coding principles of earlier-generation AC-2 coders [6], but with a powerful new provision for dynamic adaptation of time and frequency resolution of the filterbank. The fixed-resolution AC-2 coder can be considered as a special case of the generalized algorithm, as shown later in this section.

The operational features common to all AC-2 coders are as follows. At the encoder input, PCM audio samples are buffered into sample data blocks. Each sample block is multiplied by a window function prior to being transformed into a set of frequency domain (transform) coefficients. The transform implements a critically-sampled filterbank, which implies that the number of scalar transform coefficients produced per unit time equals the number of input samples in that time interval.

After a block of windowed samples are transformed, adjacent transform coefficients are grouped into sub-bands which approximate the nonuniform-width critical bands of human hearing. Coefficients within one sub-band are converted to a frequency block floating-point representation, with one or more mantissas per exponent. Mantissas are generated by left shifting each transform coefficient within a group by an amount proportional to the exponent. The exponents collectively provide an estimate of the log-spectral envelope for the current audio block, as represented on a critical-band frequency scale.

The next encoding step is to compute step-size information for a forward-adaptive quantizer which encodes the transform coefficient mantissas. The step-size information is derived by a dynamic bit-allocation routine which analyzes the sub-band exponents for this purpose. In the final encoder stage, the coefficient mantissa bits are multiplexed with exponent bits, and optionally, error control and auxiliary data bits, for transmission to the decoder.

In the decoder, exponent and mantissa data are demultiplexed, and channel transmission errors are corrected. The received exponents are processed in a manner identical to the encoder to derive inverse quantizer step-sizes. A sub-band block floating-point expander then linearizes the transform coefficient mantissas by right-shift operations, and passes them to the inverse transform stage. The inverse transform converts each set of coefficients back into a sampled data block, which is subsequently windowed, overlapped, and summed with the adjacent windowed sample block to generate an approximation to the original audio waveform. More details concerning the basic principles employed in AC-2 audio coding can be found in [6]. A description of some of the psycho-acoustic principles employed can be found in [2].

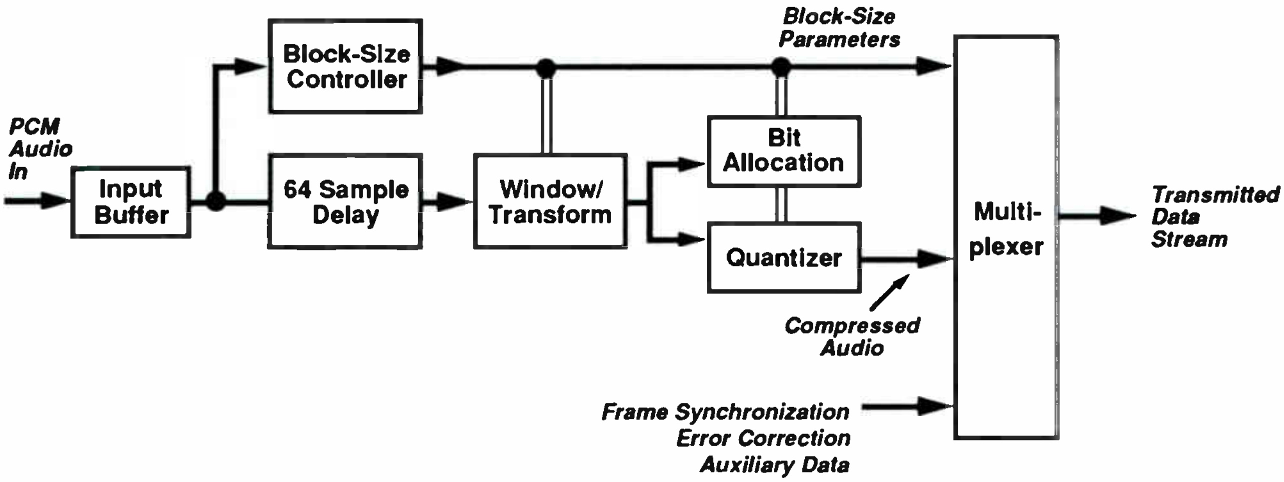

Fig. 1 Generalized AC-2 Audio Encoder

We now turn to the generalized coding algorithm, presented in Figs. 1 and 2. In the encoder, time and frequency resolution is dynamically adjusted by selecting the optimal transform block length for each 10 ms analysis interval. As shown in Fig. 1, a block-size controller preprocesses each block of input samples and supplies block-size information (a block I.D.) to each of the following processing stages. The criteria for transform block size selection are derived from temporal masking characteristics of human hearing (for example, as described in [7]). In general, if the signal under analysis is continuous in nature (or from a statistical perspective, stationary), a long block length (512 to 1024 samples) is most appropriate. Conversely, signals which are predominately transitory in nature require accurate temporal resolution, which is achieved with a relatively short block length (128 samples). The block-size controller serves as the audio waveform equivalent of an image processing "edge detector". The input signal is band limited with a 4 kHz high-pass filter, then segmented hierarchically to determine the position and rate of attack of the signal in the block. A 64 sample delay is incorporated in the main audio path in Fig. 1, allowing the block-size controller to anticipate upcoming temporal events. Side information conveying the block I.D. for the current analysis interval is multiplexed with the quantized transform coefficient data and

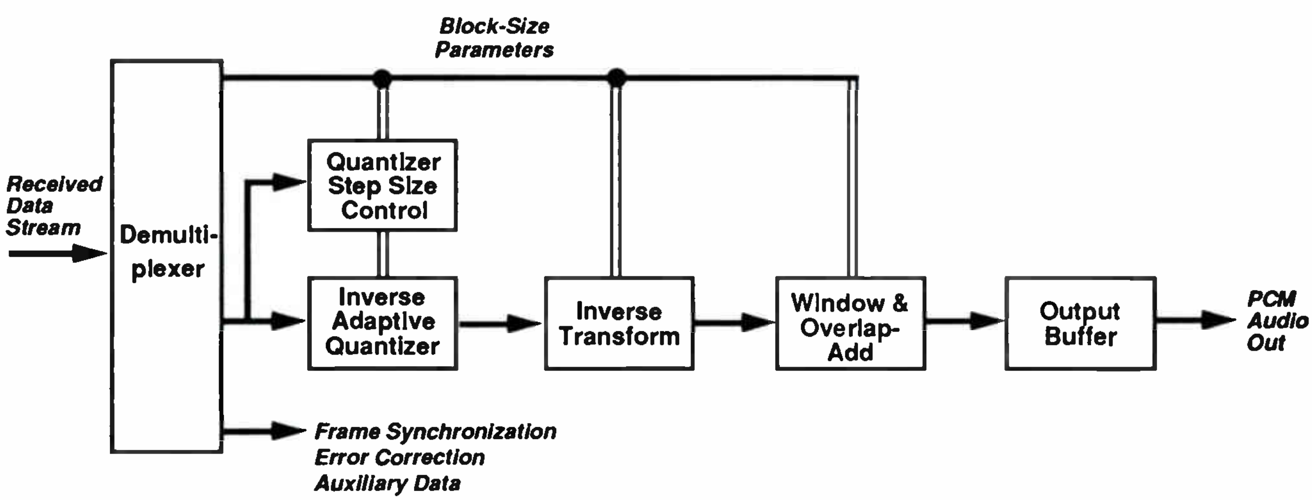

Fig. 2 Generalized AC-2 Audio Decoder

transmitted to the decoder. With error protection, the block I.D. typically comprises 1% of the total bit-rate.

In the decoder, presented in Fig. 2, the block I.D. is first extracted from the bitstream and error corrected. The remainder of the transform coefficient data is demultiplexed and decoded as in earlier AC-2 coders. Algorithmically, therefore, the only difference between the fixed and variable resolution decoders is additional logic to process the block I.D. and to allow processing a variable number of transform coefficients in the inverse quantizer step-size, coefficient linearization, and inverse transform stages. In fact, the fixed-resolution AC-2 encoder and decoder can be considered as a special case of the generalized model. When the block-size controller selects a constant frame length independently of the input signal, the fixed-resolution algorithm is implemented.

One advantage of the block length adaptation scheme shown in Fig. 1 is that enhancements to the temporal masking model can be made in the encoder without necessitating a change in the installed base of existing decoders. Secondly, perceived audio quality is improved by increasing encoder complexity while keeping the decoder complexity nearly constant. Therefore, AC-2A is appropriate for situations where the number of decoders is large relative to the number of encoders.

Our results indicate that at a sampling rate of 48 kHz, a variable resolution coder employing block lengths of 128 or 512 provide a significant subjective quality improvement compared to a fixed-resolution coder using either length. In particular, transitory signals produced by castanets, glockenspiel, and cowbells are improved during fast-attack regions of the signal, as shown in the next section. Block length transitions occur seamlessly; the decoder performs an optimal crossfade between blocks of differing length to eliminate audible switching artifacts.

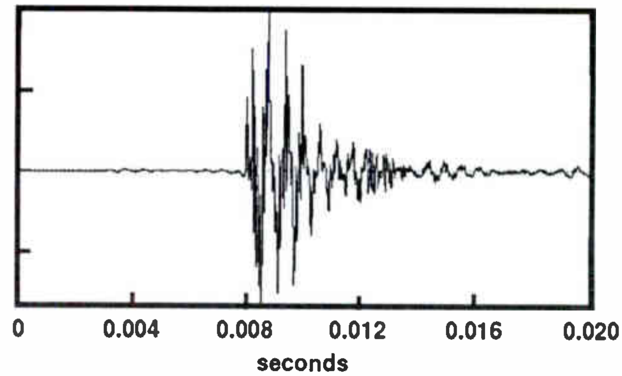

Fig. 3a Original castanet signal

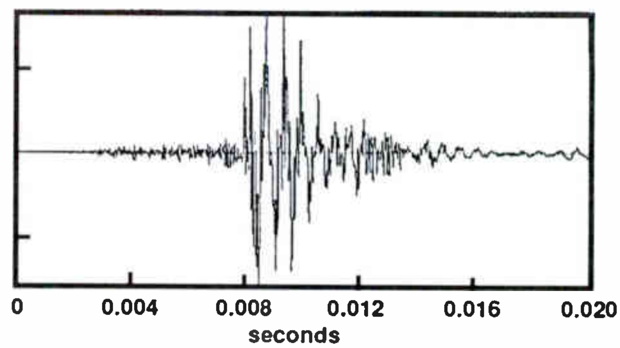

Fig. 3b Castanet coded at 128 kb/s/ch with 512 point fixed block coder

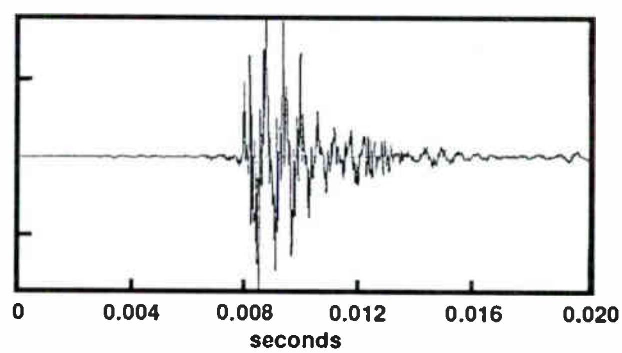

Fig. 3c Castanet coded at 128 kb/s/ch with adaptive block coder

CHARACTERISTICS OF THE ADAPTIVE TRANSFORM CODER

The design of the AC-2A filterbank strives for both optimal time/frequency resolution [9] and low complexity costs. The use of a TDAC-based transform, as described in [6], has the advantage of the low computational cost of an FFT while keeping excellent selectivity characteristics for a given block size. In this section, a description of the characteristics of the adaptive transform used in the AC-2A coders is presented. In particular, the frequency response of the filterbank is analyzed in light of error masking requirements and compared to another proposal based on sub-band filters.

Exploitation of frequency-domain psychoacoustic models has led to high compression ratios in audio coding. Perceptually relevant sub-bands are identified in the input signal, and the appropriate fraction of the targeted bit-rate is assigned for their representation. In order to ensure spectral confinement of the quantization error, the filterbank should meet specific requirements in terms of selectivity, ultimate rejection, and fall-off rate derived from the auditory masking curves [2]. For example, for a full scale steady state signal at low frequencies, the ultimate rejection required is on the order of 100 dB, the fall-off rate on the order of 100 dB per octave, and the frequency selectivity less than the minimal critical band. These characteristics are very well approximated by the long block transform. In particular the frequency resolution of this block is about 90 Hz while the time resolution is about 10 milliseconds.

In contrast, the temporal masking of transient events requires a time resolution on the order of few milliseconds. Masking curves change dramatically depending not only on the intensity of the signal and its spectral characteristics, but also on its temporal characteristics [7]. Using the long block transform to handle transient-like type of signals introduces a very audible distortion known as pre-echo, unless high bit-rates are selected. Since frequency resolution is not very crucial in this case, the short block transform, with a time resolution of about 2.7 milliseconds (frequency resolution of about 400 Hz), achieves better results (see Fig. 3c).

In Fig. 3 a-c the original (Fig. 3a) castanet sample is shown compared to the signal encoded at 128 kb/s/channel with a long block transform (Fig. 3b) and a short block transform (Fig. 3c). Notice how in Fig. 3b the signal energy is smeared through the block duration, while the attack portion is much better defined in Fig. 3c. The pre-echo of Fig. 3b is spread over an interval of ≈ 4 milliseconds and is, therefore, very audible [7].

In order to correctly represent both sharp attacks and stationary signals, the AC-2A filter bank dynamically switches between long and short block lengths depending on the characteristics of the input signal. A look-ahead mechanism allows for detection of transitory signals coming in the next frame, and selects an intermediate size transform in order to smoothly change between long and short blocks. In this fashion, the usual tradeoff between time and frequency resolution occurs on a signal-dependent basis instead of a priori, thus optimally accommodating a larger variety of signals.

In Fig. 4 a-b the frequency response of the filterbank used in AC-2A coders is shown. The long block (Fig. 4a) is used when the signal is stationary (no transient is detected). A transition block serves as a bridge to the short block (Fig. 4b) which is selected

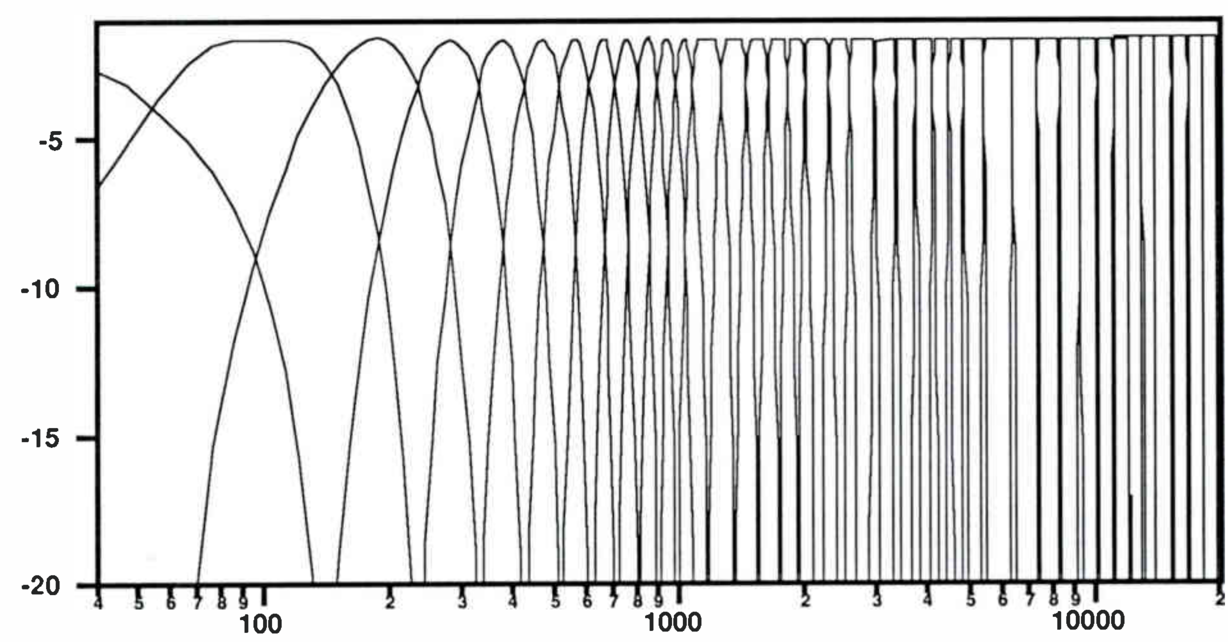

Fig. 4a Filterbank frequency response for 512-point transform

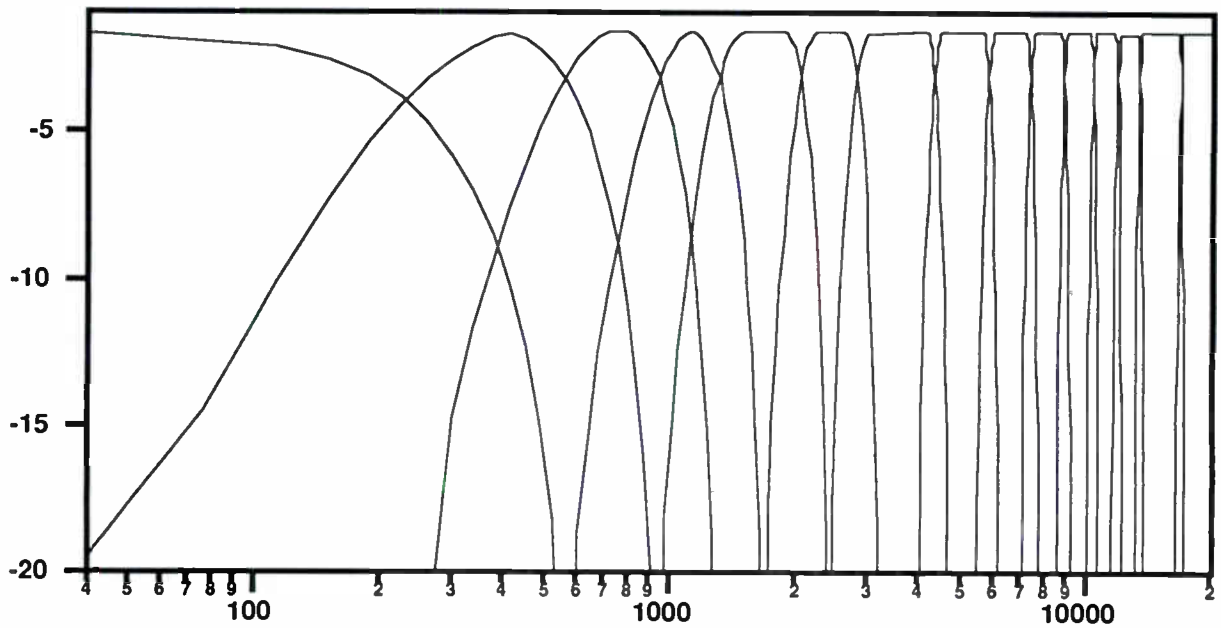

Fig. 4b Filterbank frequency response for 128-point transform

when the input signal amplitude varies rapidly in time. Notice that the first band is normalized to match the amplitude response of the other bands. The core transform is an extension of the Modified Cosine Transform/Modified Sine Transform, MDCT/MDST as presented in [8]. The input signal is overlapped by 50% and critically sampled. Furthermore the crossfade between adjacent blocks ensures a reduction of the discontinuities in the output stream from the transform.

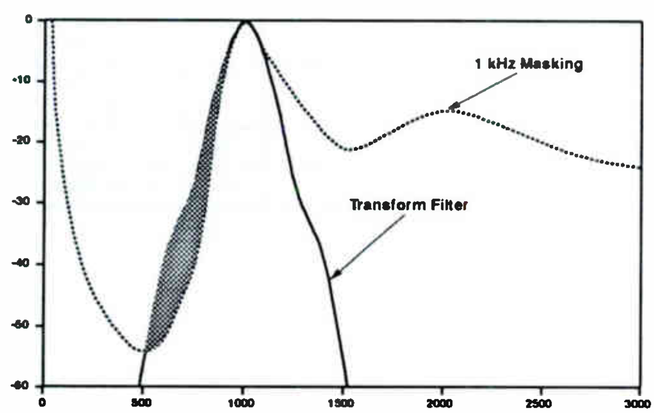

Fig. 5a Comparison between 512 pt. transform filter and 1 kHz human auditory selectivity

In Fig. 5a, the frequency response of the long block AC-2 filterbank at 1 kHz is represented together with the masking curve for a 100 dB S.P.L. 1 kHz sine wave.

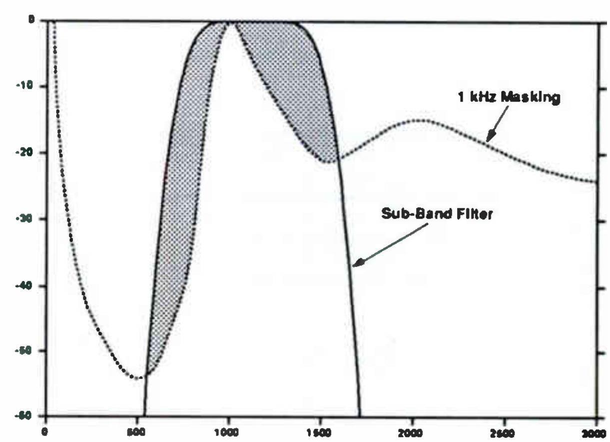

Fig. 5b Comparison between sub-band filter and 1 kHz human auditory selectivity

The shaded area indicates frequency regions where the filter response exceeds the masking threshold. Quantization noise could possibly be localized in these regions, introducing audible distortion. AC-2 frequency selectivity however, is very close to that required. Fig. 5b shows the response of the proposed ISO-MPEG filterbank. Although the fall off rate of the filterbank is very steep, its bandwidth is wider than the masking curve at low and mid-range frequencies, thus potentially spreading the quantization noise in a frequency region not masked by the input signal. Notice how the shaded regions are larger in Fig. 5 b.

CONCLUSIONS

A family of adaptable transform coders for high quality audio operating at 192, 128, 96, and 64 kb/s/channel has been presented. The AC-2 coders employ the same basic frequency domain psychoacoustics models [2]. AC-2A additionally provides a means for dynamically adjusting filterbank resolution and for incorporating temporal masking models. A transform coder with adaptable input block size is highly desirable in order to preserve the frequency/temporal characteristics of the audio signals. The AC-2A approach has significant advantages if compared with other examples in the literature since it uses a very selective window, a critically sampled transform, and flexibly adjusts to transition regions.

As shown in Table 1, encoding/decoding attributes can be traded for bit-rate within the family. Time and frequency resolution characteristics ensure effective exploitation of the auditory masking curves, as shown in the previous sections. All the examples in Table 1 operate at a sampling rate of 48 kHz except for the commentary coder which operates at 16 kHz.

The coders have been implemented using off-the shelf, 24 or 16 bit fixed-point arithmetic DSP chips. A proposed custom VLSI circuit design indicates that realization costs can be reduced to below $10. The number of Multiply and Accumulate Operations, MACs, in the fixed blocksize AC-2 encoder/decoder implementation totals to 1.06 million/sec/sample. In the adaptive coders, the transient detection mechanism slightly increases the computational complexity of the encoding system.

The last column in Table 1 classifies the AC-2 coders quality in terms of the CCIR categories as specified in [10]. In particular, for Contribution and Distribution, the audio quality of the signals reproduced after five (Contribution) or three (Distribution) tandem coding/decoding processes must be equivalent to that of the source signals represented in linear 16 bit PCM format.

The AC-2 family satisfies the needs commonly encountered in broadcasting and storage of high-quality audio. AC-2 coders are used in a number of different applications, including HDTV, DSTL, telecom and satellite links transmission, and hard disk-based recording systems.

Bit-Rate (kb/s/chan)	Block Length	Bandwidth (Hz)	Frequency Resolution (Hz)	Time Resolution (ms)	Complexity Encoder	Complexity Decoder	Delay (ms)	Quality
192	fixed	20 k	375	2.7	Low	Low	12	Contribution
192	adaptive	20 k	93.7	2.7	Medium/Low	Low	90	Contribution
128	fixed	20 k	93.7	10.7	Low	Low	50	Distribution/ Emission
128	adaptive	20 k	93.7	2.7	Medium/Low	Low	90	Distribution/ Emission
96	adaptive	20 k	93.7	2.7	Medium/Low	Low	90	Emission
64	fixed	7.5 k	31	32	Low	Low	80	Commentary

Table 1
Parameters of Several AC-2 Digital Audio Coders

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8. J. P. Princen and A. B. Bradley, "Analysis/Synthesis Filter Bank Design Based on Time Domain Aliasing Cancellation," IEEE Transactions on Acoustics, Speech, and Signal Processing, vol. ASSP-34, no. 5, pp. 1153-1161, October 1986.

9. M. Bosi, G. Davidson, and L. Fielder, "Time versus Frequency Resolution in a Low-Rate, High Quality Audio Transform Coder," Presented at the IEEE ASSP Workshop on Appl. of Sig. Proc. to Audio and Acous., Session3-4, New Paltz, New York, October 1991.

10. CCIR/CE10/TG10-2/DOC/002E

INTERACTIVE VIDEO

Sunday, April 12, 1992

Moderator:
Charles Dages, CBS, New York, New York

THE BIRTH OF AN INDUSTRY
Thomas P. Friel
TV Answer, Inc.
Reston, Virginia

***THE INTOUCH TV™ SYSTEM, A TECHNOLOGY DESCRIPTION**
Thad A. Young and William C. Laumeister
Interactive Systems
Beaverton, Oregon

PRODUCTION PROCESSES FOR INTERACTIVE TELEVISION
Thomas Kanady
Interactive Network
Mountain View, California

***PAY PER VIEW-VIDEO ON DEMAND**
Jeff Roman
Jerrold Communications
Hatboro, Pennsylvania

***NEW INTERACTIVE TELEVISION APPLICATIONS OF T-NET**
Louis Martinez
Radio Telecom and Technology, Inc.
Perris, California

*Paper not available at the time of publication.

THE BIRTH OF AN INDUSTRY

Thomas P. Friel
TV Answer, Inc.
Reston, Virginia

On January 16, 1992, the FCC authorized the use of radio spectrum for the application of 2-way interactive video and data services. This decision set the stage for the birth of an industry and the creation of a whole new way for the consumer to interact with the world. For the broadcast industry, it means achieving levels of viewer interest and involvement never before possible and for advertisers it means a whole new way to generate immediate direct responses. TV Answer (TVA) has the first patented technology based on RF digital technology transmitted via VSAT cell sites. The Hughes Network System will provide the VSAT network and transponder space and the Satellite Hub site is constructed and operable at the Company headquarters in Reston, Virginia. The TVA unit is a television set top unit that acts as a CPU and radio transmitter/receiver. The home unit includes a "point and click" joystick which activates the system for two-way communication.

HISTORY

TV Answer has been a pioneer for 2-Way TV. Fernando Morales, president and CEO,is a microwave engineer who holds several patents. The two patents that are specific to TV Answer's technology are: "Control of RF Answer Pulses in a TV Answer Back System" (7/5/88) and "Wireless Transmission From the Television Set to the Television Station" (5/27/86). These patents control a computing device with software and data cards while adding value to your TV through interactivity of both video and data services.

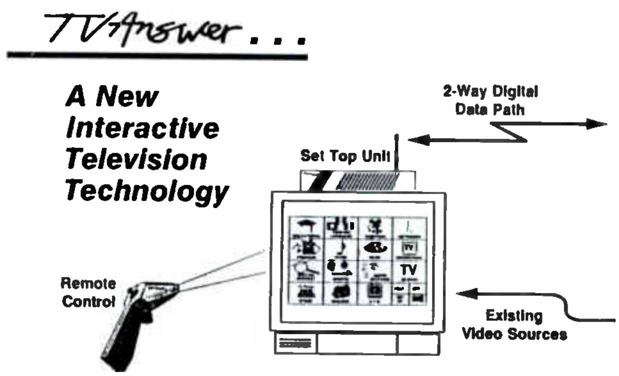
FCC GUIDELINES

Building on a base provided by long term investors, TV Answer expects to license its technology to local market operators selected by the FCC. A manufacturing partnership has been established to build the product and a satellite network is in place using Hughes Network Systems' equipment. Service provider (consumer applications) relationships are being formed. The entire system, end-to-end, works via a consumer set top unit, local cell site operators licensed by the FCC and the satellite network, including a Hub site located in Reston, Virginia.

THE PRODUCTS

The TVA Home Set

The set top unit has two-way capability utilizing the RF spectrum of 218-219 Mhz. The home unit includes a joystick for ease-of-use (if you can operate a TV remote control, you can operate this effortless joystick with cursor precision directed to a menu screen). The final memory capacity will be determined shortly.



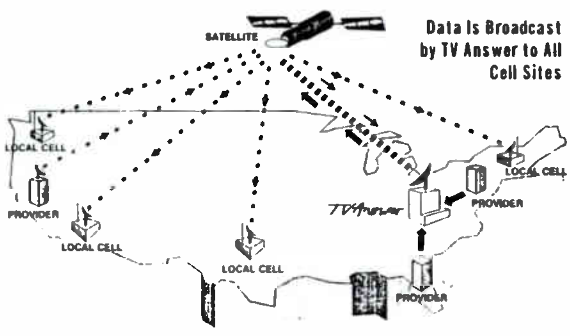
The Cell Site

The local cell site is a 1.8 meter send/receive dish which will be linked to the Hughes Network System and to the Reston Hub site. Each cell site will also have a six meter whip antenna to communicate to and from the consumer home unit.

The Hub Station

In Reston, a 6.1 meter vortex send/receive Hub station was installed as of January 1992. The Hub is 100% redundant through the IF stage and has 3:1 redundancy at the base band stage.

THE SATELLITE NETWORK



CONSUMER BENEFITS

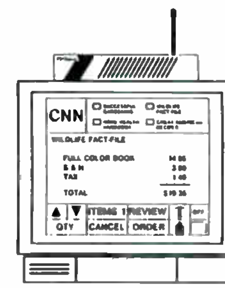
The final element in TV Answer's market-make-ready plan is provider services or consumer applications. How will the consumer benefit from this two-way TV technology? There will be advertising response, play-per-view, game show participation, news polling, and educational applications, as well as in-home data base interactivity.

THE ADVERTISER'S BENEFIT

TV Answer believes that any 800 number with the accompanying data, quantity and information content can be better captured on a screen and electronically communicated, rather than having interested consumers search for a pencil, paper, credit card, and then waiting for the next available telemarketer to take the order. TV Answer will revolutionize direct response advertising! In addition, image advertising will instantaneously provide qualified "leads." The image advertiser will be pre-screening prospects when using two-way TV commercials. The prospect reacts to the impulse for more information with a major benefit accruing to the advertiser: 1. a qualified lead, 2. demographic information, 3. measurement of expenditure versus results!

ADVERTISING RESPONSE

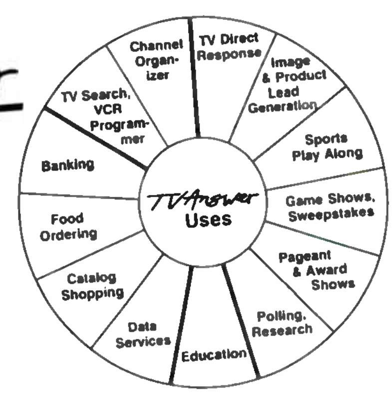
- ***TV Direct Response***
- **Image & Product Lead Generation**



CONSUMER'S PARTICIPATION

In-home player participation will give a whole new meaning to the phrase "armchair quarterback." Consumers who wish to participate in a sporting event or game show now have the ability to compete for prizes and sweepstakes with the advertiser being assured of viewership and ratings! Consumers can become quarterbacks, pro-golfers, contestants, and even beauty pageant judges - all from their own living rooms.

TV Answer USES



POLITICAL IMPLICATIONS

There is also immense political power inherent in two way TV. People can electronically attend town meetings and become citizens of the world. No longer will the apathetic refrain of "my vote won't count" be heard. The political process will be rejuvenated through real-time opinion polling. This single application will have tremendous import at the federal, state and local levels.

BROADCASTERS BENEFITS

What all this two-way TV means for the broadcasters is that your audience will be enlivened; home viewership will be enhanced; ratings will improve and for those broadcasters who seize this opportunity, revenue streams will be additive. More specifically local ad revenues will increase because national spot advertisers will go to the stations equipped to handle two-way TV. Your equipment cost could be a personal computer, a vertical blanking interval (VBI) decoder and a VSAT (very small aperture terminal) - approximately $20,000 in capital to capture national 800 screens and image advertisers.

EDUCATIONAL OPPORTUNTIES

To conclude the programming portion of this paper, let me address the educational opportunity. As the television has become the worldwide electronic hearth, it has also been labeled a vast wasteland. TV Answer will serve to irrigate this wasteland by providing real-time education. No longer will children passively learn, but rather they will re-act to on-screen queries. Sesame Street will become more active, high school and college accredited courses will be provided via the electronic hearth to the homebound, economically disadvantaged and to the thrifty. Adult fee education will become the electronic correspondence courses of the 90's! And businesses will be provided with the economic electronic training and re-training programs necessary for improving US competiveness.

THE ADVENT OF THE INTELLIGENT HOME

Finally, I want to move from programming applications to consumer applications. TV Answer is evaluating in-home electronic banking, stock information and meter reading. The advent of the intelligent home has arrived. Consumers can now order pizza, theater tickets, check travel schedules, and pay-per-view programs via the telephone. The TV Answer solution frees the consumer from the telephone and provides convenient, economic, electronic transfer of data. The benefits which accrue to the provider of these services through the TV Answer system are: 1. product differentiation, 2. immediate update on price changes, 3. elimination of human error. As TV Answer as a service matures, there will develop a natural cost/benefit, i.e, those households that value their time will be willing to pay for information and service. In return, these same house holds will achieve greater control over their daily lives through convenience.

TRANSACTION VOLUME POTENTIAL

- **800 Type (Free)**

Pepsi Superbowl	- 7 Mill Calls	
Nickelodeon	- 4 Mill Calls	
HSN	- 120,000 Calls/Day	

- **900 Type (Fee)**

Wheel of Fortune	- 4.7 Mill	@ $4 Ea.
Fcx Fantasy Park	- 600K	@ $2 Ea.
VJN	- 550K	@ $3 Ea.

JANUARY 16, 1992 - A RED LETTER DAY

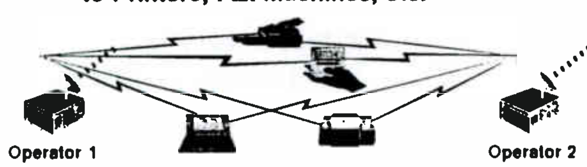
As all of you in the audience know, January 16, 1992 was a red letter day for TV Answer. Fernando Morales, his investors and his new company were among the beneficiaries when the FCC ruled on the spectrum availability for services such as TV Answer's. Service providers are beginning to sign up for this new technology, manufacturing is on track and we expect to have product in the market by the Christmas selling season of 1992! And as recently reported in Television Digest, January 20, 1992, I quote: "In a separate report to NAB board on IVDS, NAB executive vice president John Abel said it could be a great opportunity for TV broadcasters and 'momentum has been steadily gathering'. He said TV Answer had received more than 5,000 inquiries after offering to license its technology to other companies: 'Interactive video is still a wait-and-see phenomenon, but clearly ideas are being spawned, money is being spent and hardware and software are improving...NAB should stay on top of interactive technology and promote those technologies that offer broadcasting the potential for a significant additional revenue stream'."

TV ANSWER AND THE FUTURE

TV Answer stands ready to assist those among you who wish to share a continuing ten-year vision. The milestones of TV Answer are public record, we invite you to share our future milestones, our vision and our imagination.

THE LOCAL CELL SITES...

- **Provide Local Data Transmission to PC's, Laptops and Palmtop Computers**
- **Also Provide Local Data Transmission to Printers, Fax Machines, etc.**



Conclusion

Thank you for your time and attention and for allowing TV Answer to address the NAB convention.

PRODUCTION PROCESSES FOR INTERACTIVE TELEVISION

Thomas Kanady
Interactive Network, Inc.
Mountain View, California

Abstract- Interactive Network is making existing television programming interactive and continues to expand this use of its patented technology. The goal is to involve the viewer actively in the program they are watching without becoming intrusive. The focus of this discussion is the process by which our programming is created and how it might be adopted by television producers to add interactivity to their own programs. To expand the current focus on sports and game shows, several techniques shall be discussed including a sample interactive newscast.

INTRODUCTION

Interactive Network provides an interactive television medium which effectively allows viewers to play along competitively with sporting events, TV game shows and dramatic programs, express views on the topics of news, talk shows, and participate in special events like the Emmy Award ceremonies. Further applications include voting on political debates, etc.

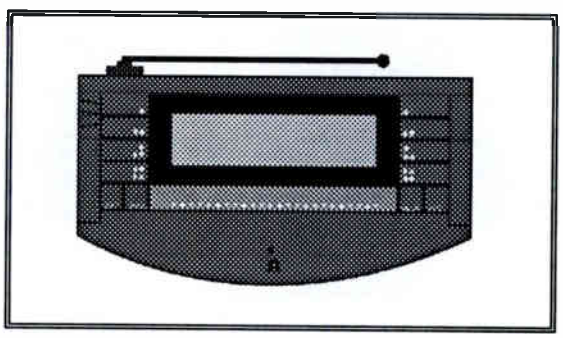
The production environment operates similar to a TV station in that our organization has programming, sports, TV related production, and traffic workgroups. The technology may be different from TV production but the concerns are very similar.

System Overview

To reach a mass market, a network of FM stations linked by digital telephone lines is employed to broadcast data to subscribers. The data is modulated on one of two possible SCA channels. This technique does not require that the return path be continuously connected. Interactivity is achieved by receiving game data that is sent synchronous with the TV programming. This information is used during game play to score the user and give feedback to the player throughout the event. This digital link is also used to download new game software and even a completely new operating system to the Control Unit.

Once the player has completed an interactive event, a short, local scoring call (under 10 seconds) is initiated using the on-board modem. The data is sent through a packet data network uploaded into the IN Central Computer System. Winners are determined in a few minutes and the distribution of scores is sent and displayed as a histogram with the player's ranking given as percentile. The high score is also displayed.

An alternate form of transmission also has been deployed to over 60% of U.S. homes. This involves inserting data on the Vertical Blanking Interval (VBI) of the PBS TV system and allows subscribers to receive service even though they may be outside the FM service area. To receive the VBI signal, an additional device is used. The "Booster" is a receiver/retransmitter that retrieves the data from a line of video from the TV station and modulates it on an infra-red signal. The control unit is equipped to demodulate data from this IR link.



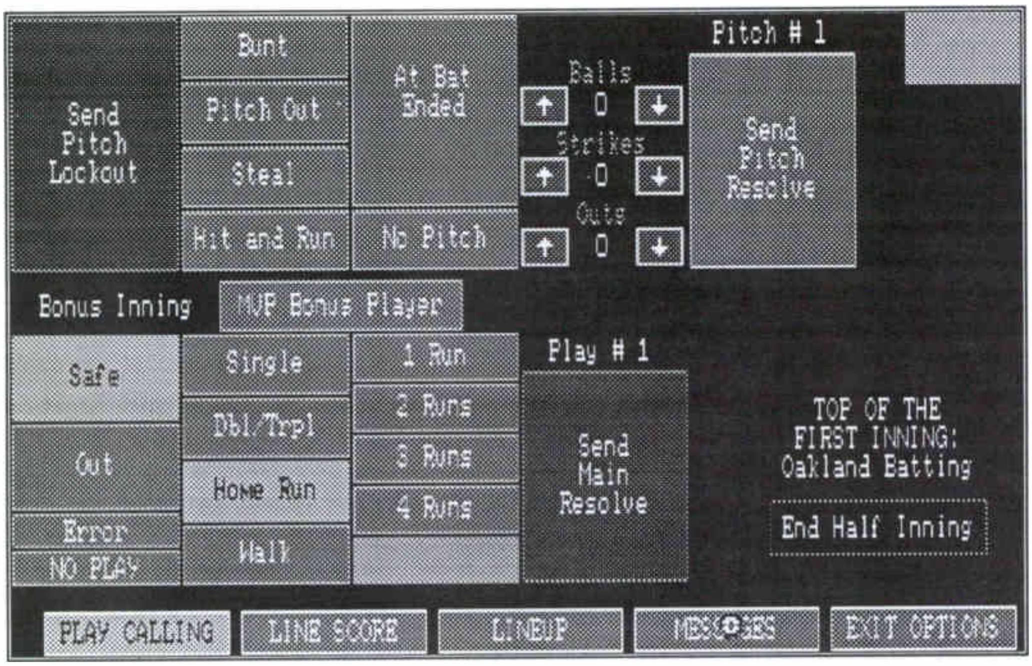
The Control Unit. Part laptop computer and part mobile data receiver, it also has a keyboard hidden under a pullout cover.

The Control Unit has many of the elements of a lap top computer, such as the CPU, memory, keyboard, display and modem. Added to this is a digital FM subcarrier data receiver, an infra-red receiver and secure microcontroller.

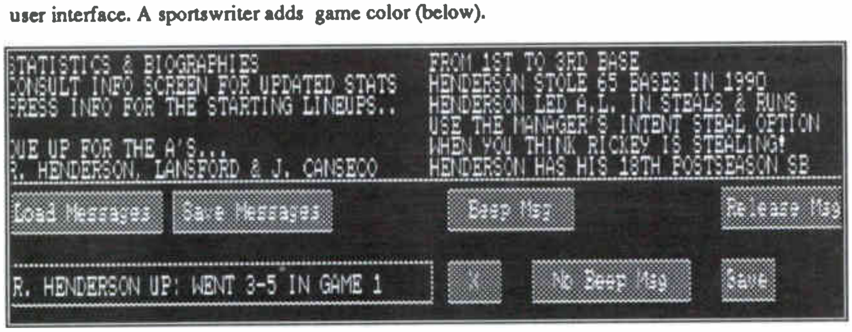
The Central Computer is a fault tolerant, real time, distributed system. All production activities center around this system by accepting input from live consoles and stored data files, providing supervisory and control functions, and prioritizating and packetizing data for transmission. In addition, a database relates scheduled programming and retrieved score/survey results to subscriber accounts.

Traffic and Scheduling

The traffic department schedules all programming on a central database a minimum of three days in advance of air time. Sybase$_{TM}$ operating on a Sun Workstation$_{TM}$ platform provides the relational database and data entry environment. The customer accounts are related to the scheduled events when a scoring call is received and are updated immediately following the scoring period. Each event has a database record with at least 20 fields of data associated with it to accomodate the versatility of programming choices available. In addition to maintaining and monitoring event traffic, all other resources including staff and equipment needs are coordinated through this group.



In The Dugout play calls are entered (above) via mouse and cursor positioning on this graphical user interface. A sportswriter adds game color (below).



The General Production Process

The production environment at Interactive Network is, to some extent, analogous to a that of a television station. Data files are cued up to air much like videotapes and live programming is carried out with much the same intensity as an on-air control room at news time or as a production van at a sporting event.

Sports Production

Live sports production is accomplished by monitoring the game and electronically "documenting" what is happening. For example, in a baseball game when the pitcher throws the pitch a time stamped message is logged and sent to the interactive player. This "locks in" the player's prediction, then if the batter hits the ball and flies out this is logged as well and the player at home is scored on the accuracy of their prediction. In a way, we are building a sentence that says "The pitcher **threw** the ball, it was **hit** and **caught** as a **fly out**." In addition to the actual game play aspects, current statistics on the team members are broadcast and available during the game for reference by the interactive player.

TV Game Shows

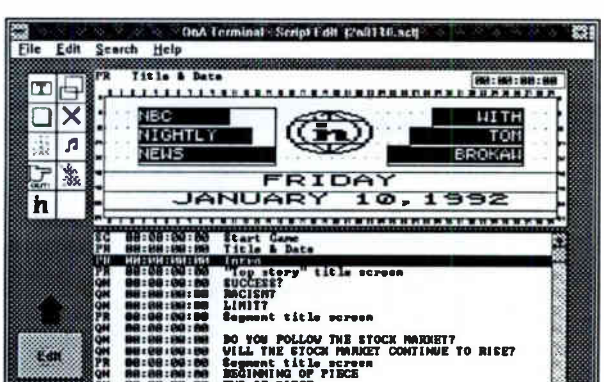
The games are produced from the timing of the show itself with some similarities to sports in that predictions are "locked-in" before the answers are revealed. Every game show is treated differently with respect to game play but similarities do exist. The data files created to simulcast with the show are visually or audibly synchronized with the actual broadcast from the TV station.

Other TV Related Productions

With its broad applications , the Question and Answer (Q&A) Shell, was developed for the production of a wide variety of programming. It was created when we recognized that a great many programs can be made interactive through a mix of information screens, multiple choice question and answer screens, and survey question screens. This has allowed for flexibility in production and can be used to create a data file to synchronize with a broadcast tape or to produce a live event such as an awards program, a political debate or a newscast.

Production of an Interactive News Program

Consider the effort that goes into a daily newscast. A great deal of research takes place to support the stories that air nightly. This kind of informational exchange currently takes place in one direction only, to the viewer. The material, however, is naturally interactive. People have widely varying levels of knowledge on the subjects, want to know more on a topic of interest to them personally than a station may have air time to support, and have opinions to express. This forces the station to set a baseline level of information about any topic and limit the detail



The Q&A edit tool allows quick reference to screens via an outline format.

into which they delve. Additionally they don't get the opinions from their audience (though some stations are experimenting with dial-in opinon polls). For example, the NBC Nightly News is currently being made interactive through a process of providing additional information on the topics, asking survey questions, and quizzing the audience on their knowledge of current events.

Q&A Producer Station

The scripting process was initially done on a text editor with only spartan features. This required a steep learning curve and gave no feedback on what the final screen format would be like. To ease this cumbersome technique a "What You See Is What You Get" graphical user interface (GUI) was developed. this allowed non-technical staff writers to quickly create and format scripts. In addition to easily manipulating screen elements and time code the Q&A Producer Station functions as a real time interface for live productions. This approach allows features to be accessed via pull-down menus with point and click ease of execution.

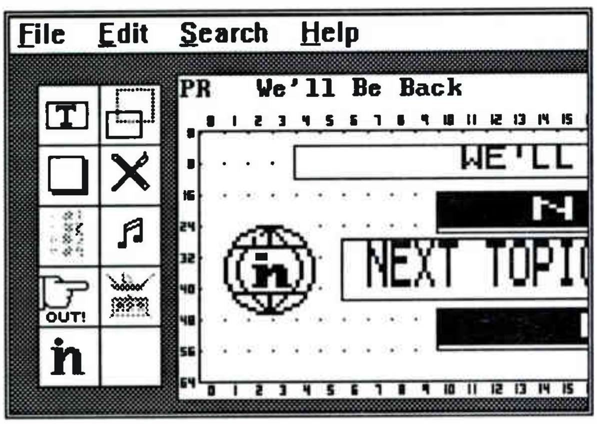
As a platform for building such an interface, Microsoft Windows$_{TM}$ Ver. 3.0 running on an Intel 80386 based computer was chosen. This commonly available multitasking GUI is expected to be upwardly compatible with high end workstations in the near future.

Q&A Shell is an application, like our other game programs, that is downloaded to the Control Unit prior to start time for any given event that is scheduled to use it. The purpose of the producer station is to allow access to the features of the application. Features will be added to the station as they become available in the down-loadable application. For example, one pending feature is **Interactive Advertising**, whereby a particular commercial can be made interactive through a context switch in the Control Unit software, allowing Q&A to take over a baseball game during paid advertising time and then relenquished when the commercial ends.

The basic features include a script tool with each production element Q&A supports, a time code manupulation tool, cut and paste editing, file merge (so multiple writers can create scripts) and a live production interface.

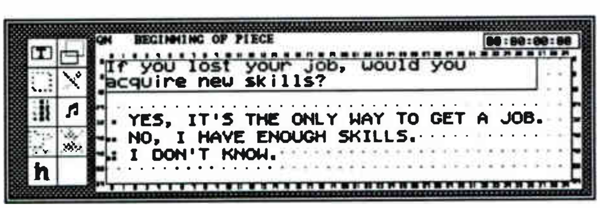
Script Interface

For editing purposes and ease of access an outline of the script with associated time code is provided on the EDIT screen. By scrolling to the appropriate item and clicking on EDIT a full screen image of the message with alignment grids and layout boxes is revealed. Text can be positioned on the grid. The shape of the box can be changed and the text will automatically word-wrap within the new shape. Graphics can also be selected and placed on the grid. Attributes such as double high , double wide , and inverse "video" are assigned through a pop-up menu.



The Features of Q&A are accessed through the symbols to the left of the screen and through pulldown menus above.

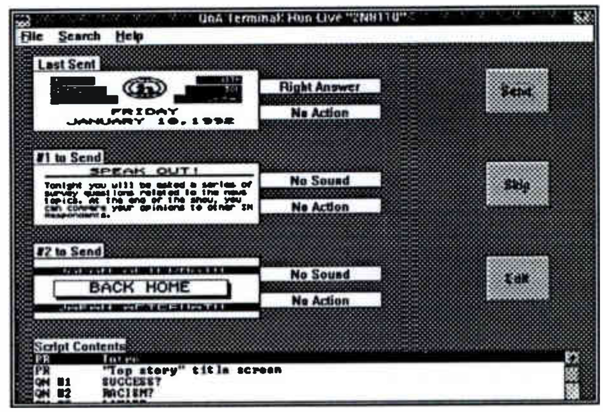
Various message types exist such as questions and multiple choice answers with point values assigned in the case of game production, survey-only questions in which more than one answer can be given, information screens with graphics, and a selection of sounds which can be added to any screen or sent without a screen. An asssortment of other message types for functional purposes, i.e. begin game, start survey call, start scoring call,etc. are also used.



Well, would you? Questions like these are the basis of interactive news and talk shows.

Live Production

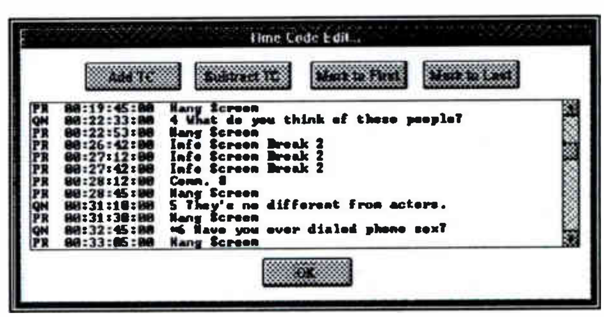
To accomodate the needs of live production the same outline format is used to access messages to be sent. The edit screen gives way to three screens indicating the last message and the next two to go. New messages and editing of existing ones is still accessible from this screen. Delays in the transmission are minimal since the performance was optimized for Local Area Network usage by integrating TCP/IP in the architecture.



The live interface lets the producer cue screens to send while viewing the last screen sent.

Time Coded Applications

While sports, game shows and other programs are produced live, often with the cooperation of the producer a production tape can be available before air time. This enables all script timing to be accurately made through time code. An on-board time code reader/generator card provides support for LTC or VITC formats. The internal PC clock can also be used to generate relative time code for the script.



Time code can be modified to fit edit changes and then tested by playing back synchronous with video via a time code reader card.

The cut and paste features allow a block of time coded material to be moved as a sequence to accomodate edit changes in the final production videotape. When the sequences are established the timing can be adjusted through a dialog box.

Putting It On The Air

To complete the process the interactive event must air in sync with the TV program. In the case of a live program, the producer station is connected through TCP/IP to the central computer system and attached to the actual event via the event ID number which is specified in scheduling. From that point on the producer can send messages to the Control Units that are tuned in for the broadcast.

To air a time coded event, a binary image of the script is created and loaded onto the central computer system with an input filename matching that which is determined in scheduling. once the program begins, synchronization is monitored with off air signals of both the data and the program video and audio. Timing adjustments are made via an interface to the supervisory and control functions of the central computer system that allows time adjustments in 2/10 second increments.

Conclusions

The production of interactive programming at Interactive Network utilizes a broad mix of technologies and skills in computers and telecommunications. The goal of these, however, is to simplify the process for those who write and produce the programming, since they are generally creative rather than technical staff members.
As new areas are explored in interactive television and new features added, the same basic concepts will be applied. The composite results will continue to be enriched by these building blocks as they are applied with fresh approaches to an ever-widening range of interactive television programming.

Acknowledgements

Many thanks to those at Interactive Network who helped make this paper possible. They are: Dr. Robert Brown, V.P. Engineering; Ed Hodapp and Robert Friele, Designers of the production interfaces; Dave DeAndreis, Production Supervisor; Mark McKeown, Senior Sports Producer; Fred Atton, Network Traffic Coordinator; Annabelle Hester, Staff Writer; and Malcom Klein, Director of Programming, for his editorial comments. Also, thank you to Gordon Currie at Microsoft for helping me print.

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INTERNATIONAL BROADCASTING

Sunday, April 15, 1992

Moderator:
Louis Libin, NBC, New York, New York

***WARC REPORT**
Ben Fisher
Fisher, Wayland, Cooper & Leader
Washington, District of Columbia

***THE EUROPEAN BROADCASTING UNION: STUDIES OF ADVANCED SYSTEMS**
George Waters
Technical Center
European Broadcasting Union
Geneva, Switzerland

***THE ASIAN-PACIFIC BROADCAST UNION STATUS REPORT**
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EUROCRYPT, A SUCCESSFUL CONDITIONAL ACCESS SYSTEM
Vincent Lenoir and Vincent Michon
CCETT
Cesson-Sévigné, France

DIGITAL TELEVISION BROADCASTING DEVELOPMENTS IN EUROPE
Terry Long
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Winchester, Hampshire, United Kingdom

*Paper not available at the time of publication.

EUROCRYPT, A SUCCESSFUL CONDITIONAL ACCESS SYSTEM

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Cesson-Sévigné, France

Abstract

EUROCRYPT is a conditional access system for broadband networks. It is used in Europe on the cable and satellite networks in association with the MAC broadcasting standards. It offers all the commercial modes that have been identified for Pay-TV. Additional functions are included. It is a simple, safe, evolutive and user-friendly system, entitlements are transmitted in the signal with a large capacity. Its implementation uses the PC2 smart card. It can be used with other broadcasting standards. Since it has been specified, in March 1989, its success increased, and we hope that it will become the reference system for enhanced, high-definition and digital television systems, and other new broadcasting standards.

INTRODUCTION

The development of new broadcasting media in Europe through the launch of Direct Broadcasting Satellites and the constant progression of cable television (6 million households connected, and 750000 subscribers in France by the end of 1991) is leading increasing numbers of operators to offer new television programs. This extension of services requires to offset production and broadcasting costs. The traditional modes of financing (advertising and licence fees) are no more sufficient and must be completed by direct financing of the users by means of subscription and pay-per-view. This commercial approach is coherent with the evolution of the behaviour of the user to pay-per-event purchase.

FRANCE TELECOM desire to improve the viability and the development of these services, and to reduce the heavy investments that were necessary led it to propose an open conditional access system for all broadband networks. CCETT was mandated to define a system that was independant of the type of media used. This system is known as EUROCRYPT. Three years ago, it had just been specified. Now, we can say that it's a success.

THE ARCHITECTURE OF A CONDITIONAL ACCESS SYSTEM

A conditional access television system contains three distinct functions :

Scrambling and descrambling of the programme components

This function depends on the signal type and coding. However, all techniques require the synchronised use at the transmition and reception sides of a **control word**, which initialize the scrambling/descrambling process.

System operation

The aim of this function is to transmit to the receiver the data it requires to descramble the program. Real-time operating messages are broadcast, including the cryptogram of the control word, and where appropiate, data inherent to the program. The control words are decyphered by the receiver **security device**, which contains the service operating key.

There are two possible configurations :

- the operating message contains only a cryptogram of the control word : in this case, only the service operating key is required in order to decypher the control word, and management of the conditional access system needs to organize the distribution of the service operating key ;

- the operating message contains the cryptogram of the control word and data inherent to the programme (cost, duration, identification). In order to decypher the control word, the security device will include the operation key and entitlements that must be compatible with the program parameters. Management of the conditional access system relies on the distribution of the service operating key and entitlements. This provides the system with greater flexibility and has been choosen for Eurocrypt.

Security is ensured by the device and the frequent replacement of the control word, for which the transmission has to be synchronised with the program. The security device may be buried, i.e. integrated in the descrambling receiver, or removable like the smart card favored by the CCETT for reasons of simplicity, cost, and difficulty of piracy.

Entitlement management

It consists of renewing entitlements in the customer's security device. Customers contact a management centre, which sends entitlement management messages containing the new rights and confidentiality if required. Management messages are protected by the management key.

The transmission of entitlement management messages doesn't need to be synchronized with the program. The messages can even use a different network from the one used for broadcasting purposes.

THE EUROCRYPT SYSTEM

The Eurocrypt specifications have been established in cooperation with industrial partners and program providers. The following commercial modes are offered :

Subscription : customers purchase a program for a given period, the program provider can diversify his programming by offering a range of themes (sport, news, etc.) or various programming levels.

Pay-per-view : customers purchase the program when viewed. It can be :

- **advanced purchase** : customers buy their program in advance from the management centre,

- **impulse purchase** : customers decide to buy the programme at the last minute. This is possible only when the transaction is local, the customers have a credit balance authorised by the management centre, and the user's purchases are checked at a later time to ensure payment of those furnishing the services.

Pay-per-time, the fee is based on the length of the session.

Additional functions are included, such as :

- use of a **maturity rating** for the program

- **blackout** of geographical areas, and **replacement** of the service for the blacked-out receivers

- transmission of **personalised messages**,

- **local viewer control** of the security device, by means of a "parental code" (PIN code)

THE SYSTEM CHARACTERISTICS

The specific characteristics of the EUROCRYPT system are :

A single and general purpose conditional access system.

The Eurocrypt system is independent of networks and broabcastings standards. This has been made possible by the system functional layer structure which clearly differentiates between the scrambling, operating and management functions.

The system organisation and responsability hierarchy.

Eurocrypt has been defined to allow, with minimal constraints for the program providers, the sharing of the system and of the smart card. This is possible by the hierarchy of the management functions in the smart card. Three levels of decision are distinguished : the issuing authority, the program provider, and the user.

The **issuing authority** is responsible for the security of the system. It opens and closes the zones that are allowed in the smart card to each program provider.

The **program provider** manages its own resource and its own customers with several marketing methods. Several commercialy-independant program providers may coexist in the same security device, each one working indepently from the others.

At the last step, the **user** has a local control on his security device and a great flexibility in purchasing methods.

The broadcasting of entitlement management messages

The Eurocrypt system uses over-air addressing methods based on individual, shared (groups of users), or entire audience messages. A efficient grouping method is used, which allows the addressing of millions of customers in a very short delay.

With a data-capacity of 15 kbits/s for over-addressing, the cycling time for addressing one million customers is reduced from 10 hours (with individual messages) to 1 mn 40 s (with shared messages)

A very high degree of security :

A high performance security device is used : the PC2 smart card. It contains the confidential keys (service operating key and management keys) as well as the access entitlements acquired by the customer. In addition to the purely security aspects, it makes it possible to standardize the descramblers, which are the most expensive part of the system

A double protection of the control words is made. Customers must have both the service operating key and an entitlement which is compatible with the programme parameters in order to be able to return the control word. The security is increased by the frequent change of the control word (every 10 seconds).

A user-friendly and ergonomic system

Most parameters are broadcast and stored in clear (e.g. subscription periods, program provider identification, etc...). The use of an easy man/machine dialogue is possible for fonctions such as reading of the stored entitlements, prevalidation of impulse pay-per-view programmes, maturity rating choice, PIN code modification, etc...

The system evolution

A flexible coding method has been choosen in order to introduce new functionalities, and new security mechanisms if necessary.

STATUS OF DEVELOPMENT

Since 1989, Eurocrypt with D2-MAC/Packet broadcasting standard has been selected by many cable and satellite operators all over Europe for public or professionnal services. In France, it is used by France Telecom for its cable network. 13 cable networks are now open. Several others will be open before the end of this year. For satellite networks, it has been choosen by three of the major european pay-TV operators : in France Canal+ on the TDF1/TDF2 DBS satellites, and in Scandinavia Kinnevik and Fimnet

A pay-per-view experiment is made on the Berlin cable network. It is managed by the European RACE research project. Other experiments are planned by France Telecom on its cable networks.

The equipment developments so far include :

- Mass product decoders : Visiopass (France Telecom), Decsat (Canal+), Nokia (Kinnevik), Uniden, Amstrad, Pace, etc.... More than 1.5 million decoders have been ordered to the major consumer electronics manufacters
- Professional encoders and decoders : Matra, Tandberg, and Thomson
- The PC2 smart card, either in EPROM or EEPROM technology
- Several management centres for France Telecom, CANAL+ and Kinnevik
- PC2 card issuing machines.

Two years ago, France Telecom presented the Eurocrypt chain in an exhibition in La Defense. Now it offers Eurocrypt via the VISIOPASS service.

A complete test chain has been developped in the CCETT. The CCETT is responsible for the validation of the Eurocrypt equipments.

TV Eurocrypt, a British company will be responsible for the promotion and labelling of the Eurocrypt equipments.

The evolution and the standardization of the specification is now undertaken in French and international organisations.

The system compatibility with other networks was foreseen since the start, enabling re-use of a maximum numbers of equipment developped. Studies are currently underway for a large number of networks, including HD-MAC, full channel digital mode, contribution networks, digital TV, DAB, RDS, ...

As a conclusion we can now hope that it will become the reference system for new broadcasting standards, including enhanced and high-definition television systems, on cable and other networks.

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DIGITAL TELEVISION BROADCASTING DEVELOPMENTS IN EUROPE

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1. INTRODUCTION

European observers have been following with great interest the progress being made in the United States aimed at the introduction of advanced television technology and, in particular, proposals for digital terrestrial HDTV broadcasting.

Within Europe, considerable experience has already been gained with digital bit-rate reduction and modulation techniques, both for video and audio applications. European standards have been developed for the digital exchange of component television programmes at 34-45 Mbit/s and at 140 Mbit/s, as a result of work in several European collaborative projects and the CMTT. A separate project, EUREKA VADIS, aims to develop the technology for the distribution of digital video systems at up to 10 Mbit/s, with a view to standardisation within ISO. In addition, much work has already taken place on systems for digital audio broadcasting (DAB), including some very successful demonstrations of the feasibility and ruggedness of the system.

Following the success of DAB, and aware of the recent flurry of activity on digital terrestrial HDTV in the USA, studies in several European laboratories (see ref. 1, for example) have more recently turned towards the development of a system of digital television for terrestrial broadcasting for Europe.

Resulting from these studies, Europe now has plans for a major four-year collaborative R & D project for the development of the emission and networking aspects of digital terrestrial television involving 18 partners from industry, broadcasting, and research institutes in seven European countries.

This project (called dTTb) will embrace a range of possible applications in the UHF/VHF terrestrial broadcasting bands, including services for mobile and portable reception of 16:9 pictures, as well as extended definition reception on fixed antennas, and extensions to provide HDTV. Associated with these developments will be the establishment of a broadcasters-manufacturers high level Strategy Group, to provide overall co-ordination of European activities relevant to digital terrestrial television, and to devise appropriate service introduction strategies and time-scales.

The general purpose of this paper is to provide an overview of digital television broadcasting developments in Europe. Following a brief review of the terrestrial broadcast scene in Europe, and a summary of other existing relevant digital projects in Europe, the paper concentrates on specific digital television issues and gives a description of the European dTTb project.

2. EUROPEAN PERSPECTIVES ON DTTB

In considering a digital terrestrial television system for Europe it should be borne in mind that the broadcast situation in Europe is radically different in some respects to that prevailing in the USA, and that this can lead to substantially different systems requirements and implementation timescales. For example:-

(i) In general in Europe there is a much greater emphasis on achieving "National" coverage than in the USA. This implies a much higher density of transmitters in the European national networks than is common in the USA. The exact situation varies from country to country as indicated in Fig.1.

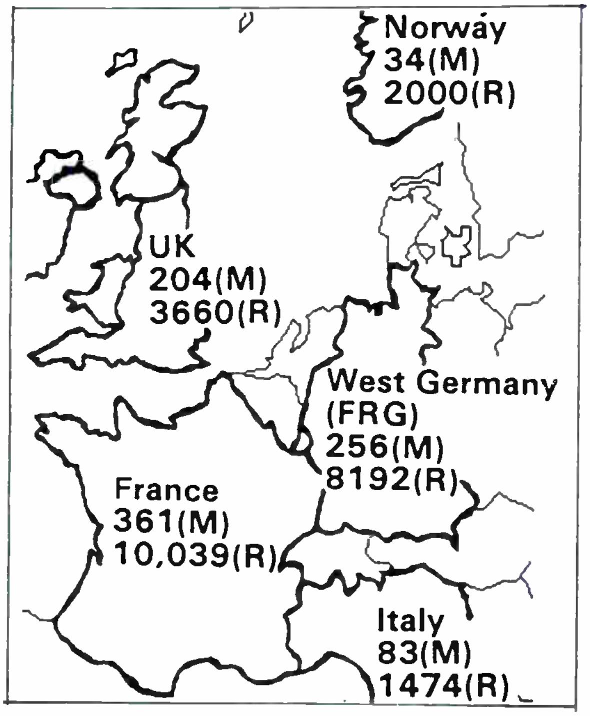

Fig.1 Terrestrial TV: Main Stations (M), and Relays(R) in five European countries

In the UK for example, virtually the whole population (around 99.4%) can currently receive four programme services with a single UHF aerial, and great emphasis has been placed on maximising this coverage. A fifth UHF programme channel is shortly to be made available, but although this too will claim national coverage, it will in fact serve only about three quarters of the population, since, using conventional analogue PAL, there is not enough space in the well-used UHF spectrum to allow for large numbers of new transmitters.

In **France**, which has a similar number of TV households to the UK, six national television channels are provided via terrestrial transmitters, but the coverage of these is by no means uniform, and for some of the channels is supplemented by cable and satellites. Three services, Antenne 2, TF1, and France 3 cover 99.9% of the population, whereas Canal Plus, which is available only to subscribers, has 87% coverage, and La Cinq and M6 both cover less than 70% of the population. Additionally, a few local commercial TV stations also exist, notably in Paris, Lyons, Toulouse and Annecy.

West Germany provides two national programme services (ARD-1, ZDF) via terrestrial transmitters, and various regional programme services, each providing over 99% population coverage. A significant number of viewers can also receive the RTL Plus and SAT-1 programmes off-air. In addition to the state-run networks, a number of private networks and local stations make the German television landscape very complex.

Viewers in **Italy** have a wide choice of programme services, with two of the three RAI stations providing over 99% coverage, and RAI-3 almost 90%. Three private stations also provide coverage of around 93% of the population, whilst six other programme channels provide substantial coverage in many of the larger towns.

In **Norway** the story is very different, with only one terrestrial channel available over air, from the national broadcaster NRK, which achieves over 99.6% coverage. The situation is also very different in the **Benelux** countries, where cable television networks are extensively employed.

Thus the frequency planning constraints can be quite different within each country, and from country to country, to create a situation that is, in general, more complex and difficult from a digital television point of view than in the USA.

(ii) It is also true, that with the exception of the Benelux countries, cable television is very much more developed in the USA than in Europe. It is not at present very clear to what extent this has bearing on the question of digital terrestrial television, other than the fact that today few Europeans have access to a large number of programme channels, thus providing an opportunity for the launch of new and alternative services such as those from DTH broadcasting satellites.

More importantly, perhaps, the limited number of terrestrial programme channels available in most European countries makes the idea of a future switch-over to an all-digital mode of network operation an extremely attractive idea.

This is because the use of digital transmission based upon a multiple carrier technique known as 'Orthogonal Frequency Division Modulation' (OFDM)[2] could enable each of the allocated terrestrial broadcast channels to support, individually, a nationwide broadcast network.[2]

A network based upon the re-use of a single transmission frequency, to obtain area or national coverage, is known as a 'single frequency network'. In principle, the use of such techniques could increase the number of terrestrial television programme channels transmittable in most European countries by a factor of about ten. In practice, it is probable that some compromises will be required, but even in the most pessimistic scenarios there could be an expansion of three to four times in the number of programme channels available from the allocated terrestrial spectrum - channels that can provide 16:9 pictures of high quality and support rugged reception on portable receivers.

While an increase in the number of channels might be obtained by other means, the attractive qualities of rugged portable reception (for example, to allow viewing on the patio or in any area of the house) should not be underplayed.

(iii) In contrast to the situation in the USA, the provision of 16:9 wide-screen pictures to the home by non-digital means is at an advanced stage of development in Europe. Satellite services in Wide-MAC are commencing today, for example, and it is envisaged that services on the HD-MAC standard will begin to be introduced by 1994. Additionally, the development of a 16:9 aspect ratio Enhanced PAL system for compatible terrestrial transmission by a consortium of broadcasters and industry (known as the PALplus group) is sufficiently advanced today to forecast service introduction dates in the mid 1990s.

In one sense, these developments could be seen as directly competitive with digital terrestrial television; in another, they may be seen as preparing the ground for a wide-screen market to be served by digital television. Either way, they represent important factors that must be taken into account in considering the requirements and timescales of digital terrestrial television.

(iv) On the question of time scales for digital television, the view of most experts in Europe is that it is very unlikely that digital HDTV terrestrial broadcasting systems can be developed to the point of service introduction in less than 8-10 years. Experience with MAC systems (the development of which began in 1981) supports this view, as does the forecast of some experts that the silicon area requirements for a digital receiver of true HDTV quality* might be four or five times that of HD-MAC in the same IC technology.

Given this perception, the emphasis in Europe is to create a framework for the development of 16:9 aspect ratio digital television. This is to embrace, in the medium term, EDTV and conventional TV quality for fixed position and portable reception respectively, within a family of standards which is extensible to true HDTV quality services in the longer term.

3. RELEVANT EUROPEAN COLLABORATIVE DIGITAL TELEVISION PROJECTS

Much research work into different aspects of digital communications technology has already been carried out in Europe, and the results from several European collaborative research projects are likely to be relevant to future digital terrestrial television systems.

It is eight years since the first European collaborative research project to consider the digital coding of HDTV picture signals for transmission purposes began. The COST 206 project studied problems related to the subsampling and interpolation of digital picture signals, and examined the practical difficulties of implementing the circuitry required. Much was learned about the high-speed digital architectures needed for such signal processing, and this led to the successful development, in LSI, of an HDTV codec to provide contribution-quality signals for transmission at a rate of 565Mbit/s.[4]

*True HDTV quality is taken to mean a quality approaching the interim HDTV studio standards as defined in the CCIR[3].

Although such high data-rates may be available for inter-studio traffic, there will also be the need for lower bit-rate local digital delivery systems. Part of the work involved in the Eureka-95 project was to develop a system for digitally coding and distributing various types of MAC signals at bit-rates of about 140 Mbit/s. The coding method used a variation of standard predictive coding techniques to implement a system which can be used with D-MAC and D2-MAC as well as HD-MAC, and which can cope with scrambled or clear signals.[5]

Arising from work in an Experts' group of CMTT, was a method for reducing the bit-rate of Rec 601 component television from 216 Mbit/s to the range 34-45 Mbit/s. This activity was heavily supported by European industry and broadcasters and led to the development of a full European standard from ETSI (European Telecommunications Standards Institute). Codecs conforming to this standard are currently being manufactured for contribution networks all over Europe.

As HDTV develops in Europe, there will be similar needs to distribute the signals at far more modest bit-rates than the 1Gbit/s of the source pictures. The European project HIVITS (HIgh quality VIdeophone and high definition Television Systems) is involved in developing hardware to apply similar coding techniques to HDTV, and also in investigating coding formats suitable for delivering these signals to the home. HIVITS forms part of the RACE programme (Research & development in Advanced Communications technologies in Europe), whose objective is to introduce a broadband communications network throughout Europe.

Another European project, Eureka-256[6], has developed a digital coding technique for the transmission of HDTV signals via satellite, using a modular DCT-based system similar to that of the ETSI standard. Experiments suggest that at least 45Mbit/s is required to maintain true HDTV quality. The project is continuing with the aim of developing suitable digital coding for a proposed digital HDTV satellite broadcasting service at 22GHz.[7]

Again planning to use digits for television, but for conventional 625 line TV rather than HDTV, is VADIS (Video Audio Digital Interactive System), a European project which is aimed at developing the technology required to allow full 625 line resolution pictures to be carried at around bit rates of 5-10 Mbit/s. If such compression can be achieved reasonably economically, then near-studio quality digital pictures could be carried over telecoms networks and on terrestrial UHF channels and satellites.

Using new techniques for digital radio transmission which have been developed over the last few years by a combination of European broadcasters and PTTs, a system with the name of Digital Audio Broadcasting, DAB, has recently been demonstrated. This has the major advantage that it could be used equally well from satellites and from terrestrial transmitters.

DAB uses a combination of different techniques to provide its rugged signals, utilising the latest ideas in digital signal processing, bit rate reduction, and digital modulation.[3]

The first step in constructing a DAB signal takes place prior to modulation; a digital audio coding technique called MUSICAM is used to reduce the data rate needed to carry the digital audio signal. These reduced data-rate signals are then transmitted using coded OFDM.

The experience gained in all aspects of digital broadcasting from these many different European projects will act as a strong foundation for the proposed digital terrestrial television collaboration.

4. THE dTTb PROJECT

4.1 The Project Consortium

In recognition of the growing interest in and importance of digital terrestrial television, a series of discussions was held between Broadcasters and the Consumer Electronics Industry during the summer of 1991 to review the situation and establish a plan of action. As a result of these discussions, a major European collaborative project was defined to provide "the required technologies for the broadcasting of digital signals at a bit-rate which allows for the broadcast distribution of digitised and compressed television signals being developed in other (European) projects." [See section 3 above] The consortium behind this major four year project involves the following partners:-

Centre Commun d'Études de Télédiffussion et Télécommunications (CCETT)
Télédifusion de France (TDF)
Independent Television Commission (ITC)
British Broadcasting Corporation (BBC)
Radiotelevisione Italiana (RAI)
Ente Publico Retevision (Retevision)
Institut für Rundfunktechnik GMBH (IRT)
EBU Technical Centre (EBU TC)
FI der Deutschen Bundespost Telekom (FI/DBPT)
Centre Norbert Segard (CNS)
Nederlanse Philips Bedrijven BV (PRL) (Philips)
Laboratoire Électronique Philips (LEP)
Thomson Consumer Electronics AS (TCE SA)
Deutsche Thomson Brandt (DTB)
Thomson-CSF/LER (TCSF LER)
SGS Thomson (STF)
SGS Thomson (Italie) (STI)
Thomson Consumer Electronics R&D France (TCE RDF)

4.2 Project Objectives

The broad objective of the project is to establish the technical basis and standards for a range of digital broadcasting services in Europe.

Included within these considerations is the possibility of creating the basis for the evolution of the European terrestrial broadcasting networks towards an all-digital mode of operation in the longer term. Of importance to this concept is the idea of 'simulcasting' a 16.9 aspect ratio digital service of 'matching' coverage but higher quality than that currently obtainable with the existing services. Also of importance is that the system should be extendible to provide services to true HDTV quality when this becomes economically feasible.

Other considerations include:-

(i) the need for good quality (if lower resolution) reception on low-cost portable and mobile receivers.

(ii) the need for a graceful quality degradation to extend coverage and avoid 'drop-outs' at the edge of the service area in unfavourable propagation conditions.

(iii) the desirability that the transmitted signal be recorded on domestic digital video cassette players using a simple transcoding process (to avoid quality loss of tandem de-coding and re-coding processes).

While the above considerations are integral to the project, it is recognised that there will be marketing and strategic issues that will need to be resolved in the high level European Strategy Group that is being established to provide overall co-ordination of European studies, including particularly co-ordination between the dTTb project and those European projects dealing with source coding.

The dTTb project therefore will draw upon the work of the source coding projects, and concentrate on modulation, channel coding and the broadcast systems. The project is divided into in two main phases:

Phase 1: A broad study which will investigate all possible data rates within existing terrestrial VHF and UHF channels, in order to determine which digital video broadcasting services and products are feasible. This study is to ensure that account is taken of possible future extensions to provide more TV channels and/or higher picture quality.

Phase 2: As a first step towards defining a service/product, demonstrators will be constructed for broadcast TV programmes and will be aimed towards portable, ie plug-free, or mobile receivers. These demonstrators will provide for investigation of network configuration aspects, extendibility towards future services, frequency allocations aspects, chip area analysis, and aim at a service with a quality above that of current PAL/SECAM systems and will include a 16:9 capability.

From these studies and developments, it is the objective to create a framework of hierarchial system standards which will allow a range of services to evolve, as the market demands, within a single European standard.

It is envisaged that the first phase of the project will form part of the European RACE programme under Task 813 covering efficient digital communication over bandwidth-limited media.

4.3 Key Technical Issues

The project objectives outlined above present a formidable set of development tasks, given that in many key areas (especially that of realistic over-air trials) the fundamental proving work has yet to be done. As already described, however, experience with the European DAB project (Eureka 147) has shown that rugged terrestrial broadcasting using relatively high bit-rates and heavy error-protection with Orthogonal Frequency Division Multiplexing (OFDM) is feasible.

The bit-rates required for EDTV and HDTV are nevertheless considerably higher than those required for DAB, and the scope for introducing a heavy degree of error protection in the narrow-band channels is limited. The question therefore remains as to what bit-rate can be supported in the terrestrial channels by the most appropriate modulation and channel coding scheme(s) over the service area of interest, taking into account interference from the existing service transmissions.

Initial results from the SPECTRE project being conducted by the Independent Television Commission (ITC) in the UK*[1] suggest that, for most regions of the UK, it would be possible to duplicate the coverage of the four 'national' networks operating in PAL using low power digital transmissions in the 'taboo' channels. Furthermore, it is suggested that these services should be of considerably higher quality than PAL - probably of a quality approximating to that of 16:9 aspect ratio enhanced-MAC. These results are based upon laboratory tests of a system utilising QPSK and OFDM with 'spectral holes' at the interfering PAL vision and sound carrier frequencies[1,8] as shown diagrammatically in Fig.2.

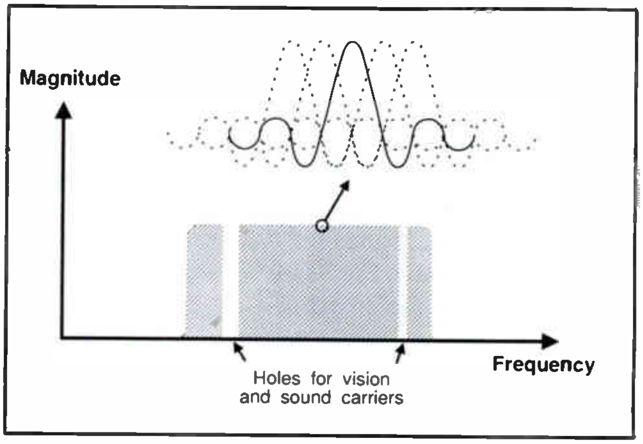

Fig.2. The OFDM spectrum

Field trials of the system, in the presence of the normal high power PAL transmissions, using the main transmitting masts in the south-west region of the UK, are planned for later this year. Taking these results as a basis, the key questions for further study will be:

(i) Can a modulation/channel coding scheme be developed which provides a higher net video bit rate than the QPSK/OFDM scheme reported above, whilst maintaining the same overall coverage?

(ii) Noting that there is excess received power density in the inner part of the service area, can the digital system be designed to have a more spectrally efficient characteristic with graceful degradation of quality at the edge of the service area?

(iii) How can the system be designed to improve the quality of portable and possibly mobile reception?

(iv) What quality levels can be supported by the system in different parts of the service area?

(v) What criteria and methodology should be applied to the subjective assessment of the quality levels available?

*This project is being carried out under sub-contract by National Transcommunications Ltd. (NTL)

Without attempting to provide specific answers to these questions at this stage (this after all is an major part of the work of the first phase of the project) it is worth making some comments regarding possible approaches in two important areas.

The first area relates to a hierarchical picture coding approach in which the two dimensional picture resolution is split by a band of filters into a number of bands and each is individually coded - as is illustrated in Fig 3.(see[9] and other papers from the same conference).

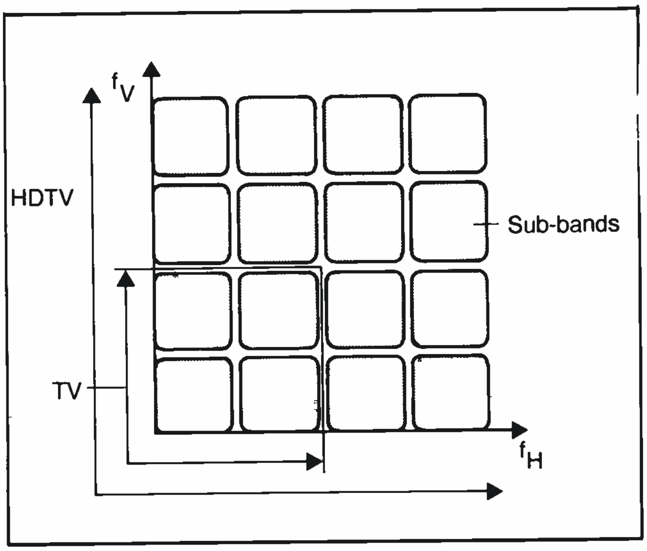

Fig.3. Basic Principle of sub-band Hierarchical Decoding.

The use of such an approach, together with appropriately matching channel coding to provide greater error protection to the lower resolution elements, could go a long way to providing the answers to questions (i) to (iii) above. The feasibility of improving the spectral efficiency and providing the required graceful quality degradation with such an approach (as is illustrated conceptually in Fig 4.) has been investigated and reported on by Uz, Ramchandran and Vetterli.[10]

A important aspect of this approach could be a considerable improvement in the quality of portable reception in a significant part of the service area. For small portable receivers, many with liquid-crystal displays and battery operation, high-definition processing would be a burden both in terms of chip complexity and power consumption. What is required here is to be able to easily find the more ruggedly coded lower resolution parts of the bit-stream and process these only at lower bit-rates.

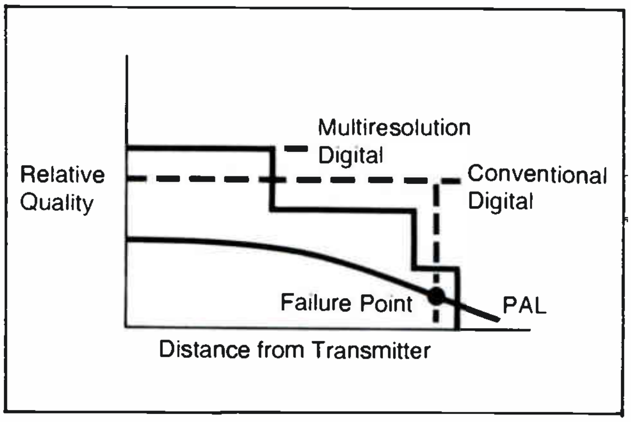

Fig.4. Multi-resolution Transmission

The second area of importance is related to the quality levels achievable at the useable bit-rates likely to be available in different parts of the service area. At present, we have no scientific way of precisely assessing the picture quality obtainable from the high compression coding schemes under consideration.

Following low bit-rate compression, picture quality is highly dependent upon the information content of the scene. In a properly designed system almost perfect pictures are obtained from most scenes, but a softening of resolution occurs on scenes whose 'information content' calls for bit-rates exceeding the capacity of the channel.

The frequency and duration of the periods of reduced resolution will depend on the maximum resolution of the source and reproduction system, the efficiency of the compression algorithm, and the 'statistics' of the varying information content of the picture sequence.

At present, we have no idea what the viewer will make of this situation. If, for example, 96 per cent of scenes are reproduced with full high definition resolution and 4 per cent with reduced resolution, will the viewers' enjoyment be unacceptably disturbed? If so, would it not be better to provide a service of enhanced definition which only suffers a reduction of resolution in 1 per cent of scenes?

What surely matters is the overall impression the viewer will have gained at the end of a typical programme, and we believe that the assessment methodology should be based upon this principle. It is also fundamentally important that the information statistics of the picture sequences used in the compression codec assessment are typical of the programme type as established by an extensive analysis of over-air programming as transmitted today.

A practicable method of low bit-rate codec assessment that would appear to meet the above requirements has been described by Lodge.[11] The method is based upon the automatic measurement and analysis of the 'scene criticality' experienced by the codec under test using picture sequences with information statistics representative of typical broadcast TV. The 'scene criticality' measurements can be related to the CCIR subjective quality scale to characterise the performance of the codec
(a) for different picture source standards at a fixed transmission rate and/or
(b) for a given source standard at different transmission rates.

Subsequent tests based upon the subject assessment of 'whole programmes' will then confirm either:

(a) which source standard will provide the best overall quality - where the available bit-rate is fixed by coverage considerations

or (b) what bit-rate is essential to achieve the service quality requirement of a particular source standard - where coverage considerations may be of secondary importance.

A fundamental aspect of the project, therefore, will be the establishment of the information statistics for broadcast material of different types, and the further investigation and development of the methodology of meaningful subjective assessment along the lines proposed by Lodge.

4.4 Project Organisation and Scope

Under the direction of a Project Board chaired by CCETT, with two members each from industry (Philips and Thomson) and broadcasting (ITC and CCETT), the project is organised into seven Task Forces. The associated work packages are described below:-

(i) Project Organisation and Management

Responsible for the internal and external management of the project as well as finance, promotion of project goals and liaison with the high level European Strategy Group.

(ii) Systems Aspects and Requirements

Responsible for overall systems requirements including specifically:

- the service quality and coverage requirements for different applications
- frequency planning studies and preparing the basis for field trials (including single frequency networks)
- optimising the use of the channel through a consideration of a number of trade-offs involving assessment of source coding performance; available data transmission allocation, target receiver features and costs, potential for future evolution
- interoperability with Integrated Broadband Communications (IBC) including assessment of the constraints imposed by the ATM and STM models of transmission, and the need to transcode from the broad-band to the narrow-band media
- the final definition of the overall system.

(iii) Channel Coding and Modulation

This central work package includes a consideration of single carrier systems employing adaptive equalisation, as well as multiple carrier systems such as OFDM.

The types of carrier modulation under study include QPSK, 8PSK, 16SDK, 16QAM, VSB-4PSK. These studies will involve both computer simulation and hardware simulation employing the systems most appropriate to the different applications as well as systems able to serve more than one application simultaneously (eg fixed position and portable reception).

The above studies will take account of a number of impairments in assessing system performance including:

- CCI/ACI from PAL B/G/I and SECAM and from digital terrestrial television itself

- noise and multipath propagation

- linear and non-linear distortion in the channel and in MATV systems

- the effect of varying parameters such as the number of carriers and the overall bit-rate.

Error management studies will address optimum strategies for each application and for multi-application systems so as to achieve an appraisal of best overall spectrum efficiency.

(iv) Multiplex Structure

Responsible for studies of the multiplexing architecture to include:

- multiplexing of services including the allocation and re-allocation of different services with distinct levels of quality and protection, as well as the possible combination of channels to obtain higher bit-rates and resolution.

- multiplexing of the vision, sound and ancillary data service components, and the synchronisation of the delivery of multiple data streams

- interworking between different networks

- Access Control and security

(v) System Tests and Measurements

Responsible for the standardisation of test conditions and procedures within the project and for organising the objective and subjective assessment of system performance.

Also responsible for carrying-out field trials,

- Initially, with an experimental system developed during phase 1 of the project, involving a wide range of bit rates and quality standards

- In the second phase of the project, field trials of the demonstrations developed under work package vi.

(vi) The development of Laboratory Prototypes and Demonstrators

As a first step towards the development of a service, product laboratory prototypes and demonstrators will be constructed for television broadcast demonstrations and field-trials of portable and mobile reception. The development processes here will lead to a formalisation of the internal and external interfaces, the production of acceptance test programmes and procedures, a realistic implementation of service and receiver features, and a realistic basis for complexity and cost evaluations.

(vii) IC Evaluation and Specification

Responsible for the assessment of the cost of production of the key system components. This will involve an initial specification of the system architecture and modes implementation in "1998" IC technology. The results of evaluations of this work package will influence the considerations of other key work packages and provide a realistic basis for the costing of various system options. This, in turn, will provide a realistic basis for forecasting the likely service introduction dates for services of differing quality levels.

5. CONCLUSIONS

The European Digital Terrestrial Television project, described in this paper, will lead to the eventual goal of replacing existing analogue transmissions with completely digital broadcasts. The experience already gained, both in the United States and in the many previous European collaborative digital projects, will provide a strong foundation for this new and exciting work.

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DIGITAL AUDIO BROADCASTING I

Monday, April 13, 1992

Moderator:
Kenneth Springer, NAB, Washington, District of Columbia

***INTERFERENCE TESTS FOR DAB IN THE FM BAND**
Kenneth Springer
National Association of Broadcasters
Washington, District of Columbia

THE CURRENT CONTEXT FOR DIGITAL RADIO: CLIMATE, OPINION, AND ACTIVITIES IN THE INDUSTRY
Skip Pizzi
Chairman, Committee for Digital Radio Broadcasting (CDRB)
Broadcast Engineering
Overland Park, Kansas
Robert Culver
Technology Group Chair, CDRB
Lohnes and Culver
Washington, DC

***AUTOMOTIVE IMPACT ON DAB SYSTEM NEEDS**
Mark Kady
Delco, Electronics
Kokomo, Indiana

***CANADIAN EUREKA TEST RESULTS**
Stephen Edwards
Canadian Association of Broadcasters
Toronto, Canada

***AMERICAN DIGITAL REPORT**
Edward A. Schober, P.E.
Radiotechniques Engineering Corporation
Haddon Heights, New Jersey

***DELIVERY METHODS FOR DAB**
Perry Spooner
EMCEE Broadcast Products
White Haven, Pennsylvania

***EUREKA REPORT**
Dr. George Plenge
IRT
Munchen, Germany

*Paper not available at the time of publication.

THE CURRENT CONTEXT FOR DIGITAL RADIO: CLIMATE, OPINION, AND ACTIVITIES IN THE INDUSTRY

Skip Pizzi
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Washington, DC

Movement toward an eventual digital transmission system in radio broadcasting is both a technical and a social process. To be successful, such a system must be acceptable to producers and consumers in the broadcast marketplace. As technical and regulatory progress toward this goal continues, work must not take place with too narrow a view. A commensurate study of the business and societal impacts of digital radio broadcasting must concurrently take place. It is toward this end that the Committee for Digital Radio Broadcasting (CDRB) has directed its most recent efforts, as summarized herein.

THE CLIMATE

CDRB has undertaken a process of study for assessing industry viewpoints and sentiments on various issues surrounding digital radio delivery. The first step has involved a simple "ear to the ground." Through this approach, the basic concerns of the industry and issues worthy of further assessment have been determined. These are detailed below, to the best of the Committee's knowledge.

This industry overview formed the basis for the next step in the process. A detailed survey for the industry on the subject of digital radio has been developed (see Annex A). It has already received pilot distribution and revisions, and will be distributed widely among U.S. broadcast professionals over the next several months. Results will be presented in October 1992 at the SBE National Convention in San Jose.

Meanwhile, CDRB has also conducted preliminary discussions with a consumer audio publication regarding the circulation of a second survey, this one directed toward assessment of the potential radio *audience's* predisposition toward a digital service (see Annex B). Although CDRB's preferred target date for results from this consumer survey is NAB 1993, discussions are ongoing, and a completion date has not yet been finally determined.

Recent influences

Several events of the last year have shaped the issues of digital radio somewhat. These include the FCC decision in October 1991 to seek an S-band allocation for U.S. digital radio service at WARC-92, the Canadian L-band propagation study (Summer '91) and L-band Eureka-147/DAB tests (Fall '91), and the formation of a Digital Audio Radio (DAR) subcommittee by the Electronic Industries Association's Consumer Electronics Group (EIA/CEG), also in October 1991. Other noteworthy activity has taken place among the various format proponents, but most has been cast with a tone of heightened propriety and non-specificity.

The Canadian L-band propagation tests showed that L-band application of a terrestrial digital radio service would have about the same coverage as current VHF FM service at the same transmission power and antenna height. The tests were system-generic, and did not involve any particular digital radio format. These results were confirmed by subsequent Canadian tests of the Eureka 147/DAB format at 1497MHz, however.

The Canadian and FCC actions seem to have had mutually exclusive results. On the one hand, the Canadian tests showed relatively positive results for L-band application of digital radio service, while the FCC effectively ruled out the use of this band for that purpose in the U.S., at least temporarily. (The commission has officially stated that a "post-WARC" conference will be held on purely terrestrial digital radio matters, at which time "all bands" will be re-opened for discussion of allocations. When this conference will occur, and who/what it will involve have not yet been detailed. It is important to recall that the WARC-92 agenda item on digital radio refers to "Broadcast *Satellite* (Sound) services and complementary terrestrial" frequencies. Although the exact nature of the latter is subject to some interpretation, the WARC agenda clearly does not address *exclusively* terrestrial, (i.e., non-satellite-related) digital radio allocations.)

The FCC action, attributed mostly to U.S. Department of Defense objections to reallocation of L-band away from their uses in aeronautical testing, broke apart what had been seen as a "western alliance" for an L-band digital radio allocation going into WARC-92 (including Canada, U.S.A., Mexico and Brazil). It has also turned the attention of the U.S. radio industry toward "in-band" systems, because the S-band is generally considered disadvantageous for digital radio applications. While this is held to be a laudable development by many broadcasters, remember that *L-band* was also considered technically unfeasible by some until targeted studies were conducted. Before S-band is entirely ruled out for technical reasons, similar tests specific to digital radio use should also be undertaken. To date (1/92), no organization has stepped forward to pursue such testing. (This is understandable, since such testing would be unlikely to be proposed before a WARC decision on S-band, anyway.)

A related concern on this matter is the proximity of S-band proposals to the operating frequencies of microwave ovens in the area of 2350-2450MHz. The FCC's S-band proposal avoids most (but potentially not all) of this possible interference by supporting a lower band than some earlier proposals had suggested, 2310-2360MHz. (Note the 10MHz overlap at the upper end of the proposed band.)

On another front, the EIA/CEG's DAR subcommittee has begun its work (see below), and has laid the foundation for standardizing a digital radio broadcasting system in the U.S. under its auspices. Although the EIA is generally seen as a trade association of electronics manufacturers, broadcasters seem well represented on the subcommittee and its component groups.

A look over the shoulder

On the satellite front, a U.S Court of Appeals ruling in March 1991 has at least temporarily stymied the Radio Satellite Corporation (RSC) in its attempt to establish a satellite digital radio service for the U.S. by 1994. In December 1991, VOA held demonstrations of a satellite-delivered digital radio service in Washington, DC. VOA's plans for the service are primarily geared toward eventual replacement of the current short- and medium-wave delivery used by their services outside the U.S. Meanwhile, 1991 also saw the official introduction of DBS-delivered digital radio service in Japan.

None of those events seem particularly troubling to U.S. broadcasters, but their concerns have mounted somewhat because of the domestic growth of digital radio service delivered via cable. In a few areas of the country, aggressive marketing of these services has borne some fruit, although the true potential impact of such service on broadcast radio remains unknown.

Current concerns

With the risk of satellite competition at least temporarily subsided, broadcasters have turned their attention to considering the impact of various in-band scenarios upon the marketplace. In general terms, in-band methodologies have been broken into two camps: those that would place the digital signal within the broadcaster's existing channel (so-called "in-band, on-channel" [IBOC]), and those proposing use of an adjacent or other "interstitial" channel within the broadcast band for a separate digital carrier ("in-band interstitial" [IBI]).

An IBOC method seems relatively acceptable to FM broadcasters, if it can function without interference or other sacrifice of their existing coverage. A more recent caveat expressed in this scenario is that the IBOC digital signal be *permanently constrained as simulcast-only*. These broadcasters seem willing to sacrifice their own potential to establish a secondary future service (thereby moving away from the AM/FM transition model), for assurance that their competitors will not do the same.

Some AM broadcasters have voiced fears that an IBOC system would not include them, or would give FM broadcasters a better shake. Other AM broadcasters remain open to *any* proposal that will put them in better competitive condition, even if it involves a move in frequency and/or simulcasting.

The move toward in-band has also calmed fears of high implementation costs and shared transmission "pods," but estimates of transition expenses for digital radio are still hard to come by. The question of IBOC or IBI still has some bearing on implementation costs, because the latter involves a second transmission system (and potentially a second site). One IBI proposal also still includes a shared transmission paradigm.

Others are now questioning just how good the performance of an in-band system can be, given the taste of a wide-bandwidth, out-of-band format's demonstrations. Multipath resistance will have to be excellent.

Finally, some broadcasters are beginning to give concrete consideration to applications of a digital delivery system's auxiliary data capacity. The tight operating margins of the current radio marketplace encourages exploration of other possible sources of revenue from related service. Existing SCA data and paging systems have set the context for this analysis, and the expected implementation of the RBDS (Radio Broadcast Data System) standard in 1992 should further this thinking.

CONSUMER ISSUES

"Does the listener really want this service?" has been a consistent question since the earliest discussion of digital radio. While there is never a definitive answer before the fact when new products or services are introduced, consideration of some contextual issues may be helpful.

Research conducted by the Recording Industry Association of America (RIAA) suggests that in early 1992, sales of CDs will eclipse those of cassettes. It has long been the case that CD sales outstripped those of LPs, but the proliferation and convenience of cassettes kept them the sales-leaders. Industry analysts also felt that consumer's assessment of the cassette's audio quality was not sufficiently lower than the CD's to warrant a choice of the latter. This assumption has now proven false, and consumers seem now to have chosen almost purely on the basis of quality. The higher cost of CDs is apparently not a sufficient deterrent to this choice, showing that consumers are willing to pay more for this better quality.

Figure 1 shows the sales volumes in units (net after returns) of LPs, CDs and cassettes for the last decade. Note the projected crossover now occurring between cassettes and CDs. (RIAA figures show that *dollars spent* on CDs

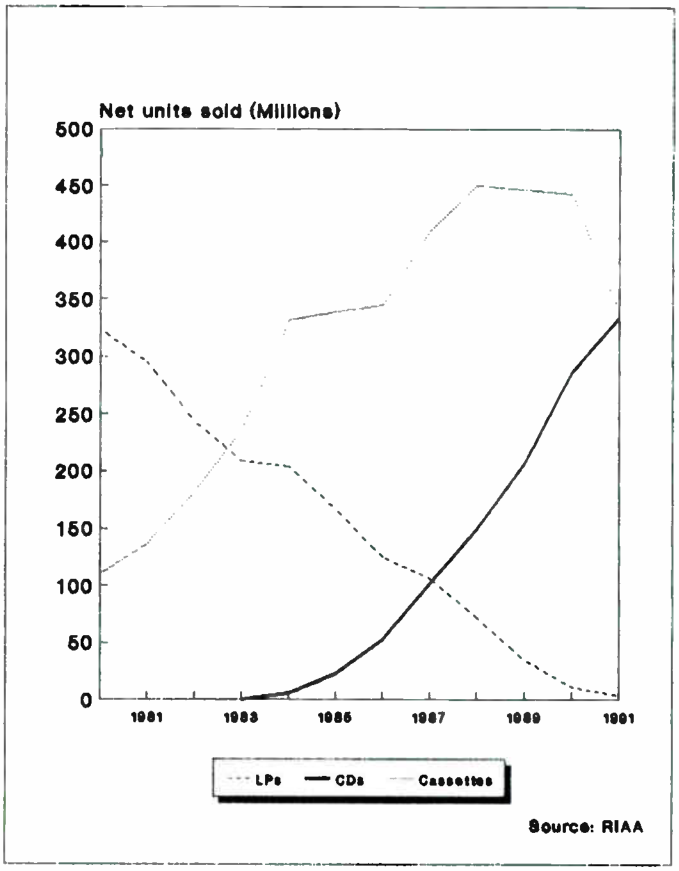

Figure 1. Sales comparison of major recorded music formats, 1980 to 1991. (Data for July-December 1991 is estimated.)

have already outpaced those spent for cassettes, with that transition taking place in mid-1991.)

Moreover, Figure 2 shows that these transitions have taken place with a total U.S. marketplace penetration for CD players of *only 25%*. The potential for further growth seems enormous.

Both these surveys suggest that consumer's preference for digital audio is strong and growing. Analog radio's perceived value may continue to drop in this context.

A consumer survey specifically geared to evaluation of this issue may further enlighten the broadcast industry. CDRB plans to undertake this process in cooperation with other organizations. (See Annex B.)

WORK TOWARD STANDARDIZATION

During 1991 there were several tests and demonstrations of digital radio. The only fully functional broadcast system was Eureka 147/DAB. It was demonstrated over-the-air on UHF channel 15 at NAB '91 in Las Vegas (its first U.S. on-air demonstration), and at L-band (1497MHz) in Toronto during November and December.

Other system proponents reported continuation of their development, and planned further interim tests and demonstrations of their competing systems. Notable in this category were USA Digital's demonstration of a prototype of their in-band/on-channel Project Acorn FM system at NAB '91 in Las Vegas, and a showing by LinCom Corporation (now associated with Strother Communications) of their prototype for an in-band/adjacent-channel FM system at a Washington, DC press conference in October 1991. The VOA also demonstrated satellite-delivered digital radio in a preliminary form at a Washington, DC presentation in December 1991, noted above.

EIA Action

Meanwhile, the Electronic Industries Association's Consumer Electronics Group (EIA/CEG) organized a Digital Audio Radio subcommittee (R-3 Audio Systems DAR Subcommittee) at a meeting in October 1991. That subcommittee and its various working groups have become the *defacto* standard setting body for digital radio in the United States.

The scope of the EIA activity on "DAR," as they refer to it, can be described as follows:

- Define DAR service objectives
- Study source and channel coding methods for broadcasting
- Propose one or more systems for testing
- Recommend system for adoption as a standard

The structure of the subcommittee consists of a steering committee directing the work of the following Groups and Subgroups:

- WG-A - DAR Systems
  - SG-1 - Source Coding
  - SG-2 - Channel characteristics and coding
- WG-B - Laboratory and field tests
- WG-C - Costs

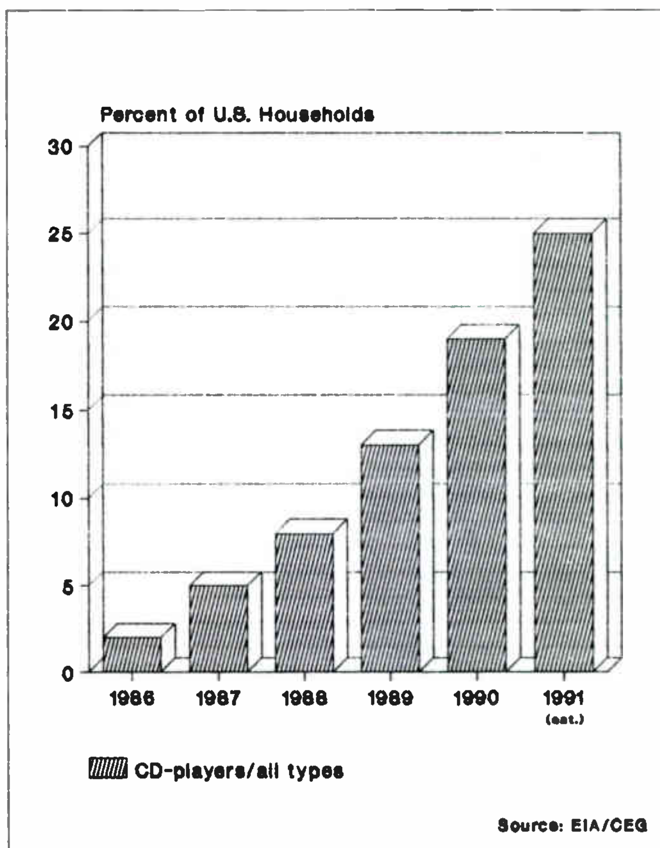

Figure 2. Total CD-player market penetration among U.S. households.

· WG-D - Liaison
· WG-E - Objectives and Evaluation

The first task of WG-E, defining the DAR objectives, has been completed. WG-A has organized its tasks and staff, adopting the following approximate schedule:

· Mid-92 - Receive positive commitment of interest from all system proponents wishing to take part in EIA's DAR testing program.

· End-92 - Receive technical description for the proponent systems to be presented for testing.

· Mid-93 - Receive hardware and begin testing.

· End-93 - Deliver recommendation for DAR standard system.

The EIA's DAR subcommittee activities are open to all participants. Representation and contributions from broadcasters, manufacturers and consumers is vital for a thorough analysis of digital radio proposals, and a valid system recommendation.

CONCLUSION

The work toward establishment of a digital radio service will necessarily continue for several more years. Gradual refinements and decisions, such as those made in the past year, will incrementally shape the next generation of radio broadcasting's final form. During this process, it is imperative that broadcasters remain informed and give voice to their opinions throughout. Only then will an acceptable and viable standard be achieved, one that allows broadcasters to prosper while serving the American public in an improved and expanded manner.

Now that some broad directions have been set, and basic market conditions are known, an appropriate next step involves further study of some detail. The upcoming surveys that CDRB proposes to undertake are a contribution toward that end. The attached Annexes are included for your perusal and comment. CDRB encourages further industry input on the strategy and scope of these surveys. Please direct remarks to the authors, who welcome and appreciate any such interest.

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ANNEX A

CDRB INDUSTRY SURVEY

CDRB (Committee for Digital Radio Broadcasting)

DIGITAL RADIO SURVEY

This Survey is designed to gauge your personal knowledge, reaction and expectations regarding Digital Radio as it is being proposed. Please feel free to comment on the questions or survey style as necessary, by including side or end comments or by contacting CDRB Technology Group Chairman, Robert D. Culver at Lohnes & Culver; address 1156 15th. St. N.W., Suite 606, Washington, D.C. 20005, telephone (202) 296-2722

1. I am generally aware of Digital Radio broadcast technology developed in Europe and as discussed for implementation in the United States.

   ___Yes ___Somewhat ___No

2. I am familiar with the following "U.S." digital radio proponents (list complete as of September, 1991).

Eureka 147/DAB	___Yes	___Somewhat	___Not at All	___No Opinion
Stanford Telecom DAB	___Yes	___Somewhat	___Not at All	___No Opinion
Kintel/Power Multiplexing	___Yes	___Somewhat	___Not at All	___No Opinion
Acorn/USA Digital	___Yes	___Somewhat	___Not at All	___No Opinion
Mercury Digital/MFM	___Yes	___Somewhat	___Not at All	___No Opinion
Synetcom/ Digital FM•S	___Yes	___Somewhat	___Not at All	___No Opinion
LinCom DAB	___Yes	___Somewhat	___Not at All	___No opinion
American Digital Radio	___Yes	___Somewhat	___Not at All	___No Opinion

3. I support the following digital radio questions/policies as indicated:

A. Digital radio should:

Be reserved <u>only</u> for present broadcasters.	___Definitely	___Maybe	___No	___No Opinion
Give <u>preference</u> to existing broadcasters.	___Definitely	___Maybe	___No	___No Opinion
Allow new broadcasters equal access.	___Definitely	___Maybe	___No	___No Opinion
Be implemented as a common carrier service.	___Definitely	___Maybe	___No	___No Opinion
Include provision for direct satellite delivery.	___Definitely	___Maybe	___No	___No Opinion

B. Implementation should utilize:

The present AM <u>and</u> FM Bands.	___Definitely	___Maybe	___No	___No Opinion
FM Band <u>only</u> accommodating AM B'cstrs.	___Definitely	___Maybe	___No	___No Opinion
New spectrum:		___Yes	___No	___No Opinion
If yes, then: ___Present TV Bands	___Above 1GHz	___Above 2GHz		___No Opinion

C. Digital radio should be constructed using:

A single existing transmitter site <u>without</u> gap fillers.	___Definitely	___Maybe	___No	___No Opinion
A centrally located transmitter with <u>some</u> gap fillers.	___Definitely	___Maybe	___No	___No Opinion
Many small transmitters in a "cellular" style.	___Definitely	___Maybe	___No	___No Opinion
A satellite/terrestrial hybrid system.	___Definitely	___Maybe	___No	___No Opinion

4. Please indicate your expectations of the following Digital Radio technical aspects over the range indicated.

		strongly agree	agree somewhat	no opinion	agree somewhat	strongly agree	
Best audio quality should be like:	"CD"	____	____	____	____	____	"FM"
AM/FM digital stations should be technically equal:	Yes	____	____	____	____	____	No
Multipath audio effects should be:	Eliminated	____	____	____	____	____	Improved
Extra data capacity should carry:	Program	____	____	____	____	____	Data
Non-broadcast "Data" capacity should be:	Large	____	____	____	____	____	Small

5. Implementation of Digital Radio in the U.S. may require sharing of new or existing transmitter facilities. Please indicate the degree to which you might find the following scenarios acceptable.

Only the present broadcaster should use his existing facilities. ____Agree ____Somewhat ____Disagree

Share present facility for in-band use with other broadcasters. ____Agree ____Somewhat ____Disagree

Yield your facility and move to another to share with others. ____Agree ____Somewhat ____Disagree

Abandon your present traditional facility to program on an "audio channel" within a multi-broadcaster shared network. ____Agree ____Somewhat ____Disagree

6. How do you feel about the prospect of building and using a totally unique Digital Radio facility? For example, an out-of-band system, shared with others, and using numerous transmitter sites. ____Very Worried ____Curious ____Interested ____Would Support It

7. Digital Radio system development should be through the <u>combined</u> efforts of the many, now separate, system proponents. ____Definitely ____Possibly ____Not at All

8. Digital Radio systems proposed for the U.S. should be tested at an independent test center (like the advanced TV test center). ____Definitely ____Possibly ____Not at All

Feel free to include any additional comments here or on another sheet:

____

____

____

____

Industry ______________________ Position ______________________

<u>Optional Information</u>:

Name ______________________ Telephone no. ______________________

Organization/Affiliation ______________________ Fax Number ______________________

Address ______________________

City/State/Zip ______________________

Thank You! Please contact CDRB officials at any time to discuss Digital Radio.

ANNEX B

CDRB PROPOSED CONSUMER AUDIO SURVEY
Draft 1.0, 27Jan92

A. Radio Ownership

1) Do you own at least one non-portable home radio? ___Y ___N
2) Do you own at least one portable radio? ___Y ___N
3) Do you regularly drive a car equipped with a radio? ___Y ___N

B. Radio Use

1) Considering all your radio listening, which of the following best describes your habits?
   ___Listen exclusively to *FM* stations
   ___Listen mostly to *FM* stations and occasionally to *AM* stations
   ___Listen approximately equally to *FM* and *AM* stations
   ___Listen mostly to *AM* stations and occasionally to *FM* stations
   ___Listen exclusively to *AM* stations

2) What kinds of programming do you listen to on *FM* radio? (check all that apply):
   ___Music
   ___News/Weather/Talk
   ___Sports
   ___Other (describe:___________________________)

3) What kinds of programming do you listen to on *AM* radio? (check all that apply:)
   ___Music
   ___News/Weather/Talk
   ___Sports
   ___Other (describe:___________________________)

4) Considering all your radio listening, which of the following best describes your habits?
   ___Listen to radio mostly in the car
   ___Listen to radio mostly at home
   ___Listen to radio mostly at the office
   ___Listen to radio mostly on a personal system (with headphones)

C. Other audio equipment ownership

1) Do you own a home CD player? If so, what types? (check all that apply:)
   ___Home unit (component type)
   ___Portable unit (personal or "boombox")
   ___Car system
   ___Do not own a CD player

2) If you answered "yes" to #1, are any of those CD players of the *CD-changer* variety?
   ___Yes ___No

3) If you answered "no" to #1, do you plan to purchase a CD player?
   ___Yes, within the next 6 months
   ___Yes, within the next year
   ___Yes, within the next two years
   ___No, or no plans to purchase

4) If you answered "yes" to #1, about how many CDs did you purchase in the last year?
   ___5 or less
   ___10 or less
   ___20 or less
   ___50 or less
   ___more than 50

5) Do you own a cassette deck? If so, of what type? (check all that apply:)
   ___Home unit (component type)
   ___Portable unit (personal or "boombox")
   ___Car system
   ___Do not own a cassette deck

6) If you answered "yes" to #5, about how many pre-recorded cassettes did you purchase in the last year?
   ___5 or less
   ___10 or less
   ___20 or less
   ___50 or less
   ___more than 50

7) If you answered "yes" to #5, do you regularly use the recording feature of your cassette deck? If so, for what purposes? (check all that apply:)
   ___Yes, to record CDs
   ___Yes, to record LPs
   ___Yes, to dub prerecorded cassettes
   ___Yes, to record off the radio
   ___Yes, to make personal audio recordings
   ___No, never use recording (or not equipped)

D. Comparative Assessment of Radio Quality

1) Using the list below, give your impression of each system's audio quality in the column at right. Use only one letter choice on each line.

a) AM radio	Best___
b) Audio cassettes you've recorded	___
c) CDs	___
d) FM radio	___
e) LPs	___
f) Pre-recorded audio cassettes	Worst___

2) What would you like to see improved in future radio broadcasting systems? (check all that apply:)
   ___Less background noise
   ___Less interference
   ___Less breakup/loss of signal
   ___Better frequency response
   ___Wider dynamic range
   ___No improvements required
   ___Other (specify:_______________________________________)

3) Would you like to have access to national radio signals broadcast by satellites? If so, would you pay for them?
   ___Yes, would want them and would consider subscribing
   ___Yes, would want them, but not on a subscription basis
   ___No, would not want them under any circumstances

4) What other radio broadcasting services would you consider useful in the future? (check all that apply:)
   ___CD quality audio
   ___Pay-per-listen concerts
   ___Over-the-air album delivery (credit card/phone orders)
   ___Over-the-air computer software delivery
   ___Text messaging on your receiver
   ___Still video accompaniment
   ___Interactivity (respond to questions or make purchases)
   ___Other (specify:_______________________________________)

5) If a new digital radio service offered CD-quality audio with programming similar to today's radio or better, and receivers were comparably priced, would you buy one?
   ___Yes, definitely
   ___Yes, probably
   ___Probably not
   ___Definitely not

THANK YOU!

VIDEO PRODUCTION AND POST PRODUCTION

Monday, April 13, 1992

Moderator:
Al Petzke, WTVO-TV, Rockford, Illinois

BUILDING TECHNICAL FACILITIES FOR A NEW GENERATION OF GRAPHICS PRODUCTION FOR *ENTERTAINMENT TONIGHT*
Robert B. Kisor
Paramount Pictures
Hollywood, California

MOVING PICTURES ON AIR—THE CREATION OF DYNAMIC GRAPHICS
John Woodhouse
Quantel Limited
Newberry, Berkshire, England

***DRIVING TOWARDS PC-BASED POST PRODUCTION**
John Sergneri
Autodesk Inc.
Sausalito, California

BRIDGING COMPUTER GRAPHICS AND HIGH QUALITY VIDEO
Danielle Forsyth
Tektronix, Video Products Operation
Wilsonville, Oregon

**FOLD IT OR FIX IT:
THE CHANGING FACE OF SPECIAL EFFECTS**
Martin Stein
Ampex Corporation
Redwood City, California

MOBILE UNIT ONE. . . FIRST STOP: 1992 WINTER OLYMPICS
James M. Herschel
CBS, Inc.
New York, New York

***A PRODUCER'S GUIDE TO DIGITAL COMPOSITING—
THE MAKING OF THE GLORIA ESTEFAN VIDEO**
Ronald B. Fenster
Limelite Video
Miami, Florida

*Paper not available at the time of publication.

BUILDING TECHNICAL FACILITIES FOR A NEW GENERATION OF GRAPHICS PRODUCTION FOR *ENTERTAINMENT TONIGHT*

Robert B. Kisor
Paramount Pictures
Hollywood, California

Paramount Pictures recently completed an update and expansion of the technical production facilities for Entertainment Tonight. The new facilities incorporate state-of-the-art digital tape machines, paint boxes, still stores, character generator, and a magneto optical audio disk recorder/editor into the existing high level production facility. These devices along with their powerful creative talent enabled **E.T.** to create a new three dimensional look to inaugurate their tenth season on the air.

BACKGROUND

Entertainment Tonight is a day and date entertainment news show that has always led the way in on-air graphics production. The show's production facilities had not seen any significant change or update since they were first built on the Paramount Pictures lot in 1984. Built as a state of the art facility with three channels of digital effects, six one-inch tape machines, paint box, character generator, and still store, graphics technology had advanced beyond the 1984 installation. In addition the volume of daily graphics work had increased significantly extending the room's production capacities to its limits.

NEW LOOK

The start of the tenth year of production generated a need to create a "new look" for the show and update both the graphics technology and audio capabilities of the facility. A new three dimensional look was designed, but three problems needed to be resolved: What equipment would be required to implement the "new look", how to integrate this equipment into the existing facility while keeping the operation manageable, and how to schedule the installation of the new equipment into a production facility that works five days a week, 52 weeks a year.

NEW EQUIPMENT

The search for new equipment started at last year's NAB. The shopping list included a new still store system, additional paint facilities, a new character generator, a digital video playback device and a small digital audio workstation. None of the decisions regarding what equipment to purchase was obvious, although multiple vendors were considered in each category with price and function

as a basis for the evaluation. The cornerstone of the installation was the still store system. E.T. has over 40,000 archived still pictures and anticipates to increase that number significantly over the coming years. It was important that the new system allow for a large number of on-line stills with capacity to expand. The installed system consists of two dual output still stores interconnected via SCSI bus allowing the disk storage of both systems to appear as one. In addition a magneto optical disk drive is connected for archiving. The system has a well integrated search and retrieval system based on user defined database information. The database information is embedded in the still picture file and travels with it for historical and archival purposes. The system has digital video and key inputs and outputs to interface with the paint system or outside devices. And finally one still store has limited cut and paste capability. It was further decided that to truly enhance the graphics operation, a new generation paint box should be added to the existing paint box. The final system allows for both still stores and the new paint box to be interconnected via SCSI bus. This feature significantly increases the efficiency of the operation by allowing transparent transfers between still stores and paint box without the need of any data transfer between uncommon systems. The system configuration allows for approximately 2000 on-line stills with the ability to expand by adding additional or larger disk drives and it provides for two simultaneous paint box and one limited cut and paste operations. The character generator that was installed is a true dual channel device that allows for the editing or updating of either channel while the other one is on air. The device also has the capability of creating transparent backgrounds and characters.
Custom three dimensional moving icons were designed for the "new look". The intent was to store these digitally for playback and integration into the daily production. The first choice to accomplish this goal was a digital disk recorder. Although completely adequate for this application, its auxiliary uses were limited. The final decision was to use two composite digital video tape machines. They were integrated into the system to not only provide for icon playback, but also to provide the availability for editing, dubbing, or any other production or post production application.
Not only was attention being given to video production, additional audio facilities were also proposed. This was accomplished by installing a small digital audio workstation. The system is configured with two on-line magneto optical disks for audio storage. The unit has one analog, one digital, and one digital optical stereo input, as well as two independent analog and digital stereo outputs. The system is interfaced to a Macintosh computer for ease of operation and increased user friendliness of the editing functions.

SYSTEM INTEGRATION

System integration required interconnecting new graphics equipment installed in one building with the production control room located in another building while allowing the still stores to be controlled from either location. The new system configuration contained 35 video sources for a 24 input switcher and seventeen key inputs were added to the existing fifteen. An addition was made midway into the design phase when production required that a logo be keyed into all segments played back into the show from any of the eleven vtrs in the system. The challenge was that this linear key had to be performed prior to the video entering the switcher since it could not be certain that a keyer in the switcher would be available at all times. The solution to the graphics interconnection was to install all the equipment in the graphics area and then gen-lock that facility to the stage, running equalized and isolated video cables between the two facilities. The paint box and still store operating positions in the graphics room have full monitoring capabilities which include a 900 line 20 inch color monitor, waveform monitor, and vectorscope at each of three locations. Additionally, four small color monitors were provided in the still store position to monitor its four outputs. An output of the existing stage routing switcher was connected to the graphics room so images could be captured from any source on the stage. A second still store operating position was installed in the control room so the operation could be controlled from there during production and an A/B switch was installed to switch the RS422 control lines between the two sets of still store control panels.

The video switcher was able to directly accommodate all the new key sources. Monitoring of the new video sources required the addition of a rack in the monitor wall of the control room. This would require physical construction since the existing racks were built into the wall. In order to accommodate the additional video sources, a timed 20 x 10 sub-router was installed ahead of the production switcher with a multi-bus control panel located at the Technical Director's position. Eleven of the inputs of the sub-router were the video tape machines, the other nine were from miscellaneous sources. Nine of the outputs of the sub-router became video switcher inputs, two of these were connected through stand alone single channel linear keyers to provide the logo key. The key video and key cut inputs to the keyers were taken from existing sources. The output of the keyers were then taken to the switcher. A serious timing problem arose when the video input to a keyer was switched from one vtr to another since the path length from the sub-router output direct to the switcher was significantly shorter than the path through the keyer and the keyers could not gen-lock. There was not enough time during production to manually reset the timings of the tape machines as the keyers were switched in and out of the path. To resolve this, an automated TBC controller was installed which allows for precise re-timing of multiple vtrs in just a few seconds.

The audio installation was straight forward since audio console inputs and outputs were made available for the digital workstation inputs and outputs.

INSTALLATION

Entertainment Tonight produces a new show every day 52 weeks a year with the exception of major holidays. The system as designed was going to require adding new video tape machines, graphics equipment, still stores, a character generator, video routing switcher, and the rearrangement of the existing video and key switcher inputs. In addition, the graphics room had to be enlarged to accommodate the additional equipment and the control room monitor wall had to be enlarged for monitoring of the new video sources. It was obvious that this work could not be completed in one weekend. Therefore a four step process was implemented. First the graphics facility would be enlarged and the control room monitor wall would be expanded by one rack. The graphics room work required moving two inside walls and providing additional technical power. The control room work required opening up a wall, adding the equipment rack and then finishing both ends of the monitor wall to maintain good acoustics and aesthetics. Once this was completed, the new graphics equipment was wired and tested off-line. The second phase was to cut-over the graphics equipment on a weekend. This process was completed with little difficulty. The problems didn't arise until Monday morning when the new still system showed the beginning of what turned out to be hard disk failure. This prompted an emergency return to the old still system which had remained in place. Once the new system was repaired, it was put back on-line and phase three was commenced. Phase three consisted of pre-wiring the new control room monitors, digital tape machines, character generator, and sub-router. Phase four was the critical path, the final cut-over and integration of all components of the new system. It was still clear that the scope of work required more than just Saturday and Sunday. It was agreed that Friday's show would be pre-produced allowing one more day for installation. In that three day period, over 110 switcher and key inputs were installed or moved, and system timed. These were in addition to the hundreds of other audio, video, and control cables that were required for monitoring and distribution. Miraculously Monday's show was produced on schedule with all equipment operational.

CONCLUSIONS

The work that was undertaken for this update required cooperation and understanding from both engineering and production personnel. The team effort that resulted insured the successful completion of a very difficult project. The final system has allowed Entertainment Tonight to maintain its position as a leader in television graphics production.

MOVING PICTURES ON AIR
THE CREATION OF DYNAMIC GRAPHICS

John Woodhouse
Quantel Limited
Newberry, Berkshire, England

Abstract - For the past decade the broadcast industry has been integrating electronic graphics into a wide range of productions. The technology to deliver these graphic images moves on, and is now able to turn the static graphic images into dynamic graphic images. Todays productions demand a fast, flexible graphics system able to create and transmit graphic sequences within the tight timescales of live production. By devising an intuitive control structure and utilising the latest technology, these demands can be achieved.

INTRODUCTION

There has been an evolution in the use of graphics in TV production. The changes have been brought about by advances in technology making it easier to create the graphics and deliver them for on air transmission. As technology continues to advance greater possibilities for the creation and application of graphics emerge. The first revolution took graphic composition into the medium of television itself, the age of electronic graphics was born. By using video framestores and digital painting functions, television graphics gained a freedom and speed of creation never seen before. The development of graphic images has continued, both in terms of technical possibilities and in production style. There was however a limiting factor to their development, they were static images in a moving medium. It is a natural development that a moving medium should have moving graphics. Recent developments have enabled the creation of such moving 'dynamic' graphics, within a time-frame which allows them to be used in the exacting conditions of live broadcasting.

This paper identifies some of the technology being used to put moving pictures on the air.

Animations

The word 'animation' can be applied to many things; from illustrated cartoons on film to 3D computer rendered images...to name just two. These two examples have something in common, they require a great deal of human effort and take a great deal of time to compose. In the lean environments of today's productions neither commodity is in abundant supply. As finished pieces they could be included in a live broadcast, but could not be created specifically for a live broadcast within the tight timescale of live production. So here we have the first requirement for the on air machine - speed of production.

Animation can also imply movement of images. Again the technology exists to move video images, we are all familiar with Digital Video Effects. We are concerned with moving static graphic images to create dynamic graphic images. Just as the creation of the graphic required precise manipulation and control, the move too will require accurate composition. Therefore the on-air machine must have a flexible, accurate control structure - the second requirement.

Every day we see animated graphics packaged and formatted to fit the look of a show. Titles slide in and reveal, pictures zoom out, montages build...and so on. This is all engineered to be achieved within the financial and time constraints of the production. Given enough time and resources these multi-layer animations could be achieved with existing facilities, what is changing is the timescale and efficiency with which the production can now be completed.

Traditional Suites

The task of creating multi-layer graphics has often been assigned to the post production suite. Within the suite an array of equipment is on-hand to tackle all editing needs from the simple cut to the most complex multi-channel digital effects sequence. The suite is general purpose but evolved from the need to edit tape in a linear environment. Several people would be involved with the creation of the graphic sequence leading to a high cost of production. A further draw back with analogue suites is the generation loss involved with multiple layers. Most graphic sequences require three or four layers or passes which, to maintain image quality, is beyond the limit of analogue suites. Digital suites offer an alternative but these raise the cost of production even further.

The demand for dynamic graphics has been created by technological advances in the equipment itself. In the past it could have been argued there was insufficient demand for dynamic graphics to consider building a dedicated suite. This is now no longer the case and specialist graphic areas abound. Even so dynamic graphic sequences have kept there distance from on air situations such as news. Technological advances are bringing graphic production timescales within reach of on-air demands.

Video Storage

A prime requirement for a dynamic graphic machine is the ability to replay the frames created at real-time speed. This 'store' would need to record graphic images frame by frame, then replay the sequence at 30 frames per second. The first attempts at this used a VTR as the 'store' using an edit controller to perform single frame recording. This approach was successful but produced considerable wear and tear on the VTR and even more so for the tape, often resulting in drop-outs on the finished sequence. VTRs like the conventional edit suites are designed to work in the linear domain and are not suited to the frame by frame working methods of animation.

Replacing the VTR with disc drives is one solution to the problem. The drives can be engineered to look like VTR's having real-time play and record and timecode for individual frame location. These Digital Disc Recorders as they are known still function as linear read/write devices. There advantages include faster access time for frame working, they cost less than a VTR and they store information in the digital domain, so allowing multiple layering of the graphic sequence.

The development of disc drives continues providing smaller units with higher capacities. The latest parallel transfer drives can be used to provide video storage in the order of minutes. Unlike DDR's these drives offer real-time random access to the stored video. With these drives as a storage medium an edit is simply made by re-addressing the store instead of re-recording the video. This type of non-linear storage clearly has advantages in terms of speed, and their random access ability makes them well suited to the graphics environment.

Alongside the development of disc storage there have been major advances in solid state memories, especially dynamic RAM. The 1M Bit RAM has been with us for several years, the 4M Bit RAM is now widely available. A 4:2:2 component digital picture requires some 700K Bytes of storage or 5.6M Bits, we are not far from having a framestore on a single chip. With such low power consumption and the use of multi-layer circuit boards the packing density is governed by the physical size of the chips themselves. To date packing densities of over 500 chips on a single 15 X 9 inch board can be achieved. Using 4M Bit chips this represents over 380 frames of video, which equates to 12.5 seconds of real-time video. Obviously this is too short for editing purposes but with typical graphic sequences lasting 5 or 6 seconds, is sufficient for dynamic graphics

Both disc and RAM provide suitable mediums for video storage. The higher capacities available with discs make them more suited to the edit and compositing environments while the lower cost and smaller size make RAM more suited to dynamic graphic applications. The system integration of the RAM is made easier by the absence of interface hardware such as disc controllers. With very little effort the RAM can be accommodated within the rackframe of the graphics unit itself. The result is a very compact, cost-effective, workstation which needs little supporting equipment.

Image Processing

To complete the dynamic graphic on time requires a number of images to be processed within a defined time frame. For post production the graphic time required becomes one factor in the overall production timescale. In the live on-air environment the production time is fixed, defined by the on-air time, so the graphic processing must fit within this time-frame. For post production several hours may be required to render images from a 3D computer system, perhaps running through the night. This time-span is simply not available to on-air graphics production.

The ability to produce graphic sequences within the short timescales available demands that the images are held in video form within a real-time store, and the operational approach be flexible. As we have seen a RAM based store combines a real-time store with a non-linear environment for frame by frame composition. A flexible operational approach can be achieved with an intuitive control structure and dedicated hardware processing.

Operational Integration

Each of the functions required to produce the graphic sequence will have their own control requirements. The graphic composition will require an interface with the designer allowing him to draw and create the graphic elements, source material will need to be brought into the workstation with a VTR interface, and so on. If the workstation treats each of these as separate unconnected tasks much time will be lost swapping between environments and exchanging images. To produce a free-flowing work area, which blurs the boundaries between control environments, requires a sophisticated software control structure.

Dedicated Hardware

One of the more demanding functions of a graphics system is to re-position or re-size an image. Digital Video Effects machines comprise a large proportion of hardware to render the compression or move in real-time. If we are able to move away from real-time processing then significant savings can be made in hardware, which will reflect through in cost savings. The systems CPU will have the power to perform the interpolation processing for the moving image. However it will not have the necessary speed to meet the on air demands. The solution obviously lies between these extremes, allowing the CPU to oversee the process while dedicated hardware interpolates the image. Using Application Specific Integrated Circuits the hardware component count is reduced, allowing a reduction in power consumption and an increase in reliability. With this arrangement a 3D image compression and position can be rendered in fractions of a second per frame. Left to the system CPU alone it would take several seconds per frame.

Graphic sequences usually require a number of elements to move on screen simultaneously. In a post production suite this is the result of a coordinated digital effects operation requiring multi-channel effects, multi-passes or both. Such sequences are difficult to set up and can be time consuming to execute. The digital 4:2:2 processing will ensure there is no generation loss, but the speed will be unable to meet the on-air demands.

A solution to this problem is to multi-layer within the graphics workstation itself, using keyframe effects to define the move and the dedicated hardware to process the images frame by frame. With such dedicated processing available the sequence will be rendered in near real-time, fast enough for the demands of live production. Should the individual elements of the sequence interact with each other then very careful planning of the layer moves will be required. To some extent this reflects the working practice of the linear edit suite performing multi-passes. Overcoming this restriction would speed up the overall production even further. The various graphic elements will be stored in the library of the system, although the access time of the library disc is in the order of 1/2 a second retrieving multiple elements each frame would slow the processing down. If however the elements were transferred to a real-time store access would be instant and the composition time would be determined by the processing speed of each element. Again using 4M Bit RAM chips as the store such a solution has been used, reducing the production time still further.

On Air Transmission

The graphic sequences may be composed around live footage, possibly incorporating live elements within the sequence. The system must therefore be able to control a VTR to transfer this live footage to the real-time RAM store. The finished sequence may simply be layed back to tape for subsequent editing with other programme material in an edit suite. Or the sequence may need to be edited directly into the transmission tape. The workstation will therefore require frame accurate editing abilities. The main function of the workstation is the composition of dynamic graphic sequences, there seems little point therefore of incorporating full edit control protocol into the operating software for the sake of simple insert edits. Fortunately VTR manufactures incorporate the protocol within their machines, so the workstation can act as the slave machine for the edit replaying the sequence on cue from the VTR.

For the most pressured on-air situations the workstation may have to replay live to air. Under these circumstances the real-time RAM store becomes the transmission output of the workstation operating from external cues.

Summary

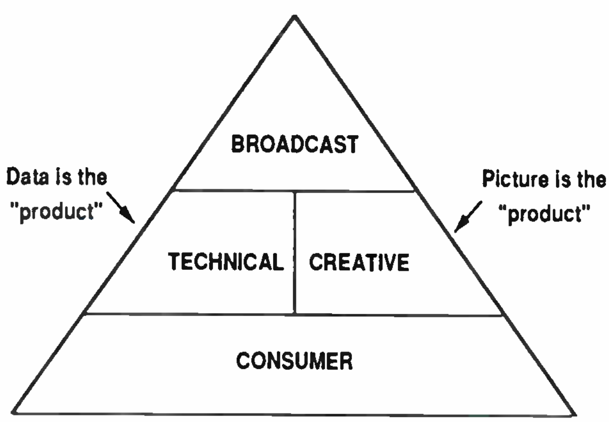
Dynamic graphic sequences are finding applications throughout broadcast and post production. Before the advent of digital composition techniques such sequences were unavailable to live productions. The development of dynamic RAM storage and the use of Application Specific Integrated Circuits have enabled the design of a self contained workstation. The flexible operating structure and dedicated hardware enables the creation of dynamic graphic sequences within the tight timescales demanded by the on-air transmission environment, and within the financial budget of todays productions.

BRIDGING COMPUTER GRAPHICS AND HIGH QUALITY VIDEO

Danielle Forsyth
Tektronix, Video Products Operation
Wilsonville, Oregon

The worlds of high performance computer graphics and high quality video have long been separate; dominated by different vendors and used by professionals who specialize in one of these complex technologies. The standards in each area reflect the historical distance between these two visual mediums. But, convergence between graphics and video is underway; computer users and consumers are forcing these two technologies together.

Computer users have a broad range of video requirements. These needs are best understood when segmented into the following applications:



While computers are moving into the Broadcast/Production segment, dedicated, single purpose controllers continue to dominate this on-line, performance sensitive market segment. Computers are used extensively for off-line animation and image manipulation and have been responsible for some of the most high impact advertising and films ever seen.

The needs for High Quality video in the technical market vary widely. In most applications, this market is *Data Driven*—the quality of the picture will be sacrificed to show the information more accurately.

The Creative/Production market consists of corporate communication professionals, advertising agencies and small production facilities. For these creative users, the picture is the product and picture quality is critical.

Office automation applications are not traditionally quality sensitive. Price sensitivity is high and picture quality will be sacrificed in order to communicate more information (ie. teleconferencing). In the future, this capability will be built directly in to all computing systems.

This paper focuses on the bridging of computer graphics and high quality video. Since high quality video is the focus, emphasis is on the Broadcast and Corporate/Production user.

THE TWO WORLDS

In order to discuss possible solutions to this integration dilemma, it is necessary to look at where graphics and video are today, what is causing the convergence and technological advances which will enable integration in the future.

Computer graphics is the creation, storage, and manipulation of models and pictures in a computer. Color graphics computers have been around for several decades but their popularity soared while prices declined in the mid-1980s. Computer graphics is very different than video. It is completely digital, varies in resolution—from 640x480 to 1280x1024—sometimes even higher, is non-interlaced, does not have specifications for motion (although most computer graphics systems have scan rates between 60 and 72 hz) and comes in a wide range of file formats.

High quality video is not clearly defined. In general, it means that the picture survives post-production, duplication and distribution and adheres to the required video standards. The formats vary widely and most facilities use a range of different format equipment. Signal/noise ratio and frequency response provide the quantitative evaluation of the video signal but the real test is considerably more subjective; does the picture look good?

With all of these differences, a new common standard is the obvious answer. But computer users and consumers can't wait for HD-TV. The demand for graphics and video is being driven by the market. The insatiable appetite of consumers for visual stimulation and the obvious benefits of high impact visual communication for promotional purposes and training are driving vendors to integrate these two technologies. There are other factors which have accelerated this process and forced computer and video companies to work with new users and new system requirements:

• The availabilty of easy to use, low cost computer based software applications for authoring, editing, character generation, 2D paint and animation have transformed low cost desktop computers into integrated production systems.

• Competition and technological advances have rapidly decreased prices of computers, storage, input and output devices and allowed a much broader range of users to have access to this computing systems.

The new non-technical users demand access to all data types—graphics, video, audio and text and require that their productions be output on a high quality, universal, dynamic medium: VIDEO. Some applications are developed specifically for distribution on computer (ie. kiosks). This works well for interactive presentations where the cost of the computer can be shared by a large number of users.

The challenges of integrating this wide range of hardware graphics and video formats is formidable but the integration goes far beyond the hardware. True integration of graphics and video requires integration at all levels—hardware, system software, application software and input and output devices.

The Standards—Computer Graphics Systems

Although there are a lot of standards in the computer world, few are universal. Generally, the process is for companies to form consortiums, consortiums to develop standards and standards to be proposed to a wide range of standards organizations. Computer companies then choose which standards to adhere to and which standards to enhance. A subset of computer standards are shown below:

- Operating System (DOS, UNIX,..)
- User Interface (Presentation Manager, Windows, Motif,..)
- Graphical Programming Interface (GL, PHIGS,..)
- I/O Bus (AT(ISA), EISA, Nu-bus, VME,..)

The Standards—Video Systems

The standards for video have evolved over a much longer period of time and most are well documented. New tape formats continue to be added and the move to digital video is evident. At present, the cost of digital video is prohitive for many facilities and the distribution system is still analog. Because of this, most computer users want to produce component or composite analog output from their computer graphics system.

Although graphics and video are converging (from user's perspective), they remain miles apart technologically and will not be integrated until a new, separate standard is universally adopted.

There is a wide array of enabling technology which is being developed to allow diverse data types to be integrated on the desktop computer. Several video chips have appeared on the market which will enable computer companies to inexpensively add video capability to the computer, but high quality video still remains a challenge. Rapid progress in compression has enabled users to have access to video on their computer while still using existing storage and networking systems. New storage and networking standards are gaining acceptance for storage of these new data types.

BRIDGING GRAPHICS AND HIGH QUALITY VIDEO

There are a number of different ways to get video into a computer and out of a computer. Each has its strengths and weaknesses. The primary areas which need to be evaluated in any graphics/video system are:

- Video Signals Supported
- Signal Quality
- Price
- Performance—hardware and software
- Computer Platform
- Applications Supported
- Level of Integration

The goal of production facilities is generally to produce the highest quality imagery as quickly as possible. Signal quality, performance and integration in the computing system are essential.

The ideal "bridge" :

- Provides the user with the necessary video signals

  All required video formats should be supported. Ideally, the image does not have to be rendered for a particular format. This way, if the output format changes, the recording time is minimized. If the video image is imported into the computer, the "translation" should be fast and integrated.

- Delivers maximum signal quality for all video formats

  For each required video format, the quality needs to be perfect so that the imagery does not have to be returned. Since many production professionals do not have an in-house engineering staff, quality must be assured by the graphics/ video integration product.

- Meets the required price point

  This may determine everything else. The price, quality and performance tradeoffs need to be carefully weighed since the quality output of any production facility is usually the final measure.

- Provides required overall system performance

  Performance can be measured in many ways. The total time for the animation or authoring development is probably the ideal measure but the number of variables is enormous. It is important that the system be evaluated as a system with equal emphasis on all areas of development:

Storyboard
Model
Paint
Composite
Choreograph
Render
Record

- Runs on (or works with) the chosen computer platform

  There are a wide range of platforms available for production applications. Most popular are the PC, Mac and Amiga. Silicon Graphics dominates the high end of the 3D graphics market. The platform decision is usually based on the application, capability of the system and price.

- Integrates with Application Software and overall system

  Not only does the application need to run on your chosen platform, the video system needs to understand or translate the file format which it produces. There are no real standards for file formats and the translation process takes time. Tools and utilities to test and "hide" the video make it easier to use for non-technical users.

There are three general types of video systems available for computer graphics systems:

1. Low cost, industrial quality systems

2. Scan converters

3. Frame accurate video systems

Each of these will be described during the presentation.

FOLD IT OR FIX IT: THE CHANGING FACE OF SPECIAL EFFECTS

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Abstract — During the past ten years, special effects systems have evolved from being video image positioners for the elite to multi-function 3D picture benders available to a wide range of users over a wide range of prices, capabilities, and output qualities. As they've evolved, the designer's dilemma between answering the demand for new effects and fresh new looks - and the desire to maintain transparent image quality has intensified. A comparison of various post production applications and their demands on special effects systems design will be discussed.

BACKGROUND

To better understand where we are and where we're going with respect to the usage and utility of special effects systems, a short historical perspective on how these units have evolved is always useful.

Last year, ADO® celebrated its tenth anniversary. At the risk of sounding immodest, I think it's fair to say that since its introduction in 1981, ADO has been a significant player in the industry, setting the standard for digital video effects at both the high-end and low-end of facilities' business.

The ADO introduced excellent video quality and transparency to the incoming video signal. Through a wide variety of image manipulations, the quality of the output is maintained very close to that of the input, allowing these devices to become an integral part of practically all post production projects. At the time of its introduction, the ADO's smoothness of motion when moving an image from one side of the screen to the other and transparent video quality immediately attracted the attention of post product firms and broadcasters.

The ADO allowed users to change the geometry of the incoming video picture, make it bigger, make it smaller, and move it around in 3D space. These capabilities sharply contrasted with those of the equipment on the market at that time, the Vital Squeeze-Zoom and the Quantel DPE-5000. These units, both in use in the early '80's, were low-quality, limited resolution 2-D devices used primarily by the networks to shrink the picture down or reposition it.

The original ADO, now known as the ADO 3000, was a state-of-the-art high-end device designed for larger post facilities and the networks. A broadcast version, the ADO 2000, was introduced four years later. In 1986 the ADO 1000, the first field-based version, arrived. Since then, mid-range and entry-level systems have been introduced, and prices have fallen dramatically, with the point of entry being as low as $20,000.

In the early '80's, digital video effects devices were perceived as a creative tool to add visual interest to a variety of productions. However, system owners quickly realized these digital effects systems had a tremendous potential to correct a broad range of errors in source video that would otherwise require re-shooting at considerable expense. (I refer to such situations as expanding a frame of video to eliminate a boom mike, fixing a skewed camera angle, or even the seamless replacement of one entire portion of video with another.) The acceptability of such "fix-it" capabilities depends on their degree of transparency; as such, transparent video signal quality has become a key design consideration in most of today's special effects systems. Viewers must never guess that the video they're watching has been manipulated in any way.

FIX-IT VERSUS CREATIVE APPLICATIONS

Today, the application of digital effects systems is almost evenly split between visible creative effects and invisible fix-it operations. However, invisible fix-it operations demand much higher quality than momentary special effects. In order to make their systems affordable, manufacturers have been forced to optimize their system architectures for either a broad range of visual effects or transparent video quality. Before I discuss the evolution of technologies that enhance image quality, I want to expand upon the differences between fix-it and creative applications.

Simply put, Special Effects Units (SEUs) allow the user to change the shape and position of input video, either for the purpose of correcting some problem with the source video (fix-it) or for creating a visually appealing transition or scene (creative application).

For instance, one capability common to all SEUs is rotating or "spinning" the input video in a 2D plane. This feature is helpful both in creating creative effects and fixing source video. On the creative side, we've all seen a line of text enter the screen spinning rapidly, then stop to communicate some message. The spinning motion attracts our attention, so that we'll focus on the words when they finally "land". In this case the source video — the words — comes from a character generator with the SEU applying the rotation.

Fix-it

On the fix-it side, imagine that the input scene is an interview with a company's CEO. After taping a 20-minute speech, we review the tape before distributing it to 50,000 employees, and notice that the blinds behind the CEO are not parallel to the top edge of the TV screen, something not noticeable during the shoot, but distracting during final viewing. In this case, rather than re-shoot the speech, the entire scene can be rotated by a SEU, thus using the "spin" feature as a fix-it application.

Fix-it applications are also necessary when unwanted video paraphernalia appear in the scene—an overhead microphone, a book lying on a table, and so on. By slightly expanding the video, the unwanted items can be placed outside the visible screen area.

Let's say that during our CEO's speech, a fly is noticed on the lapel of his sport coat. Obviously, this undesirable bug in the middle of the scene requires video patching. The SEU can be used to crop a piece of cloth from the suit, and place it over the insect, fixing the entire scene without a reshoot.

Another case of "fixing" is the case of a TV movie that is supposed to take place in Washington. During screening, we notice in a close-up that the phone number on the dial of a pay-phone has the Los Angeles area code. The correct number can't just be pasted on with a paint system, because the dial spins while the actor is dialing the number. The new number can be passed to the SEU, which can precisely size and locate it over the Los Angeles number and can realistically simulate the exact speed and rotation of the telephone dial.

Creative

On the creative side, the most popular effect is the over-the-shoulder graphic during a news broadcast. Although the graphic is usually created with a paint system and stored in a still-store, the final on-air positioning is left to the SEU. Another popular creative application is positioning words from a character generator. Examples are the words "All New" at an angle in the corner of the screen, spinning transitions, or an entire slate "flying away" to bring up the next set of scores.

Other common creative effects include push-on, mosaics for the secret witness use, and page turns.

Quality Impact on Fix-It

One critical qualitative difference between creative and fix-it applications is how long they are actually seen by the viewer. Creative effects sizzle onto the screen and typically exit quickly; on the other hand, most fix-it jobs must satisfy viewers' scrutiny for extended periods of time.

Earlier, I used the word "transparency" in association with SEU quality. Normally, this means the degree of opacity of an image. But in the case of SEU quality measure, it means how close in quality the transformed video is to the original input video.

Let's use the CEO with the crooked blinds as an example. If we could actually go back in time and rotate our camera by the one or two degrees necessary to straighten the picture, the actual quality of the new video, set side by side with the old video, would be exactly equal to the old video. The fact that the first take was incorrect would obviously be "totally transparent" to the viewer. Now, if we set the new video as rotated by our SEU next to the same old version, how will it compare? If the unit is totally transparent, it will look the same as the rotated camera. But what if we notice jagged edges on the blinds, introduced by poor interpolation or anti-aliasing on the part of the SEU when rotating? Transparency has been lost, the blinds aren't realistic, and the fix can't be accomplished.

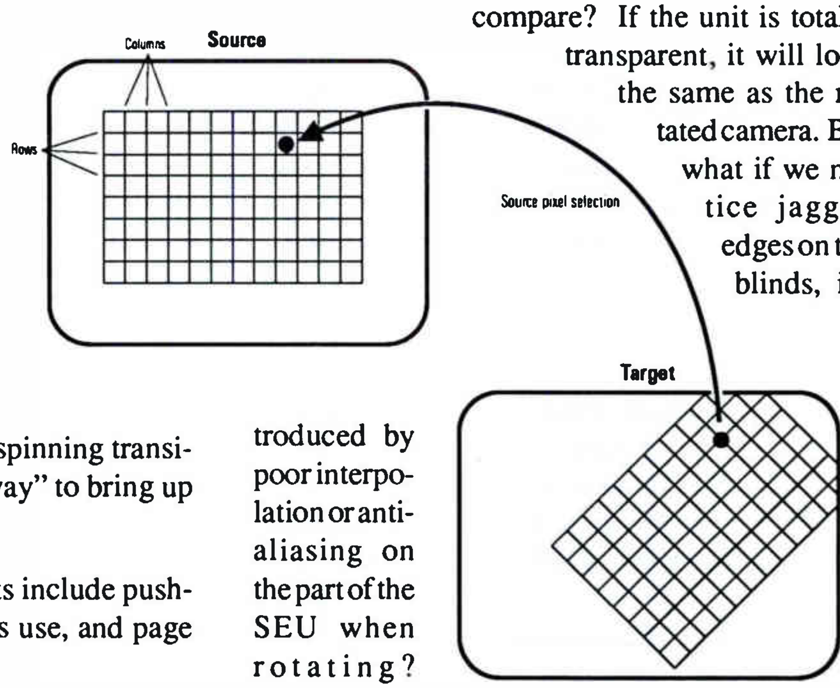

Figure 1

THEORY OF SEPARABLE ARCHITECTURE

Digital effects devices employ a number of methods for maintaining transparent signal quality. Among these is the separate handling of vertical and horizontal filtering. This approach, called "separable architecture," provides greater image clarity when dealing with a wide range of frequently-used effects, such as rotation and resizing. This is the architecture on which the Ampex ADO family of special effects systems is based.

There are several factors that contribute to the "transparency" of special effects devices, perhaps the most important being the image "filtering" approach used in the system architecture. Filtering is the process of using the luminance and chrominance values of the input or "source" image and attempting to calculate the appropriate corresponding values for the transformed output or "target" image. (Source and Target are terms coined by Ampex to describe the pre-transformed video image as it enters the SEU, and the post-transformed image, respectively. Examples of source and target images are shown in figure 1.)

The Challenge

As stated earlier, the goal of transparent processing is for the relation of the target image to the source image to remain as close to the "real life" metaphor as possible, i.e., the rotated image mimics a rotated camera, a reduced image mimics a zoom out, etc. The challenge, of course, is that the SEU is one step removed from the real life process when compared to the camera. The camera is always scanning an infinite resolution image (real life), while the input to the SEU is divided into a finite number of rows (scan lines) and columns (pixel timing). Referring to figures 1 and 2, it's apparent that for any single pixel in the target image, there probably won't be a precisely corresponding pixel in the source image. A transformation such as rotation almost guarantees that trigonometric calculations will yield an answer that is between pixel locations in the source image.

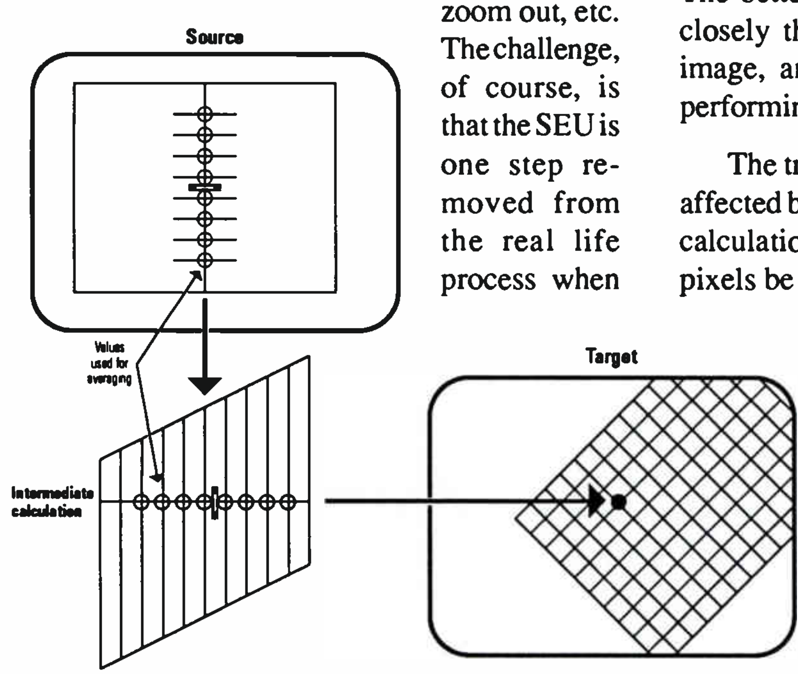

Figure 2

The approach to determining an in-between pixel and establishing its value (filtering) can be very simple or very sophisticated. The simplest approach is just to pick the nearest pixel (appropriately called "pixel-picking"), which results in a picture that has been visibly distorted and relies on the viewer's visual sensory system to reconstruct detail within the image. More sophisticated systems try to create more pixels in the source image by dividing up the space between input pixels and trying to assign values to these new locations by averaging the values of the surrounding pixels — a process called subpixel averaging. The better the reconstruction process, the more closely the source image will be to a real life image, and the better chance the SEU has of performing like a camera.

The transparency through the SEU is directly affected by the "magnitude" (more is better) of the calculations (how finely can the space between pixels be divided; how many neighboring pixels can be accessed to determine the value of the new target pixel). The calculations, in turn, are limited by the built-in constraint—for a real-time system— that they all must be completed within the interval between successive video fields.

Separable Architecture

One approach to geometrically increasing the number of calculations while only linearly increasing system complexity—and product costs—is Ampex's method of processing horizontal and vertical picture information as two separate processes, which we call separable architecture. This approach yields high quality at a reasonable implementation cost because, at a processing cost of only 16 neighboring pixels, the information content of 64 neighboring pixels is obtained. As figures 2 and 3 show, column pixels are calculated using eight neighboring pixels, which are then used as the input for a similar calculation on the "row" pixels, yielding the equivalent of 8 x 8 = 64 pixels worth of information. To provide the same magnitude of calculations in a non-separable device, the information from 64 neighboring pixels would have to be considered to calculate each resultant pixel in the target image. The cost to accomplish this is very high or, alternatively, each calculation must be mathematically trivial. Neither is an acceptable solution.

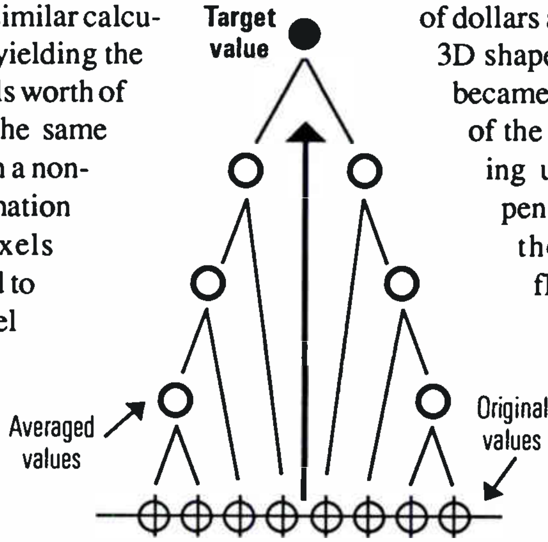

Figure 3

An obvious question would be that, if this is such an advantageous approach to high quality, why doesn't every professional SEU use it? Well, one reason is that separable architecture is at the core of several Ampex patents. Another is that this approach requires that a complex 3D transformation be broken down into two intermediate 2D transformations, with the constraint of only looking at rows in one calculation and columns in the other. While this is very complex for transformations involving perspective and warping, it's even more involved where overlapping pixels are required, such as in page turns or other true 3D shapes.

THE CURVILINEAR BREAKTHROUGH

Since the early 1980s, manufacturers have dabbled in effects where the surface of the video raster assumed the shape of curved surfaces rather than freely oriented planar surfaces. The first of these, the Quantel Mirage, allowed creation of almost any shape, with live video mapped to the surface, but at a very high cost both in terms of dollars and ease of use. The creation of 3D shapes for use in video applications became the almost exclusive purview of the 3D animation systems, providing unlimited flexibility at the expense of non-realtime creation. Although time-consuming, the flexibility of the 3D systems prevailed. In addition, curvilinear 3D shapes weren't typically requested for scene transitions, so the requirement to map different live video sources to the same shape over and over again never really materialized. The real-time special effects systems answered the demand for mapping video on 3D shapes by providing features for creating shapes made up of flat sides—such as cubes or polyhedrons—all of which can be accomplished with standard 3D, flat plane perspective, utilizing clever masking techniques, multiple special effects units' channels, and multiple passes through the effects device.

The one curvilinear effect that emerged as a heavily-requested transition was the 3D page turn. Because the turning of a page to reveal new information is a metaphor readily accepted by the viewer, using it as a transition became popular immediately. In the short term, Ampex ADO users accommodated requests for this effect by using carefully calculated warp parameters or combined switcher wipes and ADO effects. While both approaches satisfied a large portion of the demand, their implementation tied up extra hardware, and control of the effect was number intensive and limited.

Software versus Hardware

Once an application-specific solution became necessary—one that made effect programming easy and yielded consistently believable results—it became apparent that the separable architecture of ADO prohibited either a direct software-only implementation, or a general purpose "curved-surfaces" hardware module. The calculations to determine the intermediate, one-dimensional solutions (see figure 4) to problems involving curved and hidden surfaces proved too time-consuming for a software implementation, and too "case sensitive" for a general purpose hardware solution. While a general purpose solution was possible with a non-separable architecture, the implementation that maintained ADO video transparency levels was deemed too costly for most of the target users.

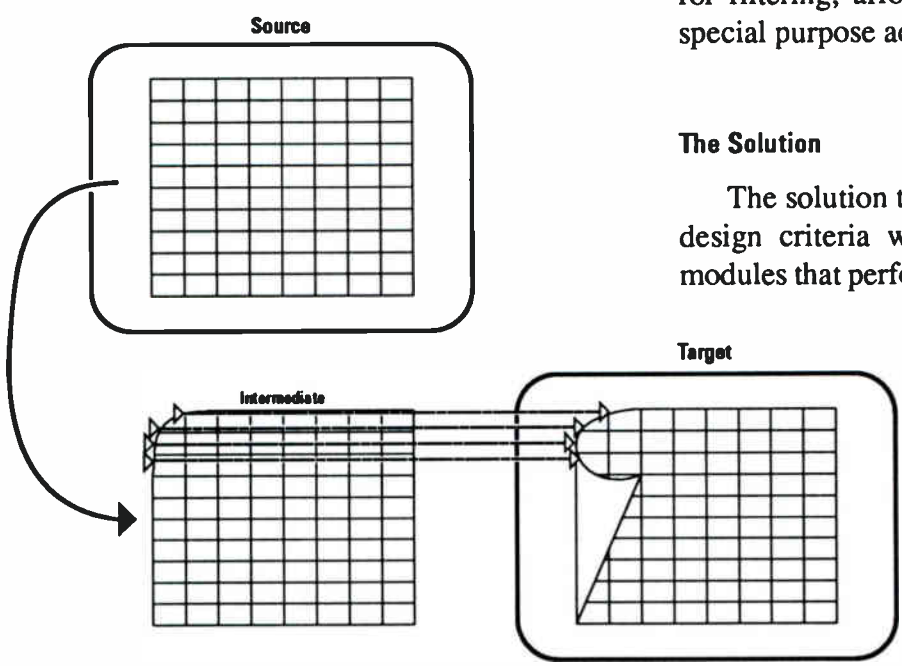

Figure 4

Within a given target price range, the most difficult decisions in SEU design revolve around the filter/memory/address generator core. The other major functional blocks of an SEU - analog input/output, user console, and frame stores - are extremely important to the total utility of the system, but are relatively easy to modify if user feedback demands it. The core of the system however, is very difficult to change and dictates the quality level and extent of effects possible with that generation of SEU. The filter/memory/address generator core may be partitioned in many different ways. For example, one approach might emphasize a flexible, general purpose address generator module, which would facilitate many different effects and shapes, at the expense of the filter size, which would limit overall picture quality. Another approach, using separable architecture to facilitate very large pixel neighborhoods for filtering, affords additional hardware for a special purpose address generator.

The Solution

The solution that met the greatest number of design criteria was special purpose hardware modules that performed specific page turn geometry tasks within the constraints of a separable architecture machine. Using this approach, dedicated page turn boards interface directly to the standard row/column processing hardware to both perform the geometric 3D surface calculations for the shape of the page and modify the horizontal pixel contents based on hidden surface computations.

This approach to curvilinear effects yields an application-specific page turn capability with the quality benefits of separable architecture. In addition, since the page turn calculation process is independent of the normal scale/rotate process, the "page" can be treated as a true 3D object that can be rotated in 3D space. This means that the page can be tilted back and rotated back-to-front while being turned. Also, since the front page/back page processing is part of a single calculation step, the mirrored back of the page is calculated with the front of the page in a single pass with a single channel, making operations easier and more cost-effective.

While the processing workload for the general case of 3D curved surfaces is too intense for efficient implementation in today's separable architecture environment, recent advances in transputer and DSP technology are likely to provide acceptable general purpose platforms for ADO real-time, computer-intensive operations.

FUTURE EFFECTS

A related advance I'm already seeing evidence of is computer products facilitating overall, cost-effective improvements in SEU performance. For instance, ADO already uses Ethernet to communicate between control panels and signal systems, allowing more flexible system configurations. Off-the-shelf floppy drives are allowing efficient effects storage. Compact, affordable memory permits increased on-air effects. Faster CPUs allow more parameters to be processed within the one-frame real-time constraint.

I also anticipate more powerful general purpose processors facilitating further cost reductions in high-quality SEUs. Although general purpose processors are used in today's SEUs for user interface and geometric calculations, special purpose, parallel processing is still required to obtain the 1200 mips required for real-time address generation in a high-quality environment. As general purpose processors cost effectively reach the 1200 mips threshold, new effect types will be software-based, with hardware power and expense adding complexity rather than specific capabilities.

Eventually, entire graphic/video general purpose workstations will be able to handle the special effects processing workload and allow the special effects application to be a software-only solution.

I have welcomed this opportunity to address this audience and explore the application of digital effects devices and the technologies they deploy. I have highlighted the concepts of separable architecture as a means of preserving transparent image quality, as well as the most recent advances that allow the incorporation of curvilinear effects to satisfy both creative and quality requirements.

MOBILE UNIT ONE. . . FIRST STOP: 1992 WINTER OLYMPICS

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Abstract- The 1992 Winter Olympics in Albertville, France provided an ideal backdrop for CBS to utilize its newest mobile unit...MU-1 as a core element in the coverage of this event. From its inception, MU-1 was designed to meet its Olympics requirements as one of the studio control rooms in the International Broadcast Center as well as providing CBS with a state-of-the-art remote production vehicle after the Olympic use. This vehicle represented a major departure from past CBS mobile units: in its design approach, cost constraints, construction and size. The resulting vehicle was so different from the past that it mandated the numbering for the CBS fleet to be reinitialized. This paper describes the design goals of the project and their implementation, the project management techniques used to insure meeting the delivery and cost targets, and the structural aspects of the trailer design required to meet federal and state highway weight and size regulations.

INTRODUCTION

The planning for the 1992 Winter Olympics called for the use of two mobile units as studio control rooms in the International Broadcast Center. Studies showed that the use of mobile units, as compared with building two complete control rooms on-site, would be cost effective. For economic reasons it was decided that we would use one of our existing units for one of the rooms. Since CBS's capital budget plan called for a replacement of our oldest in-service mobile unit in 1992, it became apparent that by accelerating this project we could satisfy the Olympics requirements and also replace our 1974 vintage unit in a cost effective way. This existing mobile unit contained several items of equipment that were relatively new, but the unit had been on the road for many years, and its body and chassis were exhibiting considerable wear and tear due to over-the-road use. The physical size of this unit also precluded any expansion of production facilities. With this background in mind, we set out to accomplish this project with unique and new approaches from the budgeting process through the design and implementation. This paper is not intended as an in-depth technical discussion of mobile unit design, but rather to share some of the special problems that were encountered in implementing this project. It is hoped that the reader may derive some benefit from CBS's experiences.

BUDGET AND COST RESTRAINTS

A project of this magnitude requires the expenditure of several million dollars. With the present economics of the broadcast industry, commitment by the company for such a sum is not made without a very strong justification. A project manager responsible for preparing a budget of this magnitude must do his homework in assembling both the technical and financial data. At CBS we took the approach that this new mobile unit was not needed just for the Olympic control room, but rather to replace an existing mobile unit at the most economical cost and to use this unit at the Olympics to offset the nonrecoverable cost associated with building an on-site room. To justify the economic replacement of our oldest in-service mobile unit, (MU-6), we established the ground rules shown in Table 1.

Table 1. MU-1 Budgeting Ground Rules

1. MU-6 would have to remain in service until the Olympics coverage was complete. This meant that we would not incur any cost associated with renting a replacement unit during the Olympics.

2. We would reuse as much equipment as practical. While MU-6 was 16 years old, not all of its equipment was of this vintage. Over the life of this unit several updates and replacement of equipment had occurred.

3. We would not replace the B unit. CBS mobile units are actually a pair of units: a production unit designated as the A unit and a utility or B unit. Since this unit was not a production unit, it was felt that with prudent maintenance we could keep it in service.

4. We would not replace the towing tractors since we already had a replacement program for our tractor fleet and the current MU-6 tractors had a few more years of service left.

Ground rules 3 and 4 immediately realized us a savings of several hundred thousand dollars. Rules 1 and 2 seemed to be diametrically opposed since we certainly couldn't reuse equipment from the old unit while still keeping it in service during the Olympic coverage. However, the Olympic project was leasing some equipment instead of purchasing, and we adopted this approach to the mobile unit project where practical. We also approached ground rule 2 with the understanding that we would only reuse equipment that was less than two years old. We started our budgeting by first putting together a list of capital items required to build a new unit. Items that fell within ground rule 2 were identified and listed as no charge items. The items in this category are shown in Table 2.

Table 2. Reused Equipment From MU-6

1. A two channel DVE
2. Eight cameras and lenses
3. Two frame synchronizers
4. Two D-2 VTR's
5. Three 1/2 inch Beta VTR's
6. A time code generator
7. Two sync generators and an autochange unit
8. One color corrector
9. Two 450 MHz radios

This was an impressive list of equipment and it represented a reduction of over a million dollars in our capital budget. It was planned to install and test all system wiring for these devices in the new mobile unit; therefore, it would be a simple matter to provide and install rental or leased units where these items were required for Olympic use. Only the DVE and cameras were required for Olympics use, and leasing arrangements for this equipment was coordinated with the Olympics project. The mobile unit's video tape system and RF radios were not required to be operational for Olympics use; therefore, this equipment would be installed after that event. The mobile unit did not require a master sync generator system for Olympics, since it would be slaved to the plant system; therefore, a surplus slave generator was temporarily installed. After the Olympics it would also be a relatively easy retro-fit of the MU-6 equipment into the new mobile unit since all system wiring would already be in place. The final budget was a joint effort of engineering, operations, production and financial personnel that clearly spelled out the cost involved, the prudent reuse of existing equipment and the savings that could be realized by undertaking this project in conjunction with the Olympics project.

DESIGN GOALS

This project was undertaken with the stated goal of using unique and new approaches to address financial as well as technical issues. As was outlined in the previous section, some unique approaches were used in the budgeting and justification of the project. In establishing the parameters for the technical design goals, there were several factors that necessitated the same approach. Some of the design factors that were established are as listed in Table 3.

Table 3. Special Design Factors for MU-1

1. The vehicle design had to be maximized to accommodate additional equipment and personnel while still conforming to Federal and State size and weight restrictions.

2. The floor plan layout had to be optimized for the most efficient production utilization.

3. The mobile unit would be parked inside the International Broadcast Center building during the Olympics; therefore, the environmental system had to operate without adding noise and excessive heat load to the building.

4. All equipment had to operate at both 50 and 60 hertz.

5. The ac power system would be limited to a 200 amp service.

6. The electronics design needed to be state-of-the-art; however, the equipment selected had to be readily available so as not to jeopardize the Olympic schedule.

7. The mobile unit schedule had to coincide with the Olympic construction schedule, which required the vehicles to be parked inside the Broadcast Center in France no later than mid-October of 1991.

The following sections highlight these factors and describe the CBS techniques that were used to satisfy them.

THE VEHICLE DESIGN

While most audio, video, and communications requirements present few problems to the skilled broadcast engineer, his experience does not generally encompass the special knowledge associated with the complex design of a large mobile unit vehicle. The vehicle design is probably the hardest part of a mobile unit project since it involves new disciplines shown in Table 4 that are interrelated with each other and that also affect the electronics design. There are companies that specialize in some or all of these areas, and their expertise should be utilized; however, the project manager must acquire a good understanding and knowledge of how each of these items will affect the design.

Table 4. Vehicle Design Disciplines

1. Vehicle layout and design
2. Federal highway weight and size regulations
3. Vehicle weight distribution
4. Structural analysis
5. Environmental systems
6. AC power distribution

VEHICLE LAYOUT AND DESIGN

The new MU-1 represents the fourth generation of expanding side mobile units for CBS. Our first expanding unit was built in 1965 on a self-contained 35 foot long International Harvester chassis. The body of

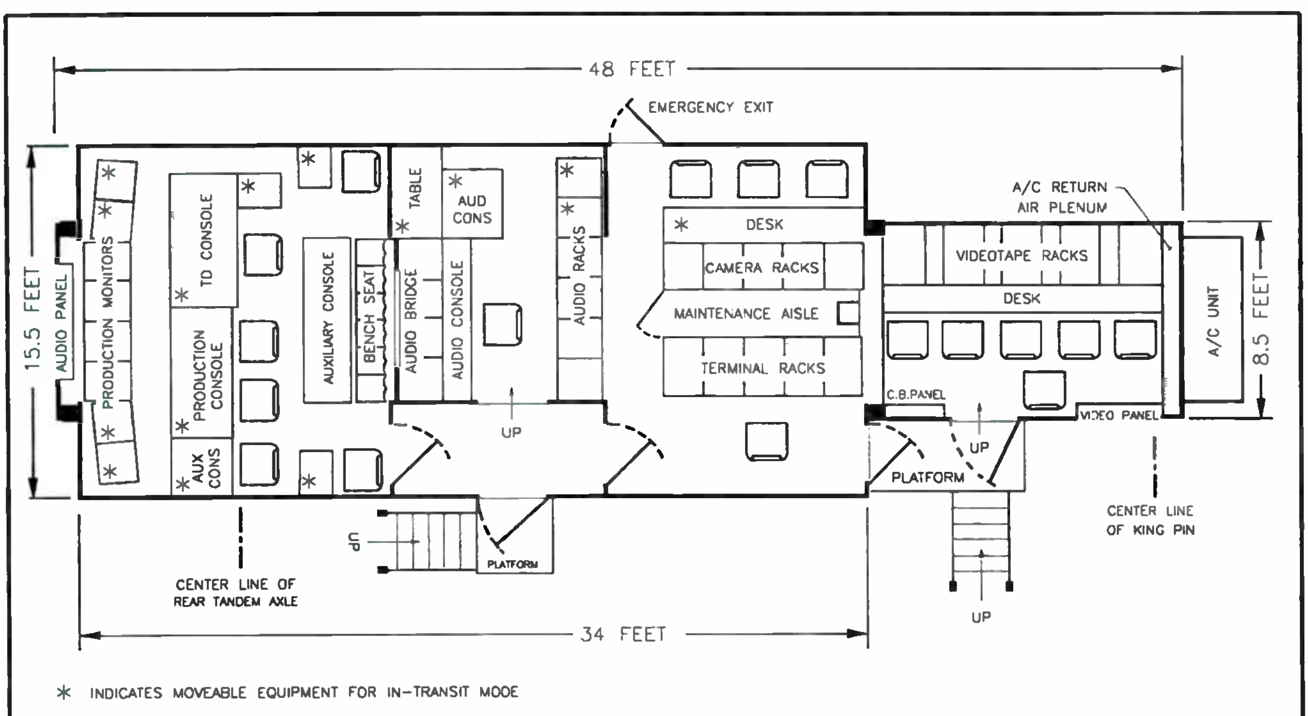

Figure 1. MU-1 Operational Floor Plan Layout

this unit was 28 feet long and the production area walls of this vehicle expanded out 18 inches on each side. Four units of this design were built and were numbered MU-1, 2, 3 and 4. These units were retired from service in the late 1970's. In 1974 CBS built its first of three expanding side 40 foot long trailers. This design also contained a double expandable wall in the production area, which increased the production area width from 8 feet to 12 feet. These three mobile units were numbered as MU-6, 7 and 8. In 1983 CBS commissioned two new 45 foot long expandable units which had one long and one short expanding side. These units, MU-11 and 12 had an increased width of 14 feet in the production area as well as personnel operating space in the video tape area provided by the longer expanding wall on the roadside of the vehicle. With this prior background of vehicle design experience at CBS, it was not difficult to undertake the design task for this project. The new CBS unit shown in Figure 1 is a 48 foot trailer with two long expandable sides. This increased the width of the expanded area to 15.5 feet and resulted in a significant increase in interior floor space for both equipment and personnel. This new unit was such a dramatic improvement over our past designs that we decided to reinitialize our numbering sequence for mobile units, and begin again with MU-1.

FEDERAL HIGHWAY SIZE REGULATIONS

With the start of the design for MU-1, we were sure of one thing, and that was the new design would use the expanding side concept. Meetings were held with both our technical staff and production personnel to gather their input concerning the layout as well as technical equipment needs. Three major requirements were universally voiced at these meetings: more production space, more space in the audio area and more technical equipment. A mobile unit layout can have a variety of relationships with regard to operating area positions. One thing that has remained constant in the CBS mobile unit designs since 1965 has been the relationship of the production area and the audio room. What works for CBS is a layout with the audio room directly behind production. There are other options for this relationship, but what is important is that your design must make your production people comfortable with their working environment. Since our personnel were comfortable with this arrangement, we used this as the start of our layout design. Figure 1 shows the layout as it finally evolved. This layout was the result of many meetings, and it illustrates the best compromise of many diverging wishes and opinions.

The layout requires some explanation as to how design goals were met and modified. As can be seen in the layout, the overall size of the vehicle was maximized to get as much equipment into the space as was possible. While federal highway laws allow longer trailers, it was determined that a 48 foot trailer was our practical limit due to access to various sites we travel to. We also limited the height to 13 feet due to the same access limitations. We effectively created a larger trailer by externally mounting the air conditioning unit. This was done after carefully checking swing clearances on our existing tractors and it is something that must also be considered when replacing a tractor in the future. We also took advantage of the 8.5 foot maximum width allowed under the federal regulations. As can be seen in the layout, two long expanding sides that expand out 3.5 feet on each side give a maximum amount of interior floor space.

The added interior space was used to increase the production area facilities of our previous design by adding two additional monitoring racks and a second production console behind the main production console.

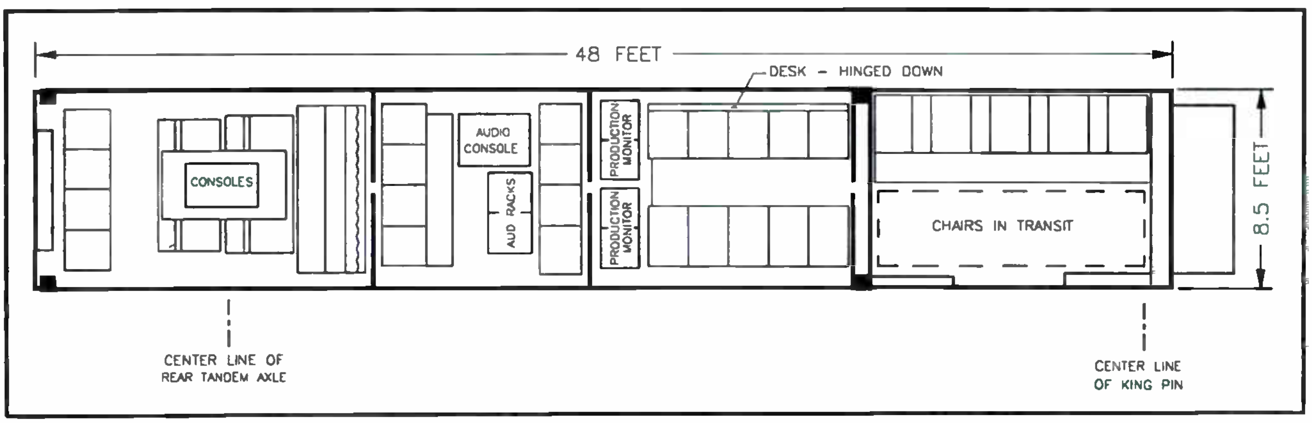

Figure 2. MU-1 In-Transit Floor Plan Layout

Figure 3. Curbside View of MU-1

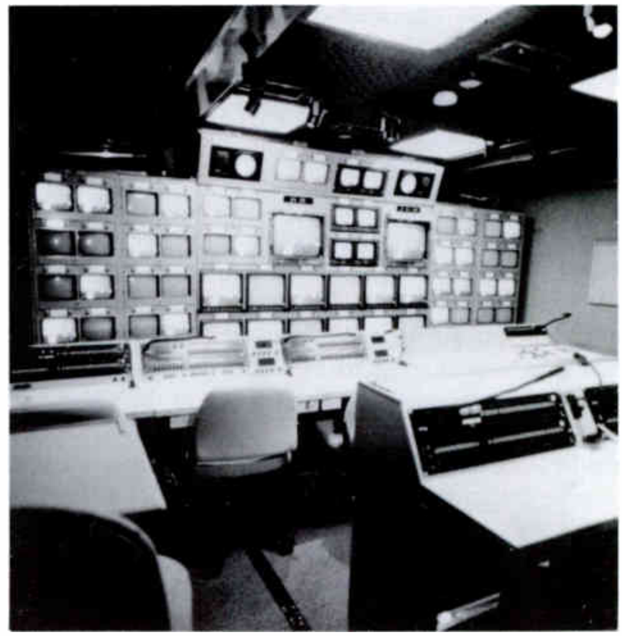

Figure 4. View of Expanded Production Area

We also added two removable wall mounted desks and seating for two additional production personnel that can be added for larger remotes. Additional space was created in the audio room by the use of the long expanding side. The long expanding side also provides space for a vestibule at the main entry, which isolates the operating areas from the entry traffic. The added floor space created by the expanding sides also provides operating space in the terminal and video areas. The video tape area was located in the forward non-expanding portion of the vehicle. Access to the video tape area was made convenient to production personnel since one can walk from tape to the expanded part of the unit via a single step instead of walking up and down the exterior steps. Figure 3 shows an exterior view of MU-1 and Figure 4 is a view of the expanded production area.

WEIGHT DISTRIBUTION

One of the most important elements of mobile unit design is the weight distribution of the equipment contained in the unit. Federal and state highway regulations limit the axle weights that can legally operate on highways and roads. These weight restrictions can be confusing since there are several variants that must be considered along with the axle loads, and these include tire ratings, bridge load calculations, axle spacings, etc. What is important to remember is that weight distribution calculations must be based on the in-transit configuration of the vehicle. Figure 2 shows the in-transit configuration of MU-1, and a comparison with Figure 1, which illustrates the operating configuration, should make it very clear how these two modes would differ as to axle loading because of the different locations of equipment for over-the-road transportation. During the design of MU-1 we paid special attention to weight. During the vehicle design several weight saving techniques were employed during the chassis and body manufacturing, and they are listed in Table 5.

Table 5. MU-1 Weight Reduction Items

1. Low profile tires
2. Aluminum hubs and rims
3. High tensile steel chassis I beams (higher profile but lower weight per running foot)
4. Lightweight honeycomb floor material
5. Aluminum vertical supports
6. Aluminum roof bows
7. Anti-skid coated aluminum sheet roof (in lieu of aluminum tread plate)
8. Aluminum electrical conduit

It was estimated that the Table 5 items resulted in a total savings of 3,000 pounds as compared with a conventional design. During the electronics design a computer program was developed to keep track of all equipment weights in each rack including hardware and cable. Each rack, console, and other hardware weight was automatically entered into an axle weight calculation. We also verified the actual weights of all equipment before installation by actual scale weighing, and this information was used to continuously update our axle loading data base. In this way we were able to make some adjustments in the location of equipment thereby transferring weight from one axle to another.

STRUCTURAL ANALYSIS

The structural analysis of the vehicle went hand in hand with the weight distribution. We were attempting to build the vehicle chassis and body as light as possible without jeopardizing the structural integrity of the vehicle. In order to verify the vehicle manufacturer's design, we contracted the services of an independent structural analysis consultant to perform a computer model analysis of the chassis and body design. As a result of this analysis, he was able to make some recommendations concerning the original design, which corrected some possible marginal structural points. This was a worthwhile investment since it gave us confidence in the integrity of the design.

ENVIRONMENTAL SYSTEMS

CBS opted to purchase a custom built air conditioning system for MU-1 as opposed to a packaged standard unit. This decision was based on our past experience with both types of units. We have used packaged units in the past, but our experience with them is that they are not sufficiently ruggedized to withstand the constant over-the-road environment. Another mitigating reason for opting for a custom unit was our special design goals which required both 50 and 60 hertz operation. This could only be satisfied by a custom unit. We also had the requirement for operating inside a building without adding heat or noise to the surrounding area. This requirement was satisfied by the custom unit, which was built with plumbing and controls for cooling the unit's refrigerant by means of an external heat exchanger, rather than the unit's internal condenser fan. For the Olympics we built a custom water cooled heat exchanger that was connected to the mobile unit's refrigerant system as well as the Broadcast Center's cooling water system. With this type of system the noise producing condenser fan is not running, and there is no heat buildup caused by an air-to-air heat exchange of the condenser coil since this system is completely bypassed. We chose to build the heat exchanger as a separate unit because this mode of operation would only be used occasionally, and we didn't want to add unnecessary weight to the mobile unit. The custom heat exchanger was located on the floor directly below the fifth wheel area of the trailer. It connects to the mobile unit's air conditioner by means of precharged flexible refrigerant hoses. It was also connected to the building's circulating water system by standard plumbing.

Because of the size constraints of a mobile unit, it is difficult to provide a zoned temperature control for each individual area of the unit. Personnel and equipment need different temperatures to satisfy their optimum operating environments. The new MU-1 uses a technique to fulfill this goal, which is a workable compromise. Air conditioned supply air is delivered through a ceiling supply duct, which utilizes the entire width of the fixed portion of the trailer. The equipment air is directed through each equipment rack, which is essentially made into a closed cooling cabinet by the use of panels and doors. The return air is drawn into an under-floor plenum, which mates with a vertical plenum at the front bulkhead of the trailer. The master thermostat is set to satisfy the equipment cooling requirements. The master thermostat uses a capillary type sensor, which is located in the return air plenum approximately 10 feet in front of the air conditioning unit. With this arrangement the equipment air can be controlled to supply air at the desired temperature. This temperature is usually cooler than personnel requirements. Since personnel air is supplied from the same common supply plenum, there is a conflict with regards to air temperature. However, this cooler air is tempered by electric heaters located directly behind the area air supply registers. The heaters are zoned with individual thermostats for five areas of the mobile unit: production, audio, terminal, camera video and video tape. This system acts like a multi-zoned system without the complications of providing a zoned duct system.

AC POWER FREQUENCY

Because of the Olympics use in France the specification for power frequency was a dual standard of 50 and 60 hertz. The voltage for this frequency was a 208 volt primary stepped down to 120 volts for each supply phase. In specifying the audio/video equipment we made sure that all equipment purchased for the project had dual frequency operating capabilities. We did not experience any frequency related problems with any of the electronic equipment purchased. Other electrical equipment such as motors, ac contactors, transformers, etc. were also specified for dual frequency operation. All of this equipment performed without any problem. There was some concern during the design phase of the project regarding flickering of fluorescent lights at 50 hertz. We took a conservative approach and ordered our fluorescent fixtures with high frequency ballasts. The lighting operated with no discernable flicker. A dual power frequency did not prove to be any serious design problem as long as all equipment was specified and purchased for dual frequency operation.

AC POWER SYSTEM

Prior CBS mobile unit designs allowed hook-up to either a 208 volt, 3 phase source or a 220/240 volt, single phase source. This dual power source design was originally based on the days when it was common to hook up to different power sources on a week-to-week basis. A design for this type of system requires some type of patching or switching system to balance the phase loading depending on the type of power source. Achieving an even power balance for both types of sources is difficult, and there is usually a compromise made in favor of a single phase system, which results in neutral current when connected to a 3 phase source. During our initial design planning, we made a survey of the remote sites and discovered that sites with single phase only power availability were a very small percentage of our total remotes. When we weighed the advantages of designing a power system for 3 phase only, we came to the conclusion that this was the right time to change our design philosophy for this item. Therefore, MU-1 was designed for connection to 3 phase power sources only. CBS's contention is that for those few places that still only have single phase power we would either rent a generator or schedule one of our dual source mobile units to cover that event. The advantages of a 3 phase only power system are shown in Table 6.

Table 6. Advantages of 3 Phase Power

1. Eliminates a patching or switching system.

2. Easier to balance equipment loads.

3. Air conditioning amperage loads are distributed over three phases instead of two.

4. Three phase motors, compressors, and other equipment are generally smaller and lighter than single phase equipment.

In our design goals we had a requirement that the system must operate off a 200 amp service. This type of power service is the most commonly found power service at the various remote locations in the United States. For this project we monitored the power consumption specs of all equipment purchased and made a power allocation by equipment rack that would result in an even amperage allocation over the 3 phases. With the choice of a 3 phase air conditioning unit the resulting power loading of all phases was evenly balanced.

One potential problem that must be addressed when using any 3 phase motor is phase relationship which determines motor rotation. Contrary to what might be expected, not all power source transformers will conform to an "abc" phase relationship. They should conform, but experience shows that you will occasionally find a power source that was not properly phase wired. If the mobile unit is connected to a mis-phased source, there could be serious problems since the motors would run in a reverse direction or cause damage to the equipment. The only equipment on MU-1 that is phase dependent is the motors in the air conditioning unit. To alert our technicians to this problem, we installed a phase relation indicator on our power panel which indicates the phase relationship on the supply side of the air conditioner main breaker. There is also a warning label to alert the technicians to check this indicator prior to turning "on" the circuit breaker. As a further back-up, we also provided a hand held indicator that can be used at the power source.

Due to the requirement of operating at 50 hertz, special provisions were required for operating the 3 phase motors of the air conditioners. At 60 hertz the motors are rated for 208 volts. At 50 hertz the motor manufacturer recommends the motor voltage to be 190 volts. In our design planning it was decided to provide a means of powering the air conditioning units from a secondary power source for those rare occasions when sufficient primary source amperage was not available. A second set of power input connectors and a changeover switch was provided on our power input panel for this purpose. This second set of power connectors provided us with a very easy and convenient way of supplying a 190 volt, 3 phase, 50 hertz power source directly to the air conditioning unit without any wiring modifications.

ELECTRONICS DESIGN

In beginning the electronics design in 1990, the question of whether the mobile unit design should be an all digital system was addressed. At that time many manufacturers had digital products on the market or new products were being introduced. CBS had many meetings with various vendors regarding digital products, and they were all very candid with us regarding their plans and implementation schedules for new digital products. After reviewing our requirements for various equipment and our own project schedule, we came to the conclusion that this was not the time to build an all digital truck. Therefore, we embarked on a more conservative approach of only using the latest state-of-the-art equipment that was already on the market. Having the mobile unit ready for the Olympic schedule was our overriding consideration. The major system equipment supplied was as follows:

Table 7. MU-1 Equipment List

1. Switching:
   - Grass Valley 300 Production Switcher, 24 Input, 3 M/E, with Master E-MEM & E-Disk
   - Datatek 2400 Programmable Stereo Router, 50 X 50 Video, 80 X 50 Audio, 10 X 50 Key Signal

2. DVE:
   - Abekas A-53, 2 Channel with Warp & Key Channel Options

3. Cameras:
   - 6 SKF-710 CCD Field Cameras
   - 3 Hitachi SK-F3 CCD Hand Held Cameras
   - 3 Additional Cameras Available

4. Lenses:
   - Canon 40 X 1
   - Canon 18 X 1
   - Fujinon 55 X 1
   - Fujinon 8.5 X 5.5

5. Video Tape:
   - 2 Sony DVR-20 D2 VTR's
   - 3 Sony BVW-75 Beta VTR's
   - 5 Ampex Controllers

6. Video Monitoring and Distribution:
   - 58 Production Monitors Plus Extensive Camera Control and Video Tape Monitoring
   - 11 Tektronix Waveform Monitors & Vectorscopes
   - 4 Tektronix 110S Frame Syncronizers
   - 2 For-A Color Correctors
   - 72 Grass Valley Distribution Amplifiers
   - 6 Grass Valley Processing Amplifiers

7. Audio:
   - Neve 6600 Series Console, 48 X 8 X 4
   - Yamaha NS10 Near Field Monitors
   - Compact Disk Player
   - Audio Cassette Deck
   - 3 Stereo Cart Machines
   - DBX Noise Reduction & Limiting
   - DBX Signal Processing
   - Lexicon Digital Delay

8. Communications:
   - RTS/McCurdy 9500 Intercom & IFB
   - RTS Camera Interphone
   - Production and Technical 450 MHz RF Radios
   - AT & T Merlin II Telephone System

This paper does not cover an in-depth discussion on the audio, video and communication design since this type of design is rather straightforward to broadcast engineers. Instead, I would like to share with you some of the new ideas we used for this project.

INTERCONNECTION OF MOVABLE RACKS

Figure 1 and 2 show the operational and in transit locations of moveable equipment racks and consoles. During the floor plan design phase of the project we did not let equipment location be dictated by the physical constraints of the fixed portion of the trailer body. Our first goal was to optimize the layouts functionality and then address the equipment interconnection. After the layout was finalized, we solved the interconnect problem. We carefully planned what equipment would be located in each rack or console so that the number of interconnects would be kept to a minimum. Wherever possible, we tried to put peripheral equipment in the movable racks so as not to impact the entire system in case of interconnection failure. Where this was not possible we planned for interconnect cables.

All movable racks and consoles contain system connecting panels and the interconnecting cables were designed to be mated with them. Video and ac power interconnects for the movable equipment was accomplished with BNC video connectors and twist lock ac connectors. These connectors are easily engaged and disengaged, and they have proven reliability over repeated cycles of insertion and removal. For audio and control circuits we used a "zero insertion force" connector with a repeated insertion specification of a minimum of 10,000 insertions. This connector can also be mated quickly and reliably. During transit all movable equipment and consoles are secured with custom

Figure 5. View of Audio Console

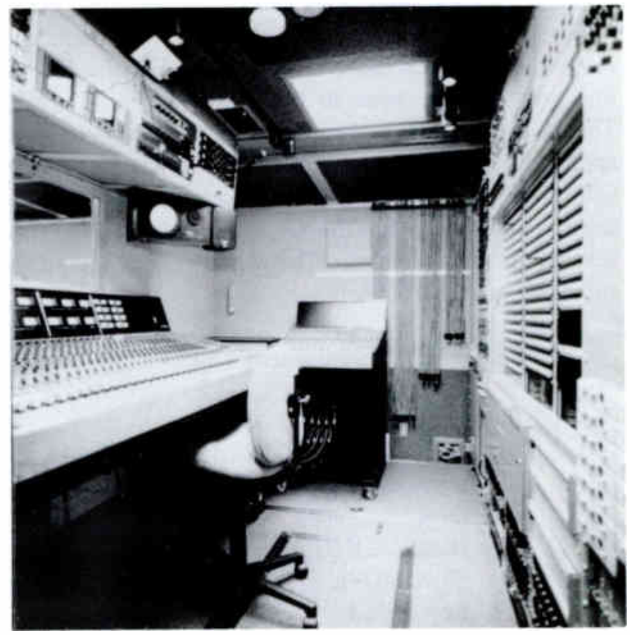

Figure 6. View of Expanded Audio Room

tie down straps that attach to built-in retainers.

KEY-FOLLOW ROUTING SWITCHER

The video router uses a 60 X 60 matrix. The video input requirements for the router was 50 inputs. The Datatek 2400 router has programming capability; therefore, we are using the extra ten inputs for key sources, which can be programmed to follow a video selection. We are using this function on our four video selectors used with the DVE. With this set-up associated key signals are automatically routed to the key channel whenever their corresponding video is selected.

CENTRAL COMPUTER STATION

Many of today's electronic equipment are addressable by means of an external computer. MU-1 has several such devices, and a rack mounted PC was provided with a selector switch to allow it to communicate to the production switcher, DVE, the routing switcher, the audio console, the intercom and IFB systems. This allows all PC addressable equipment to be programmed from a central location. As more addressable devices are added to the mobile unit they will be interfaced to the central computer.

INCREASED SIZE OF THE AUDIO CONSOLE

The audio console selected was a Neve 6600 series stereo console with 48 inputs, 8 submasters and 4 main outputs. Despite this unit's narrow front to back depth (36 inches), the physical size of this board was larger in width than could be accommodated within the closed portion of the unit. We applied the same design approach to solve this problem as was used in our other movable equipment. Neve separated the console within the width limits of their standard tub size and used zero insertion force connectors for interconnecting the two console sections. Figure 5 shows a view of the Audio room and the fixed portion of the console as well as the movable section. The fixed portion contains the stereo inputs, mono inputs, submasters, monitoring and the master outputs. This section of the console can function as a stand alone console. The movable section only contains additional mono inputs. This approach allowed us to fit a large audio console in a very limited space.

TELEPHONES

A Merlin II telephone system was purchased for this project; however, this equipment was not required for the Olympic coverage since a French phone system would be used there. We temporarily installed the Merlin system in our existing MU-6 to upgrade that mobile unit's telephone system. All system wiring and modular connectors were installed in MU-1 during its construction since many of the phone cables had to be routed through the interconnect connectors of movable consoles. This will allow us to quickly install the Merlin system presently being used in MU-6 after the Olympics. This prewiring of the phone system proved to have another advantage at the Olympics since it was discovered that the French phone system used the same 8 pin modular plugs as the Merlin system. This resulted in a very simple telephone installation at the Olympics since the mobile unit was already prewired.

SCHEDULING AND IMPLEMENTATION

This project budget was submitted to management during the summer of 1990, and we received approval to proceed in August of 1990. In order to meet the Olympic schedule for parking the unit in the Broadcast Center building by mid October of 1991, we determined that MU-1 would have to be complete and ready for shipment by mid September of 1991, thirteen months after starting the project. This may sound like a lot of time, but the lead time for building a custom expanding trailer can be rather long. Our original schedule from the trailer manufacturer for designing and building the vehicle was 7 months. Surprisingly it was only 1 month late and we took delivery of the unit in mid April 1991. Our estimate for installation and wiring of the electronics was 4 months after delivery of the vehicle, and we were estimating 1 month for testing and debugging. The design for this mobile unit was undertaken by the CBS Engineering Department's Field System Engineering Group and wired in-house by CBS's Construction Technicians. To everyone's credit, the mobile unit was completed and turned over to the Olympic's group for shipping on September 16, 1991.

PROJECT MANAGEMENT

In order to keep track of this project, several management tools were used. For the project schedule a PERT diagram was employed to track and update the schedule. The author uses his own Lotus Spreadsheet for tracking actual purchases against the project budget. This spreadsheet is broken down by project categories and gives an instant under/over summary which I find to be a wonderful tool for making decisions concerning unplanned equipment requirements that always pop up in a large project such as this. One of the simplest but most effective tools used is a Task List that shows every task, no matter how small, that needs to be done before the project is complete. This list can easily be updated on a computer to keep it current. My task list also shows other pertinent information such as who is responsible for the task, the status on any documentation required, whether the task is in process for design and/or wiring and status of any required parts.

The author was the project manager for this effort, and I know that projects of this magnitude get accomplished on time and within budget only if you have a good team working with you. Except for the vehicle portion, it was an in-house project involving many different departments at CBS who all cooperated in making it a success. I thank everyone involved for their efforts.

CONCLUSIONS

As was stated at the beginning of this paper, we started this project with the stated goal of building a new mobile unit that would use new ideas in its planning, design and implementation. From the reactions of CBS Production and Technical people who have had the chance to briefly use the unit, I feel that we have accomplished our goal. I hope the ideas I have shared with you in this paper give you the same feeling and encourage you to use and apply some of the approaches described.

UHF TRANSMISSION

Monday, April 13, 1992

Moderator:
Jerry Whitaker, technical writer, Beaverton, Oregon

*Paper not available at the time of publication.

MSDC KLYSTRON FIELD PERFORMANCE

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ABSTRACT

Varian Multistage Depressed Collector (MSDC) klystrons have been providing commercial service in UHF-TV transmitters for over 2 years. In this paper, the MSDC development program is briefly reviewed, focusing on concerns about potential problems. The actual field experience, which was carefully monitored, is related. Presently there are over 51 MSDC klystrons in use in 21 stations on the air. The accumulated operating time for these tubes is nearly 400,000 hours. A few tube failures have occurred, which are described along with corrective actions taken. Overall, MSDC performance compares favorably with that of other high-power klystrons, with no adverse effects due to the MSDC energy-saving feature.

INTRODUCTION

In the early 1980's, a few people at NASA and in the UHF-TV community had an idea, a vision, that UHF-TV transmitters could be significantly more efficient. Not just another typical incremental improvement, but a dramatic revolutionary breakthrough. The concept of "Depressed Collector" technology was not new, but previous attempts to implement it on klystrons had not been successful. A team at NASA, led by Dr. Henry Kosmahl, revisited this technology in an effort to develop highly efficient transmitters for use in satellites. The team successfully achieved a design that overcame previous problems and held promise for application on TV klystrons. Normally, klystron operation in TV service is inherently low in efficiency due to the need for linear amplification of the amplitude-modulated TV signal. While this requirement thwarted attempts to use standard approaches for efficiency enhancement, it provided the best mode of operation for the depressed-collector approach. Preliminary estimates indicated that power requirements could be cut in half. Although previous enhancements had resulted in improvements of 5 or 10%, we were then able to envision at least doubling the efficiency.

REVIEW OF MSDC DEVELOPMENT PROGRAM

A program initiated in 1984 to apply this technology to UHF-TV klystrons was supported by NASA Lewis Research Center, the National Association of Broadcasters, the Public Broadcasting System, transmitter manufacturers, and Varian. A 3-year program was envisioned, with $1 million of funding. Progress on this program was reported regularly at the NAB technical meetings (see the bibliography). By 1987, a successful model had been produced. Another 2 years were required to achieve a product ready for field operation integrated into a complete transmitter. Television station operation began in February 1990, thus beginning the compilation of over 2 years of field-operation data, which are presented in this paper.

MSDC Principle of Operation

Before describing the field-operation experience, it is appropriate to briefly summarize the principles of depressed-collector technology and review how the collector works. The collector's objective is to recover energy from the spent electron beam, so it acts only on the electron beam after it has completed its amplification task. The collector for the electron beam is constructed in segments (multiple stages), with each segment along the beam axis operated at increasingly negative (depressed below ground potential) voltage to decelerate the electrons in the beam. The electrons entering this collector region are slowed by the retarding electric field and sorted according to their velocity to impact the appropriate electrode. The important design technology involved achieving the proper electrode shape to most efficiently sort the electrons while ensuring all electrons would be collected. Another important objective was to achieve beam collection while avoiding secondary electrons that would degrade performance. By a process of computer simulation, an optimum design was achieved, which has been applied to the VKP-7990 klystron.

KLYSTRON PERFORMANCE

The performance characteristics of the VKP-7990 are listed in Figure 1. Except for the item "Figure of Merit = 130 in Typical TV Service," the performance is identical to standard four-cavity wideband external-cavity klystrons supplied by Varian and other manufacturers. The depressed-collector feature only enhances the efficiency, since great care was taken not to affect any other performance characteristic. Figure 2 is a photograph of the klystron mounted in the focusing and tuning structure.

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
- 1-dB Bandwidth 6 MHz
- Figure of Merit 130 in Typical TV Service

Figure 1.
VKP-7990 Multistage-Depressed-Collector Klystron Performance Characteristics

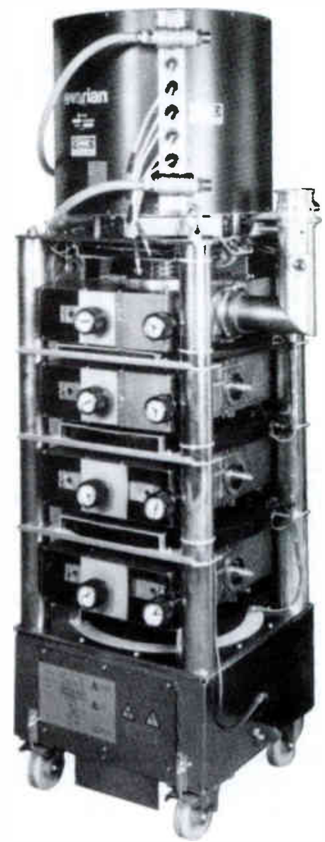

Figure 2
The VKP-7990 MSDC Klystron with Circuit Assembly

One difficulty in applying the MSDC technology results from the requirement of separate voltages for the collector stages. The VKP-7990 requires four voltages as shown in the power-supply schematic in Figure 3. In operation, the beam current divides among the four power supplies depending on signal level, as indicated in Figure 4. The power recovery results from collecting much of the beam current at lower voltages. One way to view the overall performance is to consider the power balance for the transmitter and observe how the power is distributed. This is shown in Figure 5, dramatically demonstrating the power-saving ability of the collector element.

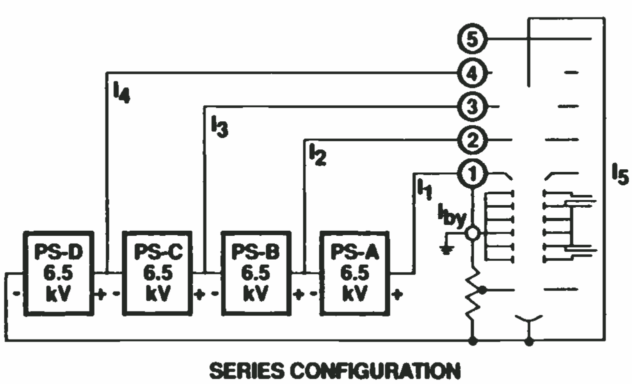

Figure 3
MSDC Power-Supply Schematic

Field Experience

As of January 1992, there are 51 VKP-7990 klystrons operating in 21 stations across the country. We have been carefully monitoring the performance of these tubes to ensure continued satisfactory operation or to promptly eliminate any shortcomings.

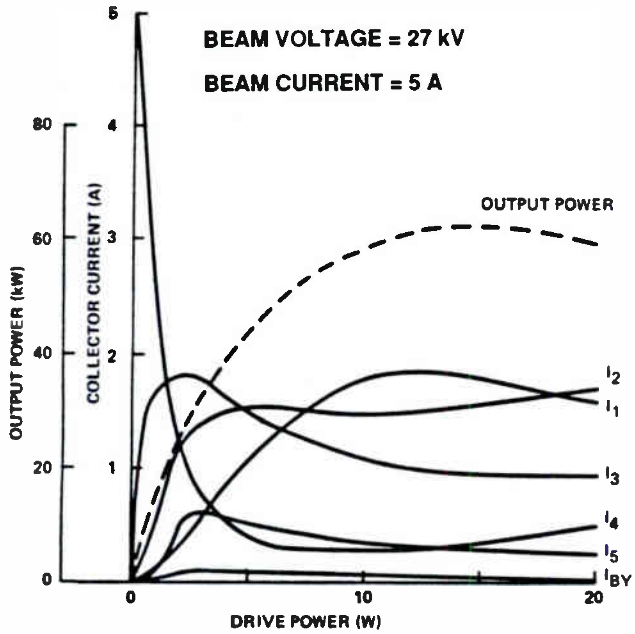

Figure 4
MSDC Current Distribution

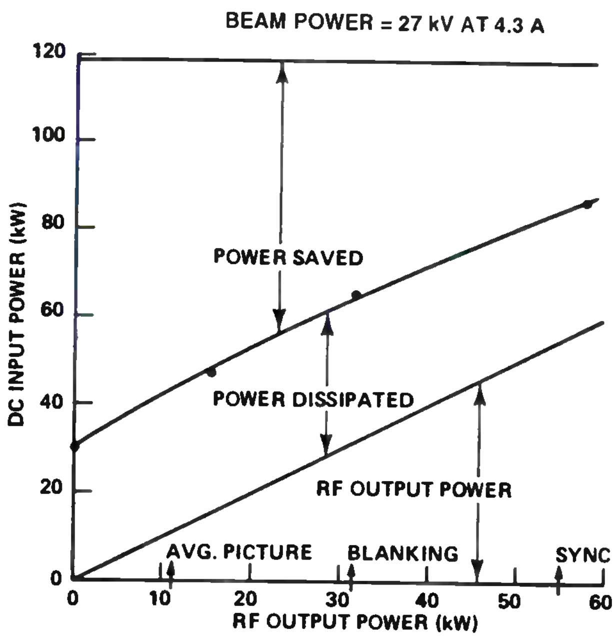

Figure 5
MSDC Power Distribution

During the development program, we identified a list of potential problem areas that we endeavored to circumvent by design (see Figure 6). Items 1 and 2 were addressed vigorously during the design effort and have not occurred on the VKP-7990. Item 3 was considered carefully with assistance from the transmitter manufacturers and has been effectively resolved. Item 4 was encountered, but in unexpected ways. Item 5 presented more difficulties than expected, requiring concerted efforts by Varian and transmitter manufacturers for an effective solution.

1. Prevention of Feedback Electrons while Optimizing Power Recovery
2. Suppression of Secondary Electrons
3. Cooling Problems
4. Mechanical Problems
5. RF Leakage

Figure 6
Potential Problem Areas

Field problems can be separated into two categories: problems occurring during initial installation in a station, and problems arising during station operation. We will address the second category first, since that is of the greatest concern for transmitter operators.

There have been five failures at operating stations. Four of these were due to breakage of the output-cavity ceramic. Careful analyses of these failures were conducted with several corrective actions implemented. There have been no failures since March 1991, attesting to the effectiveness of the actions taken. The other failure was due to voltage breakdown between collector elements on a very early model. Corrective actions early in the program have eliminated this problem.

Unfortunately, there have been many problems associated with initial station setup. The transmitter manufacturers have been very patient and understanding as we worked our way through the many problems associated with the new device. Early problems included tubes damaged in shipment as we refined the shipping container to ensure that the tubes were delivered intact. The radio-frequency interference problem was first solved by the equipment manufacturers; during the past year, we have resolved the problem with a solution that is inherent within the klystron. The coolant hose clamps were found to be inadequate; a new design was implemented and retrofitted on existing units and has demonstrated good performance. It was recently determined that there was an rf resonance at the channel 19 frequency in the collector electrodes. The problem was quickly resolved with a minor modification and has been entirely eliminated. Additional problems arose early in the program when some incorrectly wired electromagnets were provided, leading to several tube failures.

Our problem-solving efforts continued step by step throughout the program. During this work, ten tubes were destroyed, but all were replaced at no cost to our customers.

CONCLUSION

There are 51 VKP-7990 klystrons currently in operation in 21 stations nationwide. In January 1992, a total operating time of 391,000 hours had been achieved. To date, five operational failures had occurred, suggesting a preliminary MTBF of 78,200 hours. At this early stage of the program, MTBF cannot be predicted, since early failures have been few. The accumulated time for the longest-operating tube is still just under 15,000 hours. It will take a few more years of operation to get a clear picture of MSDC life expectancy, but at this point we are off to a good start.

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A TECHNICAL DESCRIPTION OF THE IOT EQUIPPED TRANSMITTER AND FIRST YEAR FIELD OPERATING RESULTS

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ABSTRACT

During 1991 the IOT tube was installed in a number of new UHF TV transmitters that are specially designed for its use. This paper will describe the technical aspects of transmitter design that accommodate the IOT and will also report on several operating stations' experience with their new transmitters. Installation descriptions and long term operating parameters will be presented. Future possibilities for 60 kW IOT applications will be presented.

I. INTRODUCTION

During 1991 a number of Inductive Output Tube (IOT) equipped transmitters were installed by Comark. Equipments using the IOT in both diplexed and common amplification applications are now in service at power levels from 30 kW to 120 kW. This paper will describe the unique features of the IOT transmitter and detail several installations from both a physical and operations point of view. Finally, the paper will describe a new UHF transmitter breakthrough that provides 110 kW peak visual and 11 kW average aural from only two parallel 60 kW EEV #IOT7360 IOT tubes in common amplification.

II. IOT TECHNOLOGY

The IOT is an evolutionary development of the earlier Klystrode® tube from Varian. EEV, the manufacturer of the IOT, focused their efforts on refining the Klystrode technology to make the IOT more user friendly.

Significant work was spent on eliminating the grid contamination found in early Klystrodes, which required periodic grid degassing cycles. Both Varian and EEV now manufacture tubes that do not require grid degassing.

It should be noted that elimination of grid contamination was based on lowering the cathode temperature while increasing its emissivity through cathode processing and design. Essentially the cathode was made physically and electrically larger. A side benefit of this was to increase the tube's peak power handling capability. Thus the IOT has power handling limitations largely based on average cathode current considerations, not on beam saturation as is found in both klystrons and tetrodes. This feature of IOT technology enables superior common amplification performance, with excellent figures of merit, at power levels close to the visual only rating of the tubes. The benefit of this will be discussed later in the paper.

EEV's approach to the development of the IOT RF system was to also make the RF hardware and tuning more user friendly. Thus, EEV developed a single tuned broadband input cavity that is based on a high voltage, low RF loss, d.c. block technology. The introduction of a physical d.c. block allowed the use of a single grounded cavity structure that eliminated the transmission line tuning system found in the Klystrode circuit assembly. Thus, the size and complexity of the input RF tuning system for the IOT was reduced to a single tuned cavity structure.

Figure 1 shows the IOT in its hardware. The input cavity is at the top of the structure and the tube is mounted with its collector down.

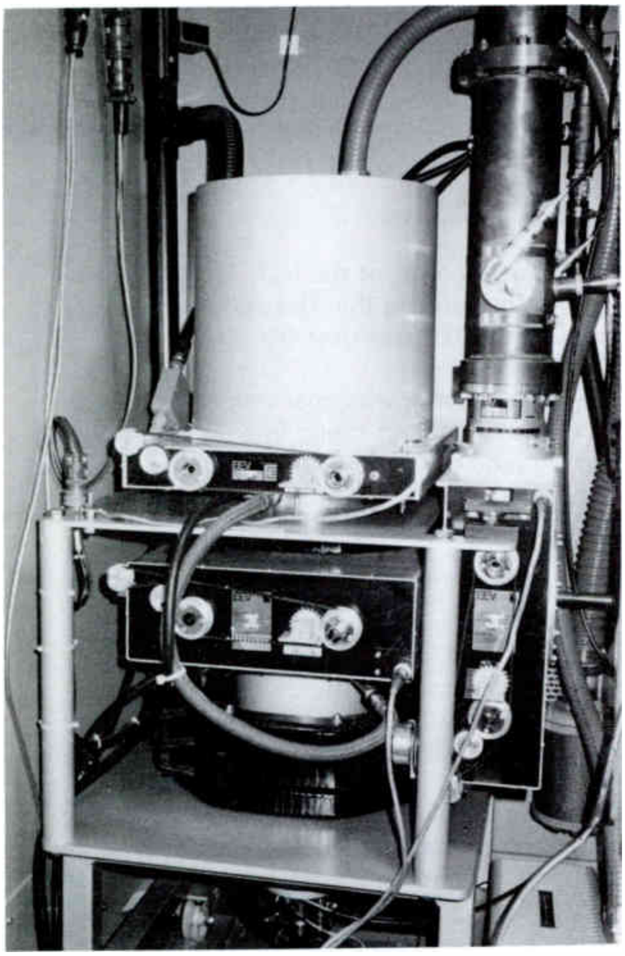

Figure 1 - IOT and Hardware

The development of the IOT relied upon much data from the field experiences of the Klystrode tubes. The use of the field results from over fifty Klystrode sockets, some in service for over three years, has resulted in a second generation tube and hardware that is more user friendly while proving to be extremely stable and reliable.

EEV also refined the tube/hardware interface so that a single model number covers the entire UHF band.

III. WGBY-TV 57, SPRINGFIELD, MA

The first IOT devices to become available were rated at 40 kW visual service. Comark's and EEV's evaluation of these devices showed them to be capable of 35 kW in common amplification. The significance of the superior cathode capability of these tubes now becomes clear. There is no need to derate the IOT by 40% to 50% for common amplification operation as is true of Klystrodes, klystrons and tetrodes.

The first field installation of IOT technology took place at WGBY in Springfield, MA. The transmitter is a Comark CTT-U-70SICR(M) using two EEV IOT7340 devices operating in common amplification. 60 kW combined visual output and 6 kW aural average aural power is achieved.

Figure 2 is a block diagram of the WGBY transmitter. The use of magic Tee combining and dual exciters provides high levels of on-air reliability through system redundancy. Dual heat exchangers, pumps and H.V. power supplies further increase redundancy levels. While the dual exciter could be arranged to configure the transmitter as two independent parallel transmitters the staff of WGBY chose to have the exciters switched in a hot standby configuration. This choice insured 100% output power in the event of an exciter failure instead of 50% power. The short time loss of an air signal during fault evaluation and switching was not considered significant by WGBY.

Figure 3 shows the dual modulator/exciter system. This cabinet was located 20 feet from the dual HPA cabinets for the convenience of the customer.

Figure 4 shows the dual HPA and solid-state driver assemblies.

Figure 5 shows the output RF system consisting of a magic Tee, a water column system load and a set of waveguide notch filters for out of band IM supervision.

The WGBY transmitters is also equipped with Comark's patent pending aural carrier corrector.

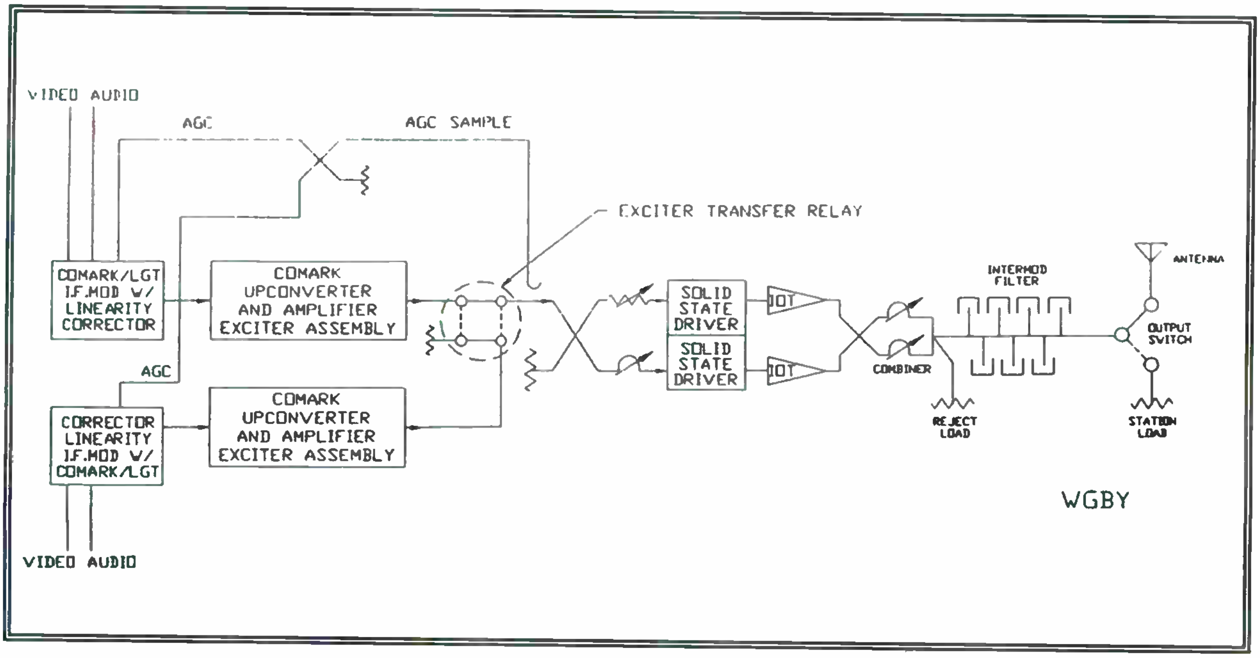

Figure 2 - RF Flow Diagram - WGBY

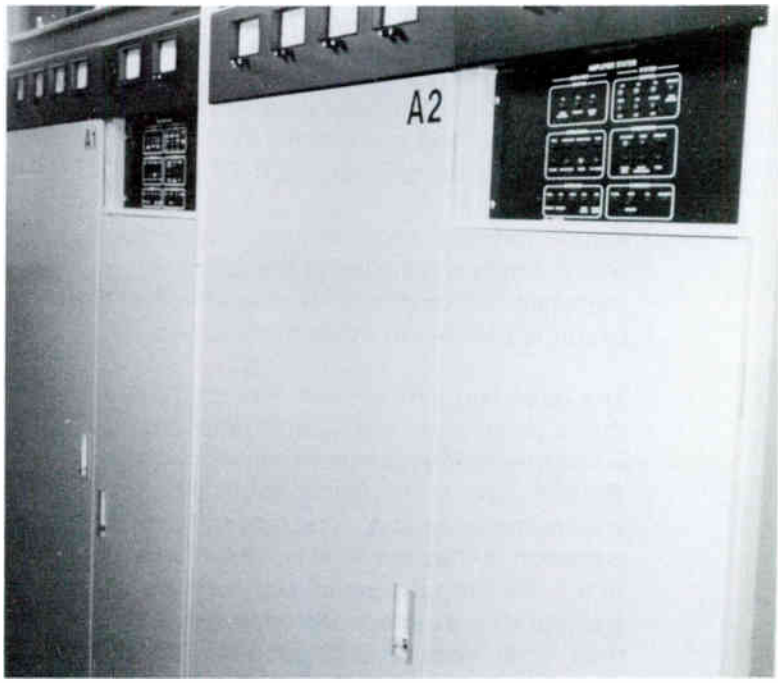

Figure 4 - Dual HPA & Solid State Driver Assemblies - WGBY

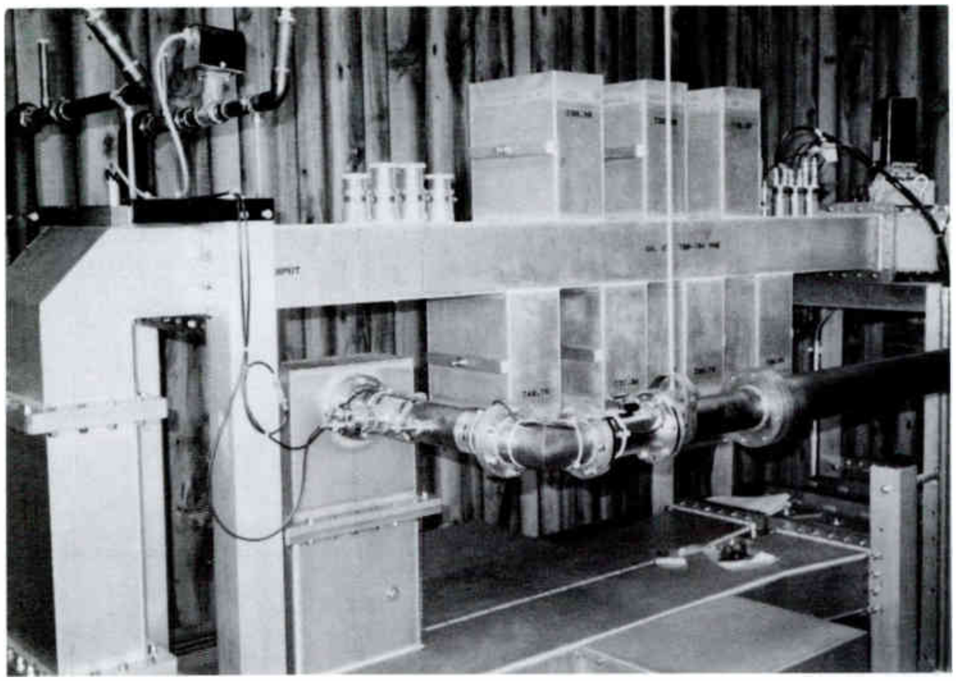

Figure 5 - Output RF System - WGBY

This corrector reduces the H sync spike contamination of the aural carrier from the visual carrier and ensures full FCC specification compliance (FCC 73.682(c)(3)).

Figure 6 shows the aural carrier corrector board which is installed in the Comark modulator's IF section. The operation of this circuit is covered in other publications.[1,2,3]

The operating data from WGBY confirmed that a common amplification transmitter inherently produces better waveforms than a diplexed transmitter, especially if the diplexed transmitter is pulsed. The reason for this condition is that the level of linearity required to amplify both the visual and aural signals in a single HPA at intermodulation levels better than -56db ensures that video waveform distortion is virtually eliminated.

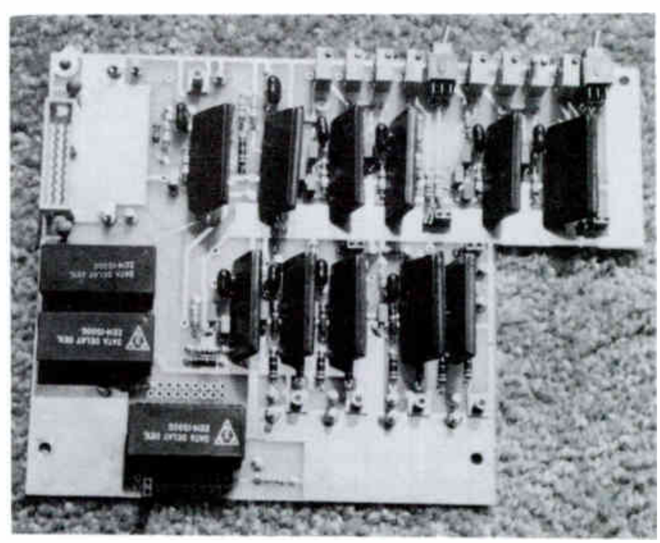

Figure 6 - Aural Carrier Corrector Board

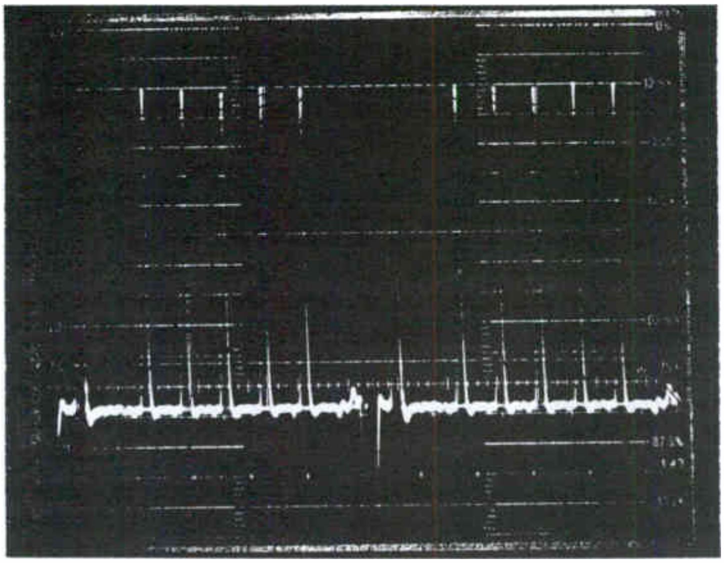

Figure 7 - Low Frequency Linearity WGBY

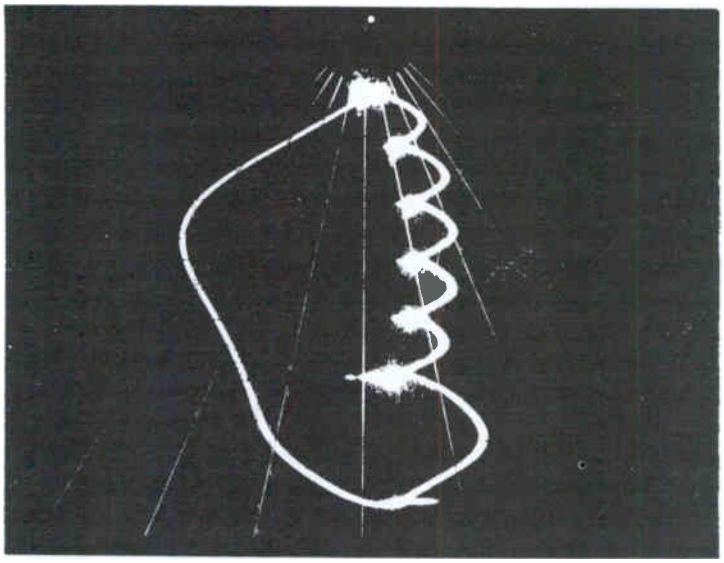

Figure 8 - ICPM - WGBY

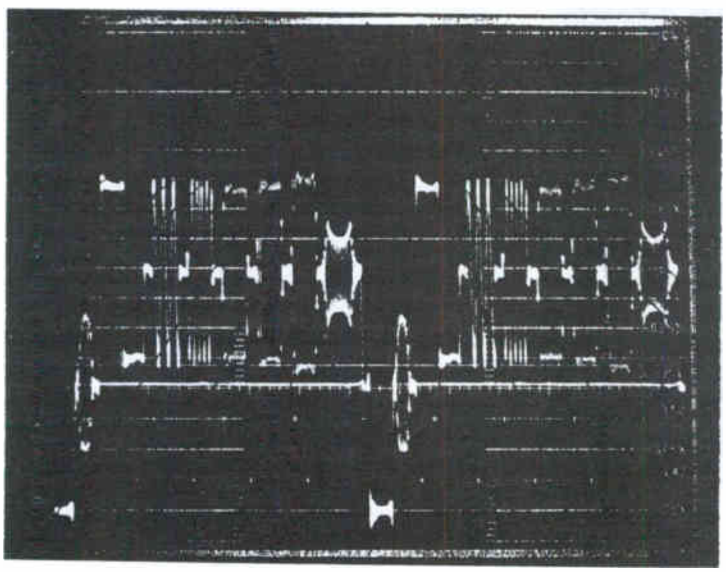

Figure 9 - Multiburst at Horizontal Rate WGBY

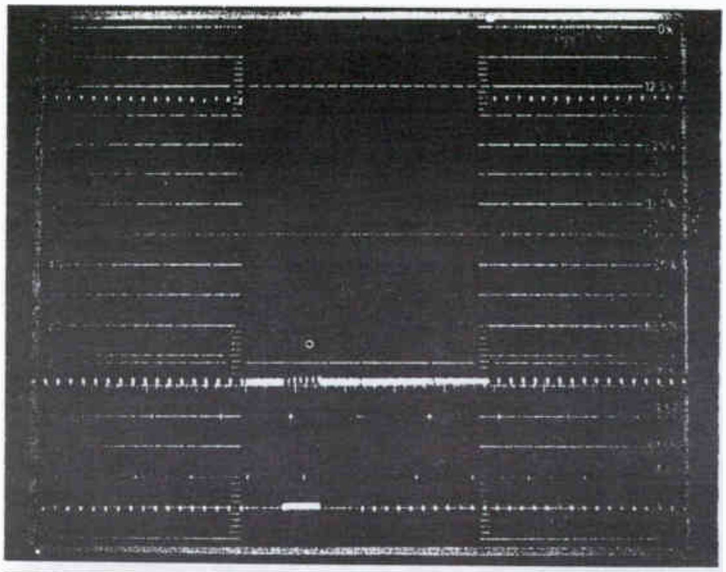

Figure 10 - Expanded Vertical Interval WGBY

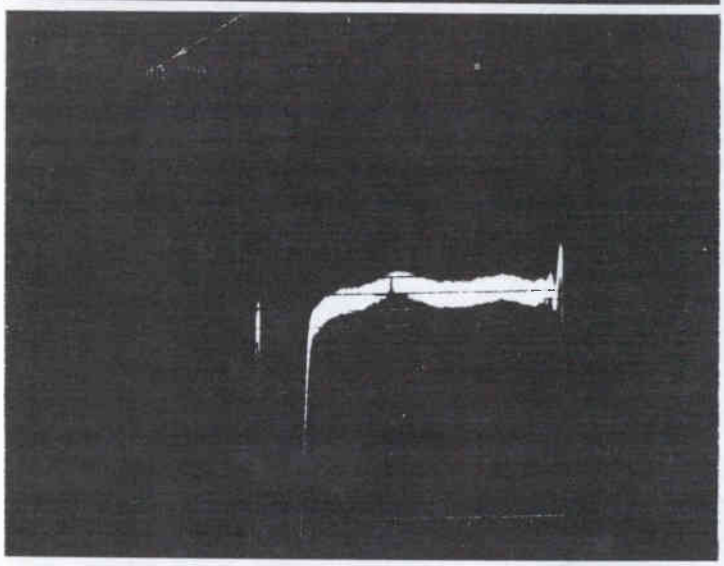

Figure 11 - Differential Gain - WGBY

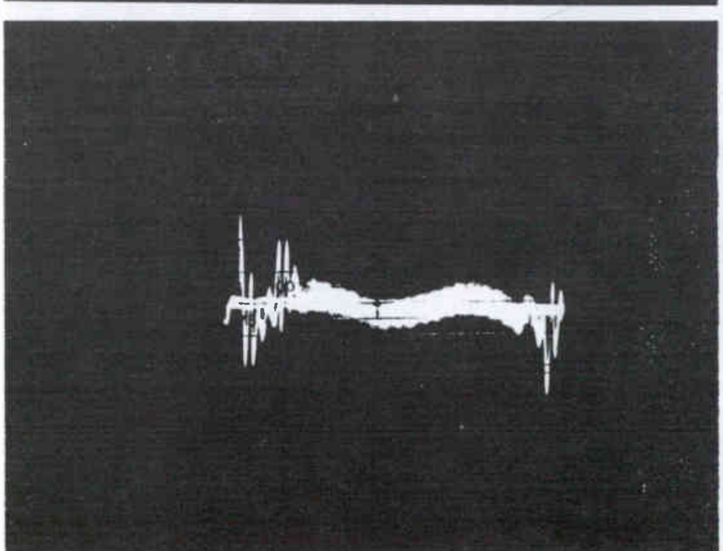

Figure 12 - Differential Phase - WGBY

Figures 7 through 12 are offered as evidence of the superior waveform performance of an IOT common amplification system. These data were taken as part of the WGBY FCC Proof-of-Performance and are incorporated into that document.

The performance of the WGBY transmitter, as evidenced by the proof data was deemed satisfactory. Subsequent audits of the transmitter's performance over a period of six months showed little change. No adjustment of linearity precorrection was required during six months of daily operations.

The plant power consumption of this transmitter is less than 85 kW under average picture conditions. This represents a Plant Figure of Merit (PFOM) of over 78%.

$$\text{PFOM} = \frac{\text{Peak Visual Output \& Average Aural Output}}{\text{Average A.C. Input to Transmitter Plant}}$$

This is the real payoff for the users of the IOT.

IV. WSTR-TV 64, CINCINNATI, OH

IOT devices provide excellent performance in diplexed applications as well as in common amplification applications. WSTR in Cincinnati, OH is operating (as of January 1992) four parallel IOT amplifiers at 120 kW. The aural amplifier is a standard wideband klystron.

WSTR was originally scheduled to operate at 240 kW, however, delays in the availability of the 60 kW IOT devices forced an early installation of the lower power tubes. An upgrade to 240 kW is now scheduled for late February 1992 (before NAB).

The first operation of WSTR with the four IOT devices commenced in November of 1991. All four IOT amplifiers came on line and have performed without difficulties since commissioning.

Figure 13 is a view of the transmitter as installed at WSTR. Figure 14 shows the four-way RF magic Tee based RF power combining system as installed at WSTR.

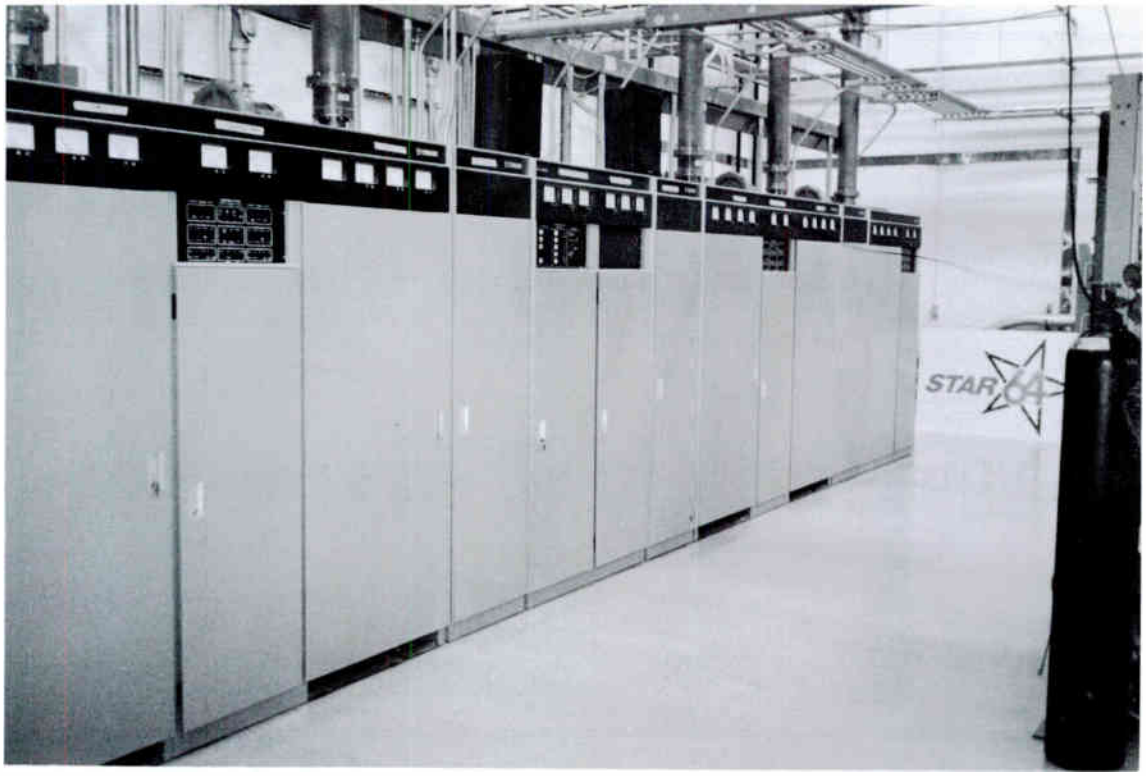

Figure 13 - WSTR Transmitter Installation

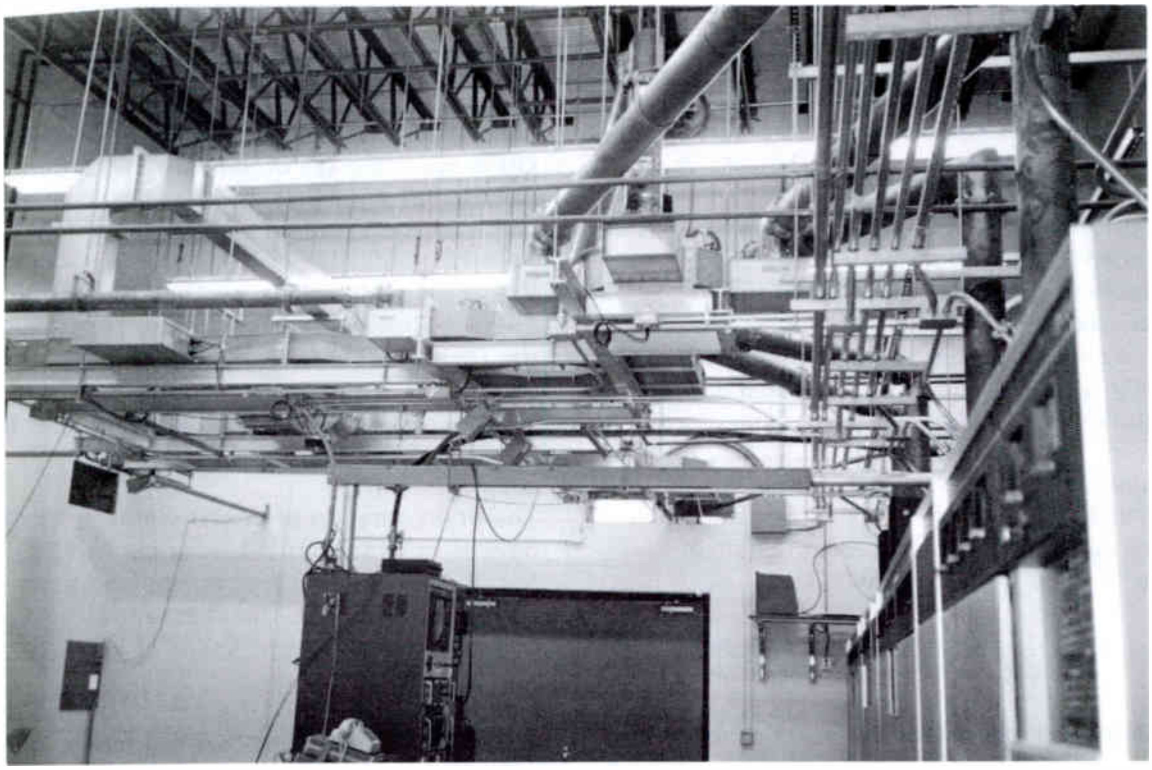

Figure 14 - WSTR Magic Tee Four-Way Combining System

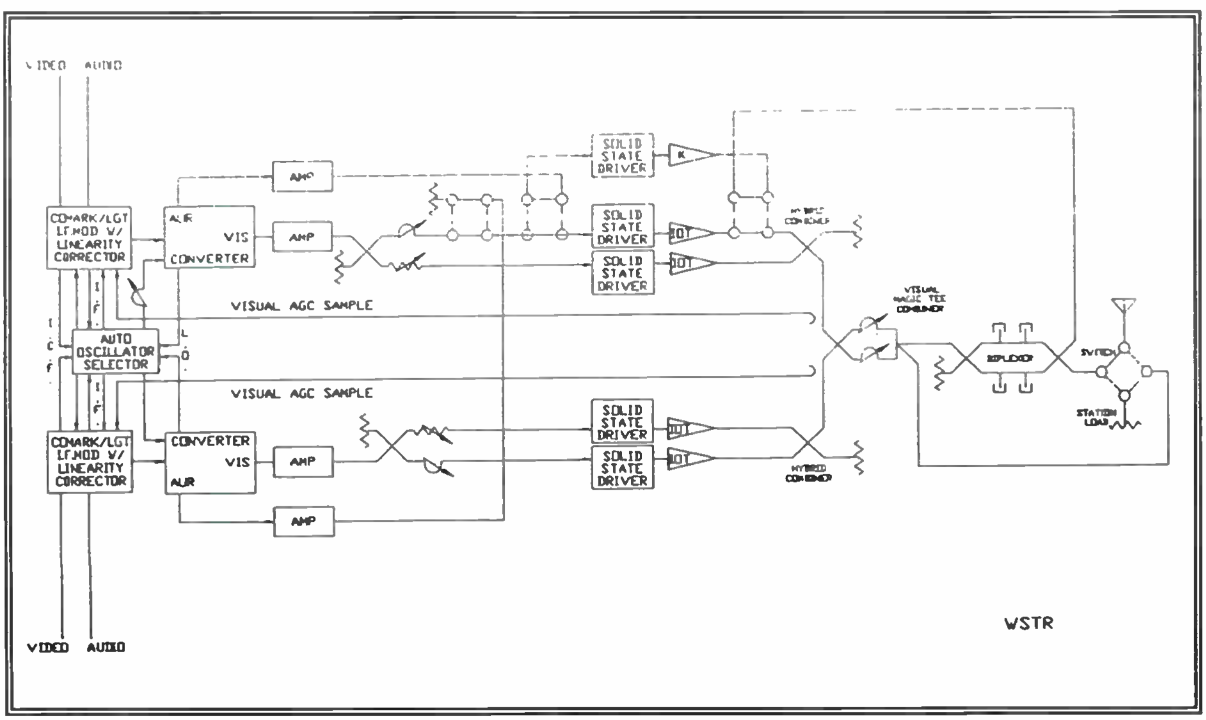

Figure 15 - Block Diagram of the WSTR-TV 64, Cincinnati, OH Transmitter

Each half of the installation is a pair of IOT devices in parallel. The two IOT pairs are combined in a magic Tee. Separate, but coherent, modulators are used to provide at least 50% output power under any single system failure. This is achieved without mechanical RF contact switching. Additional redundancy is achieved by using the magic Tee to bypass the diplexer and operate in multiplex mode or the system can be configured to operate one of the IOT devices as the aural amplifier and route its signal to the diplexer. In this configuration the transmitter will provide 50% of rated visual output. Multiple beam supplies and heat exchangers enhance the redundancy of this transmitter.

Figures 16 through 22 are offered as evidence of the linear performance of IOT devices in diplexed applications.

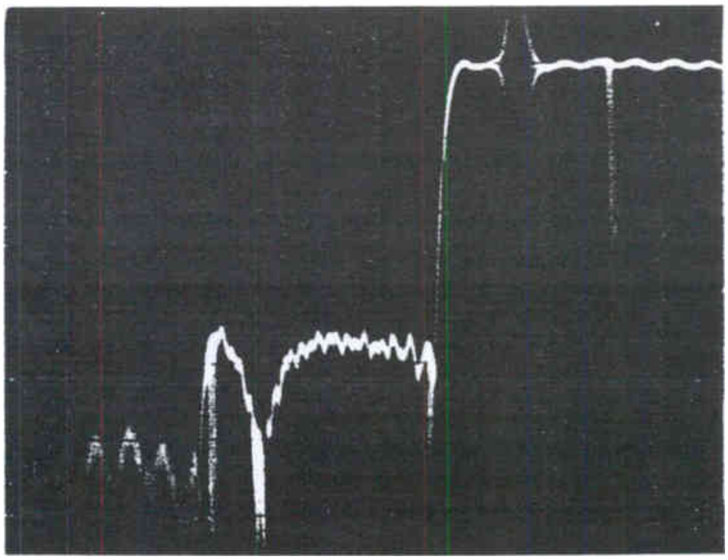

Figure 16 - Lower Sideband Response (-37db) - WSTR

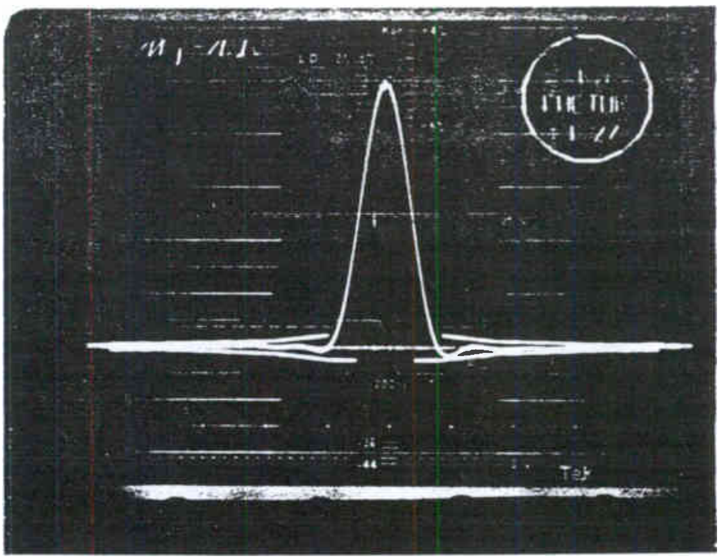

Figure 17 - 2T Pulse (1.2%) - WSTR

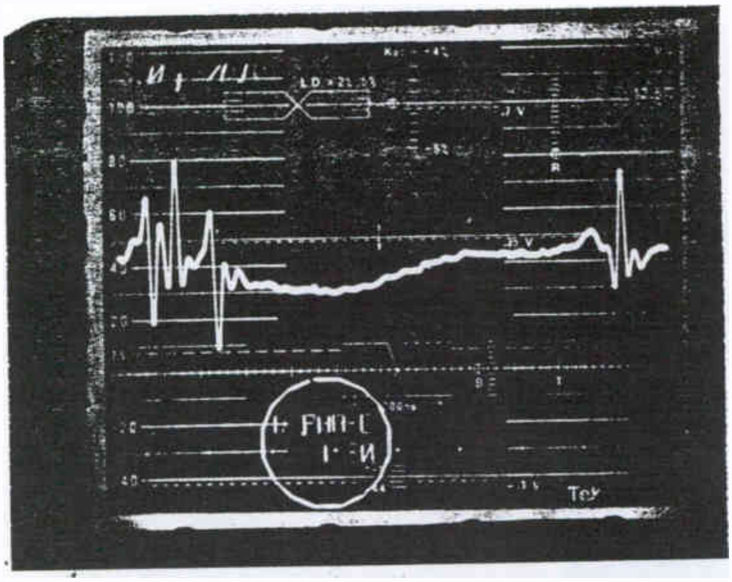

Figure 18 - Differential Phase (1.5°) - WSTR

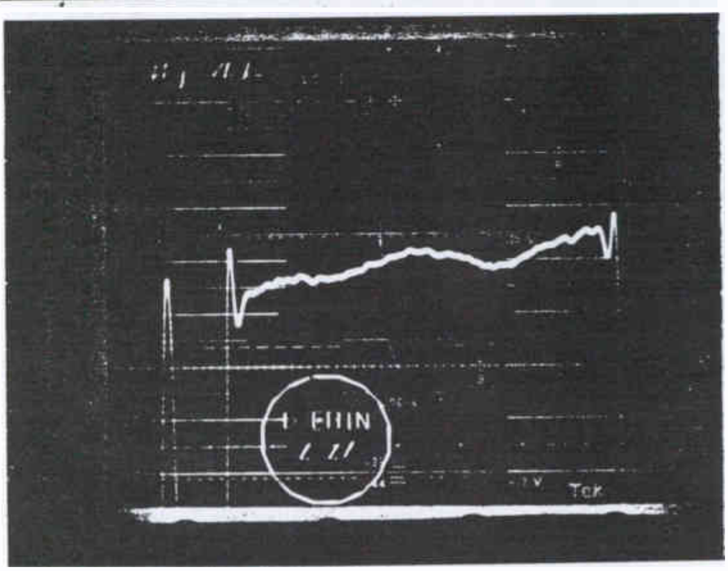

Figure 19 - Differential Gain (2.2%) - WSTR

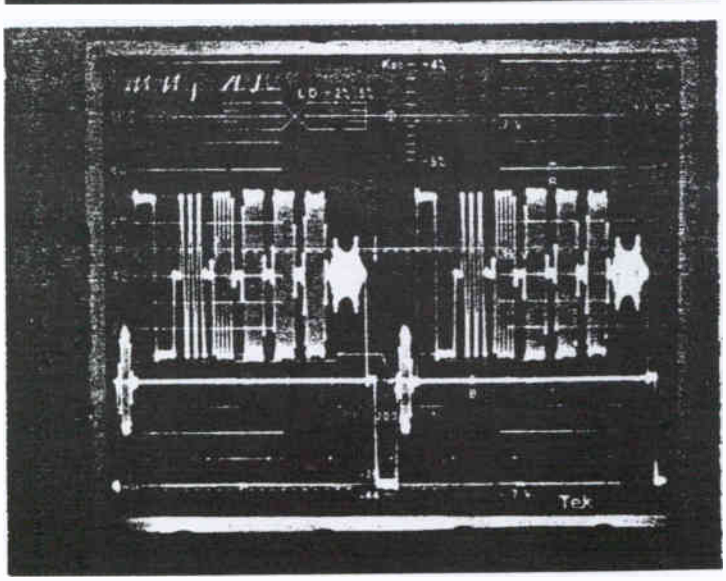

Figure 20 - Multiburst - WSTR

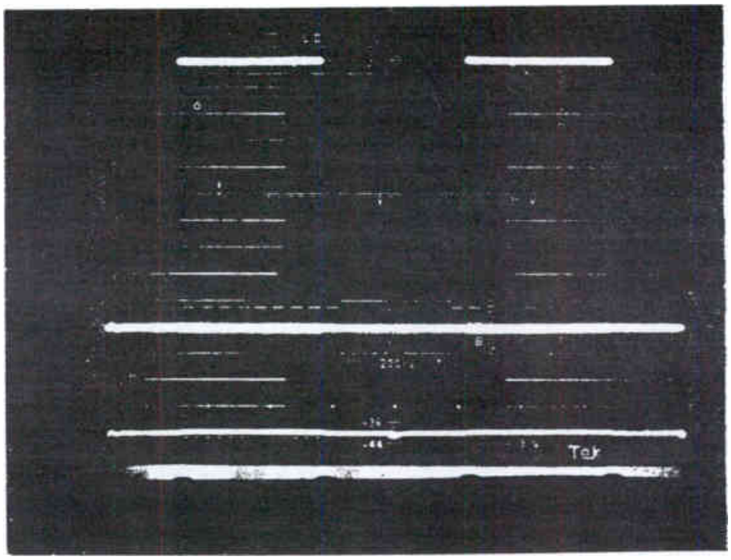

Figure 21 - Fuel Field Square Wave Response - WSTR

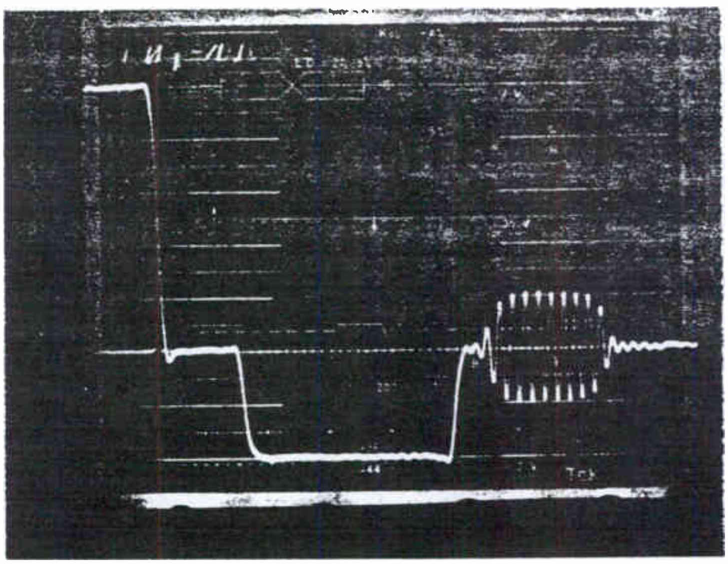

Figure 22 - Horizontal Sync - WSTR

The data on WSTR shows levels of linearity performance at figures of merit better than 120%. This is achieved without pulsing and thus waveforms do not show the typical ringing and timing errors common in pulsed systems.

The lower sideband response of the WSTR transmitter at -37db is evidence of the fact that the efficiency is not achieved by pushing linearity to the limit of acceptability.

The wide performance margins created by the waveform performance of this transmitter bode well for long term trouble-free operation.

V. OTHER IOT INSTALLATIONS

Since the first IOT system was placed in operation at WGBY in June of 1991, others have followed. Table I shows transmitters in service or under contract to Comark as of January 15, 1992.

The growth of the IOT as an accepted technology has been extremely fast. This may be based on the simplicity of the transmitter's systems - no pulsers or complex high purity water system are required. It may be based on the fact that no IOT has failed catastrophically while in service. It may be based on the small size of the IOT and the ease of shipping and handling - EEV has developed a method of shipping the IOT by overnight courier. It may be based on the pioneering work achieved by Varian and the Klystrode.

Whatever the reason for its rapid acceptance, the IOT is quickly gaining worldwide applications. At least four international manufacturers are currently developing IOT based equipment. It is therefore reasonable to ask...what's next?

VI. 60 KW IOT APPLICATIONS

With the arrival of a 60 kW visually rated IOT device, new applications become apparent. The superior peak power handling characteristics of the IOT immediately suggest a 50-55 kW common amplification based HPA. Thus, a single tube 55 kW or 110 kW, 2 tube transmitter seems likely to be developed (January 1992).

The advantages of reducing the tube count in a 110 kW transmitter by 33% means less initial cost, lower maintenance costs, lower energy costs and, of course, less cost at tube replacement time. All of these advantages are offered by the IOT while it delivers superior signal performance at state-of-the-art efficiencies from a simple transmitter support system.

TABLE I

User	Power Level	# of Tubes	Date Commissioned
WGBY-TV, Springfield, MA	60kW	2	June 1991
WOUC-TV, Athens, OH	30kW	1	Aug. 1991
WQLN-TV, Erie, PA	60kW	2	Sept. 1991
WFPT-TV, Frederick, MD	60kW	2	Oct. 1991
WSTR-TV, Cincinnati, OH	120kW	4	Nov. 1991
WPBO-TV, Portsmouth, OH	30kW	1	Under Construction
KSTF-TV, San Francisco, CA	60kW	2	Under Construction

VII. FUTURE TRANSMISSION TECHNOLOGY IMPACT ON TRANSMITTER CONFIGURATIONS

The rapid advance of digital encoding technology into the broadcast industry will place new requirements on to today's R.F. transmitting plant. In addition, competitive pressures from other video delivery systems using digital compression technology will certainly effect the broadcasters' bottom line.

In the near future, the R.F. transmitting plant of today may be called upon to handle the broadband digital signals of a video compression system which would permit four or more channels of information to be transmitted over a single 6MHz channel.

The clear advantage of IOT common amplification over pulsed klystron diplexed operation is the signal independence of the IOT system. The broadband linear Class B operation of IOT devices is readily adapted to the technology of the future. The pulsed klystron transmitter is a system that has evolved to be signal specific to today's NTSC signal format.

VIII. CONCLUSIONS

The initial installation of IOT based transmitters has been very successful. As of January 15, 1991, eleven (11) tubes are in service in five (5) transmitters. These tubes have performed without failure and are providing demonstrably superior signal quality at high levels of efficiency. In both common amplification and diplexed configurations the IOT is supported with simple straight forward systems that eliminate the need for ultra pure water and signal distorting pulsing.

A major advantage of a common amplification transmitter besides simplicity and signal quality is the signal independent nature of the system. The common amplification transmitter is simply a broadband (6MHz) linearity amplified channel capable of reproducing whatever combination of signals is fed from its modulator. The absence of notch diplexers and pulsers makes the common amplification system easily adopted to new emerging technologies that will find application in the UHF broadcasters' effort to improve his bottom line. These new technologies include digital signal compression, multiple channel sound services, HDTV and unique applications like paging and digital information transmission.

Worldwide applications for the IOT are underway with multiple international manufacturers developing IOT based systems. Air-cooled versions of the IOT are already finding international installations.

The availability of a 60 kW IOT, with its inherent peak power capability, opens the door to the development of two tube transmitters at 110 kW. The elimination of 33% of the tubes in a 110 kW rig make the IOT and its associated technology very attractive from both a signal quality and cost standpoint.

The IOT has proven to be a linear, stable device with high peak power capability. The ideal device to become the standard of the industry.

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[3]"Stereo Pilot Frequency Protection in Common Amplification Based Transmitters - The Need for An Aural Corrector" by N.S.Ostroff - Presented at SBE, October, 1991

The author wishes to thank the many supporters of IOT technology who have encourage its rapid implementation into actual transmitters. In particular, special thanks goes to Ray Miller at WGBY and Greg Buzzell at WSTR.

SOME EXCITING ADVENTURES IN THE IOT BUSINESS

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ABSTRACT

The Inductive Output Tube (IOT), first proposed by Andrew Haeff in 1938, is a device that has had to wait more than 50 years for the technology to become available to permit the construction of a useful and practical, very high efficiency, UHF-TV amplifier system. But once started (in June 1989), the project to develop it soon became very fast-moving, exciting and adventurous; attracting a small team of dedicated physicists and engineers who have produced a tube which will be used into the next century.

The technical and commercial background which finally led to the appearance of the IOT in the UHF-TV amplifier market-place is outlined. Broadcasters using UHF-TV klystrons have become accustomed to long, trouble-free life and reasonable efficiency. The IOT had therefore to provide some extra advantages such as improved user-friendliness, higher efficiency and smaller size yet without compromise as to reliability and life.

The IOT designer had, therefore, to learn from the established klystron designs. The one major component that the IOT must have and that the conventional klystron does not use is the grid. The construction process of the grid is described and the chosen solutions to some of the problems outlined.

The IOT input circuit is the key to the user-friendliness of the system. The reasons for the choice of an annular input cavity with RF chokes to hold off the DC Beam voltage are explained and some construction features highlighted.

In order to transmit the UHF television of today, some eight MHz of instantaneous RF bandwidth is needed. For the IOT, this meant the use of a novel double-tuned output cavity system, the design and evolution of which is discussed.

TECHNICAL AND COMMERCIAL BACKGROUND

It is the policy of EEV to provide the UHF TV broadcaster with the choice of up-dated conventional klystrons, ESC klystrons and IOT's for their new transmitter requirements. All have their advantages and drawbacks in a given application and EEV leaves to each individual customer the decision as to which technology is likely to serve them best. The present paper seeks to describe only the IOT and its performance and hopes to provide broadcasters with some data which will assist them in coming to their decision.

Back in the 1960's when, in Europe at least, UHF-TV was starting, broadcasting engineers were surprised and delighted to be able to transmit a few kilowatts of a decent quality signal from a klystron initially designed for troposcatter communications. No-one worried about the costs - it was a major achievement to have done it at all!

All this changed very rapidly. The focus of change was initially on the provision of high gain klystrons optimised for UHF-TV so that system reliability could be improved by using solid-state RF drive systems.

Political and military actions in the Middle East around the 1970's soon caused huge increases in oil prices which rapidly led to ever-increasing electricity prices. Suddenly, the UHF-TV klystron needed much higher efficiency, and since solid-state devices had greatly improved, gain was no longer so important. UHF-TV klystron efficiency was improved, often at the expense of gain, by 50% over early tubes by the revision of RF body designs using newly-available computer based design techniques. But the energy price pressure did not relax and the broadcasters needed even lower energy bills.

Electronic techniques had meanwhile improved and it became possible to pulse UHF-TV klystrons in order that the beam current used could be reduced from its sync (maximum) value during the picture region of each line. This gave another 50% of efficiency improvement but came at the expense of a more complicated and expensive transmitter which was more difficult to set-up for optimum performance and long term stability.

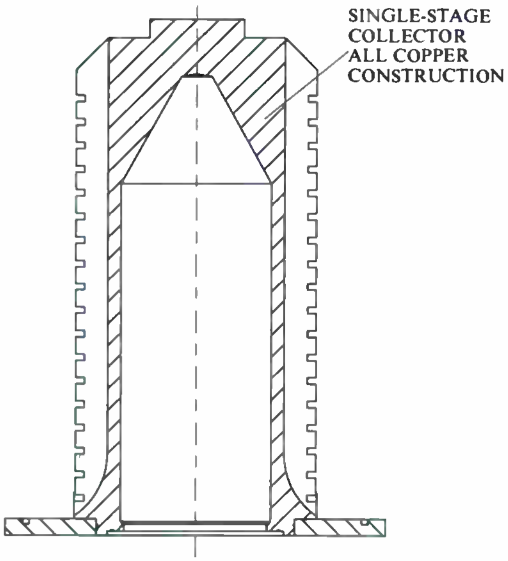

Fig. 1a.

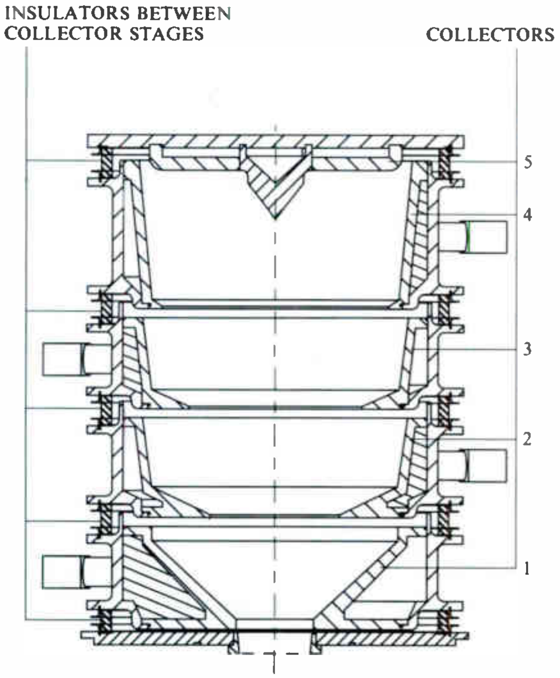

Fig. 1b.

Fig. 1. Collectors from a conventional klystron and an ESC klystron to illustrate how separate insulated stages are needed to recover spent beam energy.

Since electronic conversion efficiency in UHF-TV klystrons had been squeezed near to the limits of the physics, the next area to attract attention in the quest for further efficiency improvement was the collector electrode of the klystron (Fig. 1a). This is where the spent beam after the RF interaction is stopped and its energy dissipated as (wasted!) heat. UHF-TV klystrons soon became available with 'depressed' collectors (See Fig. 1b.) which are constructed in multiple stages and shaped so that retarding electric fields can extract and feed back into the power supply a significant proportion of the energy of the spent beam.

Energy Saving Collector UHF-TV klystrons, which retain the facility for pulsing, are now available and in service. The efficiency achieved is excellent, with Figure of Merit values around 120%, and lifetimes appear to be those expected from conventional klystrons.

The ESC Klystron was developed by the combination of the known technology of the UHF TV Klystron with that of the depressed collector. This yielded significant gains and advantages and, as always, a few problems to be solved.

The search for an alternative UHF-TV amplifier tube family was initiated. The UHF tetrode was carefully examined. It has excellent efficiency and adequate gain but a designer has to be bold to suggest pushing the power output per tube much above 20 kW because of problems of power dissipation within an inevitably very small volume.

What if a tube could be devised which would combine the small size and excellent efficiency of the tetrode with the long life, high power output and reliability of the klystron?

The first tube to attempt this was the Klystrode®, soon to be followed by the Inductive Output Tube (IOT). The Klystrode uses an input cavity system which employs internal feedback to enhance the available gain, whilst the IOT has an input cavity system which inherits the unconditional stability of the conventional klystron input system. Both tubes use a conventional klystron RF output extraction system, modified in order to satisfy the transmission bandwidth requirements. Neither tube requires the pulsing techniques to give excellent efficiency.

The aim of the IOT development at EEV was to produce a physically small 60 kW high-efficiency UHF amplifier tube which borrowed the maximum amount of established technology from the UHF-TV klystron. Target tube life was to be 'klystron-like'. The IOT development began in June 1989 and the first production tubes entered service in the Summer of 1991.

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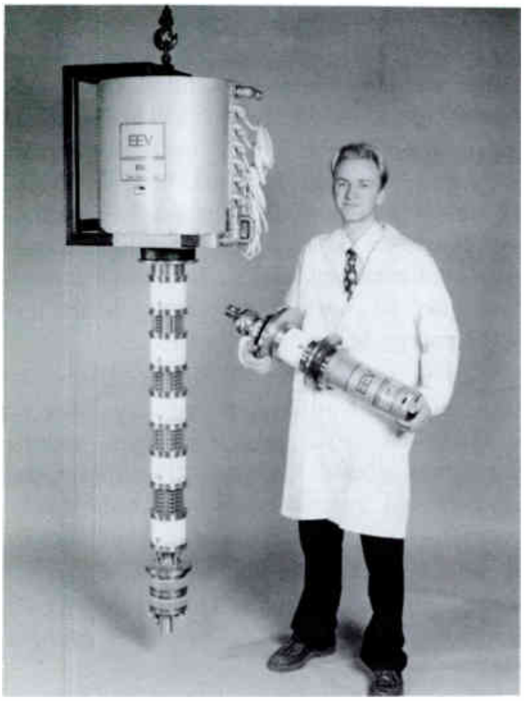


Fig. 2. is a photograph of a 70 kW 5-cavity Energy Saving Collector klystron. Standing beside this is an engineer holding a 40 kW IOT tube.

ELECTRON GUN PHILOSOPHY AND DESIGN

The barium aluminate cathode chosen for the IOT gun was taken from a conventional klystron together with its heater structure. The methods used to support the cathode, heater and necessary heat shields in the klystron gun were retained for use in the IOT. This gun and heater combination had a known and proven performance, having achieved several million hours in conventional klystron service.

In the IOT, the RF input voltage is applied between the cathode and a grid which allows extra electrons to be drawn from the cathode into a low quiescent current electron beam according to the instantaneous RF voltage appearing between grid and cathode. The resulting density modulated beam is then passed into the klystron-like RF output interaction region of the tube.

The grid parameters were established in terms of radius of curvature (the grid is a co-spherical surface with the cathode), wire size and number and shape of apertures. The tube was intended to operate in a grounded-anode mode and the computer was used to assist in the choice of gun-electrode shapes. The beam perveance and operating beam voltage (around 30 kV) were defined and the electrostatic electron dynamics computed.

Having completed preliminary electrostatic electron trajectory and beam perveance computation it was necessary to calculate the magnetic field required to focus the beam over the relatively short distance from the electron gun to the collector. This was added to the computer simulation and it was confirmed that a well-confined and laminar beam was produced and held throughout the tube.

It was decided to support the tube and its input and output cavity systems on a low trolley which would also carry the magnet coils which were designed to produce the magnetic field required. This, following conventional klystron terminology, is called the 'magnet frame'.

To assemble the IOT system, the tube is positioned between pole-pieces in the magnet frame with the electron gun uppermost. All other components of the system bolt on to the tube and are supported by the magnet frame which has small wheels to allow the whole unit to be pushed into the transmitter.

The collector was initially water cooled and water connections are accessible from the lower part of the magnet frame. Later sample tubes were fitted with air-cooled collectors which were physically larger than the water-cooled version. This necessitated support in a different design of magnet frame.

A current version of the IOT gun structure is shown in Fig. 3. The grid is clamped in place in front of the cathode, supported on a metal cylinder and isolated from the cathode by a ceramic insulator which also completes the vacuum envelope. This is the grid-to-cathode ceramic through which the RF energy from the input circuit enters the tube to apply the RF voltage to the grid.

A second ceramic insulator supports the complete gridded electron gun at the correct distance from the grounded anode. This ceramic completes the vacuum envelope and holds off the full beam voltage of around 30 kV. The IOT gun operates at cathode to anode voltages some 50% higher than conventional klystron guns and so expertise gained on very high voltage radar tubes was used to ensure proper clearances between electrodes. Stray capacitance in the grid support structure was minimised in order to minimise losses.

The grid-to-cathode space of the electron gun forms the end of a long and complex RF transmission line from the RF input connector of the input cavity system. This has a strong effect upon the final frequency range of the IOT input cavity system and needed careful consideration.

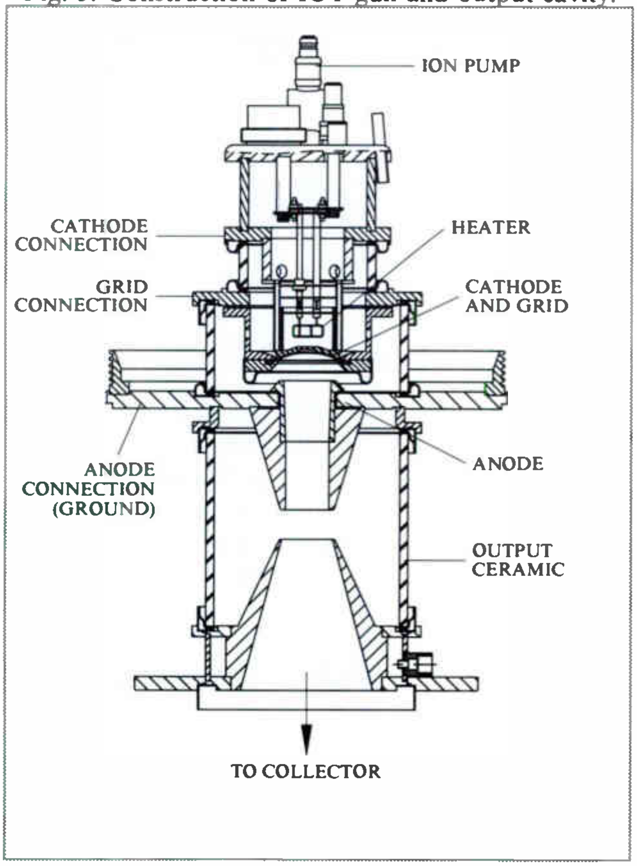
The grid-to-cathode distance is crucial to the physics of the IOT and great care was taken to establish a method of supporting the grid which would allow this distance to be maintained at the required value when the cathode was raised to its operating temperature of about 1000 °C.

At this point in the design it was necessary to consider perhaps the most important component - the grid itself. The thickness, size and shape had been defined. But what material should be used for the grid?

After consideration of the alternatives, it became obvious that a refractory metal grid (perhaps molybdenum) was feasible but that this was likely to be more expensive, weaker and less mechanically stable than the final choice - pyrolytic graphite.

Fortunately, we had full access to an in-house pyrolytic graphite production facility.

Fig. 3. Construction of IOT gun and output cavity.



PRODUCTION TECHNIQUES FOR PYROLYTIC GRAPHITE SHELLS AND GRIDS

The material chosen for the IOT grid was pyrolytic graphite. This material has the unique advantage that its strength increases with temperature up to above 2500 °C, whereas the strength of pure metals universally decreases as the temperature increases. This gives the designer the ability to produce a thin grid, with fine grid wires which may be accurately positioned yet will retain their position and shape when raised to operating temperatures in the region of 1000 °C.

A hydrocarbon, typically methane, is fed into a low pressure chamber containing a graphite rod of the correct form to produce the required graphite 'shell'. A 'shell' is typically a graphite cylinder with a closed, shaped end. The cold gas is passed into a hot zone in the reactor, which is heated by an RF eddy current system from outside the vessel. An ordered carbon structure is required, as opposed to an amorphous

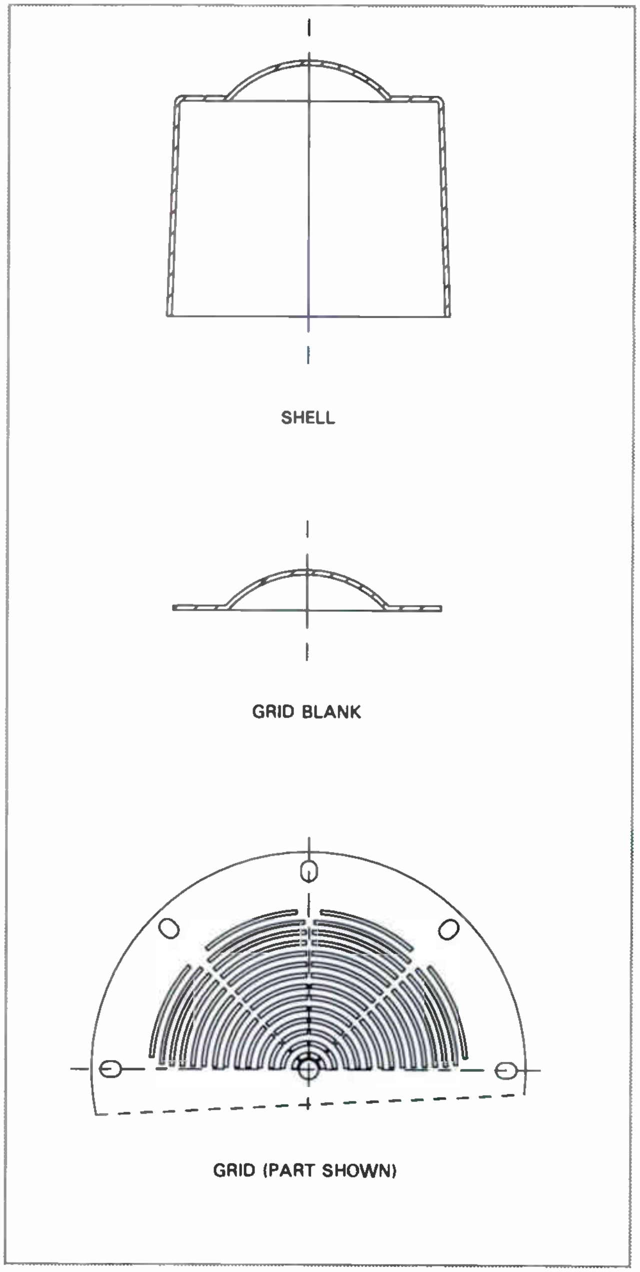


Fig. 4. Drawings showing a graphite grid shell, grid blank and a finished grid.

'sooty' structure; so the temperature is set to about 2000 °C and the pressure within the reactor to around 10 torr. The graphite shells so produced have a layered structure with very anisotropic properties and are physically very strong.

The shells are then machined to the desired shape and size and the holes are cut into the resulting grid blank using a laser under computer control. Final inspection is performed under a microscope before the grid is fitted into the IOT gun.

Considerable experience was available to the IOT team on pyrolytic graphite grids with concentric cylindrical geometry, but some extra work was necessary to establish the exact shape of mandrel necessary for the production of a spherical grid blank with an annular surround.

It was then necessary to specify and purchase the goniometer head required to modify the existing computer controlled laser cutting system to cut accurately and cleanly on a spherical surface. A grid shell, grid blank and a finished grid are shown in Fig. 4.

INPUT DESIGN CAVITY PHILOSOPHY

The initial IOT input cavity concept, which differs radically from that of the klystrode, is shown in Fig 5. A klystron-type RF cavity box with coupled sliding tuning doors is connected between cathode and grid of the tube. Since both of these electrodes are at the (30 kV) beam potential, it was necessary to maintain the body of the cavity and its tuning mechanism at ground potential by using RF chokes to prevent the leakage of RF energy, whilst holding off the full beam voltage.

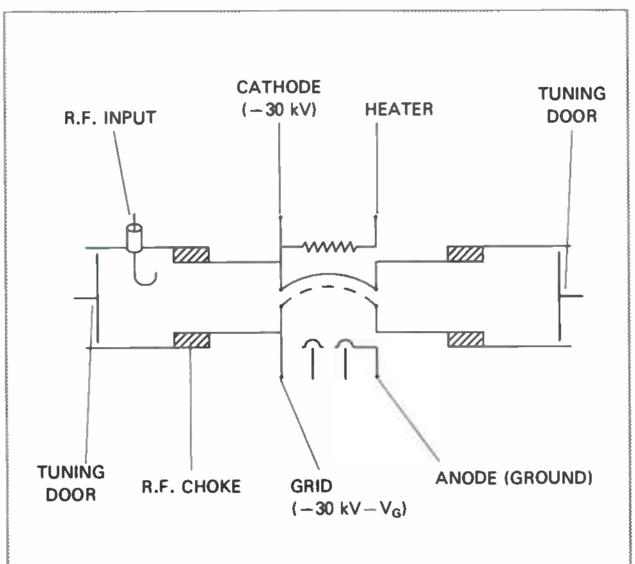


Fig. 5. Diagram showing the main components of the initial design of IOT input cavity.

It was decided from the beginning to try to cover the full UHF-TV frequency range from 470 to 860 MHz with a single IOT and input cavity. The RF chokes were designed with this in mind. They consist of silver-plated metal components, vacuum moulded with a low-loss high insulation resin compound to form a compact input cavity base which is tested to hold off 45 kV, minimum.

A prototype cavity was constructed (Fig. 6) and tested on an early sample tube. Low RF leakage confirmed that the RF choke design was successful and that RF energy could be efficiently transferred to the grid-to-cathode space without RF instability. It was confirmed

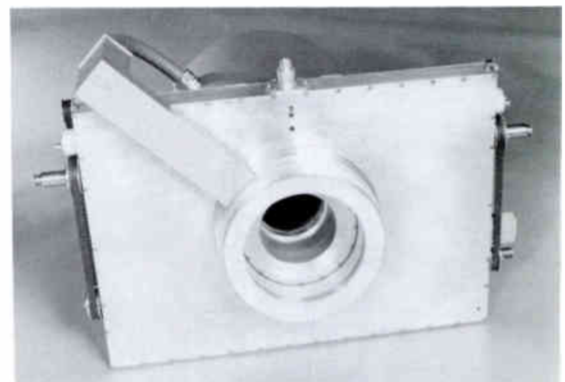


Fig. 6. A view from below of the first design of IOT input cavity.

that the gain of the IOT was about 20 dB and that good density modulation of the beam was achieved. However, the frequency coverage of the flat-box klystron type input cavity was inadequate and a new design had to be sought.

REASONS FOR THE FINAL CHOICE OF INPUT CAVITY DESIGN

The final version of the design is shown in Fig. 7. The flat box RF cavity has been replaced by a structure consisting of a cylindrical resonant cavity containing an annular sliding tuning door. The cavity is folded at the IOT electron gun end in order to make contact with the tube via the RF choke structure, which remained almost unchanged.

At one point on the grid connection within the RF choke, an insulated high voltage cable is connected and exits so that the minus 30 kV DC (and the grid bias voltage) can be supplied to the grid. This cable is screened and is fitted with RF chokes to prevent UHF or video energy passing from the IOT tube into the transmitter power supplies.

RF energy is fed via a coaxial cable to the input cavity which is excited by means of a loop antenna carried by the annular tuning door. Tests have shown that this

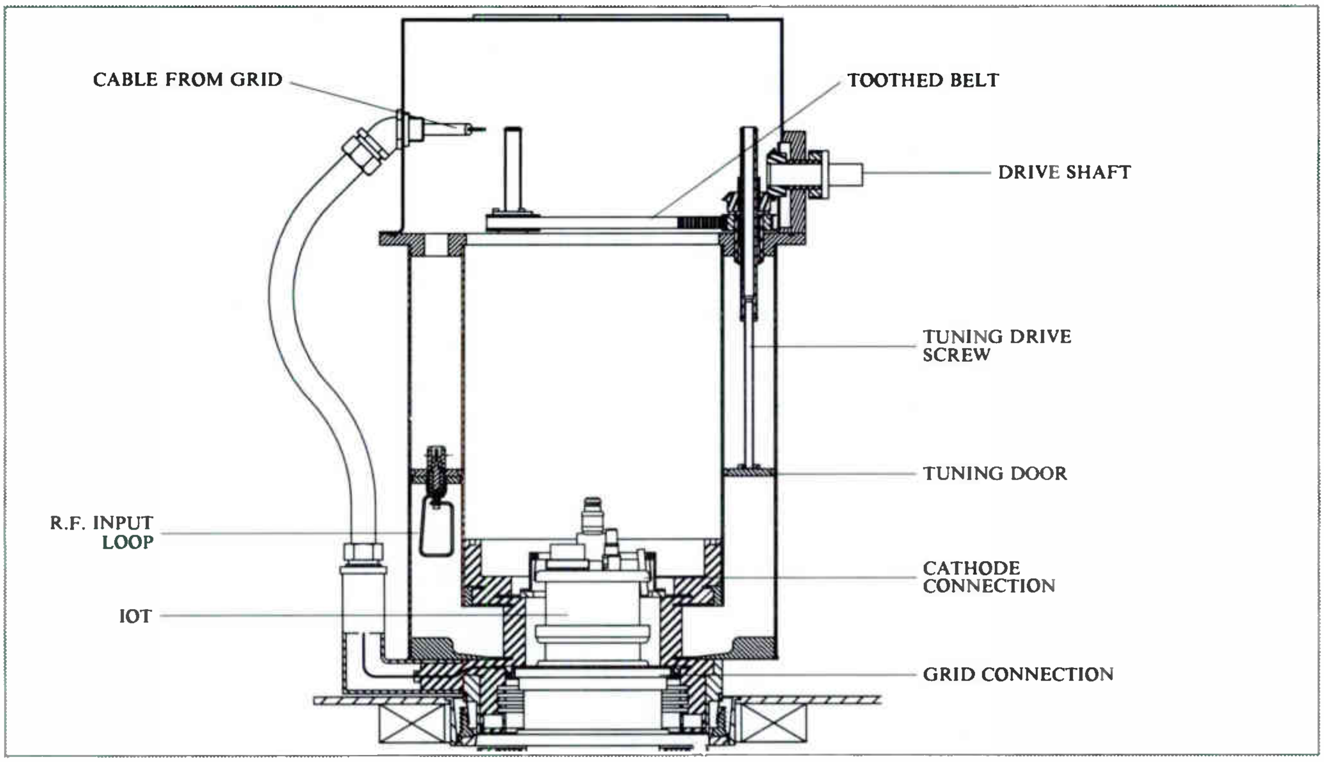


Fig. 7. shows the important features of the chosen design of IOT input cavity.

cavity, in its final form, is capable of covering the whole of UHF-TV Bands 4 and 5 in two modes. A typical tuning curve is shown in Fig. 8.

The heater, cathode and ion-pump cables for the tube were re-designed to feed to the IOT electron gun via the cylindrical space in the centre of the input cavity (see Fig. 7). The grid lead (from the RF choke area) re-enters the space above the cavity via an RF screened

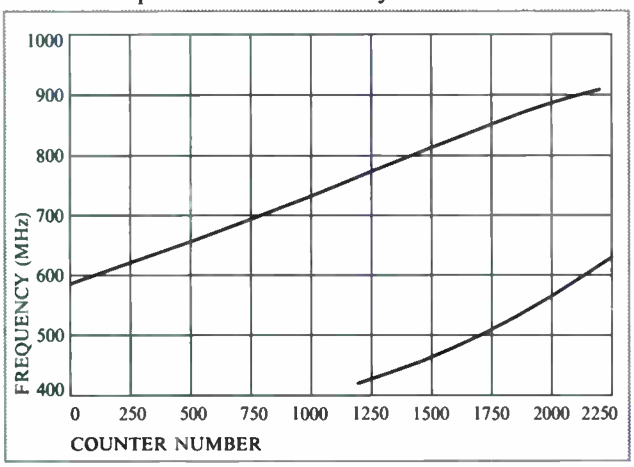


Fig. 8. Typical tuning curve for IOT input cavity.

safety cover positioned on top of the cavity. From the top 'lid' of this cover, a detachable connector panel (which becomes permanently connected to the transmitter) terminates a screened umbilical cable which carries all the power supplies from the transmitter to the IOT.

Cooling air for the IOT electron gun and grid connection area is also fed in an insulated air pipe down the inside of the input cavity.

The tuning door of the cavity is driven by three two-stage tuning screws coupled to the outside world via a rubber toothed belt and bevel-gear drive. A mechanical turns-counter is provided to relate cavity door position and frequency.

The single tuning handle is the only user-adjustable control on the input cavity, which makes for a very user-friendly input cavity system.

A DOUBLE-TUNED OUTPUT CAVITY SYSTEM FOR THE IOT

Although initial testing of development tubes was completed using a single conventional klystron output cavity, it had been realised from the beginning of the design activity that the IOT would require a double-tuned output cavity system in order to provide the instantaneous bandwidth needed for current specification UHF TV transmission. This is because the IOT lacks the intermediate cavities which, in the conventional klystron, are stagger-tuned to provide the bandwidth. It was clear that not only was a double-tuned output cavity system required but that the chosen system would need to be capable of being up-graded to provide any extra bandwidth needed by TV systems of the future.

This presented a tough design challenge since, although it is reasonably easy to design such a system to work over a restricted frequency range, it remained essential to the overall design philosophy of the IOT that the output cavity system should cover the whole of the UHF TV frequency bands with a single user-friendly unit.

After a number of proposals had been tried and rejected, a design was produced which uses what is basically a pair of conventional klystron cavities, coupled together and supported from the IOT tube and the magnet frame. The current version of the design is shown in Fig. 9. with the IOT tube in position.

the RF output feeder system via a standard three and one eighth inch EIA interface.

The resulting output cavity system has been developed to the point where instantaneous bandwidths of at least 8 MHz have been confirmed during 'hot-tests' of complete IOT systems. Further work is under way to extend our theoretical understanding of the output system and modified samples have yielded 'cold-test' bandwidths of around 20 MHz, see Fig. 10.

Since RF voltages of the order of the IOT beam voltage can be expected in the output cavity system, RF arc-detectors are fitted in both the cavities, in line with klystron practice.

Cooling of the output cavities is by means of filtered forced-air. Air enters the primary cavity, passes over

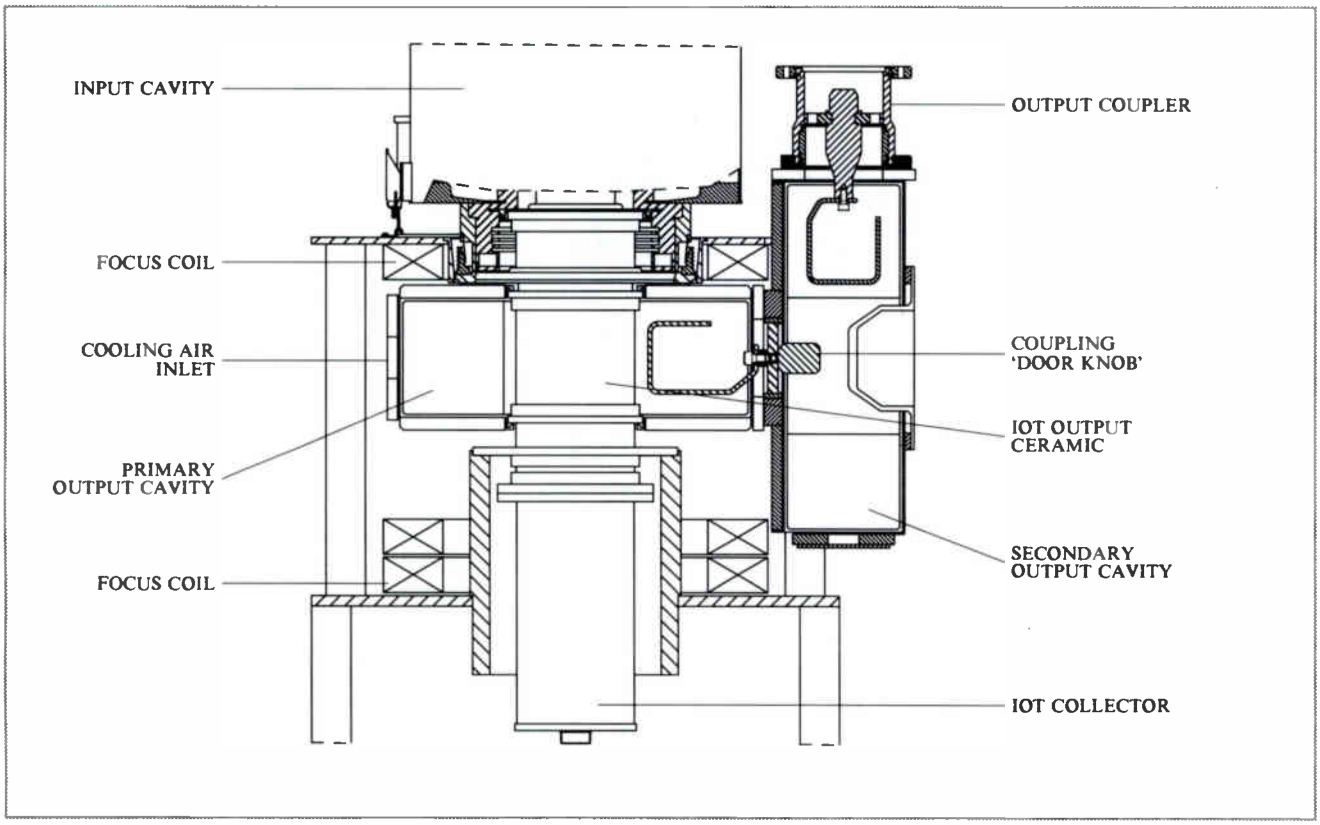


Fig. 9. showing the IOT in place in the magnet frame with the output cavity system fitted.

The primary output cavity is clamped around the output ceramic of the IOT, exactly as on a conventional klystron and contains an RF coupling loop which may be rotated about a horizontal axis to adjust the degree of coupling through a very short transmission line section (designated the 'coupling-hub') into a secondary cavity via a door-knob type antenna. The secondary cavity contains a dome structure adjusted in size so that the cavity can be made to cover the required frequency band. An output coupler, of standard klystron design with a loop antenna, connects the secondary cavity to the output ceramic and coupling loop then exits via holes in the coupling hub assembly into the secondary cavity. After cooling the secondary cavity, some of the air exits via the contact fingers on the tuning doors whilst a small proportion traverses the output loop and coupler, exiting at a stub-pipe fitted for that purpose.

The output cavity system has only two tuning and two coupling controls, all of which are fitted with either digital indicators or scales so that a tuning set-up for a given channel can be predicted and repeated. In use,

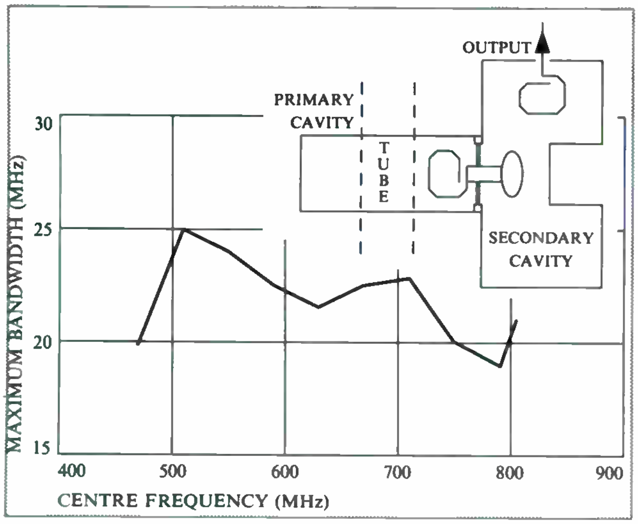


Fig. 10. Graph showing bandwidth available from experimental modification to IOT output cavity system.

after a brief period of familiarisation, most operators new to the IOT have found the output cavity system quick and easy to adjust. Typically, a few minutes only are required to set the output to a given channel.

TYPICAL PERFORMANCE OF THE IOT

The current version (January 1992) of the IOT tube type IOT7340 is shown in Fig. 11. The tube has a water-cooled output drift tube and collector and is rated at the 40 kW power level. Table 1 gives an outline of the results obtained from a typical sample tube under the operating conditions stated. The tube was driven with a linear drive signal (without pre-correction) for this test; the results therefore display the non-linearity of the IOT itself.

Channel	Low	Middle	High	
Beam volts	28	28	28	kV
Mean beam current	1.17	1.15	1.15	A
Peak beam current	2.55	2.50	2.45	A
Grid bias	–72	–72	–72	V
Grid current	5	1	0	mA
Body current	6	6	8	mA
Input power	344	399	307	W
Output power	40.4	40.3	41.8	kW
Linearity	77	77	77	%
Diff. phase	12	12	12	Deg.
Diff. gain	82	82	82	%
Fig. of merit	123	125	121	

Table 1. Typical uncorrected performance of IOT7340

Fig. 11. Shows IOT tube (above) and complete 40 kW IOT system IM7340.

Station	Channel	Location	NTSC/ PAL	Rated Power (kW)	Mode Common/ Separate	Number Of IOT's	On-Air Date	Transmitter Manufacturer
On-Air Stations								
WGBY	57	Springfield, MA	N	60	C	2	6/91	Comark
WOUC	44	Cambridge, OH	N	30	C	1	8/91	Comark
WGLN	54	Erie, OH	N	60	C	2	9/91	Comark
WFPT	62	Frederick, MD	N	70	C	2	10/91	Comark
WSTR	64	Cincinnatti, OH	N	240	S	4	10/91	Comark
Ballarat	A39	Victoria, Aust.	P	30	C	2	12/91	TTC
Albury	A39	Victoria, Aust.	P	20	C	2	12/91	TTC
Shepparton	A46	Victoria, Aust.	P	20	C	2	12/91	TTC
Horsham	A31	Victoria, Aust.	P	10	C	2	12/91	TTC
Under Installation (Mid-Jan 1992)								
WPBO	42	Portsmouth, OH	N	30	C	2	2/92	Comark
KTSF	26	San Francisco, CA	N	60	C	2	2/92	Comark
Projected Installations								
KZKC	–	Kansas City, MO	N	240	S	4	7/92	Comark
WACS	26	Dawson, GA	N	70	C	2	6/92	Comark
WNPB	24	Morgantown, W.VA	N	60	C	2	7/92	Comark

Table 2. Installed and projected transmitter base as at mid-January 1992.

THE IOT IN SERVICE

There has been considerable market interest in the IOT and production IOT systems have been shipped to transmitter manufacturers in quantity since May 1991. The installed and projected transmitter base as of mid-January 1992 is shown in Table 2. Up to mid-January 1992 more than 30 000 transmission hours have been achieved.

ACKNOWLEDGEMENTS

The authors wish to thank Steven Aitken, Sandy Anderson, Steven Bardell, Dr Edward Sobieradzki and Dr David Wilcox, whose technical contribution to the IOT Tube Project has been considerable; also Dr Alan Pickering whose work has added greatly to our theoretical understanding of the input and output cavity structures.

Thanks are also due to Comark Communications Inc. and Television Technology Corporation for data on IOT transmitter installations.

USING TETRODE POWER AMPLIFIERS IN HIGH POWER UHF TV TRANSMITTERS

Joe Wozniak
Acrodyne Industries, Inc.
Blue Bell, Pennsylvania

Over the past few years, UHF broadcasters and broadcast engineers have demanded more efficient transmission technologies to replace aging and inefficient klystron systems in an effort to contribute to the imperative of reducing the overall cost of UHF broadcasting.

Much talk and press has been devoted to more efficient "new technology" variations of the klystron--and rightly so. Tetrode technology, meanwhile, has been advanced worldwide by Acrodyne and Thompson Tubes Electroniques in higher power applications replacing heretofore klystron applications. It is now feasible to produce practical, cost effective, highly efficient tetrode transmitters for up to 30 kW of UHF power using a single tube and 60 kW of power by paralleling two tetrode amplifiers.

The purpose of this paper is to present the recent history of developments and to describe such a transmitter design, in particular, the use of combined amplification. Emphasis will be on the exciter and drive designs including the concurrent progress in solid state UHF amplification (which goes hand-in-hand with the tetrode developments) as well as the differences in the performance of the tetrode and the design of the PA stage compared to the klystron. An overview of typical operation and maintenance will also be presented.

INTRODUCTION

Baseball announcers are prone to use old cliches. One of the more hackneyed expressions is "it's a whole new ball game." Well, it may be over used but the phrase is especially applicable to what's happening in the U.S. broadcast industry today. Rising programming costs and declining revenues are squeezing profits and broadcasters are looking for ways to cut costs. Fortunately, technology provides some of the solutions. And UHF broadcasters are scrutinizing high efficiency tubes as a potential means to make an impact.

A little history

In the early days of UHF broadcasting, while tetrode transmitters were initially used to produce 1kW, 2kW or even as much as 10kW of power, klystrons soon became the amplifier device of choice for several reasons. Klystrons had the advantage of higher power handling and high gain which allowed UHF broadcasters to approach the coverage areas of their VHF counterparts especially in metropolitan areas. The reputation they have earned is that it is a reliable, long life device--but at a cost. To obtain the necessary linearity, they are operated in the inefficient class A mode.

The tetrode, meanwhile, was remembered for short life, limited power handling which required multiple device amplifiers, and poor stability. But those tubes were designed 30-40 years ago and much research and development has taken place since then, principally by Thomson Tubes Electroniques of France. About 25 years ago, Thomson developed and incorporated two production techniques which have evolved to make an efficient, long life, stable, 42kW peak visual tetrode available-- a tube designated the TH-563. These techniques are Pyrobloc® pyrolytic graphite grids, and the Hypervapotron® cooling system. Today, hundreds of sockets worldwide use these types of tetrodes. Converting to a tetrode type transmitter can cut the broadcaster's transmitter power consumption by 50%-60%.

TUBES AND CAVITIES

These are truly modern devices and RF circuits whose unique construction has accounted for proven long term stability and long tube life.

Construction

The cathode is constructed of thoriated tungsten,

® Pyrobloc and Hypervapotron are registered trademarks of Thomson Tubes Electroniques directly heated and operated at an approximate temperature of 1650 degrees C. Tungsten wires containing thorine are carbonized at high temperatures yielding a monatomic layer on the surface increasing emission capability considerably. This process also results in good mechanical stability for long life.

Three main parameters determine the behavior of the grids. They are thermal emission, secondary emission and mechanical stability. Thomson's solution is the use of pyrolytic graphite also called oriented graphite. Studied and developed in the early 1960's, it wasn't until about 1967 that suitable material was found and a process was developed to produce one-piece grid castings from which the grid structure was formed without welding.

The properties of this material have made it an almost ideal material for use in UHF grids. These are: thermal conductivity-on the order of copper; low thermal expansion; good mechanical strength which increases with temperature; low secondary emission; and low and constant electrical resistivity. Pyrobloc® grids make these tetrodes very stable, requiring very few adjustments throughout tube life.

The anode is constructed of copper whose structure depends on the amount of power that must be dissipated. Since the cathode is operated at ground potential, the anode must be insulated from ground and the insulation must be able to withstand the greatest anode potential. Anode potential is less than 10KV. High velocity air can be used in VHF tubes for up to 30kW of power; in UHF only to 10kW of power. To achieve higher output levels a combination of air and water circulating cooling systems are employed.

The cathode filament and control grid are cooled by forced air, the screen by circulating water, and the anode by the Hypervapotron® effect. This patented cooling system is a water circulating system that takes advantage of the complex boiling process that occurs within narrow slots on the outside surface of the anode and, the instant condensation of the water vapor produced. A natural pulsating action is established when the water boils, is expelled at high velocity and quickly condenses in the flow of water. A simultaneous suction draws water to refill the slot.

There are several important advantages to hypervapotron cooling. This cooling method significantly reduces the amount of forced air cooling and the cleaning necessary with high

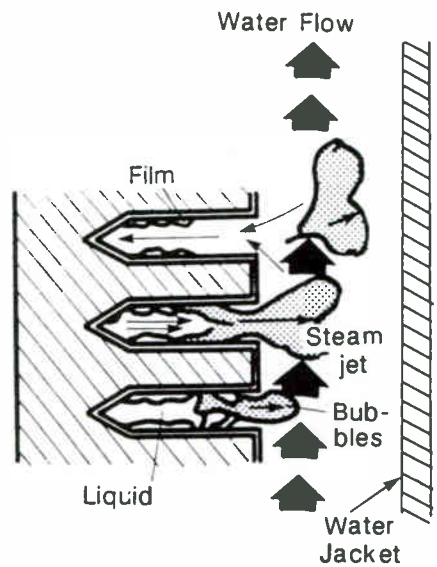


Figure 1. Anode cross section illustrating the Hypervapotron® effect

Figure 2. Thomson TH-563 Tetrode

volume air cooled systems. Ambient noise is minimized as well. A low tube operating temperature yields excellent stability and long device life.

The coaxial construction of the tube results in a small, light weight device. The Thomson TH-563 tube, capable of up to 42kW of output, is only about 7.5" high by 5" in diameter weighing only 14 pounds.

RF Circuit Assembly The Thomson cavities are 1/4 wavelength coaxial designs made of silver plated machined brass with beryllium copper electrical contacts, teflon insulators, and kapton blockers. For freer air circulation and better elasticity, improved contacts composed of wire wound on a ribbon of copper alloy are used.

A single cavity may be tuned to any UHF channel. There are two input tuning controls and three output tuning controls. Output tuning controls are tune, load, and coupling. There is a separate neutralization control which is one time factory set. The channelized cavity is typically tuned for a 12MHz bandwidth at the -1dB points.

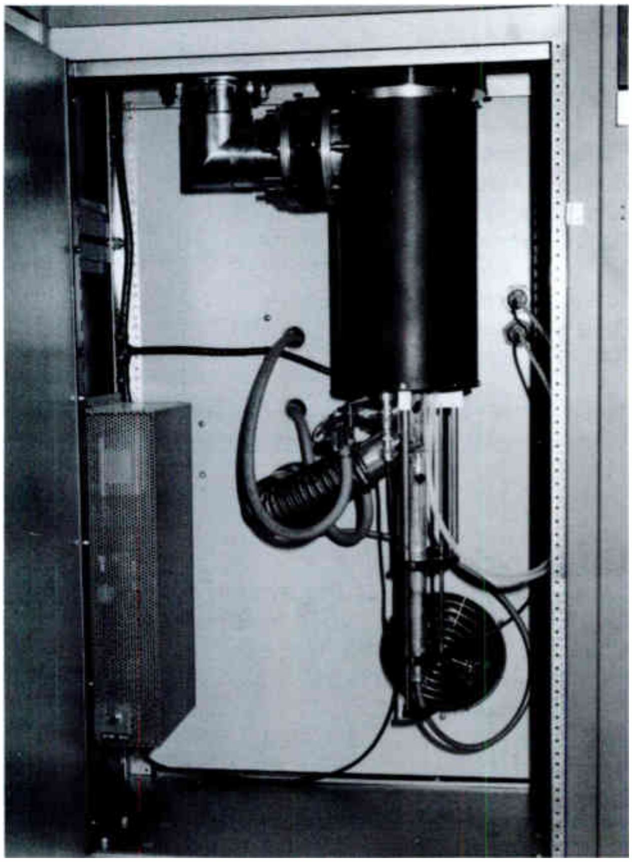


Figure 3. Thomson TH-18550 Cavity

Performance
These tubes are operated in the highly linear, highly efficient, AB_1 mode. The intrinsic linearity of the tube lends it very useful in simpler type transmitters which we call combined amplification systems. These are IF diplexed or low level diplexed transmitters that amplify the composite TV signal (aural and visual carrier together) in the final tetrode amplifier and typically, the drivers as well. Using this approach, the tubes are specified to produce uncorrected intermodulation distortion (IMD) of -48dB or less referenced to peak of sync. The Thomson TH-563 tube is now capable of producing up to 30kW of peak visual power, 10% aural power in combined amplification use. The linearity of the tube also permits the use of very simple exciters that require only a limited amount of precorrection to reduce the in-band IMD to -52dB or lower. Out-of-band IMD is lowered typically with four or five notch filters.

With a less complex exciter, these transmitters match or exceed the performance of the typical high power transmitter. In fact, the precorrection requirements of the transmitter depend almost entirely on the driver. Distortion of visual and aural parameters is typically less than 1%; just about as close to a transparent amplifier as can be imagined.

Efficiency,Gain
The anode efficiency is better than 40% (not to be confused with the nebulous term "figure of merit") resulting in low transmitter overall power consumption. Table 1 below shows measured figures from factory test data.

Peak Visual Output	For 10% Aural Power Consumption at 50% APL
10kW	21kW
25kW	46kW
30kW	55kW

Table 1. Tetrode Transmitter Power Consumption

The relative high gain of these tetrodes at 15dB has reduced the amount of drive power needed in the transmitter. The linearity, gain and efficiency of these tetrodes, simplifies the design of the overall transmitter making for a compact system as well.

In The Field Results

Hypervapotron® cooled tetrodes are designed for 8,000 to 10,000 hours life but the track record has shown that the effectiveness of this cooling method combined with the other construction characteristics yields tube life surpassing 20,000 hours.

As of this writing, seven systems have been built and delivered by Acrodyne utilizing the Thomson TH-563 tetrode. All of the original tubes continue to operate. Tube life has exceeded expectations in the two oldest systems with approximately 24,000 hours of service and 18,000 hours of service recorded. Figure 4 is an indication of the success through 1991. Operating hours continue to accumulate without a single failure.

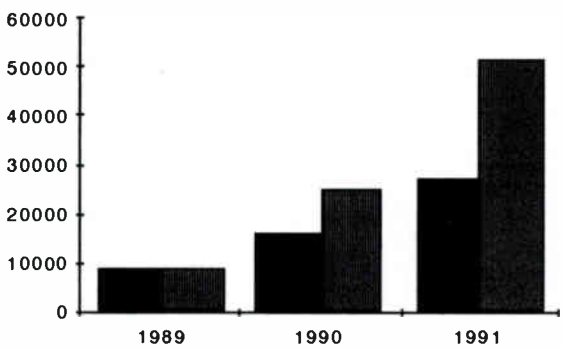


Figure 4. Estimated hours and accumulated hours of operation without failure, TH-563 tetrode

If an average life of 21,000 hours can be sustained, at a replacement cost of 16,000 dollars the TH-563 can be operated for about 2.8 cents per kilowatt hour (peak of sync plus 10% aural power).

Neither transmitter has required tuning since initial turn-on. These efficient transmitters were sold as 25kW units once again with an overall power consumption of about 46kW which further serves to minimize operating costs.

Tube Replacement

Tube replacement can be accomplished quickly because the tube is so easy to handle. The use of self-sealing connectors facilitates disconnection of the water lines to the anode. Actual tube removal and replacement can be completed in about 15 minutes and full power achieved in about 15 minutes more (this is without resweeping; it is highly desirable to sweep the amplifier to assure that proper response and gain is maintained).

TETRODE AMPLIFIER DESIGN

Power Supplies

Because of the low anode voltage, the power supplies are relatively simple and small. This eliminates the need for outdoor type or oil filled supplies.

Voltage regulation is required for the filament/cathode which must be kept at ±2% of nominal. A ferroresonant type regulator is used for this purpose. The control grid and screen grid supplies are DC regulated.

Anode voltage regulation is not critical to performance or gain, therefore, automatic voltage regulators are not needed. Choke and capacitor filtering and transient protection are necessary.

The power supplies for a 30kW transmitter are designed for the tube operating parameters shown in table 2.

Tube Parameters	0% APL	50% APL
Filament Voltage	4.9VAC	4.9VAC
Filament Current	185A	185A
Bias Voltage	-105VDC	-105VDC
Bias Current	-89mA	-39mA
Screen Voltage	825VDC	825VDC
Screen Current	90mA	54mA
Plate Voltage	8900VDC	8900VDC
Plate Current	5.6A	4.6A

Table 2. TH-563 Typical Tube Operating Parameters at 30kW peak visual, 10% aural output

Heat Exchanger/Cooling

The power efficiency of the system combined with the efficiency of the Hypervapotron® effect, minimizes the size of the heat exchanger. This is also an indoor type unit that circulates about 17GPM for 30kW output. The outlet water temperature typically reads about 80 degrees C; the temperature rise through the tube is about 6 degrees C.

A water purification system keeps the water resistivity at or above 200K-ohm-cm typically. The purifications system consists of a course particle

Figure 5. 30kW Single Tetrode UHF Television Transmitter showing solid state driver (left) and PA cabinet center and right

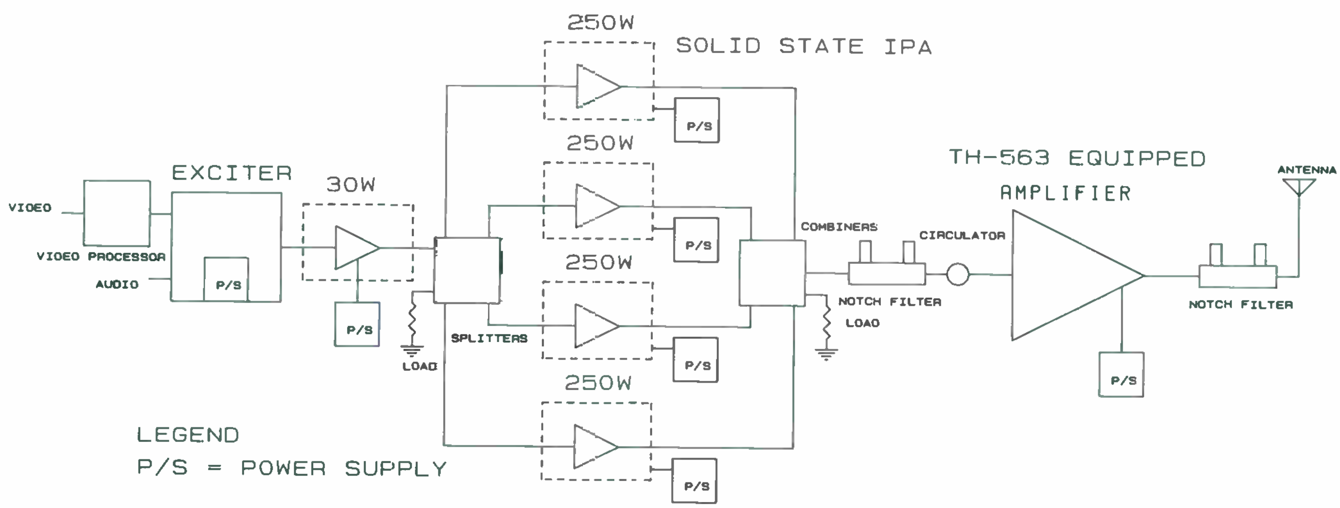


Figure 6. Block Diagram, 30kW Single Tetrode UHF Television Transmitter

filter and a chemical filter. The chemical filter removes sub-micron particles and dissolved minerals which would lend the water conductive. The amount of full flow of water passing through the filter may be varied from zero to 1GPM. Filter life varies inversely with amount of flow--typically about a year or so.

At the transmitter, the water from the heat exchanger is distributed three ways by an inlet manifold. Most of the water flow is directed to the anode; less than 1 GPM is needed for the cavity body and less than 1/2 GPM for the screen. Air cooling is also employed at low volume to cool the filament and walls of the cavity.

System pressure in a 30kW system varies between 30psi-37psi. (The high pressure interlock level is 60psi.)

For those broadcasters concerned about cold weather operation, the system could be furnished with an emergency back-up AC power generator to run the pump and an immersion heater for the worst case scenario where building power has been lost and outside temperatures are below freezing.

EXCITER/DRIVER REQUIREMENTS

While a single tetrode driver is one choice, one which offers simplicity and lower cost, perhaps of greater interest today, is the use of solid state amplification through the driver stage. Solid state design has the advantage of very long device life and multiple, parallel amplifier design reducing or eliminating single point failures that could take the broadcaster off the air. The cost to produce a solid state UHF TV driver compared to the tetrode type system is very close at 1kW of power; beyond this point, tetrode equipment which is much less complex, is far more cost effective.

Solid State Drive

Paralleled bipolar transistors operated in the AB mode provide the power density needed to produce up to 1kW of UHF power in a single 22" wide cabinet which contributes to cost effectiveness and compact design. Recently, devices have been produced with sufficient linearity to operate these devices in the combined amplification mode also allowing further cost savings. Each stage is capable of about 10dB gain minimizing the drive power required to the final IPA stage.

To drive the TH-563 tube, solid state amplification through the IPA stage is accomplished by combining four 250W modules. As in VHF solid state designs that are widely accepted today, slide-out amplifier drawers for easy access and front panel diagnostics have been employed to reduce the drawbacks that go with the complexity of solid state equipment.

Exciter

The exciter features depend on the choice of tetrode versus solid state in the IPA stage. The use of solid state drivers necessitates a few more features due to the non- linear characteristics of the transistors.

A video processor consisting of a four stage differential gain and differential phase corrector is used. An IF linearizer, designed many years ago for the purpose, pre-distorts the composite TV signal for in-band IMD. Low frequency linearity and ICPM correction are added for single tube systems.

Figure 7. 250W Solid State Amplifier Module

OPERATION AND MAINTENANCE

Today, the use of CMOS and TTL logic to control the automatic or manual start-up and shut-down of the tetrode amplifier eliminates less reliable mechanical relays and facilitates "user friendly" diagnostics and display panels.

These transmitters can be started with the push of one button and, if all system safety and self-protection interlocks are closed, the transmitter can be at full power in about 30 seconds.

Fault recycling is also employed allowing the transmitter to attempt to restart if a momentary VSWR, plate (anode) or screen fault has caused a shut-down.

Solid state driver control logic is simpler since there is no warm up or sequencing of events.

The benefits of tetrode stability, Hypervapotron® cooling, and solid state drive all contribute to lower ongoing transmitter maintenance compared to older klystron systems and other "new technology" devices. In particular,

1. RF sweeping of the cavities is not required except after removing the tube and replacing it.

2. The closed loop cooling system equipped with a water purification system eliminates the need to replace water and requires only an occasional replacement of the chemical filter.

3. Broadband solid state design eliminates the need for retuning. Transistors are replaced by simply unsoldering the old device and soldering in a new one.

CONCLUSION

To meet the demands of today's UHF broadcaster, manufacturers have virtually abandoned the standard klystron for more efficient tubes. With a proven record of success and higher power capability, the tetrode now also fits the bill offering power efficiency, low replacement cost, and long life. For best cost effectiveness, transmitters can be built in the combined amplification configuration without trade-off in performance or in separate aural and visual designs. With the developments in UHF power transistors, power levels up to 30kW, 10% aural can be achieved employing as few as one tetrode and up to 60kW with two tetrodes.

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UPGRADING UHF TRANSMISSION LINES AND ANTENNAS— TWO CASE STUDIES

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ABSTRACT

Planning consideration and problem solving are focused on in a study of two recent upgrades by UHF stations. One is a transmitter upgrade to a higher power and the subsequent changes in transmission line and antenna. The other is the replacement of the existing transmission line and antenna with new but cost effective products.

INTRODUCTION

For many UHF stations, changing demographics and/or aging equipment are creating challenges for engineers and general managers alike. Antennas with inadequate radiation patterns no longer cover the entire market area. But how critical is it to maintain the licensed ERP (effective radiated power) if the desired patterns reduce the antenna gain? And if more ERP is needed, how do you accomplish getting more power to the antenna? What if your transmission line and antenna have finally reached the downhill side of their life cycle and you not only want to replace them, but also upgrade them for lower maintenance requirements? The following case studies will describe how two stations answered these critical questions.

New Patterns and Higher ERP

WDRB in Louisville, Kentucky had an aging RF plant. Utilizing a 30 kW transmitter, 6-1/8 transmission line and a directional top mounted antenna, a peak ERP of 1170 kW was obtained. With new technologies and techniques in transmitter, transmission line and antenna designs available, an upgraded and improved RF plant was designed.

The desire to increase the overall coverage area required both higher ERP and an omnidirectional pattern. The maximum allowed ERP of 5000 kW was the goal which resulted in a need for a 240 kW transmitter. The required transmitter power dictated that waveguide would be used for the transmission line.

The tower size, a 943 ft. tall, 6 ft. face Dresser, meant that windloading would have to be kept to a minimum when choosing the new transmission line and antenna. Therefore, circular waveguide was chosen as the transmission line in order to present a cylindrical surface for windload calculations. Installation of the 12 ft. long waveguide sections would be difficult because of the tower dimensions, but it was determined that the sections could be taken in through the top of the tower and then stacked from the bottom. This proved to work out very well.

Once the waveguide and transmitter were chosen, it was determined that there would be excess transmitter capacity after reaching the 5000 kW ERP for horizontal polarization. Therefore, in order to improve reception on television receivers using rabbit ears or loop antennas, it was decided to add a vertically polarized component to the antenna. To maximize the effectiveness of the vertical polarization, a directional pattern focused on the Louisville metropolitan area was chosen. This resulted in higher vertically polarized signal levels in coverage areas where the majority of viewers were located.

A Cost Effective Upgrade

WPNE-TV in Green Bay, Wisconsin is a part of the Wisconsin ETV system. This station was the last in a series of station upgrades and presented a unique problem. The existing antenna was a series of three panel arrays mounted around a 10 ft. face tower in order to produce an "omnidirectional" azimuth pattern. It was desirable to replace this system with a single radomed antenna designed for minimum maintenance but maintaining or improving the existing coverage area. To accomplish this, a study of the effects of the tower on the antenna radiation patterns was performed. [1]

In addition to determining the pattern and mounting requirements for the antenna, a decision had to be made regarding the existing 6 1/8 transmission line. At previous upgrades, a completely new transmission line system was installed. At WPNE-TV, however, an inspection determined that the outer conductors and hangers were in good condition. Therefore, it was decided to replace only the inner conductors and connectors using a bellows style inner conductor replacement kit from Andrew Corporation [2]. After installation, the system would have the higher performance of a premium transmission line and the radomed antenna.

SUMMARY

Two recent upgrades have been described utilizing different methods to obtain the same goals of high reliability and maximum coverage to their viewing audience. The following exhibits illustrate the changes made and the benefits derived from the new systems.

Acknowledgements

Special thanks to Mr. Glen Cook of WDRB and Messrs. Bill Woods and Jim Sheetz of the Wisconsin Educational Communications Board for their input and assistance.

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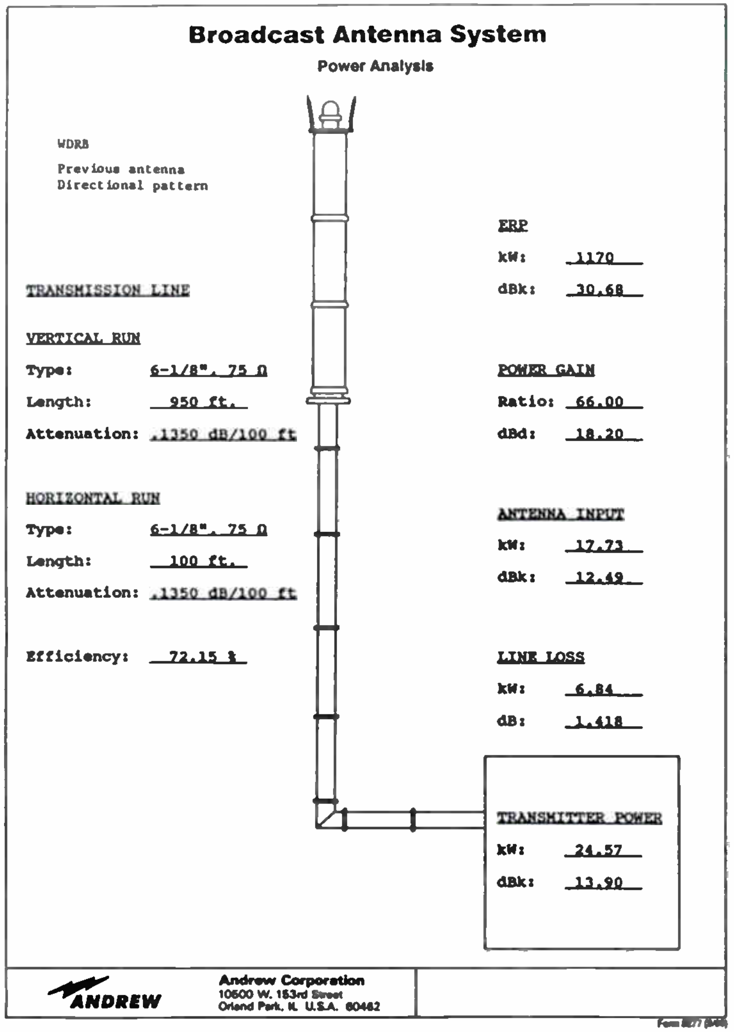

Exhibit 1

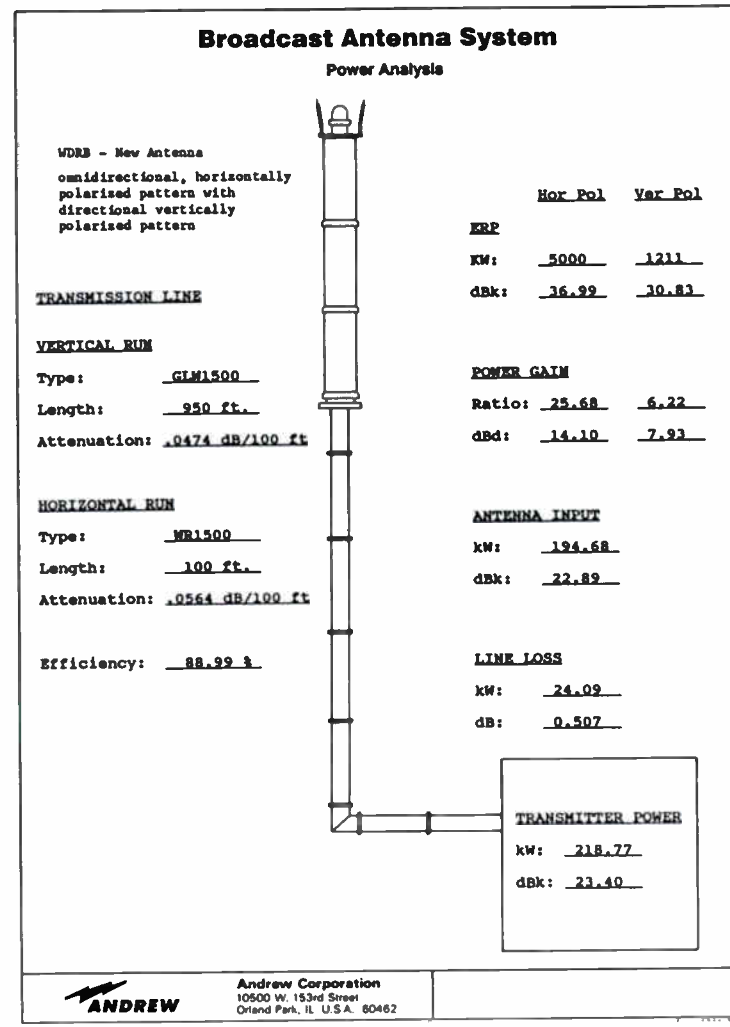

Exhibit 2

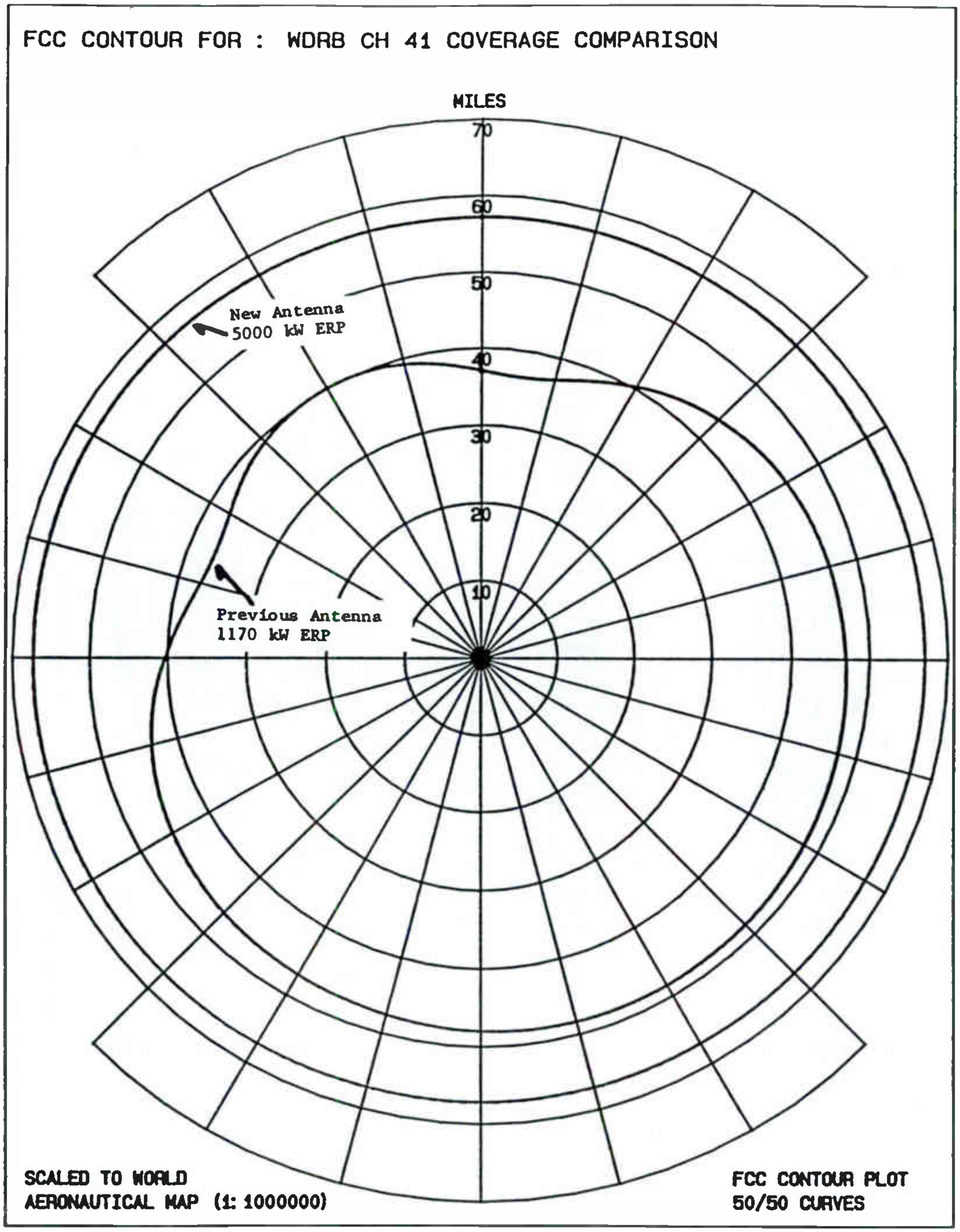

Exhibit 3

GRADE B Contour Comparison

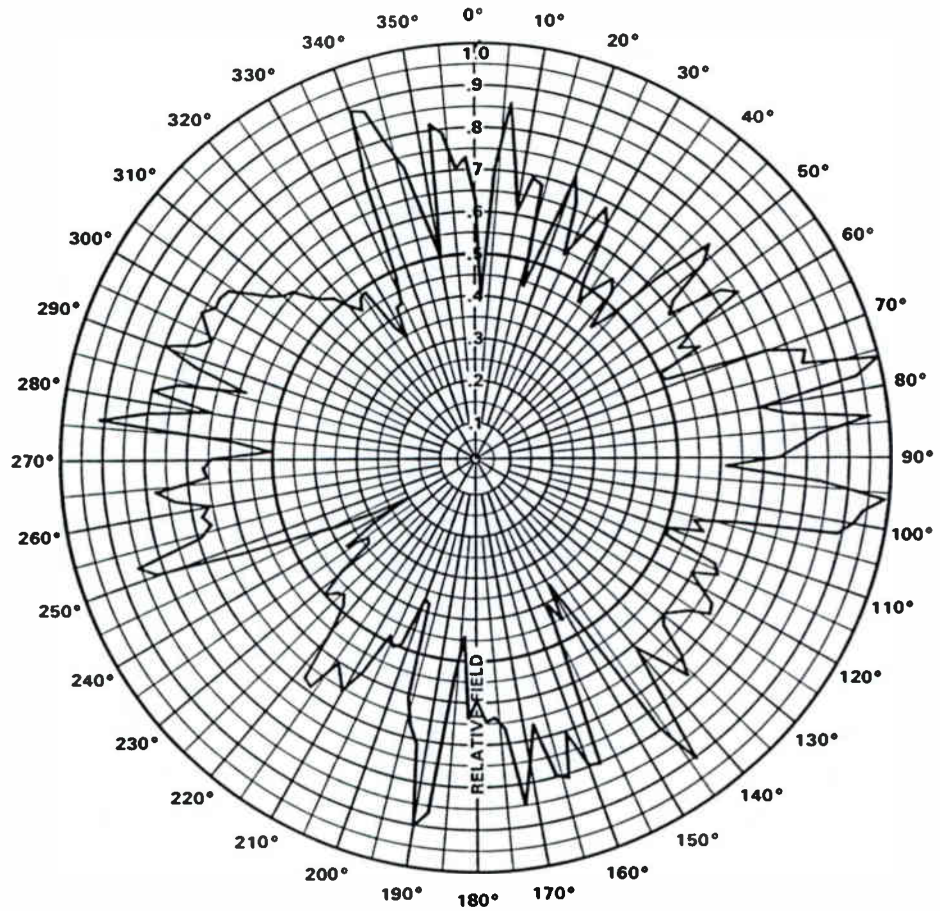

Exhibit 4

WPNE-TV previous antenna mounted as three panels around a 10 ft. face tower. Note sharp minima and ragged scalloping.

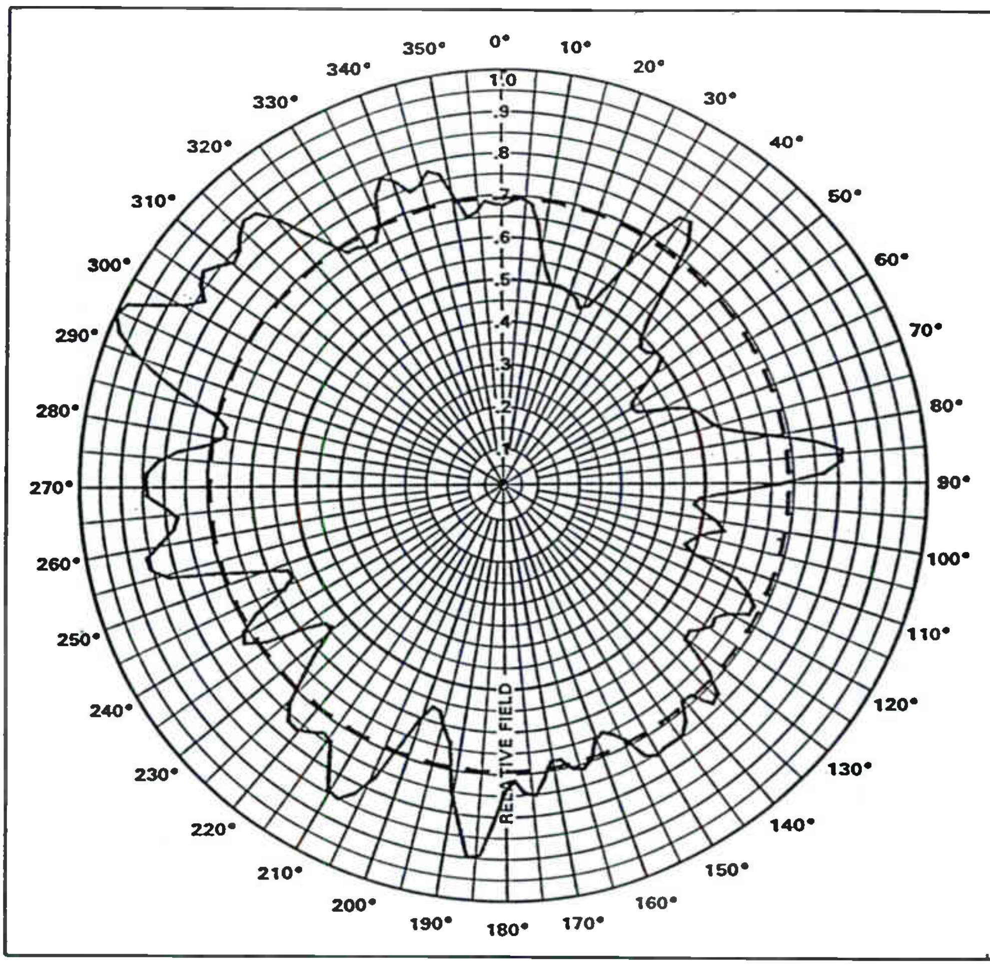

Exhibit 5

WPNE-TV new sidemounted omnidirectional slotted array. Note smoother scalloping and higher level minima. Dotted line is equivalent pattern for top mounted omnidirectional antenna.

BROADBAND UHF TV COMBINERS FOR THE AUSTRALIAN EQUALIZATION PROGRAM

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ABSTRACT

Recent regulation changes in Australian broadcasting have brought about the requirement for new UHF-TV facilities throughout the country. This program, known as equalization, has meant the changeover from VHF single channel sites to UHF installations with up to five channels.The opportunity to integrate new services both public and commercial, into common broadband antennas using multi-channel combiners has provided significant savings in implementation of the program as well as reduced site complexity.

This paper provides background on the development of television in Australia and the station design philosophies. It describes the design, construction, and implementation of these broadband waveguide combiner systems. Installation photographs and performance data for some of the nine combiner sites are also presented. It is hoped that this paper will guide those contemplating multi-channel installations and provide initiative for others to consider their benefits.

AUSTRALIAN TELEVISION

Television broadcasting began in Australia in 1956 with the introduction of both government and commercial stations in most of the six capital cities. During the 1960's the government stations, known as ABC, spread throughout most of the populated areas and even into some of the more rural areas. Today, the ABC covers more than 95% of the population in Australia. In the early 1980's a second government station was introduced. The SBS (Special Broadcast Service) began programming in the capital cities on the UHF band, and at present is continuing to spread into the rural areas on the UHF band.

At the same time a second and third commercial station had been added at each of the capital cities (Where, it should be noted, most of the people in Australia live.). In general all of the stations had independent facilities from each other and from ABC. The commercial stations did however group together, one station in each city, to form three networks (seven, nine, and ten) all with common ownership, logos, and programming. In the smaller cities and rural areas there has only been one commercial station. These generally small budget stations carry programming from a variety of sources and are typically independently owned. The regional city station's coverage markets were isolated; they overlapped with neither the capital city stations coverage nor each others. Thus these stations had no commercial competition, competing for viewers with only the ABC. In most regions transmission facilities are shared with the ABC at government owned sites, however,.they typically have their own antennas. All of these services both government and commercial were on the VHF band.

This all changed with the introduction of the federal governments equalization program. Equalization provides the regional viewers the same choice of services as the capital city viewers and in effect brought city network programming to all areas of the country. In each area the number of commercials had to be increased from one to three. Because of the lack of spectrum , all new services had to be in the UHF band. VHF stations which occupied the 88-108MHZ band were also required to move to UHF to free up space for FM broadcasting. This program was to be carried out initially in the three eastern states of Queensland, Victoria, and New South Wales

The location of the new services was entirely the choice of the commercial operaters. Generally the new commercial stations chose to share facilities with the ABC facilities and to share in the capital expenses. Telecom Broadcasting was contracted by the government to undertake all of the expansion work. It was presented with project briefs for each site specifying the number of new UHF services to be accomodated, the frequency of these and the ERP required.

Further complicating the expansion was the requirement for growth of the SBS service as well as a future spare channel. Thus the requirements for combiners varied by site from a minimum of two channels to a maximum of four channels expandable to five. Table 1 shows the sites and the required facilities.that were to be upgraded in this portion of the equalization program.

SITE	BEFORE		AFTER				
	ABC	COM	ABC	SBS	-COM-	SPARE	TX POWER
MT. DOWE	7	9	7	(28)	31 34	40	2 X 20KW
MIDDLE BROTHER	1	8	6	(59)	62 65	68	2 X 15KW
MT. SUGARLOAF	5	3	48	45	54	34	3 X 20KW
MT. BARANDUDA	1		1	30	33 39	42	3 X 20KW
MT. MAJOR	3	6	40	34	43 46	49	4 X 20KW
MT. ALEXANDER	1	8	1	29	32 35	41	3 X 20KW
GOSCHEN	2	11	2	(44)	47 50	12	2 X 10KW
LOOKOUT HILL	3	6	11	30	33 39	42	3 X 30KW
MT. TASSIE	4		40	34	43 46	49	4 X 30KW

TABLE 1

INSTALLATION OBJECTIVES

The requirement to provide up to five separate services at sites that had only contained one or two services in the past, presented significant engineering problems. Each of the sites required revamping and new equipment. It was determined early on in the project that it was not possible or logical to provide separate transmission systems for each service, they must be combined. Thus each site required the following:

**Uhf Transmitters*
**Multichannel Combiners*
**Transmission Lines*
**New or Reinforced towers*
**Broadband Antennas*
**Increased Mains Power*
**Building redesign and increased systems capacities*

It was also decided that while all this new equipment was being installed the old VHF transmitters should be removed and new, more efficient and compact ones put in their place. This required the following additional items at some of the sites:

**VHF Transmitters*
**Transmission lines*
**Antennas*

The primary installation and operating objectives for the renovated facilities consist of the following:

***Maintain high reliabity**
***Allow completely isolated operation of all services**
***Provide redundant systems for emergency operation**
***Maintain high safety levels during installation and operation**
***Install new services with minimum disruption of the existing ones**

These objectives were maintained throughout the project.

COMBINER REQUIREMENTS

The objectives generated a specific set of combiner design requirements. The designs are slightly different for each site based on the number of inputs and outputs necessary, however, the overall goals are the same. The following items were determined to be necessary characteristics of the combiners:

***Power Handling:**
The combiners must be able to handle up to five 30 kw transmitters continuosly. The peak voltages generated by these five stations could reach values equivalent to a peak power of 750 kw.

***Reliability:**
The combiner must be highly reliable and not require maintenance.

***Expandability:**
The combiner must be capable of expanding to add additional channels at a later date.

***Multiple Switched Outputs:**
The combiner must contain a splitter/switcher to allow the combined output to be fed to different portions of the antenna. Should one portion of the antenna/feed line fail the stations can remain on air with the remainder of the antenna.

***Monitoring:**
A monitoring device for forward and reflected power on both inputs and outputs is necessary to prevent component or transmitter burn-up.

***Isolation:**
The stations must be able to operate completely independent of each other.

***VSWR:**
The combiner systems must have very low VSWR, less than 1.06:1 at all inputs.

***Insertion Loss:**
The combiner must have low insertion loss, less than 0.4dB on each channel.

COMBINER DESIGN

Following review of the combiner requirements, initial design of the combiners was undertaken. Since the design and specification of multi-channel combiners has been presented before[1,2], we will only discuss the considerations for this set of requirements.

From the beginning it was clear that in order to provide the total power handling capability it would be necessary to use waveguide combiners. Although some of the initial installations could have been done with coaxial systems, the expansion requirements necessitated waveguide. Since the systems would have coaxial inputs and outputs, waveguide to coax transitions would be used to get in and out of the combiner. Waveguide provides the highest available peak and average power rating for its size as well as the added benefits of lower insertion loss and ease of filter design

SIZE	PEAK	AVERAGE
*WR1800	95MW	900KW
*WR1500	65MW	600KW
*WR1150	35MW	260KW

Figure 1
Waveguide Power Handling

Components and complete systems can be easily manufactured, tested, and assembled in waveguide. All but one of the combiner sites was built utilizing WR1500 waveguide components. The Middle Brother, N.S.W. site used WR1150.

The requirement for expandability dictates the use of constant impedance combiners that can be easily expanded by adding other diplexer modules. One diplexer module for each additional channel is required. All of the system components have to pass the entire bandwidth, including expansion channels, thus very low VSWR and insertion loss are required over bandwidths as large as 168MHz. Significant amounts of new development work went into the tuning of components such as; W/G to coax transitions, hybrids, elbows, and "magic T's". The development of new waveguide to coax transitions with VSWR's less than 1.02:1 over almost the entire waveguide band presented one of the largest challenges.

The channel spacing for all combiners was kept at a minimum of two 7MHz channels, i.e. channels 30 and 33. This relatively close spacing requires the use of four section band-pass filters. Iris coupled, Tchebycheff design filters provide low insertion loss and VSWR on the single channel passband and excellent rejection at the stop channels. Figure 2 shows a typical response plot for one of the filters.

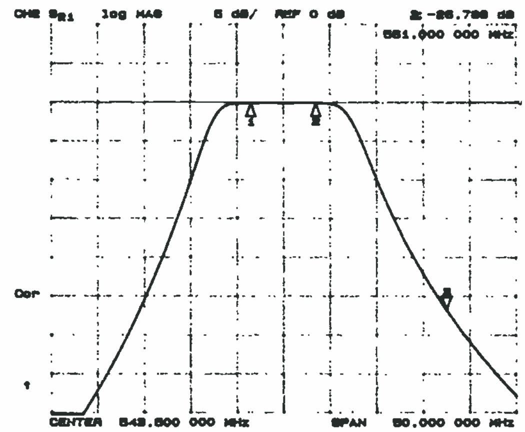

Figure 2
Typical Filter Response

The combiners utilize input filters on the narrowband inputs to provide additional isolation for this channel. This approach requires N-1 diplexer modules to combine N channels. The input filters were of the same design as the filters used in the diplexers with somewhat wider passbands.

The output systems used in the combiners consist of two different types. Four combiners utilize automated coaxial splitter/switcher systems. These systems provide four equal-amplitude, equal-phase outputs. The multi-channel signal is split four ways by three "magic T" hybrids and then fed to coax switches. The switches provide the capability to place one of the antenna/feed lines off air while the stations remain at full power on the rest of the antenna. The remainder of the systems utilize a splitter/patch panel arrangement to provide two outputs. The patch panel allows combined power to be placed through the splitter to both antennas or in a bypass mode to either of the antennas. The patch panel has three links one waveguide and two coax. Each has quick disconnects and can be easily switched.

Monitoring of the combiners is performed by two different types of systems. For the four output splitter/switcher combiners a computer controlled power monitoring system was designed. The system monitors and displays both the forward and reverse power of each input and output. This adds to a total of 18 separate power inputs in a fully expanded system. Figure 3 shows one of the completed monitoring systems mounted in it's rack with the switch control panel.

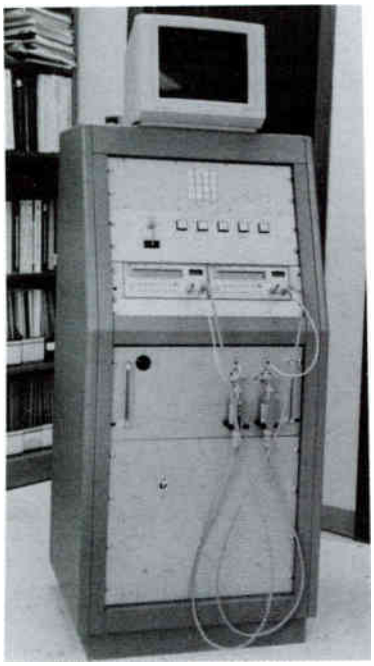

Figure 3
Computer Controlled Monitoring System

The monitor also computes and displays the VSWR for each of the locations. It has preset warning and alarm indications to alert the operator and open interlocks to shut the transmitters down if necessary. Input power readings are made utilizing standard diode detector type elements. Output power readings are made by thermocouple detectors. These avoid the problems of multi-channel signals summing incorrectly and giving wrong readings as occurs in diode detectors.

The other sites monitor input and output power with separate rack mounted panels for each of the monitoring points. Each panel provides both forward and reverse meters as well as VSWR alarms and transmitter interlocks.

INSTALLATION DETAILS

Sites with four outputs

Four sites; Mt. Tassie, Lookout Hill, Mt. Major, Victoria; and Mt Sugarloaf, N.S.W. were built with four-output splitters and coax switches. Two sites had three inputs and the others had four inputs. Input power levels of 30kW were used for all but one of the sites which had 20kW stations. A schematic of a four-input system is shown below.

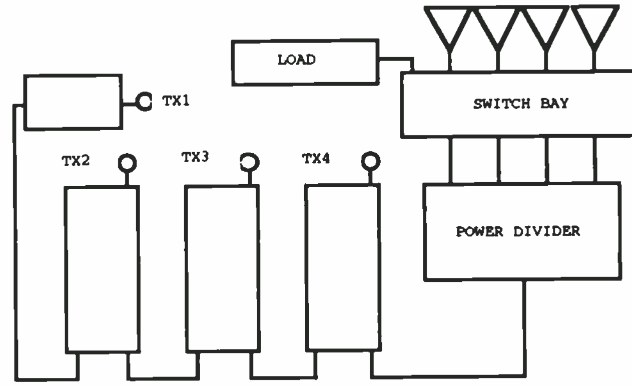

Figure 4
Four Output System Schematic

Three of the systems were designed for ceiling mount in a flat layout with expansion diplexers being added across in a horizontal fashion. The fourth system was built in a floor mounted frame. Each diplexer module was stacked vertically on top of the other with u-links as interconnects. Expansion channels are added by simply stacking on more modules.

The Mt. Tassie, Victoria site was the first to be installed in late May, 1991. This site serves a region about 150kM east of Melbourne. A new 16 bay UHF antenna was installed along with four new transmission lines and transmitters. The tower at this site required significant rework and strengthening. A photo of the site taken from the tower is shown in figure 5.

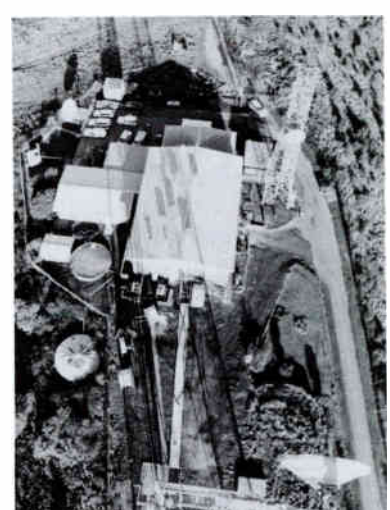

Figure 5
Mt. Tassie, Victoria Site

The Mt. Sugarloaf, N.S.W. site is located approximately 175kM north of Sydney and serves the region near Newcastle. This combiner was the first of the floor mounted designs and combined Australian channels 45,48, and 54 with expansion capability on channels 34 and 57. Each of these services uses 30kW transmitters. A photograph of the combiner as installed in the basement of the building (figure 6) shows the efficiency of the frame system and the neat installation.

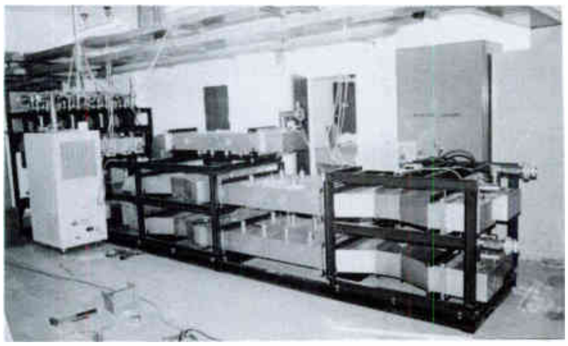

Figure 6
Mt. Sugarloaf, New South Wales Combiner

On the left in the photograph is the Altronics self-contained load which absorbs power from the splitter when a transmission line is switched out.

Sites with two outputs

Five sites; Mt Baranduda, Mt Alexander, Goschen, Victoria and Mt Dowe, Middle Brother, N.S.W. were built with two outputs and a patch panel/splitter arrangement. All five of these systems were built concurrently and utilize the same components wherever possible. The Middle Brother, N.S.W. site uses a WR1150 combiner. Floor mounted frames with vertical stacked diplexer modules are used at all sites. The frames are built in two sections which are bolted together on site, one holding the modules, and the other holding the splitter and patch panels. The two sections allow separate testing and shipping of the output system.

The patch panels used on these combiners consist of; a quick disconnect waveguide u-link to connect the combiner output to the splitter input, and two quick disconnect coax u-links to connect the splitter outputs to the antenna feed line inputs. Bypass of the splitter is performed by placing a special waveguide to coax link between the compiner output and the selected feed line input.

PERFORMANCE DATA

All of the combiners met or exceeded all of the performance specifications as outlined in the combiner requirements. The insertion loss and input isolation were significantly better than specified due to the excellent performance of the filter. The table below summarizes the performance of the nine combiners. Note that the insertion loss values include the minus three and six dB for the two and four way split output respectively. Thus these values include both the output system and the combiner. The isolation values given are the lowest obtained for that channel with respect to all the other inputs. Isolation to the other channels is greater than this value. .

SITE	CHANNEL	VSWR	LOSS DB	ISOLATION DB
MT TASSIE	34	1.05	6.22	88
	40	1.04	6.26	35
	43	1.05	6.35	50
	46	1.03	6.22	62
MT MAJOR	34	1.05	6.25	95
	40	1.05	6.30	61
	43	1.05	6.40	55
	46	1.06	6.32	48
LOOKOUT HILL	30	1.05	6.14	65
	33	1.05	6.17	55
	39	1.04	6.15	76
MT SUGARLOAF	45	1.05	6.23	83
	48	1.04	6.29	58
	54	1.04	6.17	82
MT DOWE	31	1.02	3.16	47
	34	1.03	3.18	50
MT BARANDUDA	30	1.05	3.12	57
	33	1.02	3.12	49
	39	1.02	3.10	71
MT ALEXANDER	29	1.03	3.20	62
	32	1.01	3.13	47
	35	1.04	3.14	56
GOSCHEN	47	1.02	3.19	59
	50	1.04	3.17	46
MIDDLE BROTHER	62	1.04	3.24	55
	65	1.05	3.22	58

TABLE 2
COMBINER PERFORMANCE DATA

CONCLUSIONS

The Australian equalization program has provided a unique opportunity for the installation of UHF multi-channel combiner technology at many sites within the country. The state of the art systems provide the ability to combine a large number of stations into one feed line with low insertion loss and VSWR. Completely independent operation of each station is maintained by the high isolation of the combiner. It is hoped that this paper has given the initiative for others to attempt this type of installation.

ACKNOWLEDGEMENT

The authors wish to thank Ross Thyer and Hank Prins of Quantum Pacific Pty and the many people at Telecom Autralia who helped make the combiner program a success

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ALL BAND ANTENNAS AND COMBINERS

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Abstract - Increasing antenna, tower and real estate costs, FAA restrictions, local zoning restrictions and EPA regulations have forced the broadcaster to consider sharing a common antenna system. **The benefits of co-location make multi-channel operation economical and efficient.** In some cases it is the only method available to get two or more stations on the air.

Recent advances in antennas and combiners provide these benefits to almost any combination of stations. Full service and LPTV stations both in the U.S. and Internationally have found this to be true.

INTRODUCTION

This paper will discuss the individual elements that make up an "All Band" system and their usefulness in:

* **Full Service TV**
* **LPTV**
* **HDTV Simulcast**

Among the topics of discussion are:

* **Antenna Characteristics**
  Panel Performance
  Standard Azimuth Patterns
  Quadrature Phased Feed System
  Compensated Pattern
  VSWR Performance of Array

* **Combining Techniques**
  Constant Impedance Combiners
  CIN and CIBP Types
  Star-Point Combiners
  Performance: VSWR & Efficiency

ANTENNA CHARACTERISTICS

Panel Performance

The term **"All Band,"** means an antenna that has a **VSWR less than 1.10:1 over the Lo-VHF, Hi-VHF or UHF Band.**

Figure 1 shows the typical panel that is the basis for a system. The modular panel is built of stainless steel, plated brass/copper and covered with an all weather radome. The radiating elements are flat dipoles mounted in front of a reflector. Extensive development has proven that flat dipoles are far superior in performance than tubular dipoles. The average power rating of the panel shown is 2.5kW with a single input feed and 5kW for a dual feed.

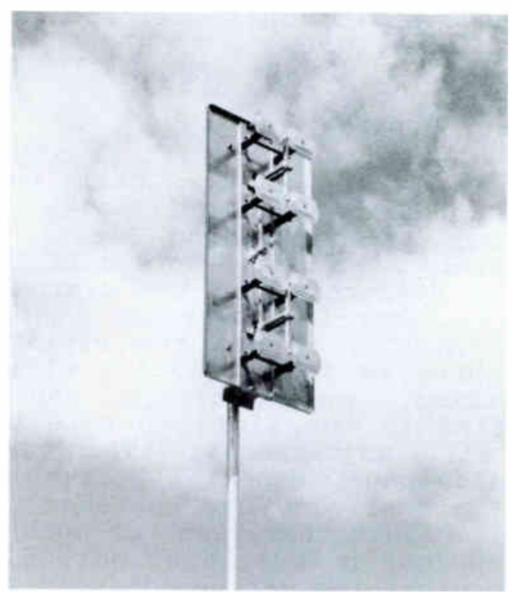

Fig.1 Antenna Panel

Production of every panel is closely monitored to insure equal quality and performance. Strict quality control methods are instituted to guarantee satisfaction.

Each panel has the same characteristic impedance. Performance over the full UHF band is shown in figure 2. This plot is from a randomly selected production unit. All panels exhibit a similar result.

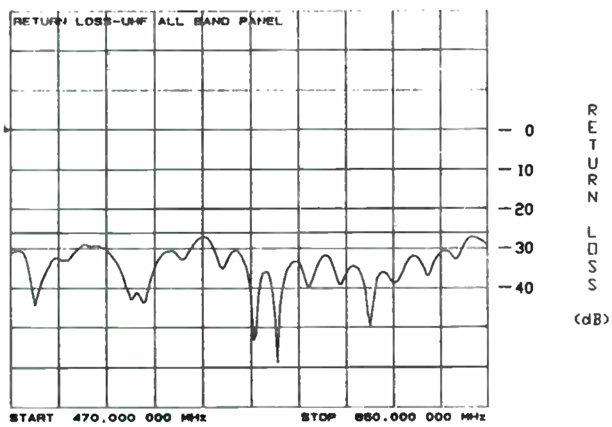

Fig. 2 Single UHF Panel Impedance Performance

Each panel has a return loss of less than 26dB over the full UHF band (470-800 MHz). This corresponds to a VSWR of 1.10:1.

Radiation Patterns

The advantage of the "All Band" panel use is that virtually any pattern configuration can be achieved. Beam tilt and null fill can be designed into the system to satisfy near-in coverage. Unlike slot antennas, where the amplitude and phase of the radiating element is fixed, panel arrays can be modified in the field to make tilt and fill adjustments. This is a definite plus, especially after you have installed your antenna and realized that there was too much (or not enough) beam tilt or null fill. These variables can be adjusted with a panel system by modifying the harness cable lengths.

Figure 3 shows the typical azimuth patterns for different antenna configurations.

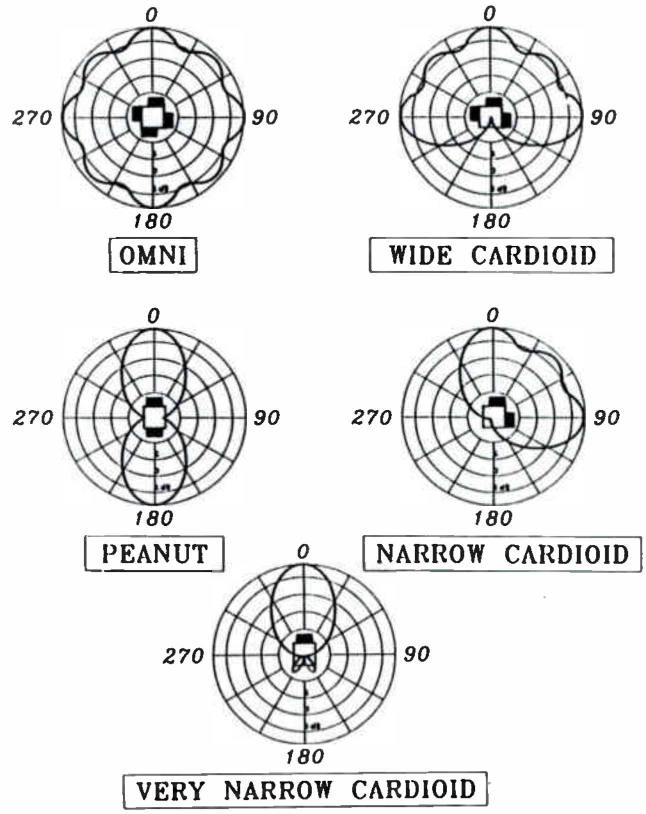

Fig.3 Typical Antenna Array Patterns

The patterns shown in figure 3 are based on an antenna array mounted on a specific tower size. The tower used for these measurements has a 24 inch face width.

Quadrature Phased Feed Systems

The complete cancellation of identical panel reflections is accomplished through the use of "Quarter-wave difference" cables.

Each antenna system is made up of panels, power dividers and feed cables. From a simple array of panels, as shown in figure 4, we can see how a properly designed harness will give reflection cancellation. The concept used is known as phase-rotation. If all the antennas are identical, in terms of amplitude and phase, then complete cancellation of the reflections will occur.

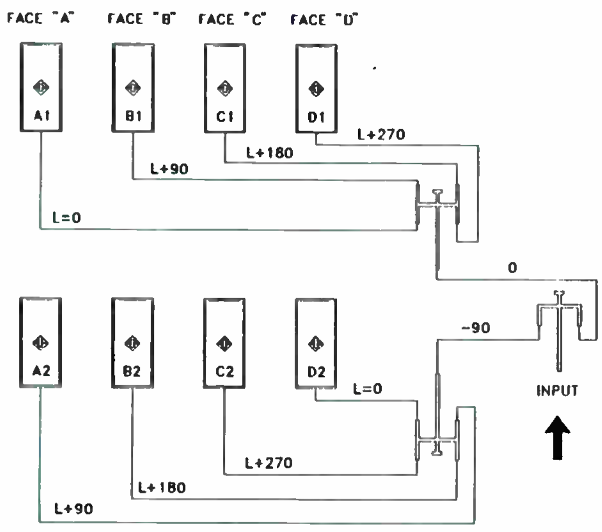

Fig.4 Quadrature Phasing of Two Bays

As shown in figure 4, each panel is fed by cables that have a 90 degree phase difference. When the signals arrive at the power divider, since they are on opposite sides of the Smith Chart, their reflections are cancelled. The phase relationship must be maintained on each face. If the cables feeding each bay are in phase quadrature, then the elements of the other bay have to be fed in such a way that the signals arriving at the element are in the same phase relationship as the bay above it. Figure 4 shows that if the two bays are fed with a 90 degree difference (0, -90), then the cables feeding the elements have to be adjusted accordingly. The example shows how this is done.

Compensated Pattern of Panel

4 Sided Tower

The way to achieve broadbanding within the feeding arrangement of the panels has been shown. Now let us consider how to recover the phase displacement introduced for matching. This will optimize the performance in terms of radiation patterns.

Figure 5 shows how we recover, with mechanical displacement, what we electrically altered. This displacement will allow us to obtain the correct vector summation and smooth out the azimuth pattern.

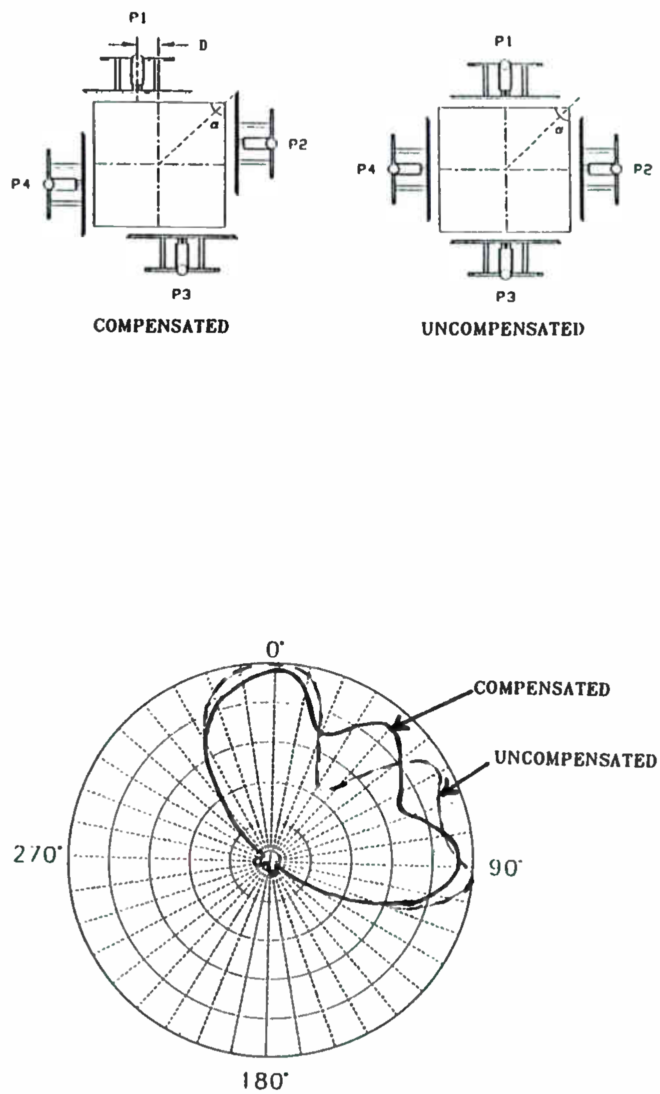

Fig.5 Compensated Azimuth and Panel Offset

By introducing this off-set, the compensated pattern shown is achieved. These results are true over the whole band.

VSWR Performance of Array

By using quadrature phasing, a VSWR of 1.10:1 or less can easily be obtained over the entire UHF band.

Figure 6 shows a schematic of a six bay array and its harness configuration.

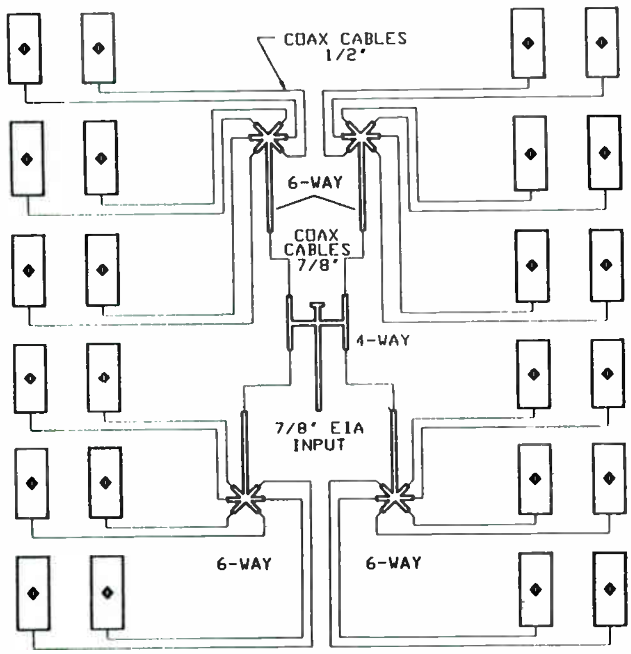

Fig.6 Antenna Feeding Arrangement Six Bay Omni Pattern

Figure 7 is a photo of this array assembled on its tower section.

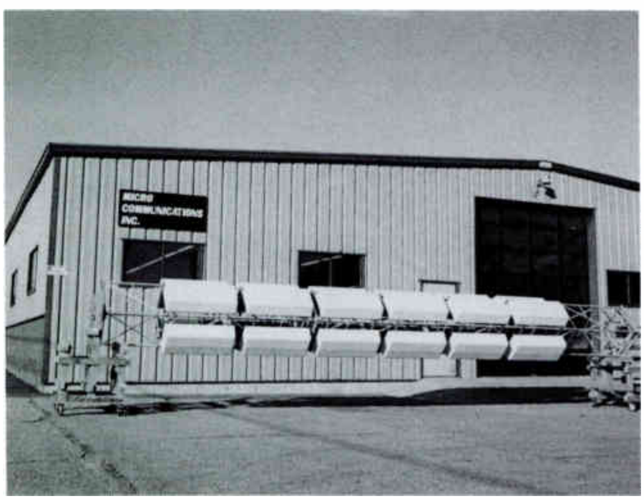

Fig.7 Six Bay Array Antenna

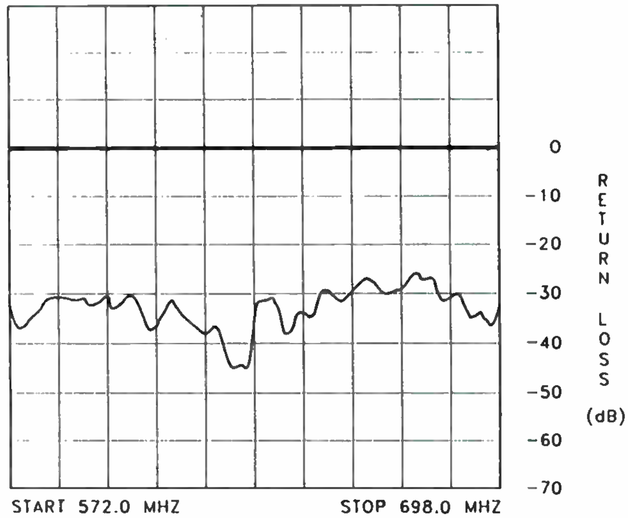

Fig.8 Final Field Results Antenna VSWR

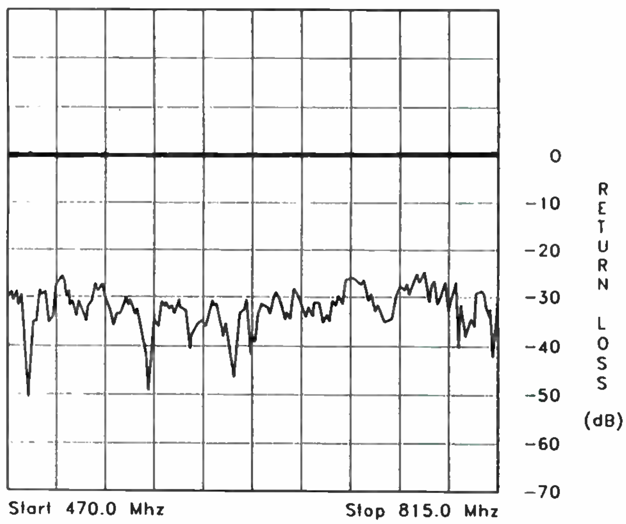

Fig.9 Final Field Results Antenna VSWR Full Band

Figure 8 shows the return loss (dB) of the band of interest. **The VSWR is less than 1.08:1 over the 126 MHz band.** Expanding the bandwidth of the analyzer further, as shown in figure 9, **the complete UHF spectrum is below a 26dB return loss, or a 1.10:1 VSWR.** The above multi-channel antenna system is radiating five channels. The results are only possible with an "All-Band" panel.

MULTI-CHANNEL COMBINING TECHNIQUES

Now that one antenna for the full band has been established, let us discuss a way to combine different channels to a common feed. MCI uses two common different combining techniques.

Constant Impedance Combiners

Constant impedance combiners use a combination of bandpass (CIBP) or reject filters (CIN) and hybrids to do the combining.

In order to understand how this type of combiner works, let us look at the individual components.

Figure 10 is a block diagram of a simple hybrid. When a signal is introduced at port A, the power splits and arrives at C and D in the phases shown. Port B is the isolated port.

Now, if we were to introduce a signal into port D and short circuit A & B, the power would be reflected back and because of the phase relationship, come out port C.

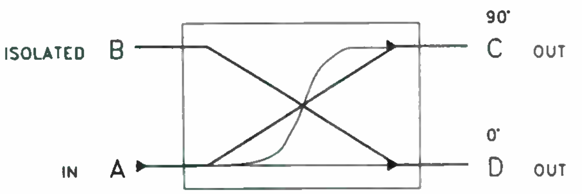

Fig.10 Hybrid Schematic

Figure 11 shows a block diagram of the CIBP combiner.

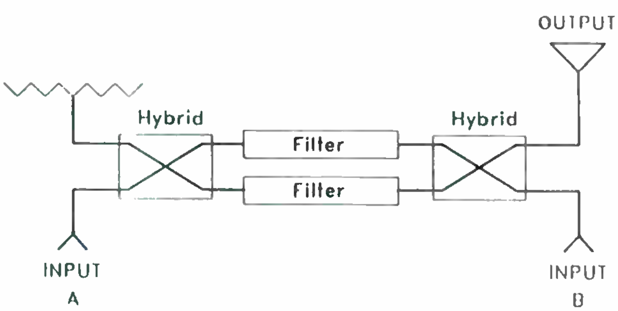

Fig.11 Constant Impedance Combiner with Band Pass Filter

If a signal is introduced at Input A in figure 11, the power splits and passes through the filters which are tuned to this signal. These two signals then re-combine and continue on to the output because of the phase relationship in the output hybrid.

A signal fed into Input B will split in the hybrid. The filter will act as a short circuit to this channel and will be reflected and re-combined at the antenna output due to the action of the hybrid.

The filter performance of two closely spaced channels is shown in figure 12.

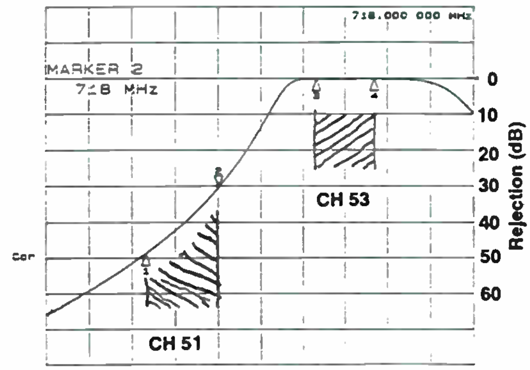

Fig.12 Band Pass Filter Response

A constant impedance notch combiner, on the other hand, works in a similar way, except the filters used are notch type instead of bandpass type.

A typical CIBP combiner results are shown below:

		One Channel Spacing	Two Channel Spacing
VSWR	:	**1.05:1**	**1.05:1**
Loss	:	**0.15 dB**	**0.10 dB**
Isolation	:	**50 dB**	**60 dB**

With this type of combiner, it is possible to design a system that can be expanded in the future. The only limitations to a system using this approach is the power handling capability of the last combiner in the chain. By using an all waveguide CIBP combiner, it is possible to combine four 60kW transmitters. The last combiner must be capable of handling 240kW.

Star-Point Channel Combiners

Star-point combiners consist of individual channel filters connected to an output tee. Each channel input has a bandpass filter that passes only that channel of interest and rejects all other channels.

Figure 13 is a typical block diagram of a 4 channel star-point.

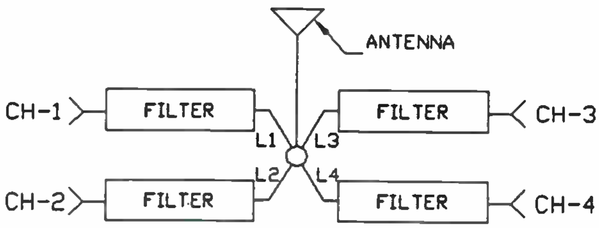

Fig.13 Star-Point Combiner

To combine the outputs of each filter, the line L1, L2, L3 and L4 must be chosen properly. At the reject channel, a perfect open circuit must appear at the tee. When the relationship between the lengths are correct, the channel will combine at the output with the insertion loss and VSWR of only the filter. Very little VSWR or loss results from the other channels. Isolation between the inputs is that of the filter.

Performance results to be expected from a UHF star-point combiner are as follows:

		One Channel Spacing	Two Channel Spacing
VSWR	:	1.10:1	1.10:1
Loss	:	0.20dB	0.10dB
Isolation	:	40dB	50dB

The star-point combiner can have performances similar to the CIBP combiner. The isolation between channels corresponds to the spacing. The greater the spacing, the higher the isolation. A disadvantage to this approach is that it is not easily expanded in the future.

MCI has computerized the design of both bandpass and reject filters for FM and TV application. A matrix wave analysis calculates the transfer properties including loss and group delay.

Summary

Now that an **"ALL BAND"** antenna system and combining network capable of multi-channel sites has been established, we can examine the usefulness in full service, LPTV and HDTV simulcast.

- **"All Band"** antenna and multi-channel technology has evolved to the point where widespread use is now possible. Therefore, tremendous **economic** and **technical** benefits result from the combination of two or more stations into one common antenna.

- **Full service** as well as **LPTV** can take advantage of this technology and minimize cash outlay.

- **HDTV simulcast** signal can be combined into the same NTSC antenna.

- The antenna is capable of radiating both channels each with the same **"foot print."**

- Broadband panel antenna systems with non-dispersive transmission lines provide a **linear phase shift** across the band and reduce group delay contribution.

- Element spacing on the panel is a half wavelength, eliminating **downward radiation**.

- Quadrature phasing provides a system where the impedance is **insensitive** to **ice** build up.

- Various allocation **plans** being studied by the **FCC for HDTV prefer CO-LOCATION.**

- **By utilizing a common antenna, only one antenna, building, tower and real estate is required.**

DIGITAL AUDIO BROADCASTING II

Monday, April 13, 1992

Moderator:
Donald Wilkinson, Fisher Broadcasting, Seattle, Washington

***DIGITAL SOUND BROADCASTING: A CCIR STANDARD**
Gerald G. Chouinard
Communications Research Centre
Ottawa, Canada

***LINCOM/SCI DAB REPORT**
Steve Kuh
LinCom Corporation
Los Angeles, California

***TERRESTRIAL DELIVERY OF DAB**
Lloyd R. Englebrecht
Stanford Telecom
Portola Valley, California

***USA DIGITAL REPORT**
Paul Donahue
Gannett Broadcasting
Los Angeles, California

***SYNETCOM REPORT**
Etienne Resweber
Synetcom
Hermosa Beach, California

Panel Discussion
Judith Gross, Moderator
Washington, District of Columbia

*Paper not available at the time of publication.

TELEVISION AUTOMATION

Monday, April 13, 1992

Moderator:
Gerald Robinson, Hearst Broadcasting, Milwaukee, Wisconsin

WHAT IS BROAOCAST AUTOMATION?
George L. Fullerton and Seth L. Olitzky
Louth Automation
Menlo Park, California

ROBOTIC CAMERA PEOESTALS FOR NEWS AT CBS-NEW YORK
Darcy Antonellis
CBS Operations and Engineering
New York, New York

***CAMERA AUTOMATION AT WJZ-TV**
Richard Seaby
WJZ-TV
Baltimore, Maryland

THE ATTC LABORATORY AUTOMATION SYSTEM
Scott E. Hamilton and Jeffrey U. Longbottom
Advanced Television Test Center
Alexandria, Virginia

***IMPLEMENTING AUTOMATION: PRACTICAL HINTS FOR PLANNING THE PROJECT**
All presenters

*Paper not available at the time of publication.

WHAT IS BROADCAST AUTOMATION?

George L. Fullerton & Seth L. Olitzky
Louth Automation
Menlo Park, California

Abstract- Automation is certainly a buzz word in today's broadcast world. But what does it really mean? Is broadcast automation simply the addition of a master control switcher or the control of devices from a computer workstation? Or should it be much, much more? Automation is defined by Webster as "a system or method in which many or all of the processes are automatically performed or controlled by machinery, electronic devices, etc." Broadcast Automation should automate specific processes and integrate these processes across the separate departments/tasks.

BROADCAST ENVIRONMENT

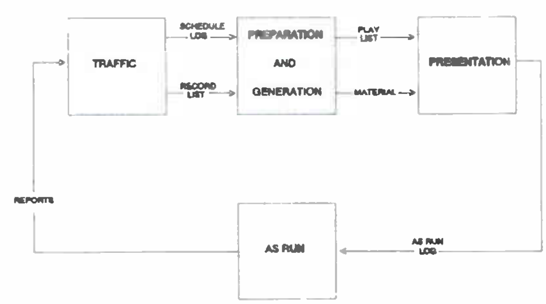

Exhibit A

INTRODUCTION

This paper will explore several different automation questions. What do today's broadcasters want from automation? What are the problems and solutions associated with conventional automation? And how can state of the art automation make the difference to you?

Many functions in broadcasting are currently automated. There are systems that address the traffic department, media preparation, on-air presentation, and as-run logging. All broadcasters have their own idea of where automation applies and what it should be, but certain ideas are shared by all. How do we best utilize various devices, such as VTRs, cart machines, and switchers regardless of age and manufacturer? And how do we pool the devices together to perform a variety of functions? Automation should allow devices to be moved from task to task without the need to re-cable either audio/video or control signals. This type of automation increases the utilization and productivity of capital equipment.

Broadcast Automation should encompass the total environment shown in Exhibit A. Similar environments for Automation Systems are the newsroom and Post Production. Automation should be made up of various modules which are integrated through an industry standard local area network (LAN). Because every station has its own specific needs in addition to standard functionality; what we call the 80/20 rule applies. This means that somewhere near 80% of the system functionality applies to all users and the additional 20% is special for each individual user.

No two broadcast environments are identical so customization is a requirement for all automation systems. A vendor's ability to deliver the customizable portions of the system in a timely and cost effective way will determine the success of the automation system. Users want their customized automation systems to increase productivity, streamline operations, eliminate lost revenue due to clipped or blown spots, and to be cost effective and justifiable. The automation system should support the end user's operation and function according to specific outlined requirements.

CURRENT BROADCAST SOLUTIONS

What are the problems associated with conventional automation?

One word sums it up: limitation. Most of the installed device control automation systems are based on the technology shown in Exhibit B. This technology was the standard for the 1980's when there was a greater dependency on hardware. This type of automation is reliable but not easily customizable since customization requires hardware changes. Device specific hardware interface products are necessary between the server and the devices themselves. Any customization requires modification to the hardware interface products. In addition, this technology runs on an external serial control BUS, generally an E/S BUS. As newer devices are added to the system, hardware interfaces must be designed and built. The increase in the number of interface hardware boxes causes system thruput to be degraded. In addition, if the automation system is networked, (additional clients or servers), a separate network is required for connecting clients to the system, causing various problems associated with managing two separate methods of communication.

ES BUS AUTOMATION SYSTEMS

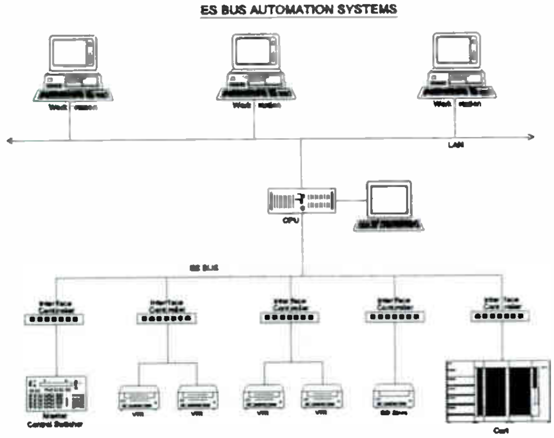

Exhibit B

SOLUTIONS FOR TODAY AND TOMORROW

What are the solutions for todays broadcaster?

The solution for the 1990's is to use new software technology coupled with new computer technology to develop customizable systems that meet all of the users needs. In contrast to the technology currently used, the systems for the 1990's are software driven (Exhibit C). Devices can be made available to perform a variety of tasks. New applications can be created and most importantly new devices can be added. In addition, new devices can be added to existing applications. The software development technology that supports this approach is called Object Oriented Programming (OOPS).

SOFTWARE DRIVEN AUTOMATION SYSTEM

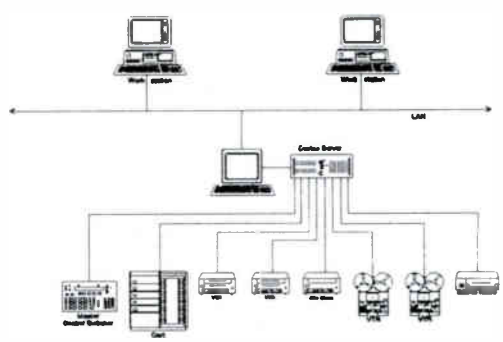

Exhibit C

New Technology

Most experts see OOPS as the future for software development. The recently announced IBM/Apple cooperative development project will produce joint OOPS applications by 1995. Object Oriented Programming is a modular approach to software development that greatly enhances productivity and shortens the development cycle. It allows for the reuse of existing software into new applications without rewriting already functional code because of new requirements of the application. One main advantage of Object Oriented Programming is the ability to encapsulate new objects around existing objects, extending their functionality upwards. In the broadcast environment, devices such as VTRs, cart machines, or switchers, can be created through software objects. In addition, LANS and user interfaces can also be created as software objects.

Once a device or component has been created, its interface to applications becomes standardized and streamlined. A VTR software object can be defined to play, record, stop, eject, fast forward, rewind, and perform any other functionality found in all VTRs. All applications can use a VTR with this software interface. It doesn't matter which type of VTR is used or how the VTR is actually controlled at the physical device level. The application is completely isolated from the device protocol and physical connections. Thus it is easy to see that if a different VTR is to be added to the application, the application itself doesn't have to be changed. The only requirement is building the software object for the new VTR.

Software objects can be independently modified without changing the program as a whole. This makes it easier to debug and upgrade a system or application. Applications can be developed and tested using software objects such as VTRs without the actual VTR being present.

OOPS technology eliminates the need for hardware interface products because all the device functions are replicated in software in both the server and the clients. This system only requires a single network (any Netbios LAN) for total control. Eliminating interface products and the need for an external BUS make a system much more powerful and able to support new and complicated devices such as a Library Management System.

How can this type of automation make the difference to you?

Flexibility

To build a flexible environment for using devices requires making the devices easily available to a new application. The encapsulation of a device such as a VTR in a software object, allows all applications to view the VTR identically. No specialized programming from application to application is needed to use the VTR. The devices can be assigned to perform various tasks such as satellite record or on-air replay. These same devices can later be dynamically reconfigured to perform other important functions. For example, a VTR can be configured for satellite record and later reconfigured for sequencer playback. This kind of dynamic assignment of devices allows the system to be very versatile and optimizes the use of various devices.

Making the devices available to a variety of applications, all of which may be running at the same time, requires the use of a device server. The server's basic purpose is to control devices in an orderly manner, handle client requests for device control and device specific information, and make device running status regularly available over the network to those applications wishing to use it. The server also handles the assignment of devices to requesting applications. A VTR that has been used for satellite feed recording could be freed up and moved to a newsroom application to archive news material. Each system server can control a larger number of devices and multiple servers can be used.

Because a network is a software object, any off-the-shelf Netbios LAN, such as Novell Netware, Token Ring, or Artisoft LANtastic can be used. Clients can be remotely connected to the system for control of devices or for monitoring privileges. The number of clients on the system is virtually unlimited. Device status is available to all the clients at a frame accurate rate, and access to the devices can be restricted to the appropriate clients.

Client/Server Architecture

Use of Client/Server technology is the key to customization in the broadcast environment. If the server model has been designed correctly then any type of application that uses broadcast devices can be created. This includes commercial playback sequencers, satellite feed recording, spot reel generation, traffic interface systems, etc. In the event that future applications require changes to the server model, the use of object oriented programming allows extensions to be built into the server without impacting the already installed base of applications.

Client/Server implementation uses remote procedure calls at the network level. This technology enables the use of a language independent application programming interface (API) and engine for any host computer interface.

Users can move information smoothly and easily across multiple technology environments, while also accommodating next generation technologies (Exhibit D).

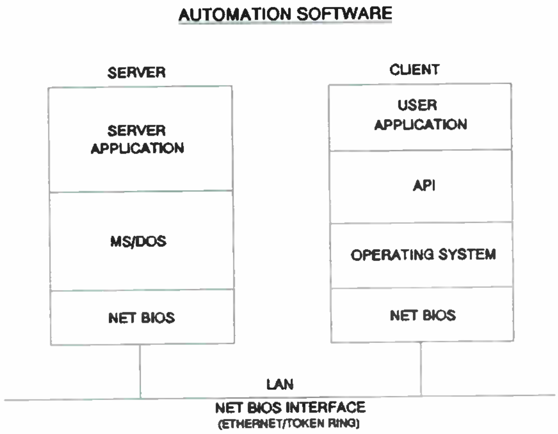

Exhibit D

Some clients may not actually use devices themselves but are created as a bridge between other applications. The as-run log application may monitor devices attached to a commercial sequencer application. The as-run log can in real-time, send back completion information regarding these commercials to a traffic system. This as-run log would only need to be customized with respect to its interface to the traffic system but not to its interface in monitoring the devices. Another example of a bridge would be software controlled automation supporting a newsroom system. The newsroom system would be a client application where the News Producer could control all the on-air devices from their News System workstation (Exhibit E).

Redundancy

Backup is an essential part of an automation system. The ability to recover immediately from hardware failure in a broadcast environment is crucial to the business of a broadcast network. Failure of one of the broadcast devices such as a VTR can be handled easily by moving another VTR into the system. The VTR may even be a different type, but the application will still run because with OOPS, a VTR is a VTR. A sequencer application

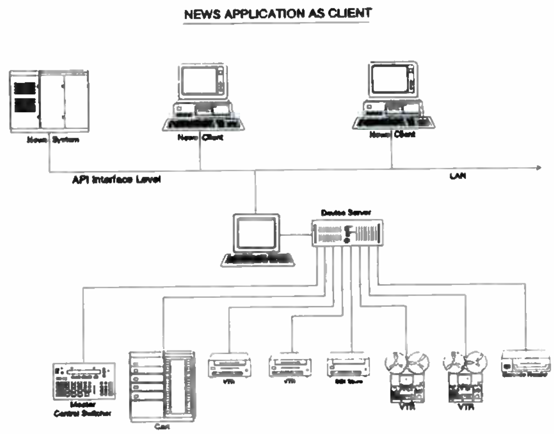

Exhibit E

which is running on a cart machine can continue to run on external VTRs or using the cart machine in manual mode without elevator support. Again, the application can move devices in and out as needed.

Hardware failure of the server itself can be compensated for by building redundancy into the client server applications. Clients whose sole purpose is to monitor the state of the system can be built. A client who detects failure in the server can itself become a server by automatically switching control lines from the failed server to itself. This requires that the redundant client be configured identically to the server. The switching of control lines can also be accomplished by manual operator intervention. A client can be earmarked to keep redundant or backup copies of crucial information running at the server. This would include the status of a sequencer list being played out by the server. In the event that the server is recovering, its current state can be reloaded across the network from a client.

Economical Configurations

This kind of automation offers broadcasters a wide range of solutions. System configurations can range from a simple server connected to a switcher and 3 or 4 VTRs, to complete management of several cart machines, many VTRs and a wide variety of other devices. Initial configurations can be limited and eventually expanded, module at a time, to provide total automation of all installed devices.

Future Proof

With any new technological advancement, there is always hesitation on the part of end users. How does one eliminate the "I'll wait for the next generation" reluctance and take advantage of the benefits of the current technology? Historically, a system would have to be redesigned or retired completely. Not so with this kind of automation. The use of object oriented programming and client server architecture renders an automation system virtually future proof. As new devices emerge and new features are required, the system can easily be expanded to meet changing needs.

Summary

Broadcast automation should be a modular, customizable, and cost effective system that integrates all the broadcasting functional areas. It should utilize state of the art technology that supports connectivity in a multi-vendor environment. Automation should meet the specific requirements of the end-user and be Future Proof in responding to changing needs.

ROBOTIC CAMERA PEDESTALS FOR NEWS AT CBS—NEW YORK

Darcy Antonellis
CBS Operations and Engineering
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ABSTRACT.

As cost containment and reduction continues to be a high priority for broadcasters, robotic devices find increasing application in the operational environment. This trend is most evident in the studio, where the traditional camera-person is being replaced by a computer-literate operator controlling multiple cameras.
This paper describes the camera automation systems installed in two studios at the CBS Broadcast Center in New York.

INTRODUCTION.

The principal value of automated camera control systems in studios is the achievement of improved productivity and the containment of costs. In earlier work, CBS deployed robotic camera systems in Washington, D.C., both in the CBS News Bureau, and at remote locations in the Capital. The experience there gained prompted the design and introduction of robotic camera control systems in two studios at the CBS Broadcast Center in New York.

The studios chosen were the CBS News Studio 47 and the WCBS-TV Studio 46, used primarily for local news on Channel 2.

This paper describes the system requirements, design, installation, and the operational experience gained in nine months of system use.

SYSTEM REQUIREMENTS.

Studio 47 is the origination point for the CBS Evening News with Dan Rather. Four pedestal-mounted cameras are employed on the studio floor, and it was required that the cameras be remotely controlled at will from the Studio 47 floor, the Studio 47 Control Room, or from an ancillary Control Room 34. In addition, it was required to remotely control two cameras in an adjoining News Flash Studio.

It was determined that the control locations on the studio floor and the Studio 47 Control Room should control all six cameras, while Control Room 34 should control the two Flash Studio cameras, and any three of the four cameras in Studio 47.

Studio 46 in Broadcast Center is used primarily for local news, but the requirements were more challenging than for the relatively fixed camera operation employed in Studio 47. In Studio 46, two different sets are used for the 5 PM and 6 PM news broadcasts. An interview set and a weather map are also installed in the studio. These facilities require that the four cameras be capable of moving quickly from one location to another during a broadcast. It was required that these cameras be controllable from the studio floor or from the studio control room.

Because of the more demanding requirements of the WCBS-TV studio, this paper details only the system installed therein, although many features and requirements are common to the CBS News Studio 47.

For full automated control of the cameras, it was required to have:

positional control of the camera pedestal on the studio floor,

pan, tilt, and elevation control of the camera head, and

zoom and focus control of the camera.

Although not required at the outset, the ability to add additional remote control facilities to the system was desirable. These might include set lighting, audio, and camera setup.

Following a careful evaluation of the responses to the above requirements from competing companies, including laboratory evaluation and on-site visits to TV stations to note how the equipment was being used, the Vinten Broadcast Company was selected as the prime contractor for the project. This decision was based in part on the successful work accomplished by Vinten in providing the robotic camera system installed in the CBS News bureau in Washington in 1990.

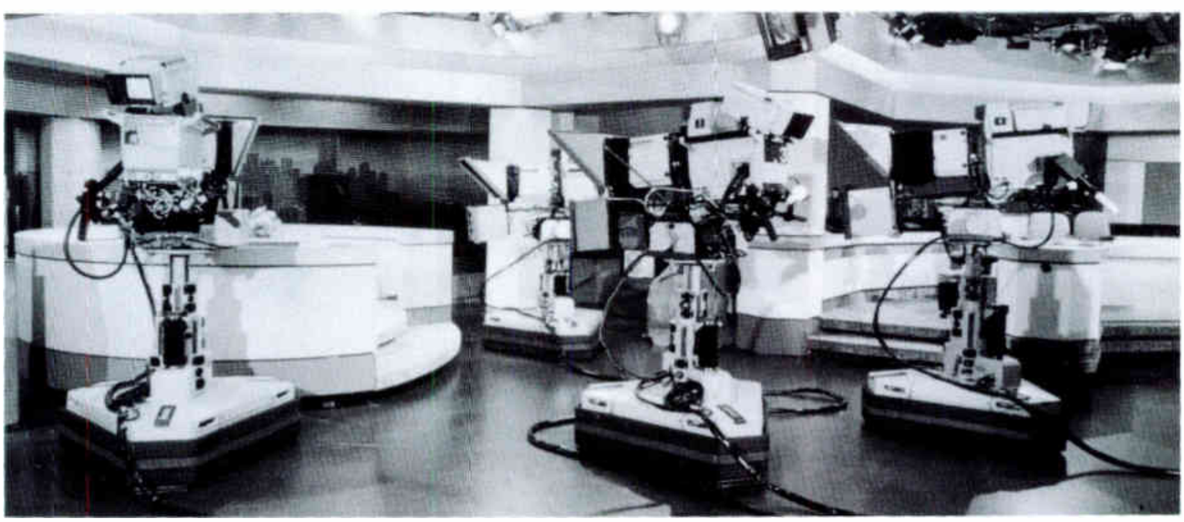

Fig. 1. Robotically controlled cameras in Studio 46.

SYSTEM DESCRIPTION.

The four cameras in Studio 46 are shown in Figure 1. The "home base" reference points for the pedestals consist of two 2-inch black lines drawn on an aluminum tape backing on the studio floor. The system then operates on the basis of cartesian coordinates for all locations of the pedestals, which can move up to 30 meters from their "home base" marks.

The requirement to control the cameras at will from the control consoles located either on the studio floor or in the Studio 46 Control Room, was met by the installation of a Vinten Data Switcher in the Control Room.

The Control Consoles (Figure 2), in addition to being well designed ergonomically, had to occupy as little space as possible, and, in the case of the Control Console on the studio floor, had to be mounted low enough to avoid obscuring the view of the set. This Control Console was mounted on wheels so that it could be readily moved around the studio.

The Control Console contains monitors and tally lights for each of the four cameras, and a Grass Valley 10X1 video switcher that is controlled by the Vinten Operator Control Panel. The output of the 10X1 switcher is fed to a "Trim Monitor", which shows the output of the camera that is under the control of the operator. It was found to be of critical importance that this monitor have good resolution and accurate framing.

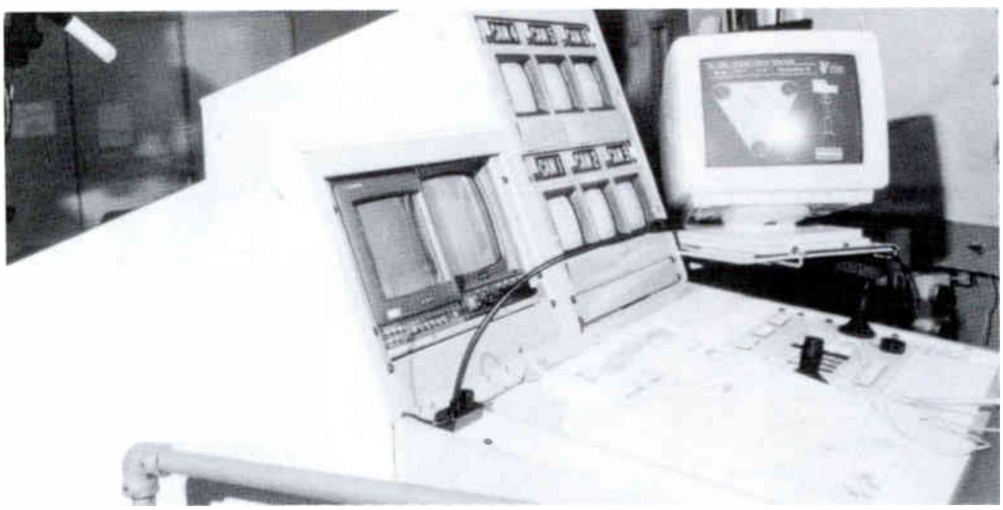

Fig.2. Control Console.

The Operator Control Panel (Figure 3) has two joysticks; one for the control of the pedestal height, wheel direction, and movement, and one for the control of the control of zoom, pan, and tilt. Focus is controlled by a separate knob.

The system stores the data for up to 500 shot positions in each camera. The data is actually stored in each camera head and pedestal, providing security in the event that the Camera Control Panel fails to transmit the shot commands. If a failure occurs in a camera head or pedestal, only that single unit is affected, and the rest of the system and cameras will continue to operate.
For routine operations in Studio 46, more than 60 shots, each of which is ascribed a number, are in common use. Simple keystrokes at a Camera Control Console initiate the movement of the pedestal to the programmed position, and the required setup of pan, tilt, elevation, zoom, and focus.

A graphics tablet installed (Figure 4) at the Control Console allows the immediate selection of a stored shot by touching a "pen" at the required location on the tablet. The tablet features a graphic layout of the studio set, including the positions of the on-air talent and icons indicating the various camera positions which have been pre-programmed. Once the shot is selected, and the pedestal has moved to the required position with the programmed positions of pan, tilt, elevation, zoom, and focus, the shot can be fine tuned with the joystick controls on the operator's Control Panel. The graphics tablet enables the operator to initiate the movement of one camera pedestal while retaining joystick control of a second camera.

A Vinten Panel Electronics Module interfaces with the Operator Control Panel, the graphics tablet, the camera pedestals, and the camera heads. The Panel Electronics Module interfaces also with a PC which generates a VGA display of the pedestal head position and the

Fig.3. Operator Control Panel.

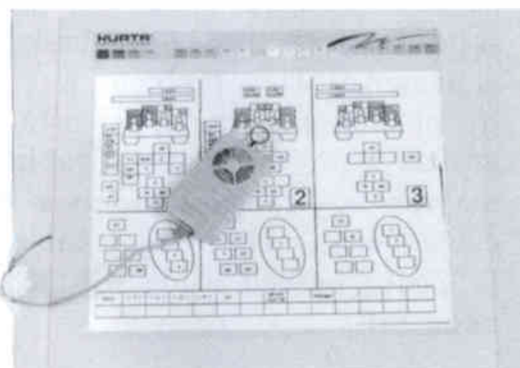
Fig.4. Graphics tablet.

alignment of its wheels, as seen in the background of Figure 2.

WIRING.

Extra cables were run between the Vinten Data Switcher in the Control Room and the studio floor to permit the connection of cameras to different A/V boxes. This feature facilitates any future expansion of robotically controlled cameras, or major changes in the layout of the studio sets.

The Data Switcher is connected to the studio Control Console by a 19-pair RS422 cable. At the Data Switcher the cable is broken out into eight separate 9-pin D-connectors. At the studio Control Console the cable is terminated in a 37-pin FK connector, mated to a CBS designed connector panel.

The Vinten robotic system is supported by a second cable running to each camera on the studio floor. It was found that these cables would become wedged under the pedestal skirt, causing the safety brakes to be activated. This problem was solved by placing plastic zipper wrap around both the camera cable and the robotics cable, to form a single cable mass. This larger cable bundle is easily pushed aside by the moving pedestal.

LENS CONTROL.

The camera lenses have been modified to achieve automated control of zoom and focus. Vinten installed a switch at the point in the lens system where an analog voltage determines the position of zoom and focus. In one position the switch allows manual control of zoom and focus, while the other position permits the robotic Control Panel to set the zoom and focus values. This switch is activated by a pendant on/off switch mounted on the panning bars of the camera.

OPERATING EXPERIENCE.

The complete system can operate in any of three modes:

- -full automation using pre-programmed shot memory,
- -automated control of cameras by the operator at one or other of the Control Consoles, and
- -full manual control of a camera by a camera operator.

For manual control, it is necessary only to activate the pendant on/off switch mounted on the panning bars of the camera. Thereafter, elevation, steering, and drive is effected by the pendant joystick on the panning bar. The pendant activates the appropriate servos. If necessary, the servos may be disengaged, and a tiller bar is then used for steering, and the head controls are operated manually.

A robotic camera system is celebrated for its ability to enable a single operator to control multiple cameras. For the director, accustomed to an immediate response to his camera commands by a camera-person manning a camera, the robotic system is unnerving at first, because while the camera control operator who now receives the commands from the director, does in fact respond, he may do so after a short delay, if he is at that particular moment controlling another camera. The console operator must necessarily decide when to give up control of the camera he is working with, and when to trim the next camera shot called for by the director. Fortunately, by the use of the graphics tablet, the operator can initiate the movement of one camera while retaining joystick control of a second camera. The operator can then quickly trim up the first camera as soon as it reaches its new location.

A very basic requirement for successful robotic camera operation is to assure a very clean studio floor. Any dirt or debris on the pedestal wheels compromises accuracy of movement. While Vinten makes roll-over bars for the pedestal to facilitate cleaning the wheels, the camera head must be removed for this operation.
To obviate this time consuming procedure, CBS is designing a pedestal jack driven by worm gears, which will raise the complete pedestal three inches off the floor to allow easy cleaning and maintenance of the wheels.

The Teleprompter display which overhangs the pedestal, has on occasion hit an obstacle when moving. If the pedestal itself hits an obstruction, the safety brakes are automatically applied, but this does not happen when the Teleprompter hits. As a precaution, it is planned to install externally mounted safety pressure switches on the Teleprompter's protruding parts. These will actuate the pedestal brakes.

One feature of the system that requires improvement is the response of the pedestal to the movement commands from the Camera Control Panel, or from the pendant joystick on the pedestal panning bar. The response to steering commands is too laggy, and a more precise steering and motion control operation is desirable.

CONCLUSIONS.

The Vinten Robotic Camera Control systems installed at CBS News and WCBS-TV have proven to be effective and reliable in the nine months of operation. They have allowed a significant improvement in operational productivity, and have demonstrated the flexibility required to meet the ever-changing and more complex demands of news broadcasts. In general, the more complex the series of camera setups required, the greater are the gains in productivity achieved.

THE ATTC LABORATORY AUTOMATION SYSTEM

Scott E. Hamilton and Jeffrey U. Longbottom
Advanced Television Test Center
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The ATTC Laboratory Automation System was developed to permit full functional access to any of the devices and instrumentation necessary to perform the required video and audio tests defined within the Objective and Subjective Test Procedures documents produced by the ACATS (FCC's Advisory Committee on Advanced Television Service). Although specifically designed to permit a highly accurate, repeatable, and auditable test environment to exist for the long period of time necessary to fairly and objectively administer the extensive suite of tests to all ATV systems, the approach taken demonstrates the value and flexibility obtainable from a fully distributed Macintosh computer network implemented with advanced software development techniques.

THE ADVANCED TELEVISION TEST CENTER

The Advanced Television Test Center (ATTC) is a private, non-profit corporation created and supported by a coalition of broadcasting companies and television industry organizations to test and report on proposed transmission systems for advanced television. The results of this work will assist the Federal Government, the television industry, and the American public in selecting among the proposed new systems and determining the necessary national transmission standard to implement the new service.

The ATTC laboratory facilities consist of an RF Test Bed which contains 3 VHF and 11 UHF television transmitter upconverters driven from standard Harris Corporation exciters; Orban BTSC stereo generators; frequency synthesizers; peak and average power meters; coaxial cable delay lines; digitally controllable attenuators; random and impulse noise sources; and RF pathway switching necessary to simulate most types of terrestrial broadcast interference and impairment conditions in a highly controllable and repeatable environment.

Videotape production facilities consist of three Sony HDD-1000 digital television recorders coupled with devices termed Format Convertors (invented by ATTC, and designed and manufactured by Tektronix, Inc.) which permit the digital recording and playback of 1125 line reference video as well as each of the unique proponent video formats. Two Sony DVR-10 D2 format recorders provide NTSC record and playback capabilities. A Sony PCM-7030 RDAT machine provides digital audio record and playback capabilities.

A Viewing Room is equipped with 25 commercially available NTSC consumer television sets and a 65 inch Hitachi 16:9 multisync projection monitor for display of all proposed formats of advanced television signals. This permits panels of expert viewers to subjectively evaluate both NTSC and ATV signal impairment conditions established through the RF Test Bed facility. The expert viewers are presented with varying levels of introduced signal impairments, precisely controlled and monitored by the automation system, to determine two critical signal values: the Threshold of Visibility or TOV (the point at which the impairment is just barely visible), and the Point of Unusability or POU (the point at which the picture is deemed to be impaired to such an extent that it is extremely annoying to watch). Statistical analysis of the expert viewers' voted responses is performed immediately to ensure that smoothed and statistically correct data is obtained for TOV and POU. Intermediate values between TOV and POU are also acquired.

The acquired signal levels of interest determined by the panels of expert viewers are replicated by the laboratory automation system for recording through either a Format Convertor on the HDD-1000 if a proponent ATV signal, or a D2 machine if an NTSC signal. The videotapes generated and assembled into randomized sequences interspersed with reference video segments are sent to the Advanced Television Evaluation Laboratory (ATEL) in Canada which performs the subjective testing procedures involving non-expert viewers.

Additional ATTC facilities consist of a photography room which is used to capture displayed images of specialized test signals applied to the proponent ATV system. The photographs taken are included among the extensive battery of objective measurements taken on each proponent system under test.

Architectural design of the RF Test Bed and automation system support facilities within ATTC began in earnest in

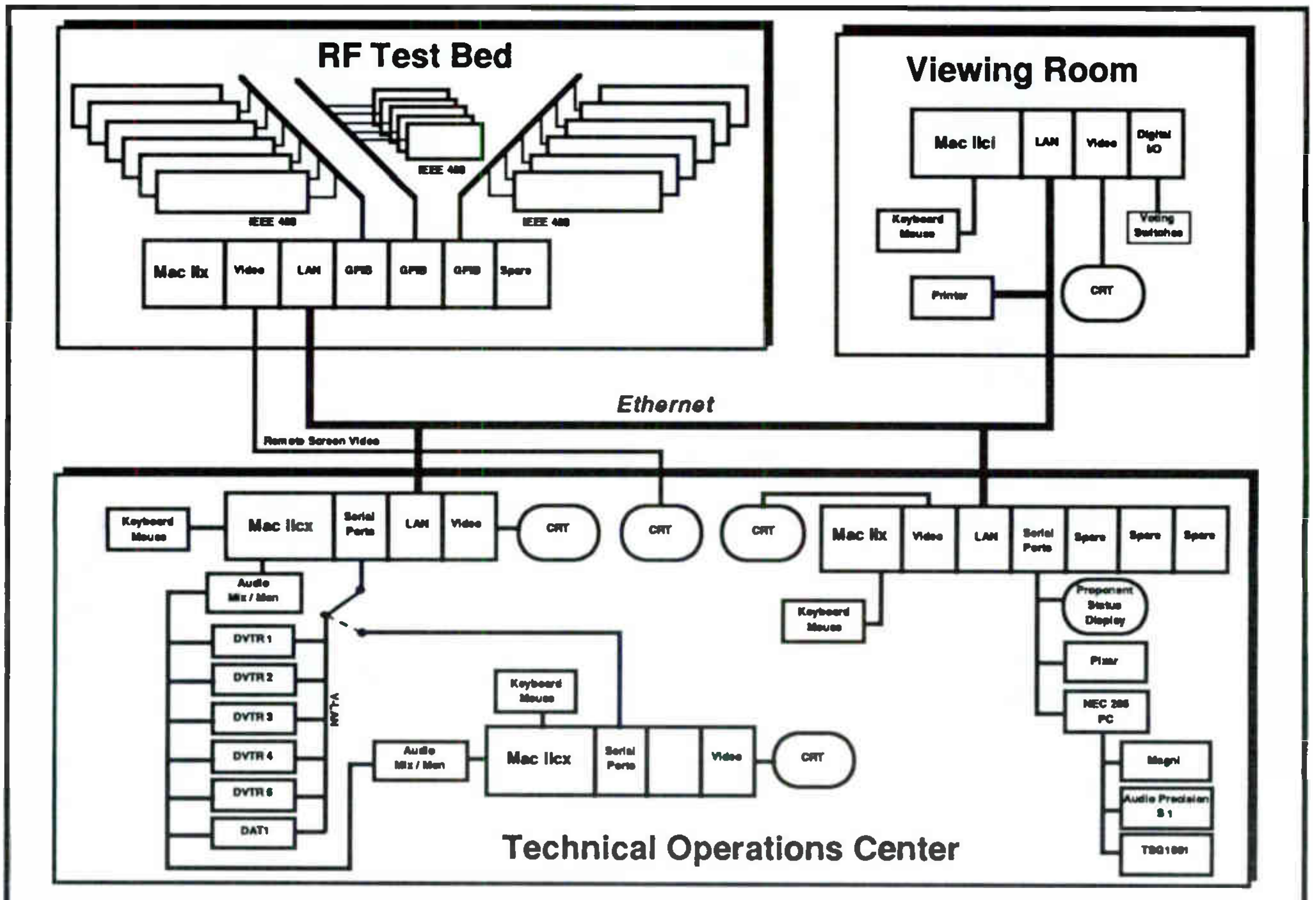

Figure 1 - ATTC Automation System Architecture

October, 1989, as the Advisory Committee identified and refined the requisite areas of proponent ATV system characterization to be performed by ATTC and specified within the Objective and Subjective Test Procedures documents.

Harris Corporation was selected as the vendor for provision of the RF Test Bed in March, 1990. A close working relationship with Harris was established to monitor the evolution of their design of the Test Bed and its instrumentation. Throughout March and April of 1990, many different approaches to an automation, control, and data acquisition system were evaluated by the authors. An approach was formulated and presented for approval by the ATTC Technical Committee on May 2, 1990.

The RF Test Bed was delivered to ATTC by Harris in the second week of July, 1990. Within twenty four hours after Harris had installed, integrated, and checked out the Test Bed, the Test Bed Computer had control of all switching functions and instrumentation. Automation system software development continued through the end of April, 1991, as fully automated videotape production and control was integrated into test control and expert viewer voting data acquisition and analysis. A dress rehearsal period was initiated in the first week of May, 1991, employing NTSC as the "proponent" under test. The dress rehearsal continued at a frantic pace until Sarnoff's ACTV system became the first proponent ATV system to be tested at ATTC on July 12, 1991.

ATTC LABORATORY AUTOMATION HARDWARE ARCHITECTURE

The ATTC Laboratory Automation System consists of seven separate computers; six are Apple Macintosh II machines, one is a PC compatible NEC 286. Four of the Macintosh machines are interconnected via an Ethernet local area network link and are termed: 1) the Viewing Room computer; 2) the Control Room computer; 3) the Machine Room computer; and 4) the Test Bed computer. A fifth Macintosh, termed the Stand-alone Editor computer, is not interconnected to the Ethernet network. The NEC 286 is interconnected to the Control Room computer via an RS-232 9.6 Kbps link. A systems interconnection diagram (Figure 1) illustrates the manner in which the five Macintoshes and NEC 286 are physically and electrically connected. The sixth Macintosh, integrated into ATTC's administrative office network, maintains a test results and audio/video tape inventory database necessary for tracking the myriad of results generated by the testing procedures for each proponent.

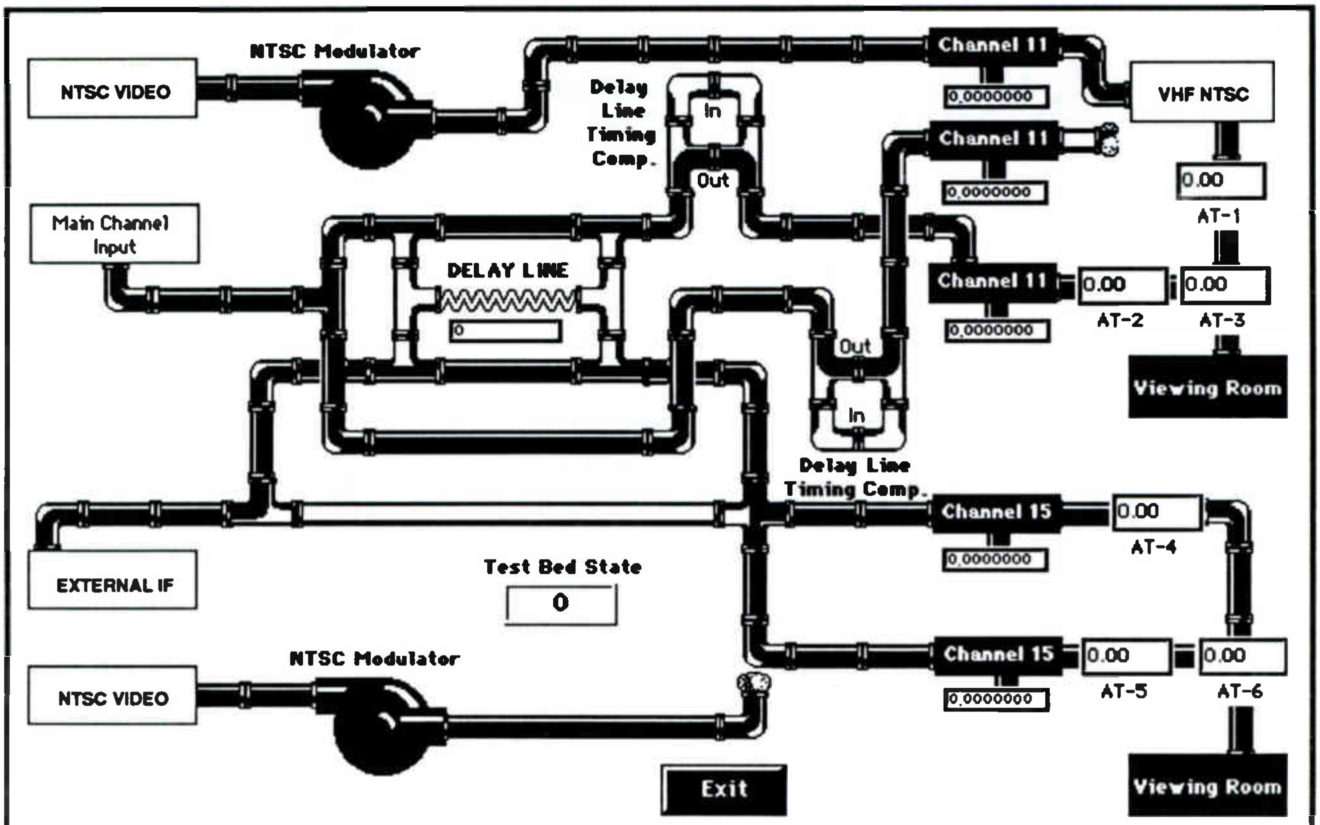

Figure 2 - RF Test Bed Path Flow Monitor Screen

Test Bed Computer

The Test Bed computer is a Macintosh IIx machine (providing up to 6 interface board slots) configured with 8 MB of memory, a 160 MB hard disk drive, a single 1.44 MB 3.5 inch floppy disk drive, an Apple 8 bit color video card, and three National Instruments IEEE-488 bus interface cards.

The Test Bed computer has the responsibility of monitoring and controlling all of the RF pathway switching within the RF Test Bed as well as the instrumentation contained within the Test Bed: two Wavetek frequency synthesizers, eight Weinschel RF digitally controlled attenuators, a Wavetek peak RF power meter, and a Boonton average RF power meter.

Three separate IEEE-488 busses are supported. One of the busses is assigned to control the eight Weinschel digital attenuators, one is assigned to the two frequency synthesizers and the peak and average power meters, and one is assigned to support the five IOTech digital IO interface boxes necessary to interface nearly 400 bits of control and tally status information involved with pathway switching within the RF Test Bed.

The Test Bed computer monitors Ethernet for incoming messages from either the Viewing Room computer or the Control Room computer that contain control or state change requests for instruments or the RF Test Bed. Each RF Test Bed and instrument change is recorded to an event log database for future analysis if required. The Test Bed computer responds to each request with a positive acknowledgment and supplies the requisite information or control activity as necessary. The Test Bed computer continually monitors (on approximately 500 millisecond cycles) all devices within the RF Test Bed to ensure that both instrument and control settings are within nominal and expected parameters. In the event of an anomaly, the Test Bed computer records the entire RF Test Bed status and checkpoints the information to an error log database. An anomaly message is also sent to both the Viewing Room and Control Room computers to notify operators of unexpected or out of tolerance situations.

Control Room Computer

The Control Room computer is a Macintosh IIx machine (providing up to 6 interface board slots) configured with 8 MB of memory, a 160 MB hard disk drive, a single 1.44 MB 3.5 inch floppy disk drive, an Apple 8 bit color video card, and a Creative Solutions four port serial interface card.

The Control Room computer has the responsibility of maintaining, at a single point within the network, knowledge of the control status of the other machines (whether the Viewing Room or Machine Room computers have control of either the RF Test Bed or the video and audio tape recorders), as well as a single copy of the acquired "test results" (the values, corrected for plant pathway losses and calibration factors, of

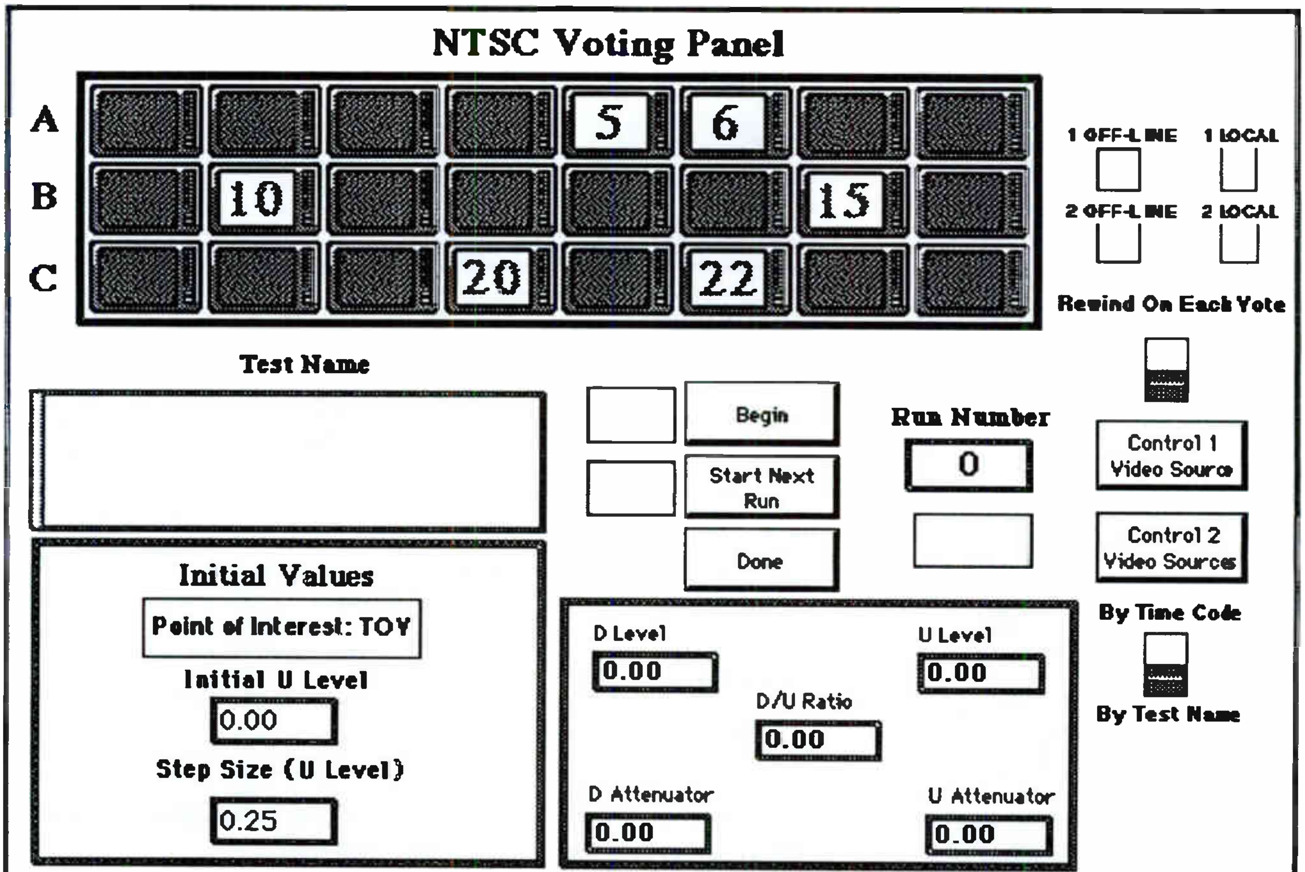

Figure 3 - NTSC Voting Process Monitor Screen

the Desired and Undesired signal RF power levels determined from expert viewer voting and ranging conducted in and delivered to the Viewing Room). The Control Room computer is able to display a simplified RF Test Bed pathway and instrument setting diagram for monitoring purposes (Figure 2). The Control Room computer also provides the information and interface necessary to maintain the Proponent's status display which indicates the current test in progress. Additionally, the Control Room computer provides the information and control parameters necessary to activate an audio test suite to be performed by the Audio Precision System One test instrument, and is capable of similar activities for the Magni and Tektronix video signal generators as well as the Pixar.

Viewing Room Computer

The Viewing Room computer is a Macintosh IIci machine configured with 8 MB of memory, an 80 MB hard disk drive, a single 1.44 MB 3.5 inch floppy disk drive, and a National Instruments 32 bit digital IO card. Adjacent to the Viewing Room computer, an Apple Laserwriter IINT printer is interconnected to Ethernet via a Rana EtherPrint box to provide network wide access to a printer.

The Viewing Room computer has the responsibility of commanding the other resources (Test Bed, Control Room, and Machine Room computers) to provide the necessary Desired and Undesired signal sources at the proper signal levels to the 24 NTSC receivers or Hitachi high definition multisync monitor. These devices are observed by the expert viewers to determine TOV and POU of either impaired NTSC (Figure 3) or ATV (Figure 4) television signals. The Viewing Room computer also acquires the voting data obtained from interface boxes or pushbuttons used by the expert viewers to identify whether or not they are able to see an impairment within the television signal. The Viewing Room computer also performs statistical analysis of the acquired voting data and generates graphical printouts of the results as well as creating ASCII data files of the recorded voting data for possible future analysis.

The Viewing Room computer, in conjunction with the Control Room and Test Bed computers, is capable of reproducing any video signal test impairment (out of the 260 possible video tests defined for any seven levels of Undesired signal at any of three levels of Desired signal). A total of 5,460 individually recallable conditions determined by the expert viewers (or any other value, as may be required) may be established at any time in the future for review, or to generate the subjective rating and quality audio and video tapes.

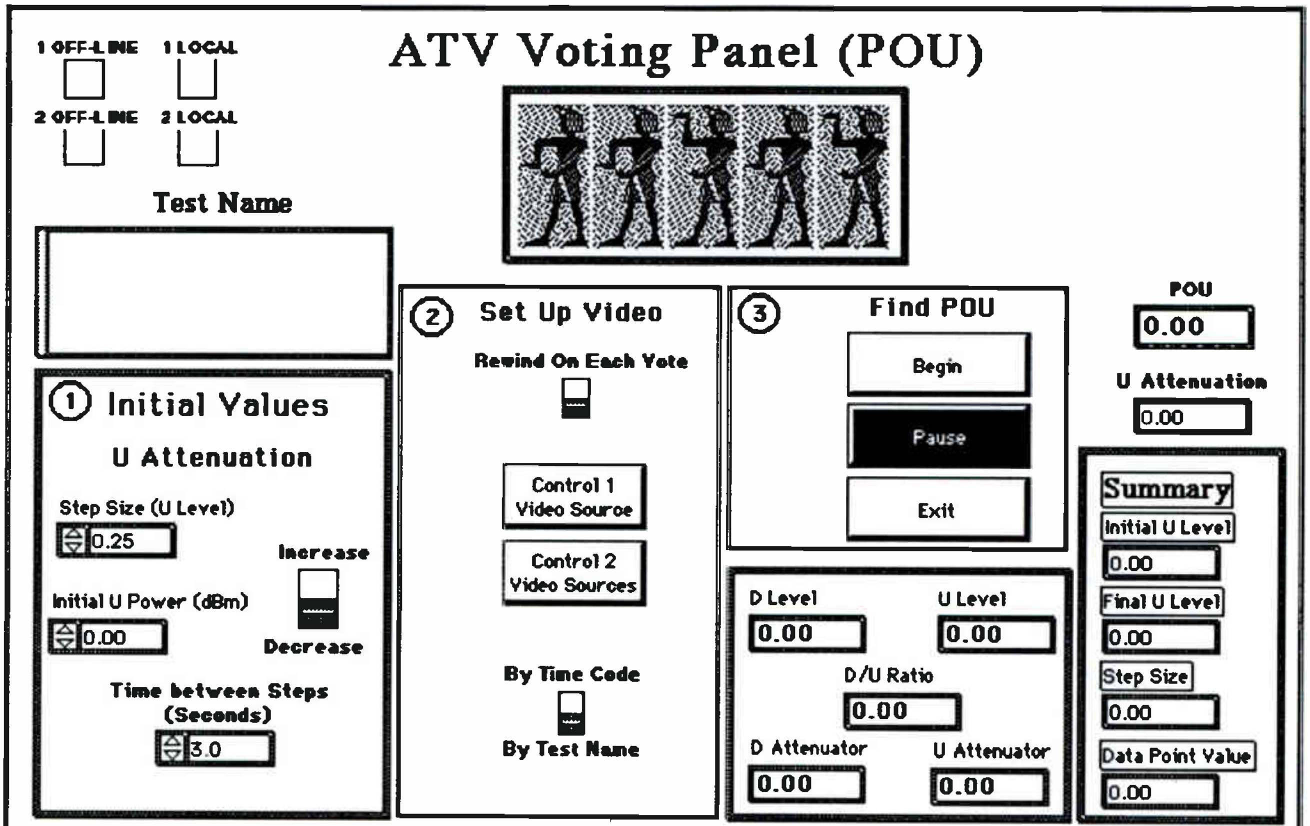

Figure 4 - ATV Voting Process Monitor Screen

Machine Room Computer

The Machine Room computer is a Macintosh IIcx machine configured with 8 MB of memory, an 80 MB hard disk drive, and a single 1.44 MB 3.5 inch floppy disk drive.

The Machine Room computer has the responsibility of controlling any or all of the six available video and au[illegible] tape recorders within ATTC upon request from the [illegible]ng Room computer or under automated control while [illegible]ming the pre-computed scripts necessary to assemb[illegible] required video and audio subjective rating and quality tapes.

The Machine Room computer implements a fully functional professional quality video and audio editing system (Figure 5) whose functionality is also available at the Control Room and Viewing Room computer locations. The Machine Room computer will automatically create and execute the complex EDLs (Edit Decision Lists) necessary to assemble the following types of HD/NTSC/RDAT tapes: 1) Parent Masters; 2) Video Subjective Rating tapes; 3) Video Quality Rating tapes; and 4) Audio Quality Rating tapes. While executing an EDL, the Machine Room computer commands the Control Room computer to establish the requisite test environment (in conjunction with the Test Bed computer) which will replicate the video or audio test and impairment conditions previously determined during expert viewer voting sessions. Signals routed through, and impaired by, the RF Test Bed are delivered to the proper video or audio recorder for creation of the tapes to be submitted for subjective analysis in Canada (necessary correction factors are applied to replicate to within ±0.25 dB of the RF signal strength originally delivered to the Viewing Room during expert viewer voting).

A unique capability, available from either the Machine Room computer, or the Stand-alone Editor computer, is that of computer synthesized speech. Synthesized speech audio, generated in software within the Macintosh computers, is converted to appropriate balanced audio levels to allow recording on either audio or video tape. The audio tracks generated are useful for identifying the individual cuts on video tape and for providing instructions or cues to subjective viewers and auditioners.

NEC 286 Computer

The NEC 286 computer is configured with 1 MB of memory, a 40 MB hard disk drive, one 1.44 MB 3.5 inch floppy disk drive, one 1.2 MB 5.25 inch floppy disk drive, 6 serial ports, one parallel port, an internal VGA adaptor, an Audio Precision System One interface board, and a Microsoft mouse. An HP LaserJet III is attached to the parallel printer port through an automatic printer sharing switch to allow ATTC's Tektronix VM700 Video Measurement system and Rhode and Schwarz Spectrum Analyzer to access the printer as well.

The NEC 286 computer has the responsibility of providing an environment necessary to execute programs developed spe-

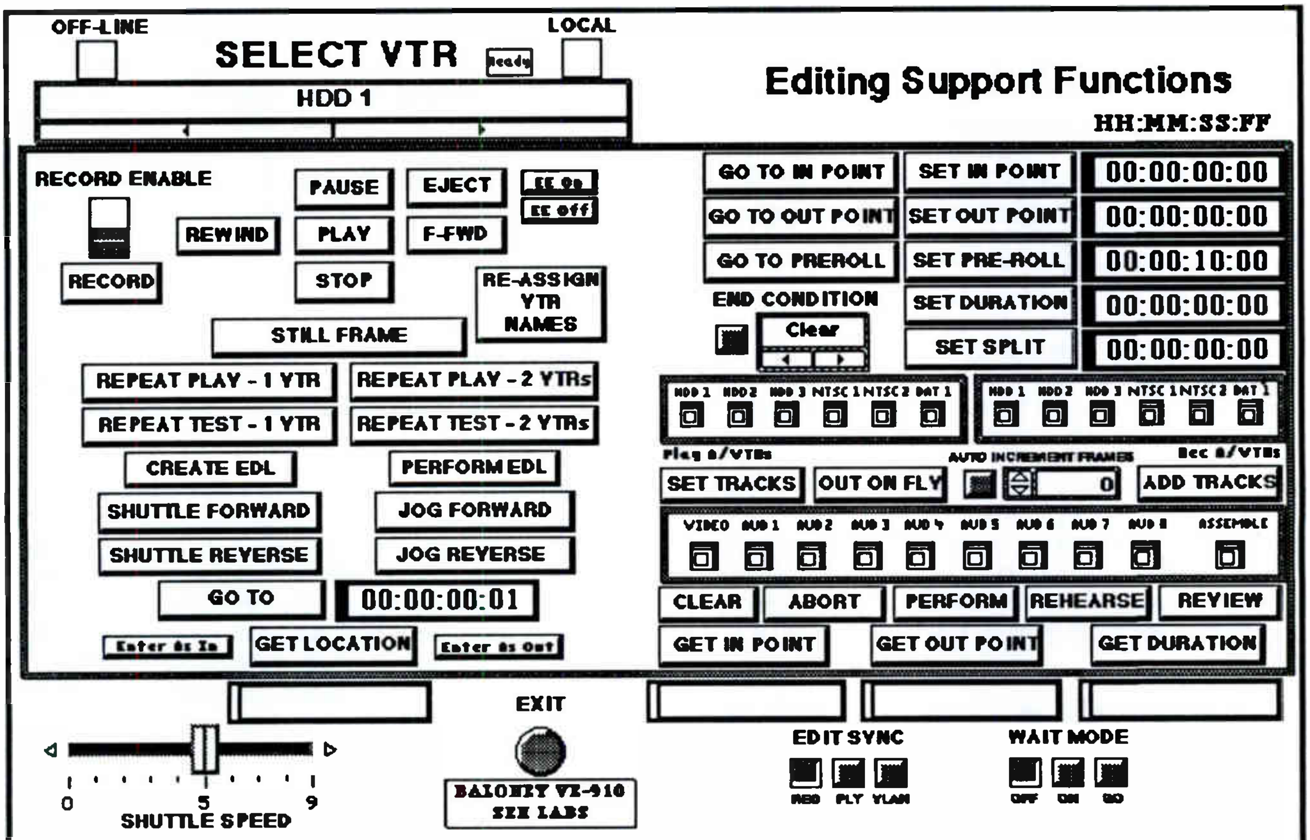

Figure 5 - Video Tape Editor Control Screen

cifically for the PC or compatibles. The NEC 286 contains a very special operating system developed by ATTC to receive command and control information from the Macintosh machines to cause execution of either Audio Precision, Magni, or Tektronix developed application software to control their respective proprietary devices. Results may be passed back to the Control Room computer if necessary. The Audio Precision developed application software permits up to approximately 1700 types of audio tests to be performed on proponent systems. The Magni and Tektronix developed software permits generation of a variety of video test signals in NTSC and proponent formats.

Stand-alone Editor Computer

The Stand-alone Editor computer is a Macintosh IIcx machine configured with 8 MB of memory, an 80 MB hard disk drive, and a single 1.44 MB 3.5 inch floppy disk drive.

The Stand-alone Editor computer's responsibility is to provide an editing support function to permit either fully automated or manually controllable generation of digital video or digital audio tapes in an off-line manner. The editing software in the Stand-alone Editor is identical to that running in the Machine Room computer with the exception that RF Test Bed communications via the Control Room computer are not supported.

Synthesized speech is generated by the Stand-alone Editor in a manner similar to that described for the Machine Room computer.

On occasion, an administrative Macintosh IIcx has been expropriated from a staff member and pressed into service within 30 minutes to provide additional editing suite capacity when needed for parallel activities.

LICS Database Computer

The LICS (Library Information Code System) Database computer is a Macintosh IIcx machine configured with 4 MB of memory, an 80 MB hard disk drive, and a single 1.44 MB 3.5 inch floppy disk drive.

The LICS Database computer has the responsibility of maintaining a descriptive record of each test result generated. The machine supports a database application program written in FoxBase. Details of the LICS system are contained within the ATTC LICS Specification document.

ATTC LABORATORY AUTOMATION SOFTWARE ARCHITECTURE

The software architecture of the ATTC Laboratory Automation system is based upon a hierarchical, fully distributed approach to functional support implemented upon a multi-layered communications protocol. Each computer within the system is responsible for the low level support of the instruments or specialized subsystems directly attached to it as well as providing access to the functionality and data of those instruments and subsystems to the other computers within the network. Each computer maintains a network connection to it's adjacent neighbor, sometimes acting as a client to a neighboring server, and conversely, sometimes acting as a server to a neighboring client depending upon whether information to complete a task is local to or remote from the given computer.

Figure 6 illustrates the enveloping rings of responsibility of each of the machines. The user interface, at the highest level of the functional hierarchy, appears at the outermost ring and is responsible for the final presentation of information to human operators.

Directly beneath the user interface layer lies the instrument and device drivers necessary to control, interpret, or display information from equipment subsystems. The instrument and device drivers are common to each of the individual computers; the same functionality of control and appearance of information therefore remains essentially the same at all machines.

The innermost ring represents the communications layer. The communications layer either deals directly with the devices attached to a local machine, or routes control information or data to the appropriate remote machine which has the subsystem under its local control.

The fully distributed control approach to the architecture permitted a very high degree of operational flexibility within the ATTC laboratory. Access to plant resources became location independent: the functionality and instrumentation within the RF Test Bed, contained in a physically isolated and environmentally controlled room on one floor, could also be exactly duplicated within the Viewing Room on the second floor of the ATTC facilities, in the Technical Operations Center (TOC), as well as the videotape Machine Room. Similarly, videotape machine control may be accessed in the TOC, the Viewing Room, or locally within the Machine Room. In effect, all of the capabilities of the ATTC facility are available from anywhere within the facility; only the outermost ring (the user interface layer), prevents or allows access to any of those capabilities. The flexibility obtained also permits a moderate degree of backup redundancy. In the event of a machine failure, it is possible to access and control facilities from an alternate location.

Object Oriented Virtual Instruments

The ATTC Laboratory Automation system was implemented utilizing a significant new software development technology: the graphical, object oriented, LabVIEW™ 2 system from National Instruments, Austin, Texas. LabVIEW is both a development and operational software "environment" currently implemented for the Apple Macintosh platform.

LabVIEW application programs are termed "Virtual Instruments" or VIs. There are three main parts to the program: the *front panel*, the *block diagram*, and the *icon/connector*. The front panel (for example, Figure 4) contains graphical representations of controls and indicators which may be extensively customized by the programmer. The controls and indicators provide the inputs and outputs, respectively, to the system and become the man-machine interface. The block diagram (Figure 7) may be considered to be the actual "program". It is constructed using selection, placement, and "wiring" tools. The icon/connector is a mechanism of creating an *object*, similar to a subroutine, out of a VI. The icon will then represent the VI on the block diagrams of still other VIs. The program development process thus becomes remarkably similar to wiring together software "integrated circuits". Very complex functionality may be constructed in a hierarchical fashion of VIs using, and re-using, multiple layers of other VIs.

Data relationships from one object (icon or VI) to the next are represented by wiring the connectors (similar to the parameters passed to and from subroutines) on the icons to each other. A fundamental concept within the LabVIEW environment is that of *data flow:* an icon/connector or VI will not execute until the data to be processed by the VI is available at all of its input connectors which are wired to controls. After the VI has performed its required processing function, data is passed out through connectors (which are wired to indicators), to other VIs or to indicators. Program execution sequencing is provided by a variety of control structures (For loops, While loops, and Frame sequences).

CO-CHANNEL INTERFERENCE, ATV INTO NTSC: A SAMPLE TEST SCENARIO

Now that most of the individual ATTC Laboratory Automation Control system components and functions have been described, it may be useful to learn how they are used in concert to perform an actual test scenario on a proponent ATV system.

A panel of experts is assembled at ATTC by 9:00 A.M., and instructed in the proper voting protocols and procedures to be used for the particular test. Another testing day in the Viewing Room begins.

The Viewing Room computer is first used to establish the test to be performed. A test control database contained within the Control Room computer is consulted via the network to retrieve pre-stored information. It consists of the proper states of all RF Test Bed switching relays, the identification of attenuators to be used for control of the desired (NTSC in this case) and undesired (ATV in this case) signal sources, the channel frequencies and calibration factors to be used and whether or not a channel offset should be established using the frequency synthesizers instead of crystal control, which power loss calibration factors should be employed depending upon the location of the proponent's equipment (Proponent Room A or B) or the Viewing Room, whether a delay line should be switched in and the amount of delay in microseconds which will also be used to properly set the cursor delay pick-off point for the peak power meter, and other pertinent information. The information from the test control database is returned to the Viewing Room computer which forwards necessary commands to the Test Bed computer so that the RF Test Bed and its instrumentation may be properly set. The Test Bed computer delivers commands to the Test Bed's 88 switching relays, verifies that all relays have been set (237 status bits) and all instruments have been initialized to their proper settings. In the event of an error, the operator is notified and an entry is made into an Error Log database maintained on the Test Bed computer for future analysis. A verification is returned to the Viewing Room computer and an entry is made into the Event Log database on the Test Bed computer to record all status and instrument settings.

Upon receipt of a successful response from the Test Bed computer, a power calibration sequence is initiated by the Viewing Room computer in conjunction with the Test Bed computer to verify the aural and visual power levels of the desired and undesired signal sources. Out of tolerance conditions are reported via error messages. Total elapsed time to complete the initialization, verification, and power calibration: about 20 seconds. Expert viewer voting and data acquisition may now begin.

Expert viewer voting determines the crucial desired and undesired ATV and NTSC signal power levels at which interference between the systems is either just barely visible (Threshold of Visibility or TOV) or has severely distorted the picture (Point of Unusability or POU). The TOV power level is especially important because it may be used to determine reception strength of a television signal necessary to overcome competing signals broadcast on an adjacent channel (one higher or lower) than the channel desired to be received, or for co-channel, a competing television signal transmitted on the same frequency as the desired channel. The determinations permit calculation of the radial distance or spacing required between television transmitting antennas, hence the number and positioning of television stations that may exist within a given geographical area that will provide an adequately receivable signal at a home's television set. Different voting procedures and statistical analysis of the voting results are required for determining TOV and POU depending upon whether an ATV signal is the desired signal or if NTSC is the desired signal.

Once the TOV, POU, and several intermediate levels have been determined by the expert viewers and analyzed by the Viewing Room computer, the requisite signal power levels are stored within a Test Results database maintained by the Control Room computer. That Test Results database is then accessed by the Machine Room computer, controlling the creation of randomized sequence, subjective rating videotapes, in conjunction with the Test Bed computer and RF Test Bed, to exactly replicate and record the viewing conditions observed previously in the Viewing Room by the expert viewers. Those videotapes are then sent to ATEL, Canada, for exposure to lay viewers who rate the recorded impaired signals relative to reference signals. The results of the analysis performed by ATEL will tell us the sensitivity of a large distribution of people to their ability to discern interference to the television picture thereby providing greater insight into inter-station spacing requirements.

The data acquired from all expert viewer voting exercises is recorded in ASCII formatted computer files for future analysis as required. Together with the TOV, POU, and intermediate power levels determined for strong, medium, and weak signal contours; Event and Error logs; paper printouts of the statistical analysis of voting runs; spectral plots; photographs; NTSC D2 and Format Converted HDD-1000 video tapes; RDAT audio tapes; expert viewer written commentary; audio test results printouts and plots; operations logs; and other engineering documents comprise a vast collection of data documenting the tests performed on each proponent system.

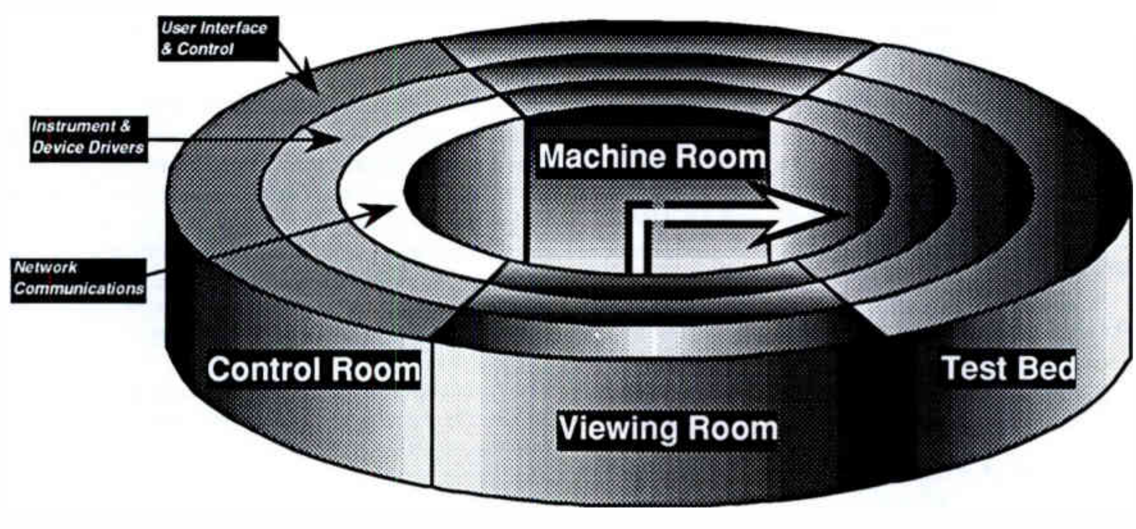

Figure 6 - Interprocess Communication

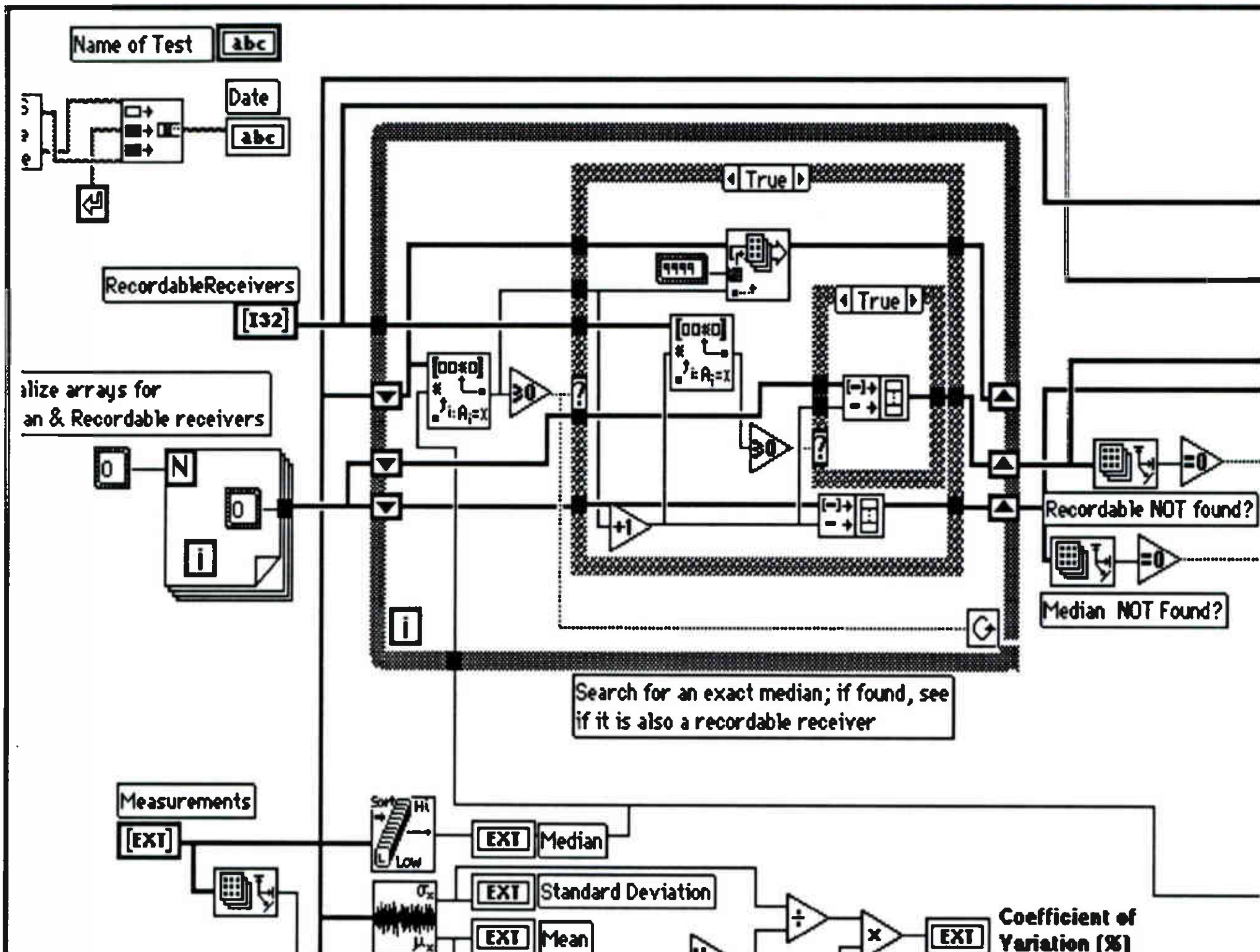

Figure 7 - LabVIEW's Block Diagram: Graphical Object Oriented Programming

Each data element is cataloged within the LICS database by reference number for future correlation and lookup.

SOFTWARE SYSTEM DEVELOPMENT METRICS

The ATTC Laboratory Automation system software consists of more than 750 LabVIEW virtual instruments. Additional development was performed in Pascal, C, and MC68000 assembly which was incorporated into some specialized LabVIEW VIs to provide unique capabilities such as speech synthesis. The control system envelope extended from the LabVIEW/Macintosh environment to include the IBM PC platform based subsystems from Audio Precision, Tektronix, and Magni, was written entirely in PC assembly language. The LICS database was implemented in FoxBase for the Mac.

Approximately 6 man months of effort were expended during the systems design and planning phase which occurred from January through April, 1990. Software implementation began in May, 1990, with most command, control, and data acquisition functions completed for the RF Test Bed by mid July, 1990.

Network communications support and low level interprocess communications were fully operational by the first week in November. Initial videotape control functions (those depicted in Figure 5) were completed by the first of December; full video and audio tape production automation was dependent upon final specification of the randomization sequences and test material to be employed to create the subjective evaluation tapes to be sent to ATEL, Canada, and was completed by the first week in April, 1991.

The nearly 1700 audio test procedure modules to drive the Audio Precision System One were delivered and integrated into the overall systems design in January. Final integration of all operational software was completed on March 15, 1991. Alpha testing continued from that date through the end of the second week in April. New software changes were implemented in April and May to address ACATS test plan changes anticipating the emerging digital ATV systems.

(Further software expansion occurred in January-February, 1992, as more new tests were added for digital systems.)

The four Macintosh machines and the NEC 286 were relocated from the software development lab to their final operational positions and linked into Ethernet during the third week of April. Final checkout and systems level debug continued up to the start of the dress rehearsal "NTSC Dry Run" within the first week of May. The LICS database machine and software were fully operational by the first week in June.

The Dry Run continued, as a test procedure verification and systems shakedown exercise, up to the start of official proponent testing which began on July 12, 1991, with the ACTV system from David Sarnoff Research Center/Advanced Television Research Consortium.

The total elapsed calendar time for the ATTC Automation Control System design and implementation was almost exactly 18 months; a total of 33 man-months of development effort, including the LICS database system but exclusive of the effort required to develop the Audio Precision System One test procedures.

The total Automation Control System hardware and commercial software cost was slightly over $55,000. That figure includes all operational test support computer hardware, LabVIEW and FoxBase software, IEEE-488 bus interfaces and IOTech digital I/O boxes, VLAN transmitter and receivers, and Ethernet network wiring components. It does not include the cost of stand-alone test instrumentation, video and audio tape recorders and Format Convertors, or RF Test Bed instrumentation.

As this paper was being written, in the second week of January, 1992, over 4100 physical test records had been generated and cataloged for the first three ATV systems. Three more proponent ATV systems are scheduled to undergo the test procedures conducted at ATTC.

SUMMARY

The ATTC Laboratory Automation System was implemented within stringent budget and schedule constraints thanks to a revolutionary new software development technology: the graphical, object oriented, scientific instrumentation and control capabilities embodied by National Instruments' LabVIEW 2 system. Combined with the readily obtainable Apple Macintosh II computer and other commercially available peripheral components, and the flexibility and speed of the EtherTalk (Ethernet) network, it has been shown that islands of automation may be successfully integrated into amazingly powerful and useful solutions.

In order to propagate the cost effectiveness of automation systems integration it is incumbent upon equipment manufacturers to "design-in" remote control capabilities into their equipment, to perform extensive validation tests on those capabilities, and to fully disclose the command and communications protocols necessary to integrate their equipment into larger, more flexible, and human usable systems. The power of today's personal computers linked into automation networks may then be utilized to provide a common man-machine interface to specialized equipment subsystems that are intuitive to operate.

As we progress into the 1990's, cost competitiveness, the flexibility to integrate and use both new and previously existing technology (as exemplified by the ATV and NTSC equipment within ATTC), and the increased productivity obtainable from user friendly man-machine interfaces to diverse manufacturers' specialized equipment subsystems may be successfully realized by employing the "software IC" techniques similar to those described in this paper.

ACKNOWLEDGMENTS

An undertaking of the magnitude of the ATTC Laboratory Automation System cannot be accomplished without the dedicated support and human understanding of family and friends which the authors gratefully acknowledge: Ms. Carole Audick, Dr. John Janovy, Jr., Ms. Jackie Rhode, Ms. Pam Shearmur, and Mr. Evans Wetmore. Additionally, a very special thanks for guidance and truly unbelievable technical support from our friends at National Instruments: Dr. James Truchard, and Messrs. Rob Dye, Brian Powell, and Greg Fowler. From VideoMedia for VLAN support: Mr. Roland Levin. From Audio Precision: Ms. Debra Brimacombe. From Weinschel: Mr. Gerry Messina. For the LICS database: Mr. Andy Gallant. Finally, Mr. Robert Niles, Capital Cities/ABC and Chairman of the Technical Committee of the ATTC who believed when others doubted, and all of the members of the ATTC Technical Committee. We would also like to thank Mr. Peter Fannon, ATTC; Mr. Charles Rhodes, ATTC; the ATTC sponsors (Capital Cities/ABC Inc., CBS Inc., NBC Inc., Public Broadcasting Service, Association of Independent Television Stations, Association for Maximum Service Television, Electronic Industries Association, and the National Association of Broadcasters); engineering staff of the FCC; many participants in the FCC Advisory Committee; the helpful ATV proponents; and most certainly the expert viewers for the opportunity to have been involved with this remarkable enterprise.

MAXIMIZING BROADCAST SIGNAL COVERAGE

Monday, April 13, 1992

Moderator:
William Hassinger, Federal Communications Commission, Washington, District of Columbia

***TERRESTRIAL MOBILE RADIOWAVE PROPAGATION— A TUTORIAL**
Richard L. Biby, P.E.,
Communications Engineering Services
Falls Church, Virginia

MOUNTING YOUR TELEVISION BROADCAST ANTENNA FOR OPTIMUM RECEPTION AND COSTS
Kerry W. Cozad
Andrew Corporation
Orland Park, Illinois

ANALYSIS OF FM BOOSTER SYSTEM CONFIGURATIONS
Stanley Salek
Hammett & Edison, Inc.,
San Francisco, California

THE MOUNT DIABLO BOOSTER SYSTEM
William F. Ruck, Jr.
KFOG/KNBR
San Francisco, California

***INCREASING FM COVERAGE WHILE REDUCING ROOFTOP EMI EXPOSURE**
Tom Silliman
ERI, Inc.
Evansville, Indiana

OPTIMIZATION OF VHF EFFECTIVE RADIATED POWER AND ANTENNA HEIGHT COMBINATIONS
Karl D. Lahm, P.E.
Lahm, Suffa and Cavell, Inc.
Fairfax, Virginia

A NEW MULTI-CHANNEL COMMUNITY ANTENNA FOR FM BROADCAST
Ali R. Mahnad, Ph.D.E.E.
Jampro Antennas, Inc.
Sacramento, California

A NEW HIGH POWERED SOLID STATE TRANSMITTER
Hilmer I. Swanson
Harris Corporation, Broadcast Division
Quincy, Illinois

*Paper not available at the time of publication.

MOUNTING YOUR TELEVISION BROADCAST ANTENNA FOR OPTIMUM RECEPTION AND COSTS

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Andrew Corporation
Orland Park, Illinois

ABSTRACT

The age of many VHF and UHF transmitting antenna systems has many station engineers and managers reviewing the needs and requirements for replacement systems. Shifting population concentrations, new tower design specifications, tower space leasing and HDTV are all issues complicating the decision making process. This paper will present various options to existing antenna system designs that will produce optimum coverage to viewers, more efficient use of tower space and cost effective alternatives when planning for future growth and technical changes such as HDTV. Measured field and laboratory data will be used as a basis for the conclusions.

INTRODUCTION

Mounting your television transmitting antenna directly to the top plate of the tower (top mount) is the ideal condition from a radiation pattern standpoint. The calculated or measured free space patterns are produced and coverage predictions are straightforward. However, top mounting the antenna is not always possible, particularly when the station does not own the tower. In this case, it is necessary to sidemount the antenna. But even if top mounting is an option, it may be more cost effective to sidemount the antenna particularly when using a directional azimuth pattern. The following information describes various conditions associated with sidemounting antennas and compares them to the free space predictions.

Why Be Concerned About Sidemounting?

Unlike top mounted antennas which typically have no physical obstructions located close to the antenna aperture, sidemounted antennas must consider the tower as part of the antenna system. Even highly directional antennas will still radiate energy toward the tower that may be only 10-15 dB below the peak signal. Two primary effects then result:

1) The tower reflects the energy in different directions rather than letting it pass through the tower structure. This results in a "shadow" of low signal level behind the tower.

2) The tower reflects the energy back toward the main coverage area. This produces a secondary signal of varying phase and amplitude which combines with the primary signal. The result is a "scalloping" of the azimuth pattern signal levels as a function of azimuth angle. The peak to minimum variation in the major lobe area can be

in the ranges of 1 dB to 10 dB.

While the effects may be of concern regarding the coverage requirements of the broadcaster, they are readily determined through the use of reduced or full scale models of the antenna/tower system. Judicious selection of antenna location can be used to minimize the scalloping effect or optimize the location of low signal levels over areas not critical to the coverage requirements, e.g. lakes, swamps, mountains, etc.

Directional Patterns

An antenna with a directional azimuth pattern, e.g. cardioid or skull, lends itself easily to a side mount application. Use of this pattern typically indicates coverage requirements over a 180° to 240° angular range. By proper placement of the antenna on the tower, the scalloping effects can be minimized.

Exhibits 1-3 show the results of model tests for a UHF antenna sidemounted on a 10 ft. face tower.

Omnidirectional Patterns

Historically, omnidirectional antennas have been top mounted or were configured as multiple panel arrays mounted around the tower to produce "omnidirectional" coverage. Due to the large face widths of typical towers used for television broadcasters, the resultant patterns were scalloped similar to the effects described previously. However, satisfactory performance was achieved with the antenna when initially installed.

Over many years of use, the panels and their feed harnesses would tend to deteriorate due to environmental conditions such as a corrosive atmosphere and ice formation or physical factors such as damaged feed lines and radiators from falling ice. These conditions result in a less than comfortable feeling about the radiation performance of the antenna.

A solution to the above scenario is to sidemount a traveling wave type antenna, enclose it in a full radome to protect the radiators and feed components from physical damage and then pressurize the entire structure to eliminate the environmental conditions degrading the antenna performance over the working life of the antenna. Although antennas can be designed to operate in an unpressurized state, the added protection a fully pressurized and radomed antenna provides is a value added feature that can save thousands of dollars in maintenance and repairs over the lifetime of the antenna.

As with the sidemounted directional antenna, model studies can be performed to optimize the mounting location of the antenna and provide satisfactory coverage of the market area. In fact, it may be worthwhile to investigate changing a top mounted omnidirectional antenna to a sidemounted antenna to reduce windloading effects on the tower. This may allow the addition of other communication services to the tower and result in additional income from the leased space.

Exhibits 4-6 show the results of model studies for a UHF omnidirectional antenna sidemounted on a 10 ft. face tower.

What About Coverage Behind the Tower?

As can be seen from the previous discussion and measurements, the tower can cause a deep shadow due to signal blockage. If there is a need for additional signal strength in this shadowed region, it is possible to install a vertically polarized dipole system (if the main antenna is horizontally polarized) on the back side of the tower. The signal to feed the dipole(s) is coupled from the main transmission line producing a *elliptically* polarized antenna system. This configuration has been successfully implemented at various sites to accomplish coverage fill-in. The vertically polarized signal can be received on rabbit ears or loop antennas as well as close in cable receive sites. It does not interfere with the main signal because of the different polarization.

A typical system configuration is shown Exhibits 7-8.

SUMMARY

It has been shown that sidemounting television broadcast antennas can result in satisfactory performance of the transmission system. In addition, coverage of the market area can be optimized through the use of model studies. The benefit of sidemounting is reduced windloads making it possible to add additional antennas to the tower, resulting in increased revenues from lease agreements. It also may provide the possibility of adding a second broadcast antenna to fulfill the needs of the proposed second channel for HDTV.

Acknowledgements

Special thanks to Messrs. Bill Woods and Jim Sheetz of Wisconsin Educational Communications Board and Messrs. Don Watkins and Don Kirby of Nationwide Communications, Inc. for their input and assistance with the described projects.

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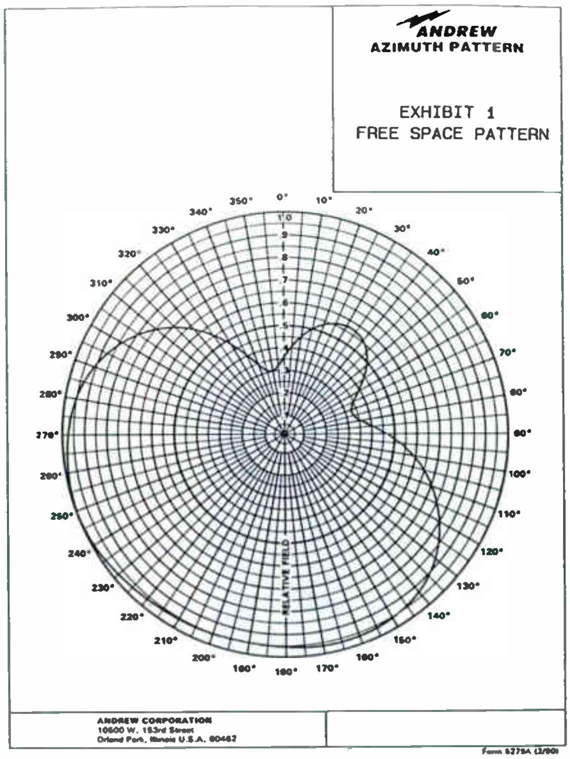
2) Cozad, Kerry W., "HDTV Antenna Placement in a Collocated Situation", presented at the monthly meeting of the AFCCE, October, 1991.



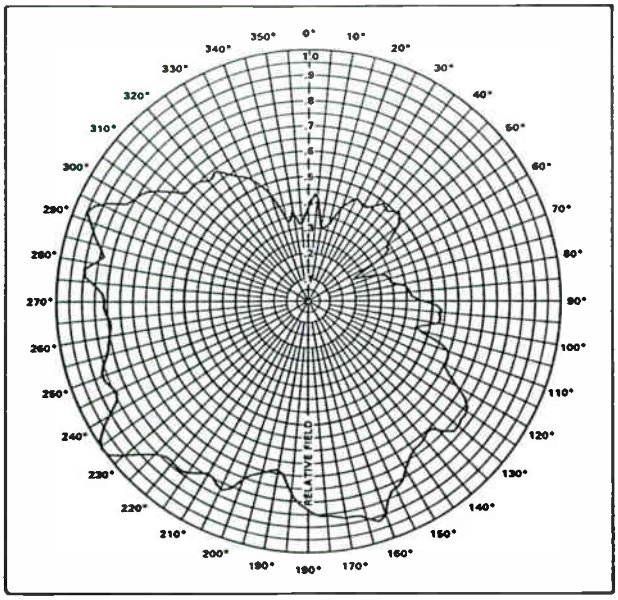


Exhibit 2
Sidemounted Directional Antenna Pattern
based on 0.385:1 scale model of antenna and tower

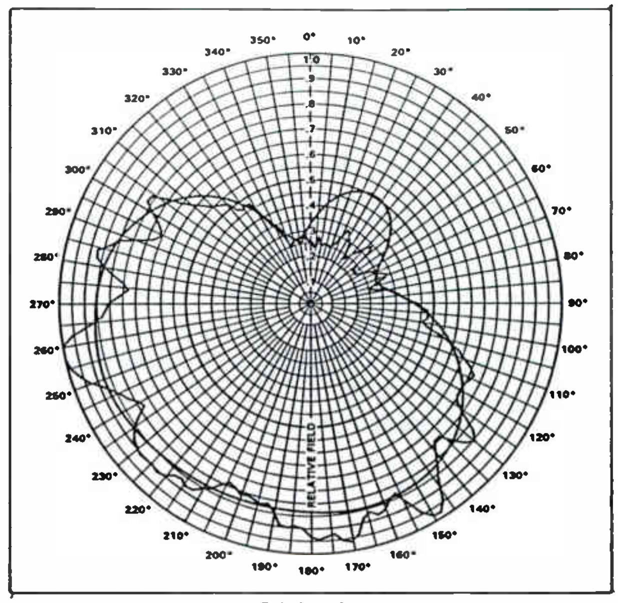


Exhibit 3
Sidemount pattern of full scale
model compared to free space pattern.

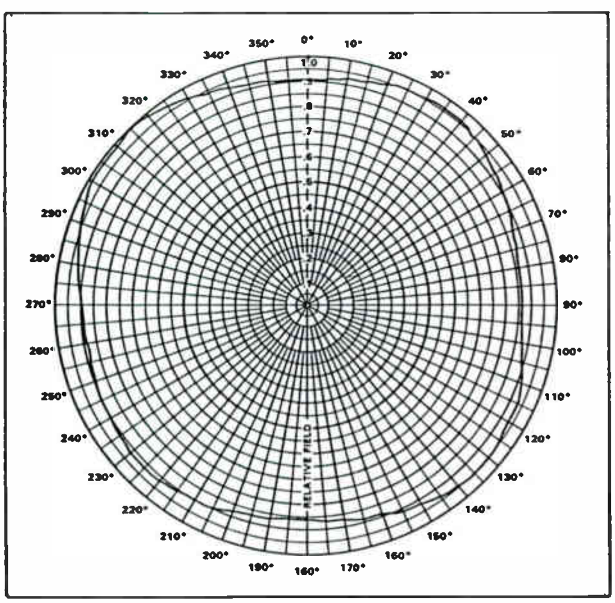


Exhibit 4
Omnidirectional Pattern
Top Mount

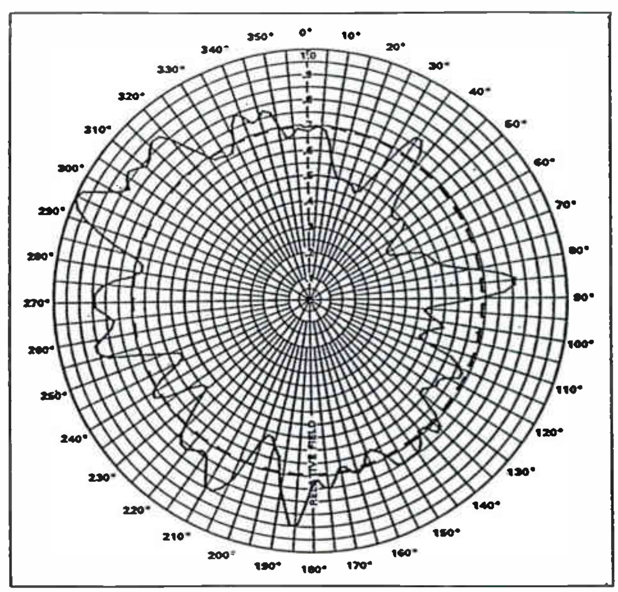


Exhibit 5
Sidemounted omnidirectional slotted array.
Dotted line is equivalent pattern for top
mounted omnidirectional antenna.

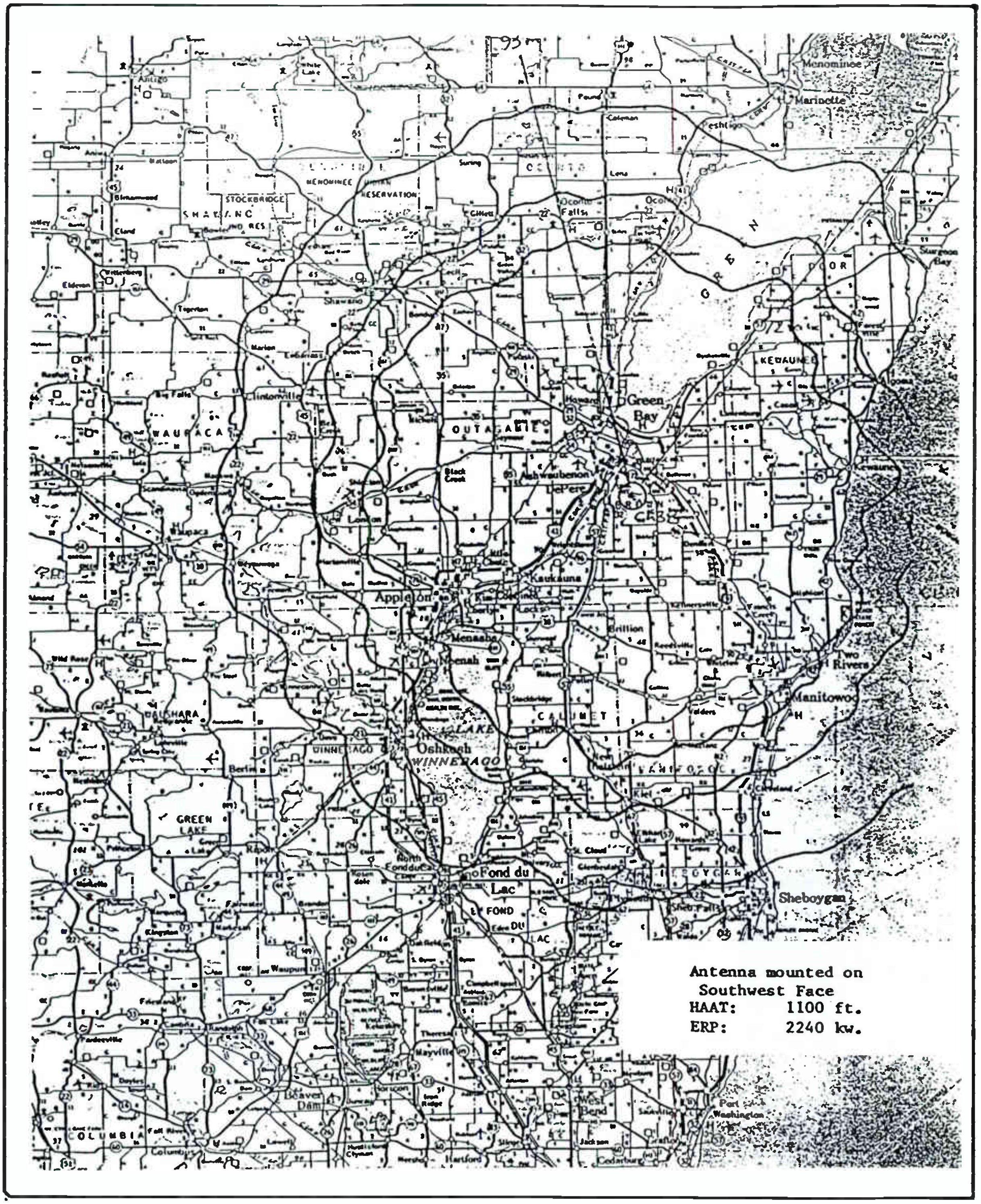


Exhibit 6
Sidemounted Omnidirectional Antenna
Service grade contours showing optimum orientation of antenna for coverage requirements

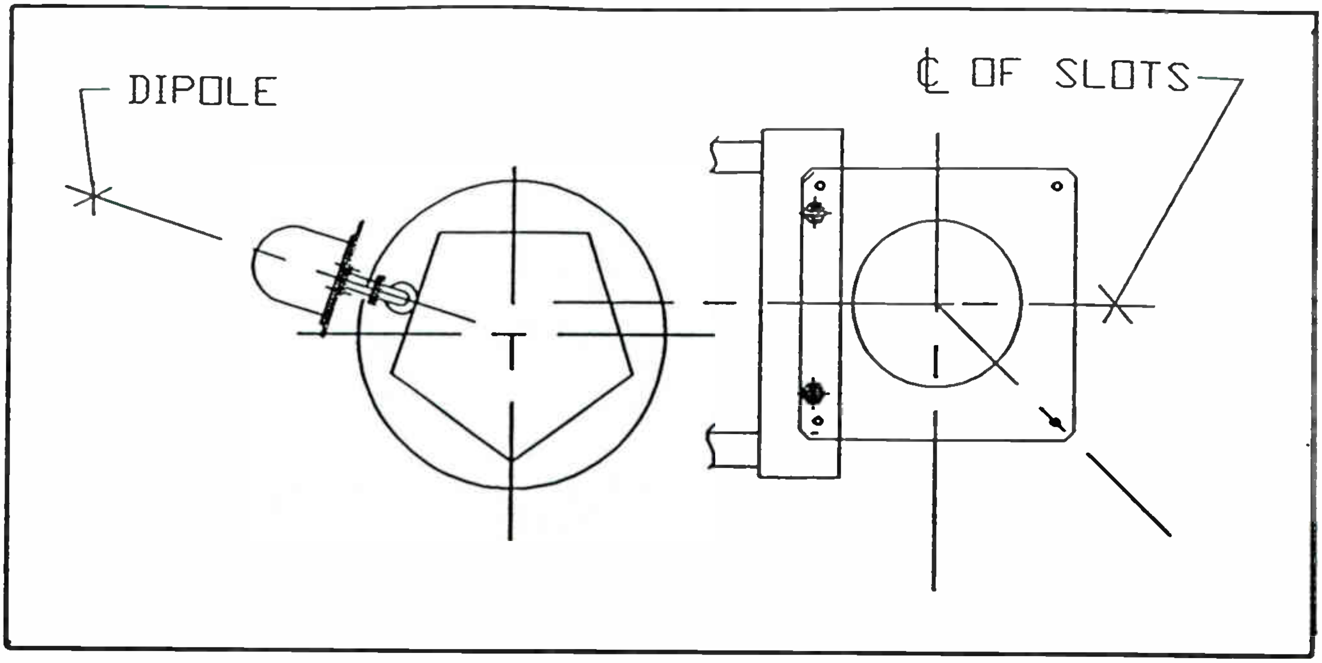


Exhibit 7
Configuration for vertically polarized signal transmitted behind a mounting support of the main horizontally polarized antenna.

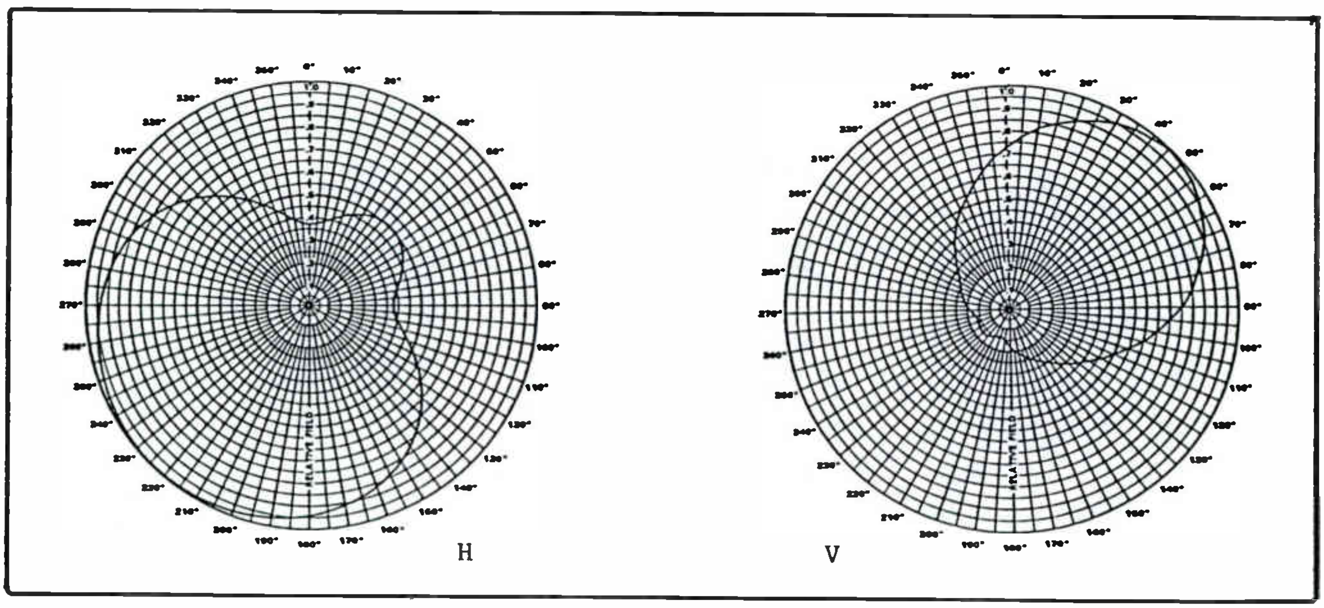


Exhibit 8
Azimuth patterns for horizontal polarization (H) and vertical polarization (V). Patterns do not take into account the power split between polarizations.

ANALYSIS OF FM BOOSTER SYSTEM CONFIGURATIONS

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Abstract

There has been little documentation of the performance of FM booster transmitters using different forms of synchronization with the signal of the main FM station. This paper describes laboratory measurements conducted to study booster systems without synchronization, with carrier synchronization, and with combined carrier and modulation synchronization. The study also examines the beneficial (and possibly detrimental) effects of adding carrier delay to reduce interference in the areas where overlap of the main and booster signals is present.

INTRODUCTION

FM booster stations are a special class of FM translators that receive the signals of a full-service FM station and retransmit those signals to areas that would otherwise not receive satisfactory service from the main signal, due to intervening terrain or other factors. FM boosters operate on the same carrier frequency as their primary full-service station. Among other requirements defined in Federal Communications Commission Rules, a booster's coverage may not extend beyond the predicted class contour of its primary station. [1]

In August 1987, the Federal Communications Commission released a Report and Order that amended its rules with respect to FM and TV boosters. [2] For FM, the amended rules allow alternative signal delivery methods and increased power. Until that time, FM boosters were limited to using direct off-air reception and retransmission methods, with a maximum output power of 10 watts. The rule changes allow, with few exceptions, the use of virtually any signal delivery method, as well as power levels of up to 20 percent of the maximum permissible effective radiated power of the full-service station they rebroadcast.

Since the rule changes were implemented, many FM stations have installed booster systems to improve coverage. Unfortunately, as has been discovered in practice, boosters can also diminish usable coverage in locations where their signals substantially overlap coverage areas of the main facility. Increased perceptions of audio distortion and signal disruption are commonly reported, because the two unequally delayed signals can create significant amplitude modulation artifacts (self-generated multipath distortion) in the overlap area.

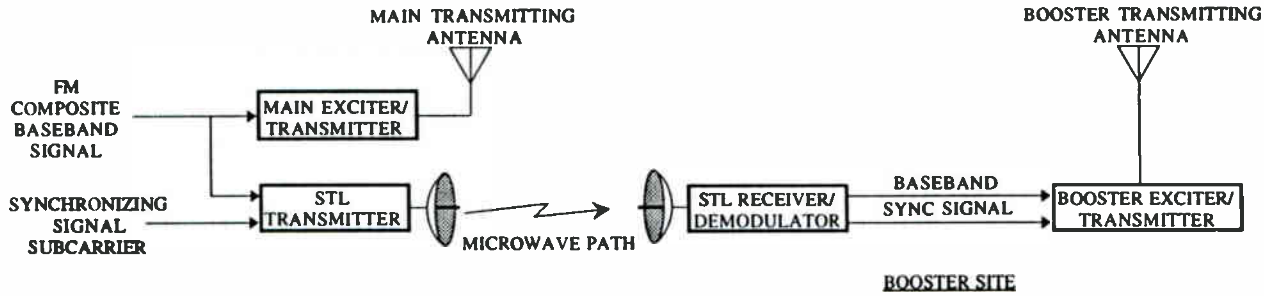
SYNCHRONIZING MAIN AND BOOSTER SIGNALS

Synchronization Methods

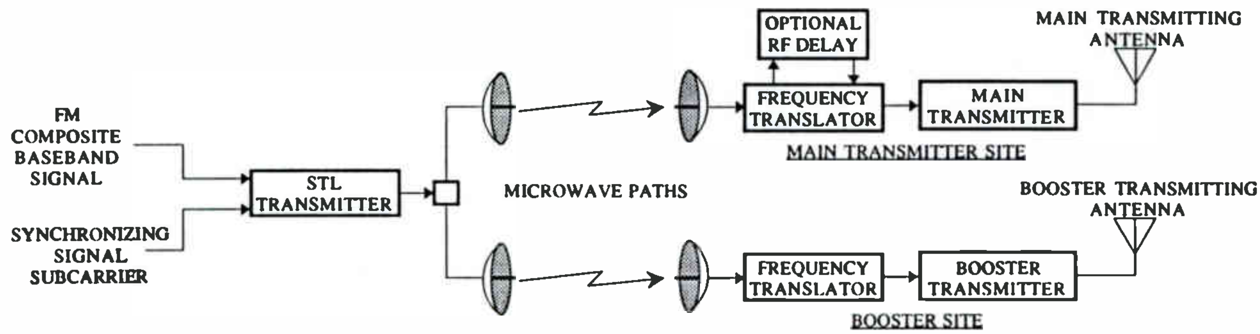
To combat these effects, some attention has been paid to synchronization of the main and booster signals, as well as to careful selection of the booster transmitter location. [3] Synchronization generally involves frequency locking the carriers of the main and booster transmitters, while adjusting the booster composite baseband modulation level to match the main as closely as possible.

Carrier synchronization. An example of carrier synchronization is shown in Figure 1a. In addition to the composite stereo signal, a microwave studio-to-transmitter link (STL) transmitter is fed with a synchronizing subcarrier. One method that has been used to generate the synchronizing signal involves dividing the FM RF carrier signal of the main transmitter by 1000 to produce a subcarrier near 100 kHz, which is extracted at the booster location and used to frequency lock the exciter carrier. Another method uses a high-stability timebase to derive the 19 kHz stereo pilot (as well as the other stereo subcarrier components), which is used at the booster exciter in a similar frequency locking scheme. The STL transmitter feeding the booster link can be located at the studio or the transmitter site of the main station, or at any other site where a stable RF sample of the main signal can be found.

Carrier and modulation synchronization. Both methods described above employ a technique that demodulates the composite baseband signal before it is used to remodulate the booster exciter. Field experience has shown that mismatch of the RF modulation components generated by the main and booster



(a) Carrier synchronization



(b) Carrier and modulation synchronization

Figure 1. Booster synchronization schemes.

transmitters (due to poor level matching, differences in equipment performance, composite overshoot, seasonal drifting, or other factors) may worsen areas of interference created by the implementation of the booster transmitter.

To minimize this effect, a system that uses a single modulator can be employed, as shown in Figure 1b. The modulator in the STL transmitter is used as the only FM modulator in the signal path, feeding both the main transmitter and booster sites. Instead of demodulating the microwave signal at the transmitter sites, it is translated, within the RF domain, to the licensed FM broadcast carrier frequency. The synchronizing signal is used to frequency lock each translator, producing transmitted RF signals that are frequency and modulation coherent.

Carrier delay. Carefully synchronizing the RF signals of the main and booster transmitters does not, however, entirely solve the problems of interference caused when the two signals are received simultaneously. Depending on terrain conditions, there could be a number of locations that receive strong signals from the two sites, but due to unequal signal propagation times and poor summation, a region of interference is still created. Adding RF delay to the main or booster transmitter signal path has been found useful to time equalize the two signals at a particular location, significantly reducing received interference.

LABORATORY STUDY DEVELOPMENT

Study Goals

Even though the described synchronizing methods have been known to improve booster system performance, engineers have relied on heuristics to select a booster site and to calculate interference zones. To provide better guidance, a laboratory study was designed with the following goals: (1) to determine the overall ability of each synchronizing method to reduce self-generated multipath (excessive amplitude components) and receiver distortion, (2) to determine the range of signal levels required to create interference zones, and (3) to determine the useful area of signal improvement when additional delay is added to time synchronize the two signals at a particular location.

Model development

The design of the booster study model assumed a hypothetical flat terrain condition. This selection yielded a simpler analysis than the assumption of arbitrary terrain

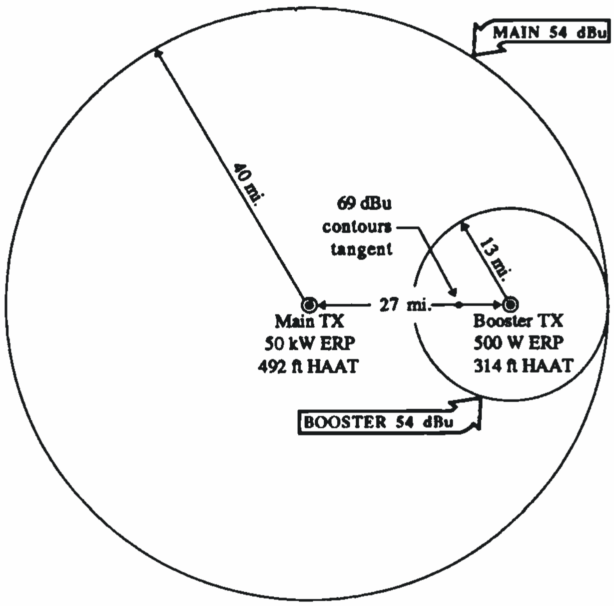


Figure 2. Booster study model.

conditions, and also produced the largest interference zones.

The defined model is shown in Figure 2. The main station was modeled as an omnidirectional Class B facility, having an effective radiated power (ERP) of 50 kilowatts and a height above average terrain (HAAT) of 492 feet (150 meters). Under the assumed flat terrain conditions, the theoretical radius of the 54 dBu class contour of this station (using FCC F(50,50) methods) extended approximately 40 miles (64 kilometers). An omnidirectional booster site was located at two-thirds of this distance, having a power of 500 watts ERP at 314 feet (95.7 meters) HAAT. The radius of the booster's 54 dBu contour extended about 13 miles (21 kilometers), so the two contours were tangent at one point and the class contour of the main station was not extended.

From this model, it was calculated that, between the two sites, the 69 dBu contours of the facilities were tangent to each other, defining the signal level at a point of presumed maximum interference. Considering that most automobiles use a receiving antenna of approximately one-quarter wavelength at FM broadcast frequencies, the following formula was derived from antenna theory to convert the electric field to a voltage:

$$V = E - 36.8 + 10 \log(R) + 20 \log(\lambda) + G - L, \quad (1)$$

where V is the voltage at the antenna terminals in dBuV, E is the electric field in dBu, R is the load resistance across the terminals in ohms, λ is the wavelength in meters, G is the gain of the antenna above an isotropic radiator, and L is the loss of the antenna lead. For this case, a 50 ohm load was assumed, along with a wavelength of 3 meters (100 MHz), an antenna gain of 5.15 dB [4], and no line loss, yielding 64 dBuV or 1.6 millivolts at the antenna terminals for an ambient filed strength of 69 dBu.

Equipment interconnection. The equipment interconnection is shown in Figure 3. A low-distortion audio generator fed the left channel of an FM stereo generator with a 1 kHz tone, while the right channel input was terminated. The stereo generator produced a 100 percent left channel-only composite waveform, containing frequency components at 1, 37, and 39 kHz, in addition to the 19 kHz pilot signal. A high-stability clock source was used to lock the stereo generator timebase to within ±2 parts per million; the 19 kHz stereo pilot frequency served as the system synchronizing signal. [5]

The composite baseband output of the stereo generator was split and fed through two attenuators, one of which was used to vary the signal level in the lower transmission path by ±1.5 dB for level mismatch analysis. The two composite signals fed identical microwave STL transmitters, both of which generated FM-modulated RF carriers (±37.5 kHz peak deviation) at 953 MHz.

After being attenuated to appropriate levels, the microwave signals were connected to an option jumper arrangement, which modified the equipment configuration as required to select the synchronization mode being tested. Option 1 independently connected each STL RF signal to the remainder of the test setup, for analysis of unsynchronized and carrier synchronized cases. Option 2 employed only one STL signal, which drove an RF power divider, for the carrier and modulation synchronized case.

The two STL RF signals were connected to the inputs of frequency translators, which converted the STL RF signals to appropriate FM broadcast signals at 100.7 MHz. In the process, the STL carrier signal deviations were converted to ±75 kHz using an intermediate frequency doubling method. The translator also contained the circuitry needed to lock to the pilot synchronization signal (or to its own internal reference if the stereo pilot was switched off). [6]

Digital time delay units were connected to each translator unit through a 2.5 MHz intermediate frequency loop-through point. [7] Translator B delay was fixed at 150 microseconds, which represented an arbitrary (but reasonable) time period for the signal to travel to the booster transmission site and back out to the 69 dBu interference zone. (At the speed of light, radio signals travel one mile in 5.37 microseconds.) Translator A delay was adjustable to time periods less than or greater than the Translator B time delay; its setting was used as a test variable.

The translator RF outputs, both of which were fixed at +23 dBm (after bypassing the final power amplifiers),

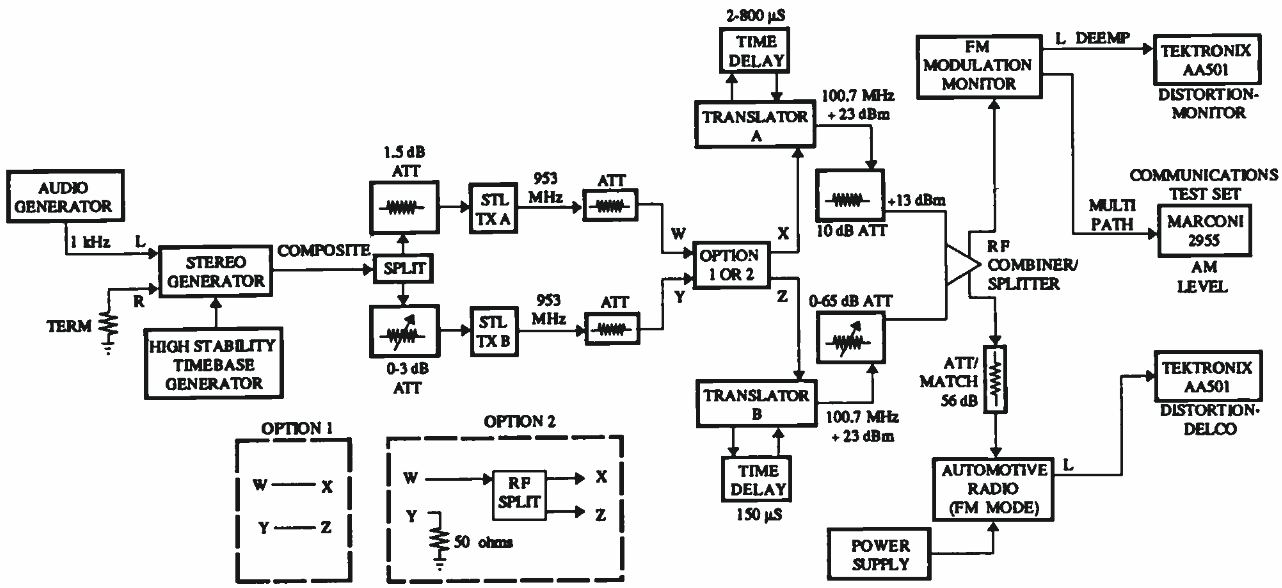


Figure 3. Equipment interconnection.

were each fed to attenuator networks. Translator A, which represented the main transmitter, was attenuated to +13 dBm by a fixed 10 dB pad. Translator B, which represented the booster, fed a 0–65 dB variable attenuator. This allowed the ratio of the two sources to be adjusted over a wide range.

The outputs of the two attenuators were summed and then split to feed a typical FM modulation/stereo monitor and a popular domestic automotive radio receiver, both of which were tuned to the 100.7 MHz carrier frequency. The 56 dB matching pad used with the automotive radio attenuated the combined main and booster signals to about 1.6 millivolts, which corresponded to the antenna terminal voltage calculated earlier.

Measured parameters. Two parameters, audio distortion and amplitude modulation content, were measured for varying system composite level matching, time delay matching, and main/booster signal level ratios. Separate harmonic distortion test sets were connected to the left channel deemphasized audio output ports of the monitor and receiver. A 30 kHz measurement bandwidth was selected. Even though audio artifacts not related to harmonic signals were likely to be present during data collection, they were included in the measurements to produce overall distortion figures.

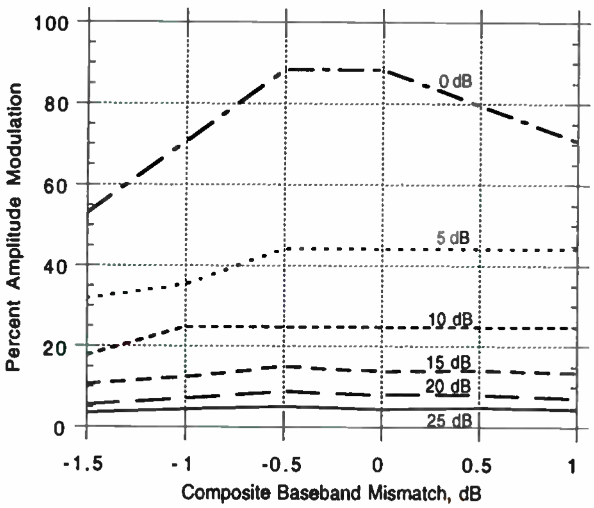
AM content, the other monitored parameter, was defined as the detected amplitude component created by the interaction of the main and booster signals. It was measured using an envelope detector circuit contained in the modulation monitor, which is normally metered to provide a relative multipath indication for receiving antenna adjustment. [8,9] The selection of this detector was based on a review of international studies of automatic deemphasis (multipath distortion masking) circuits employed in automotive receivers. [10] A storage oscilloscope, contained in a separate communications test set, was used to average the monitored amplitude component over several seconds per measurement.

DATA MEASUREMENT AND RESULTS

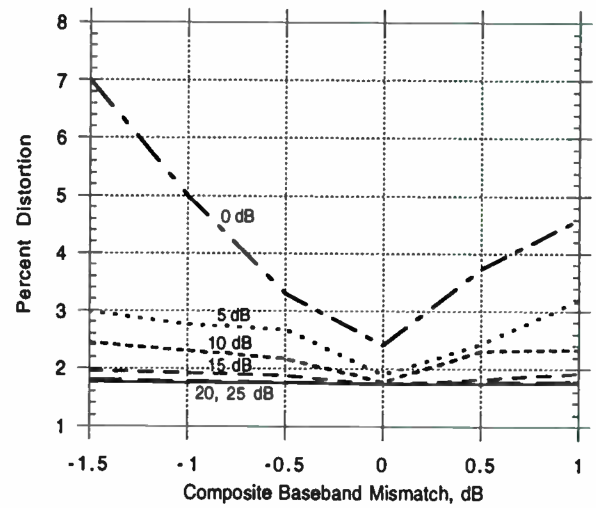
Data Collection

Study data were collected in three phases. During the first two phases, both STL transmitters were operated independently (using Option 1 shown in Figure 3). The third phase employed only one STL transmitter (using Option 2). For all three phases, AM level and distortion readings were taken as a function of main and booster signal level ratios, and as a function of the other parameters noted below. Because there was excellent correlation between distortion readings taken from the monitor and the automotive receiver, only the receiver distortion data are shown in the accompanying graphs.

Phase 1: unsynchronized operation. The first phase measured the system in an unsynchronized mode, with the translators using their own separate internal reference oscillators instead of being locked to the pilot reference (the carrier frequencies were manually matched to within a few Hertz of each other). In this mode of operation, the time delay units were unused. Data were taken as a function of composite level mismatch, over a -1.5 to +1 dB range in 0.5 dB increments. The line graphs of Figures 4a and 4b illustrate the results.



(a) AM content versus composite level mismatch



(b) Distortion versus composite level mismatch

Figure 4. Data taken for unsynchronized operation over 0-25 dB carrier signal ratios.

Figure 4a shows the measured amplitude modulation component for six selected RF signal level ratios versus composite baseband level mismatch. While the baseline AM generation was found to be about 2.5 percent for either of the main or booster systems operating independently, no significant worsening was noted until the RF signals were set to within 20 dB of each other. As the signal level ratio further decreased, the AM component rapidly increased, climbing above 50 percent when the signal levels fell within 5 dB of each other. Careful adjustment of composite level matching produced no significant improvement.

Figure 4b shows measured audio distortion for the same test conditions. No significant worsening of the receiver's inherent audio distortion (about 1.5 percent) was measured until the signal ratio fell below 10 dB, with the ultimate measured distortion reaching seven percent. Also, lower distortion was measured when the composite levels were closely matched.

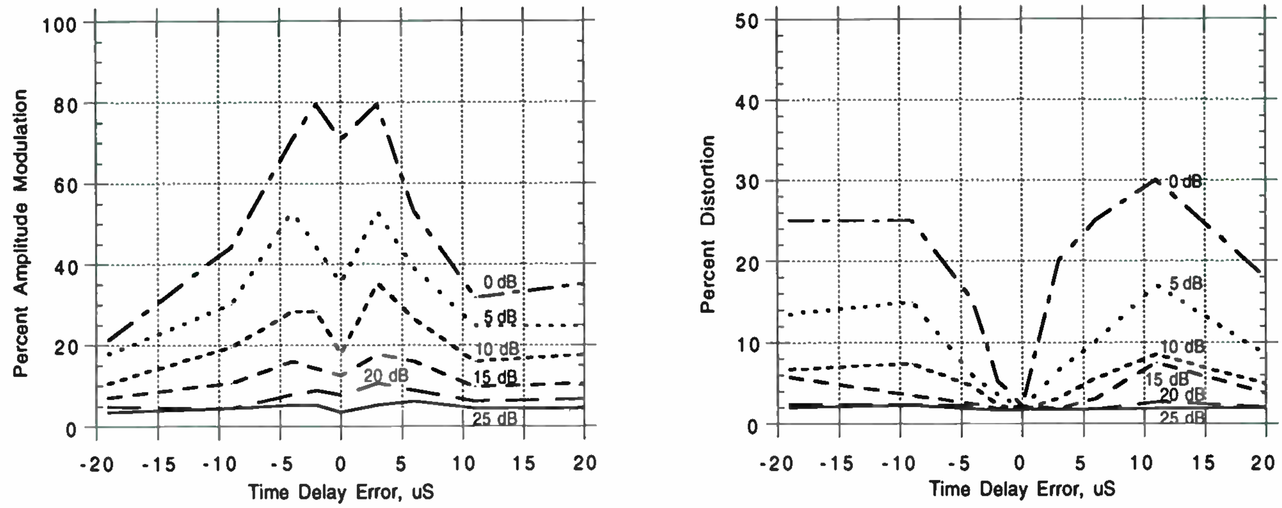
Another test was run with the carrier frequencies of the main and booster transmitters separated by 500 Hz. No worsening of the measured distortion was noted, but the AM level increased more rapidly, exceeding 50 percent at signal level ratios less than 15 dB.

Phase II: carrier synchronization. The second phase of the study measured the system in a carrier-synchronized mode, with both translators frequency locked to the 19 kHz stereo pilot reference. The variable delay unit connected to Translator A was set for nine individual delay times between 131 and 171 microseconds for each RF signal level ratio measured, which was equivalent to a relative time shift between the main and booster signals of −19 to +21 microseconds. Data were taken as a function of the relative time alignment of the main and booster transmitters, as well as the setting of the composite level matching attenuator at the input of STL transmitter A. The graphs of Figures 5a and 5b illustrate the measurement results for equal composite level matching. Figure 5a shows the AM component steadily increasing as the RF signal level ratio decreased, reaching almost 80 percent at 0 dB. Highest levels were found when the signals were nearly time aligned, with somewhat lower levels measured at at perfect time correlation, as well as when the signals were removed in time by more than 10 microseconds.

Measured distortion under the same test conditions is shown in Figure 5b. Almost no distortion increase was measured when the main and booster RF signals were perfectly time aligned, but a rapid degradation was noted at low signal level ratios when the time alignment was shifted by more than five microseconds.

The importance of composite level matching when using this synchronization method was also discovered. A 1 dB mismatch worsened and broadened the area of the measured AM component and distortion to a point about the same as was found for unsynchronized operation. For a mismatch greater than 1 dB, no substantial benefit was gained from time aligning the main and booster RF signals; the AM component uniformly exceeded 50 percent for a signal level ratio of less than 10 dB.

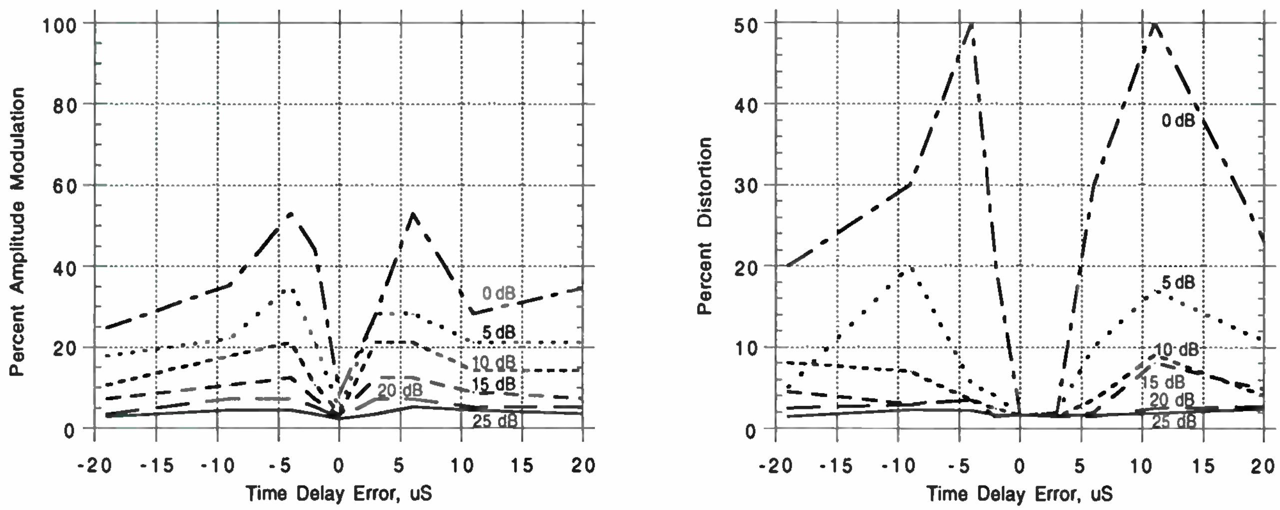
Phase III: carrier and modulation synchronization. The final measurement phase evaluated system performance using a single STL transmitter feeding both translators. Once again, the 19 kHz pilot signal was used to frequency lock the translators. Data were taken as a function of the relative



(a) AM content versus time delay matching

(b) Distortion versus time delay matching

Figure 5. Data taken for carrier-synchronized operation.



(a) AM content versus time delay matching

(b) Distortion versus time delay matching

Figure 6. Data taken for carrier and modulation synchronization operation.

time alignment of the main and booster transmitters, over the same range as the carrier synchronization-only measurements. The measurement results are shown in the graphs of Figures 6a and 6b.

Figure 6a shows the ability of this synchronization mode to reduce the generation of the incidental AM signal. The AM component found at zero time alignment error is virtually non-existent, with generally smaller peaks than were measured in the other test phases.

The range of low measured distortion also was improved using this synchronization scheme. Figure 6b shows that for an RF signal level ratio greater than 5 dB, more than a 10 microsecond region (corresponding to about 2 miles at 5.37 microseconds per mile) was reduced below 10 percent distortion, with higher distortion readings found only at a nearly equal RF signal ratio for relative time delays of five to ten microseconds.

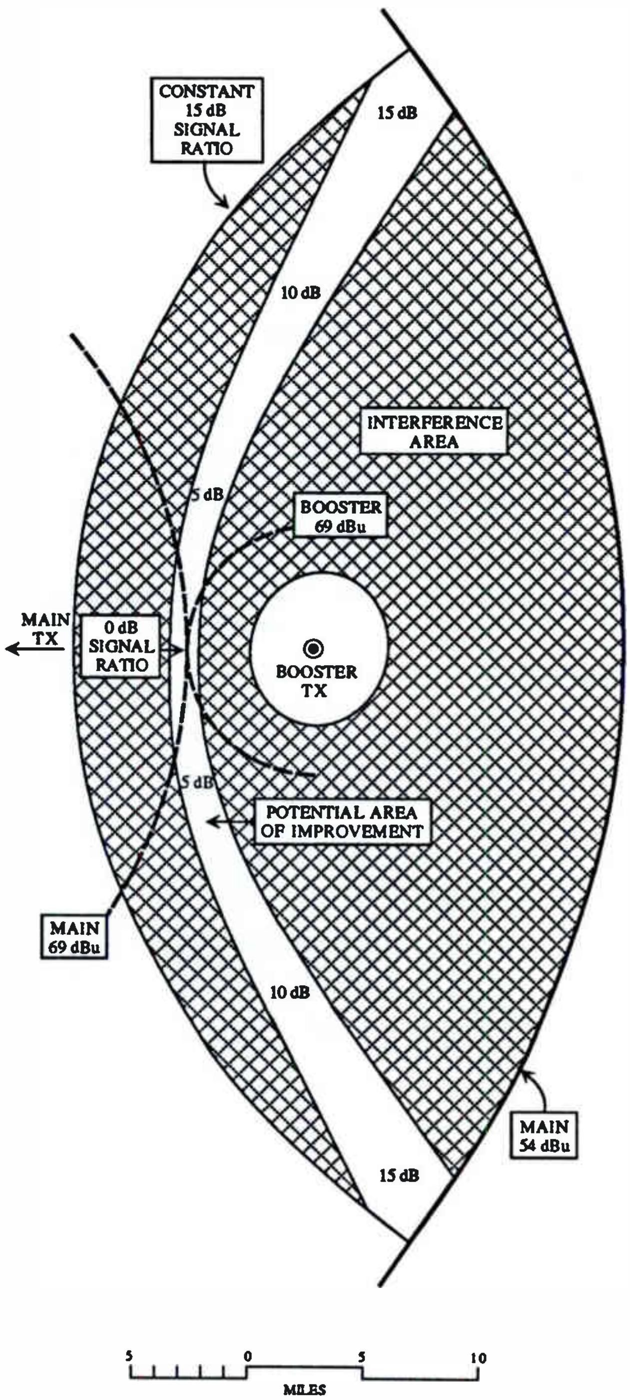


Figure 7. Study model showing interference zone and potential area of improvement.

ANALYSIS AND CONCLUSIONS

Discussion

The measurements taken of the unsynchronized mode configuration were somewhat surprising, in that the measured distortion levels remained low in all cases, regardless of the composite baseband level matching. However, the enormous AM levels found for RF signal level ratios less than 10 dB are almost certainly a dominant factor in creating the multipath-like interference zones that would be quite evident in a moving vehicle.

The carrier synchronization method reduced AM generation and improved distortion performance over unsynchronized operation, as long as the main and booster composite input signal levels and transmission chains were well matched. In practice, however, the AM component may be difficult to minimize in a predictable fashion. As with unsynchronized operation, signal level ratios greater than about 15 dB produced negligible degradation to the received signal, regardless of the relative time delay.

Carrier and modulation synchronization provided the best balance, by greatly reducing the incidental AM component and distortion when the signals were closely time aligned. From interpretation of the improved areas shown in the graphs of Figures 6a and 6b, it was determined that selective time alignment could improve a hyperbolic-shaped area. Once again, if the signal level ratio exceeded 15 dB, incident AM and distortion levels remained low when other parameters were varied.

Figure 7 illustrates an enlarged part of the study model, showing the interference zone and a potential area of improvement when time alignment methods are employed. The interference zone is defined by the shaded region between the boundaries of the main and booster contours that differ by 15 dB, out to the 54 dBu contour of the main station.

The unshaded band shows an example of the area that may be improved by time synchronizing the arrival of the main and booster RF signals. It is narrowest, about 1 mile across, when the signal ratio is 0 dB, increasing to above 2 miles for a 5 dB signal ratio and 3 miles for a 10 dB ratio. By adjusting the time delay, the improved area can be moved to any location within the interference region. In this worst-case example, the potential improvement area is relatively small compared to the total interference area. In real situations that include hills, mountains, or other high terrain between the sites, most of the interference zone likely would be located in unpopulated areas.

Booster Implementation Guidelines

Interpretation of the data indicates that booster transmitters constructed using any of the synchronization methods, including no synchronization at all, can operate without significant disruption in areas where the RF signal level ratio remains above 15 dB. In coverage areas that are well shielded by terrain obstacles, a simple unsynchronized booster may work quite well.

However, when signal overlap with ratios less than 15 dB is predicted in important coverage areas, synchronization methods can provide some improvement within the overlap region. Using both modulation and carrier synchronization provides the potential added benefit of improving system performance over time, because the demodulation and remodulation steps, prone to system drift and aging, are not needed. Additionally, the ability to "steer" reception improvement into critical areas of the interference zone, using precision time delay matching, may prove useful in many implementation situations.

Based on the study results, the following FM booster system design steps are indicated:

1. Identify the approximate boundaries of the coverage problem area from listener reports and informal drive-through reception surveys.
2. Conduct field strength surveys or use a terrain-sensitive computer analysis method to verify the location(s) and severity of the coverage loss.
3. Identify potential booster transmitter sites and antenna coverage patterns.
4. Using terrain-sensitive computer analysis techniques, determine the optimum booster site and antenna system to maximize interference-free coverage of the affected area. Use a 15 dB signal level ratio as the basis to identify interference zones.
5. If all interference zones are shown to be in unpopulated areas, synchronization of the main and booster RF signals may not be necessary. Otherwise, identify candidate location(s) that may be improved using RF time synchronization. (Typical examples of important locations include major access roads or areas of significant population.) Calculate the time delay required (at 5.37 microseconds per mile) to time align the signals in the chosen area.

It is understood that real world conditions, including multipath distortion and other factors unrelated to the booster system, can degrade the ultimate improvements suggested by this laboratory study. Future field study experience should provide additional guidance in confirming the validity of the data presented.

ACKNOWLEDGEMENT

Funding for the measurement portion of this study was provided by TFT Inc. The author would like to thank Joseph Wu, Darryl Parker, and Terry Peterson of TFT for providing laboratory space, equipment, and guidance in developing the tests.

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THE MOUNT DIABLO BOOSTER SYSTEM

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Abstract

Four San Francisco, California FM stations, faced with terrain-related coverage problems, designed a successful multi-user high power synchronous booster providing improved service to listeners in Contra Costa County.

Introduction

Although wonderful to view, the spectacular hills of the San Francisco Bay Area give headaches to FM engineers. When the area's FM stations began operations in the 1950's, transmitter locations were chosen based on the major population centers of the time: San Francisco, Alameda, and San Mateo Counties. The three main FM transmitting sites are Mt. San Bruno, on the San Francisco/San Mateo County border; Mt. Sutro, in the center of San Francisco; and Mt. Beacon, north of San Francisco in Marin County.

However, population spread to neighboring counties. Santa Clara County boomed in the 1970's when its orchards became "Silicon Valley", and a lack of space for growing numbers of families elsewhere created scores of new bedroom communities in Contra Costa County during the 1980's.

Stations that had located their transmitters to cover the first three counties found themselves unable to serve these two new areas. Santa Clara County is mostly outside of the predicted 0.5 mV/m Class B contours, and the population centers of Contra Costa County are in a narrow valley shielded by a ridge from the transmitter sites. While there is little that a station can do legally to improve their coverage in Santa Clara County, most of Contra Costa County is inside of the station's predicted 0.5 mV/m contour. Figure 1 shows transmitter locations and the predicted contours.

Booster History in Contra Costa County

In the early 1970's, KKHI installed a 10 Watt booster to cover just the city of Walnut Creek, at that time the only major population center in the county. The booster's signal improvements were obvious to other broadcasters and KFOG became the second station to use this site. Because this site was designed piecemeal, with stations being added one at a time, the combining equipment was less than ideal and the site suffered from numerous technical problems. Around 1980, the owner of the site terminated the lease and the site was moved one house down the hill.

Based on our experience with the first site, KFOG planned and managed the second location with the clear intent of having many users. The transmit combining was standardized and technical standards were implemented. The second site operated successfully for almost ten years, providing improved coverage for Walnut Creek. However, terrain limitations prevent this location from serving Concord, to the north-east of Walnut Creek. Another site, Briones Park, covers Concord well (but not much of Walnut Creek); this site is used by several other stations. A third site, Wiedemann Hill in southern Contra Costa County, has had as many as seven stations using 10 Watt boosters and a variety of antennas to cover Danville, Pleasanton and San Ramon.

Because these sites are terrain shielded from each other, and parts of the county are not served by any of the sites, less than one half of the county received improved FM reception. Not all stations are at all sites, so the amount of improvement varied by station.

Contra Costa Improvements

In 1987 Mr. Kevin Mostyn, at that time Director of Engineering, KSFO/KYA-FM, and I started looking at improving the FM reception throughout most populous areas of Contra Costa County. We decided to focus on the areas along the Highway 680 corridor, where most of the population was concentrated. Other communities nestled in folds of the hills simply could not be covered cost effectively. We spent the summer of 1987 driving around looking at possible sites with the goal of finding hills that could selectively illuminate areas with 10 Watt boosters. We learned that (1) there weren't always hills in the right places, (2) even if the hills existed, getting permission to install a booster on many sites would be difficult, if not impossible, and (3) the more transmit locations that we had, the greater the likelihood of interference caused by overlap of booster coverage.

Transmit Antenna System Design

It became apparent that the only realistic solution would be to move the booster transmit location to the top of Mt Diablo, located to the east of the valley. Mt. Diablo is the highest peak in the San Francisco Bay Area (3489 feet AMSL), has line of sight to most of the San Francisco Bay Area (at one time EIMAC operated an experimental 500 kW FM transmitter from Mt. Diablo which covered, literally, most of California), and is almost on the edge of our proposed 0.5 mV/m Class B coverage. As a result, it required a very sophisticated antenna system. The scope and cost of the project grew enough that we began to plan on accommodating other stations and informally sought other interested FM stations to share the cost. The antenna would have to cover at least the entire commercial FM band.

The four stations that participated in the first study, performed by the firm of Hammett & Edison were KFOG, KYA-FM, KYUU, and KSOL. The result of this study (Figure 2) was a booster radiation pattern useful to stations broadcasting from all three FM main transmitter sites to provide significantly improved FM reception to most of Contra Costa County with minimal impact on FM reception in other areas.

Our attempt to find an antenna manufacturer was further complicated by the requirement that we use as little vertical tower space as possible, while having some gain in the desired directions. Because there were already many antennas mounted to the single tower available, we were prevented us from using standard broadcast panel antennas.

The antenna design chosen is comprised of three pairs of log-periodic antennas. Because the antennas are not mounted with their phase centers in line, Hammett & Edison used specialized software to calculate the composite pattern. The antennas are rear-mount versions of Scala CLFM log periodic antennas with custom splitters and phasing harnesses.

We wanted the flexibility to allow each station to change their pattern slightly by using more or less power to each of the three paired antennas. This meant that the transmitter combining had to be done individually for each pair of transmit antennas. We first went to broadcast manufacturers for combining, but their equipment was designed for much higher power and was both too big and too expensive. (The entire facility had to fit into a 8' by 20' shipping container.)

Further research led us to the EMR Corporation. Unlike most two-way manufacturers who view broadcasting as a totally alien field, EMR had considerable experience with broadcasting, as their President, Mr. Bill Lieske, Sr., had put several FM stations on the air. (In fact, his first comment was "We will have to build in enough bandwidth so that your SCA's aren't harmed.") Their combining system was custom-built for us, delivered on time, under budget, and exceeded all specifications.

We left the actual booster power amplifier equipment up to the stations involved because we could not agree on standard equipment.

By this time, KYUU dropped out of the group and was replaced by KSAN. We all filed Construction Permit applications based the composite antenna pattern and started negotiations with the tower owner for use of the tower. Our companies started working on a partnership agreement to cover the cost of construction, budgeted at over $300,000.

Design Microwave to Feed Mt. Diablo

Due to the congestion of the Part 74 Radio STL/ICR band in the San Francisco Bay Area, it was not possible for each station to install its own ICR to Mt. Diablo. Therefore, we decided that a common off-air receive site would need some form of a microwave link to our booster site on Mt. Diablo. KIOI had a high power booster that was using one system with Part 74 950 MHz STL's to Mt. Diablo and we had plenty of opportunity to review this system's lack of synchronization. While the average frequency was probably accurate, the two FM signals (main and booster) were never the same during modulation, and in areas with coverage overlap, there was easily detectable interference - the station and its booster were actually interfering with each other. We also noted that it took a lot more than 20 dB desired-to-undesired before this interference would become unnoticeable.

KSOL and KYA-FM were in a hurry to get on the air and had Part 74 STL channels they could reuse. They selected another synchronization system, vastly superior to the system that KIOI was using (but not perfect.) The other two stations, KSAN and KFOG did not have a Part 74 frequency and decided to look further.

We investigated equipment regularly used by cable TV operators to link systems together, commonly called "CARS band" microwave. This equipment uses the 12 GHz band with direct up-conversion of RF carriers to the microwave frequency. Because the system includes a pilot from the transmitter to the receiver's local oscillator, there is no frequency difference from input to output. Also, because the audio is never demodulated or remodulated, the booster's signal is identical (and only time-delayed) from the main transmitter's signal.

This appeared to be exactly what we needed. Hughes Microwave was our choice for equipment, and we presented our needs to the company. While they normally sell the equipment to a different market, they were interested in our application and worked with us.

The problem was licensing the equipment. The stock equipment is only made for the Community Antenna Relay Service, and licenses are only available to cable TV operators; getting a license in this band would require waiving too many rules. Could Hughes Microwave provide equipment on a custom basis for nearby frequencies that could be licensed in Part 94? These frequencies can not be used as the last link to a Part 73 broadcast transmitter. Although a booster is considered an "auxiliary" service in Part 74, the Part 94 branch of the FCC decided that a transmitter is a transmitter and requested that we formally apply for a waiver. After extensive consultation with our lawyers in Washington, and technical assistance from Mr. Michael Newman, of C.S.I. Telecommunications, we were granted a waiver on a coordinated frequency in Part 94 (Public Notice DA 90-700 released May 16, 1990). The license granted, we then placed the order with Hughes Microwave for the equipment.

In the mean time, we had selected the KYA-FM transmitting site at San Leandro as our off-air reception point. This site has excellent line-of-sight to all three San Francisco transmitter locations and to Mt. Diablo. Our head-end equipment from Catel receives individual stations and provides a consistent output for the microwave transmitter. Because KYA-FM is also transmitting from this site, we included a notch filter for KYA-FM to keep it from overloading the Catel processors. The receive antenna is a Scala CLFM.

System Performance

A complete system diagram of what is on the air today is shown as Figure 3. Although the total system is a "one of a kind" system, all of the blocks are assembled from "off the shelf" components.

Two of the stations have been using the site since the summer of 1990. Coverage appeared to be as predicted, although the nulls between antenna directions was greater than predicted for KYA-FM (at 93.3 MHz) and less than predicted for KSOL (at 107.7 MHz). With further antenna research by Hammett & Edison simple changes were made to the systems to fine-tune the performance of the antenna system at any particular frequency.

The CARS microwave system was installed in January 1991, which allowed KFOG and KSAN to begin operation. The microwave system also allows KYA-FM and KSOL to have the option of fully synchronized transmission.

As more time was clocked gathering empirical data on the performance of the booster system, we began to detect that there was not enough signal on the ground in areas that were direct line-of-sight to Mt. Diablo. This was most noticeable in some of the more built-up areas like Walnut Creek. For example, when a four story building was between the receiver in a car and Mt. Diablo, the signal faded.

In addition, we noticed interference from the booster to our main signal in areas shielded from the main transmitter but still with line-of-sight to Mt. Diablo, even when quite distant from Contra Costa County. In fact, with the main transmitter turned off, a useable signal from the booster still covers parts of San Francisco, San Mateo and Marin Counties. These areas were far from Mt. Diablo; we concluded that there was far too much signal on the horizon.

Experiments showed that increasing the amount of power transmitted from Mt. Diablo helped the shadow areas of the booster in Contra Costa County, but at the cost of major objectional self-interference in areas shadowed from the main transmitter.

We returned to Hammett & Edison to review the antenna performance in all three dimensions, as previous studies were only made in two dimensions. We also wanted the antenna system revised to get more signal on the ground and less signal on the horizon. This could be described as beam tilt or a horizon null; they describe two different results of the same effect. Hammett & Edison modified their computer program so that it could synthesize antenna patterns in all three dimensions with asymmetric mounting centers.

As this paper is being written, the engineers in the partnership are reviewing the results of the studies.

Credits

Any project as large as this required the cooperation of many people. First I would like to acknowledge and thank my good friend Mr. Kevin Mostyn formerly of KSFO/KYA-FM for his help and work on this project. Also Walt Ellis (at that time with KYUU), Mr. Kevin Douglass (at that time with KSOL), and Mr. Chuck Waltman at KNEW/KSAN for being the rest of the "Gang of Four" that put this project together.

We could not have completed this project without the services of Mr. Robert Hammett of the firm of Hammett & Edison.

Mr. Michael Newman of C.S.I. Telecommunications helped steer us through the intricacies of Part 94 frequency coordination and license application.

Mr. Norman Woods and Dr. Thomas Straus of Hughes Microwave listened to our requests and built the special microwave system that we are using.

The redundant redundancies and the unclear obfuscations having been removed, the credit for editing this paper should go to my loving wife, Siobhan.

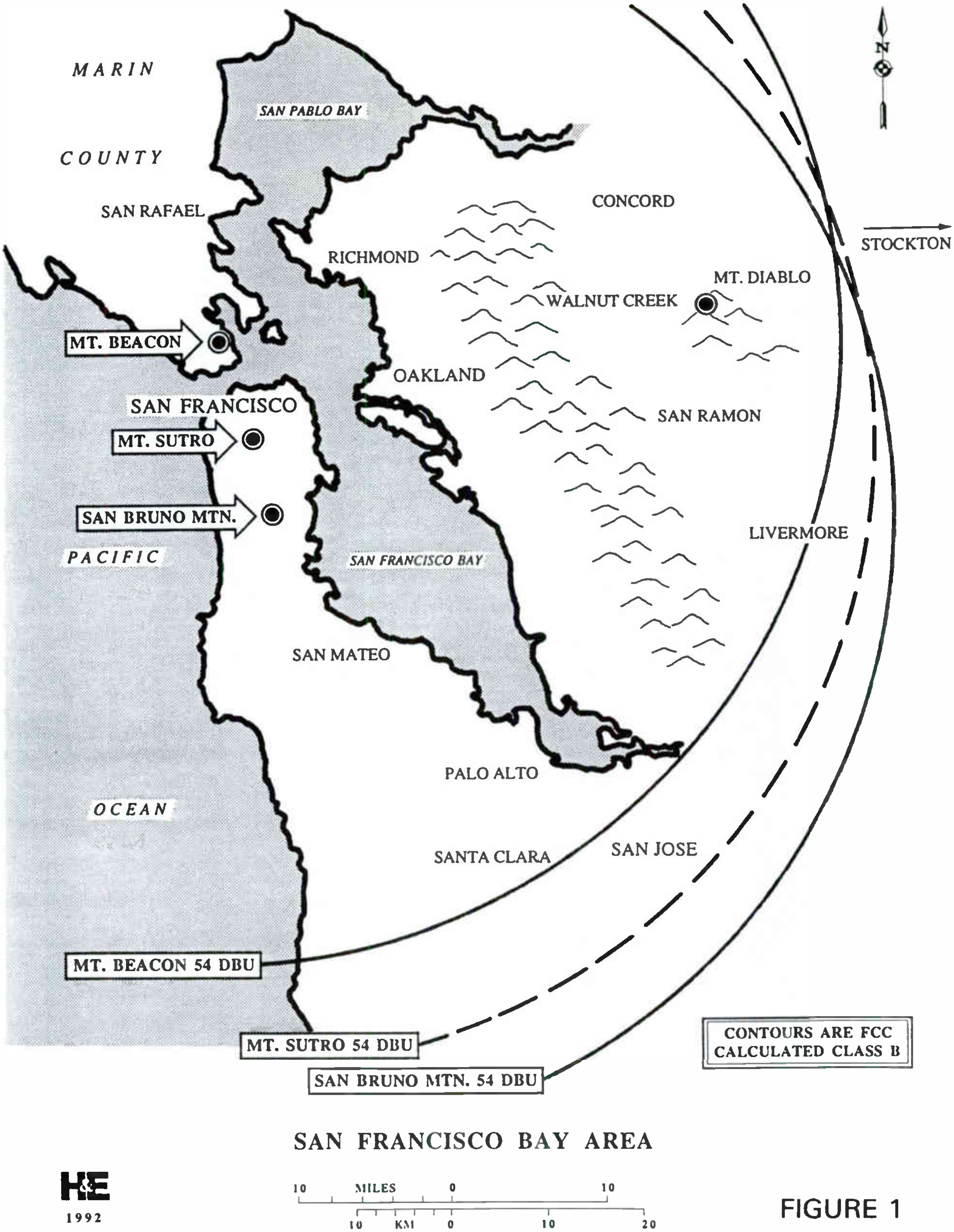
MARIN
COUNTY
SAN PABLO BAY
SAN RAFAEL
CONCORD
RICHMOND
STOCKTON
MT. DIABLO
WALNUT CREEK
MT. BEACON
OAKLAND
SAN FRANCISCO
SAN RAMON
MT. SUTRO
SAN BRUNO MTN.
LIVERMORE
PACIFIC
SAN FRANCISCO BAY
SAN MATEO
PALO ALTO
OCEAN
SANTA CLARA
SAN JOSE
MT. BEACON 54 DBU
MT. SUTRO 54 DBU
SAN BRUNO MTN. 54 DBU
CONTOURS ARE FCC CALCULATED CLASS B
N
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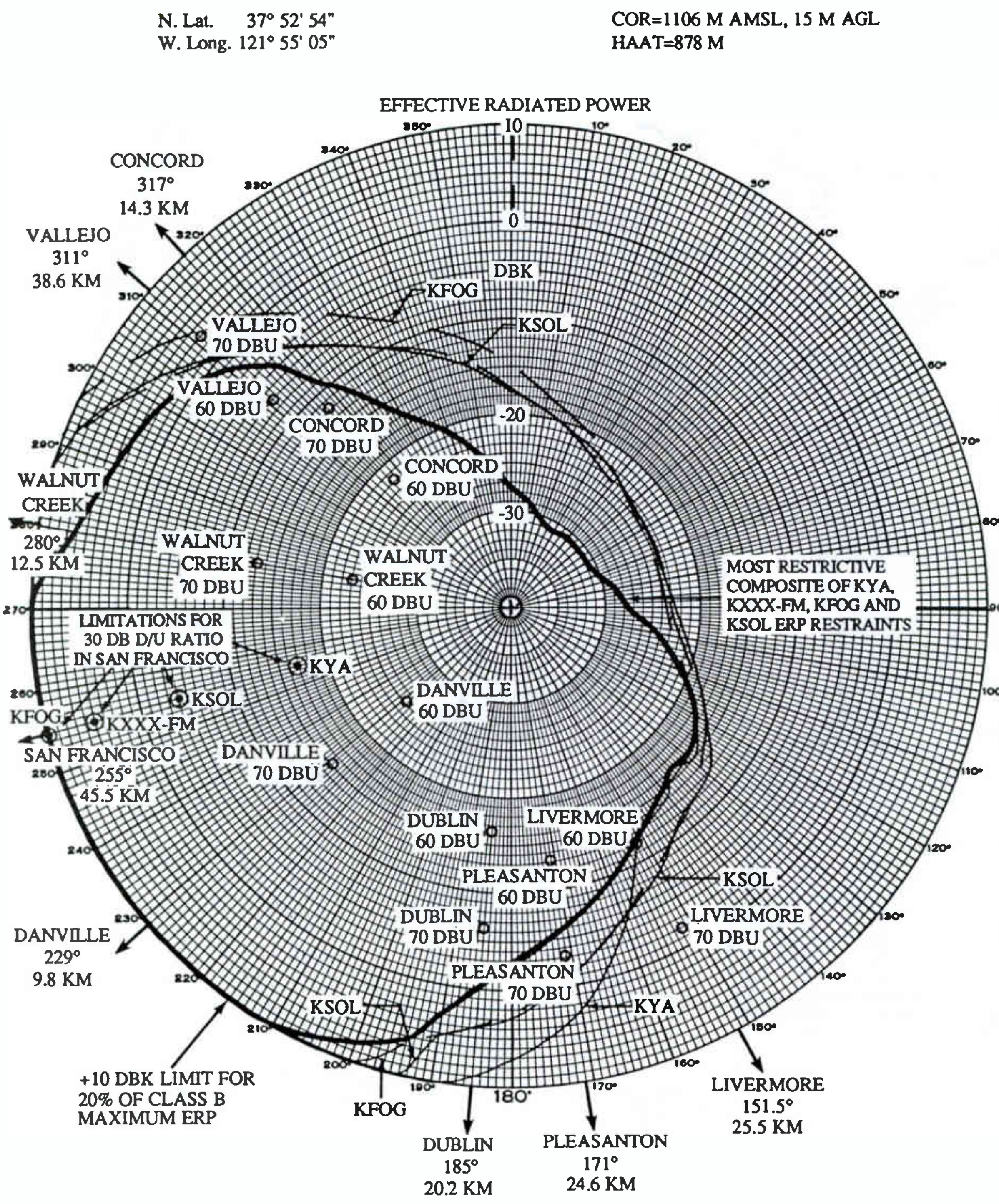
FIGURE 1



		COMPOSITE ERP RESTRAINTS FOR MT DIABLO FM BOOSTER
HAMMETT & EDISON, INC. CONSULTING ENGINEERS SAN FRANCISCO	FM STATION KFOG 104.5 MHZ, CHANNEL 283B SAN FRANCISCO, CALIFORNIA	881003 FIGURE 2

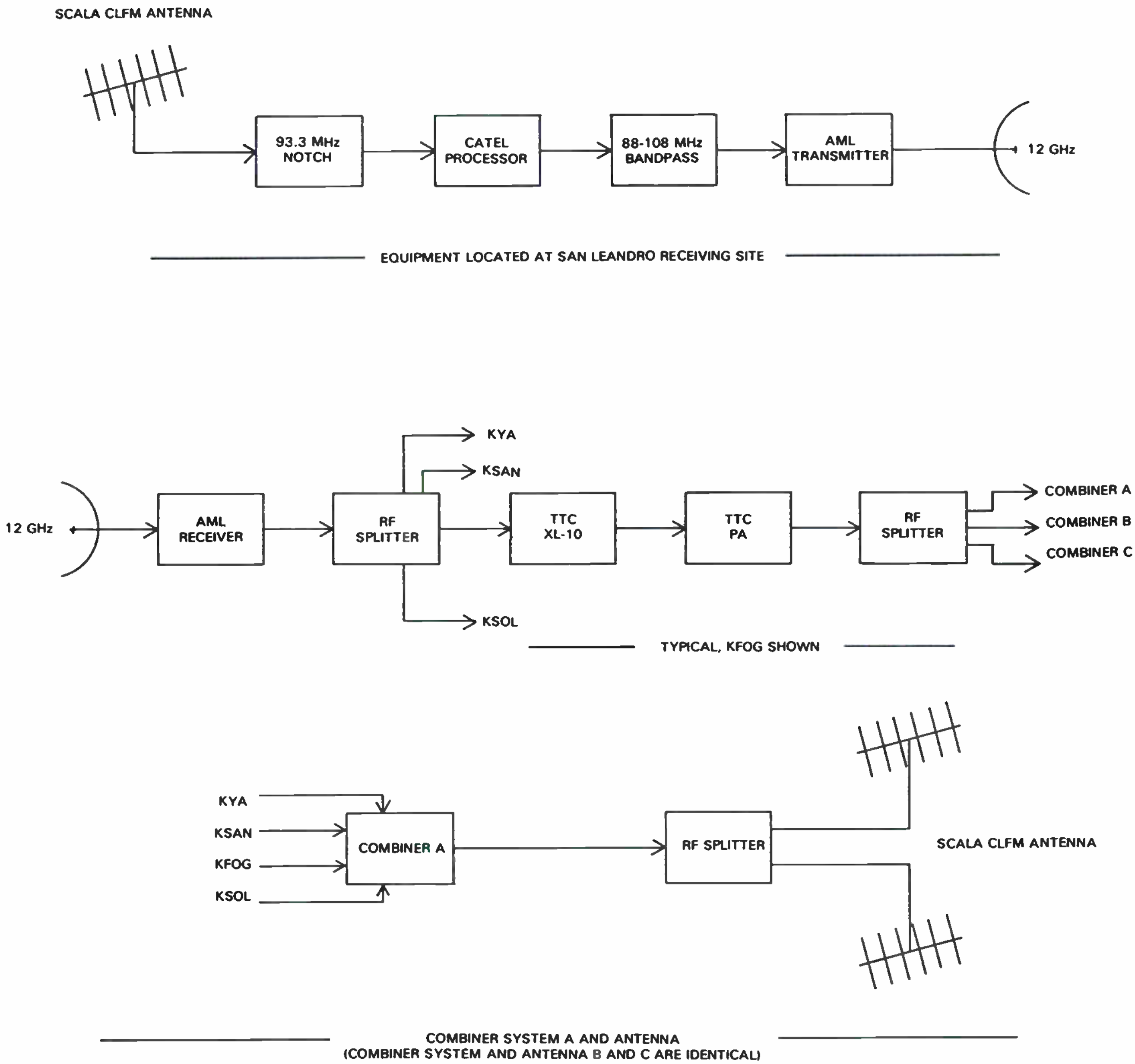


FIGURE 3

MT. DIABLO BOOSTER SYSTEM

SIMPLIFIED BLOCK DIAGRAM

OPTIMIZATION OF VHF EFFECTIVE RADIATED POWER AND ANTENNA HEIGHT COMBINATIONS

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Abstract

A method of establishing antenna height limits for achieving reliable VHF broadcast service is described using FM radio transmission as an example. FM receiver performance specifications were analyzed to determine a practical value of minimum field strength for satisfactory stereophonic reception. Antenna heights and effective radiated powers were adjusted until the specified field strength value was predicted at the radio horizon for each U.S. FM station class. The height solutions obtained define a practical upper limit for FM antenna height under average terrain conditions, as well as a field strength value that better defines the extent of FM service in the absence of interference. This technique is applicable to VHF television situations by using TV receiver performance data in place of the FM criteria used herein.

Introduction

For many years, a variety of opinions has been expressed as to what combination of transmitting antenna height and effective radiated power is optimal in VHF FM and TV broadcasting. Unfortunately, there does not appear to be any practical guidance available which is based on careful reasoning or relevant theory. Much information seems to fall into the category of "old wives' tales" or rumor.

VHF broadcasting is considered by many to be a line-of-sight service. Coverage beyond the horizon is generally expected to be unreliable. People presume that station service will expand if the line-of-sight distance from the antenna to the horizon is increased. However, as will be developed herein, there are practical limits to the amount of service expansion that may be so realized.

In order to regulate interference between stations, upper limits have been established for antenna height and transmitter power for VHF FM and TV stations by the Federal Communications Commission (FCC) and its foreign counterparts. If height is increased above the nominal limit, power must be reduced to maintain the station's service radius within the maximum range permitted for its particular class. A reduction in power lowers the density of a station's signal both at its coverage fringe and within its close-in service area. Accordingly, height increases above established maxima do not result in unlimited benefits in terms of station reach.

The premise of this paper is simple: a station's service "reach" should not exceed its coverage "grasp". In other words, the line-of-sight distance to the radio horizon should not be so great that the signal strength there is attenuated to the point where reception quality is unacceptable. We may define practical upper limits for FM antenna heights for use where obstructions do not dictate the use of higher elevations. The same technique may be applied to television station situations if TV receiver performance data is used in lieu of the FM data presented herein.

Receiver Performance Criteria

Acceptable reception is dependent upon three factors: the signal strength achieved within the service zone of interest, the magnitude of potentially interfering signals in that region, and the minimum signal strength necessary to result in acceptable performance in the consumer's receiver.

The broadcast station has control of its signal strength only. Interference is a function of the type of regulatory structure adopted by the government and the distribution of potentially interfering stations. Each station situation *vis-a-vis* interference is unique, so no generally applicable conclusions regarding the effects of interference on station service can be drawn. The performance of consumers' receivers is entirely unregulated, but can be understood by reference to relevant performance data.

The receiver performance parameter of primary interest here is its sensitivity. Sensitivity is defined as the signal level at the receiver's input terminals necessary to cause a specified performance characteristic to be achieved at the audio output of that receiver.

Sensitivity figures are usually specified by receiver manufacturers and/or independent reviewers for both monophonic and stereophonic operation. The sensitivity specification, expressed in input power terms (dBf), is usually based on achievement of a particular level of quieting of the receiver output, the latter expressed in dB. In other words, for a given receiver input signal level, say 40 dBf, the output will be free of receiver-generated noise down to, say, 50 dB below the 100 percent modulation reference audio signal level.

The dBf power unit is decibels above one femtowatt, 10^{-15} watts. How is this level related to field strength? The detailed derivation of the pertinent equation is beyond the scope of this paper, but the end result thereof is:

$$(1)\quad dB\mu = dBf + 20 \log F - 45$$

where: $dB\mu$ = field strength
dBf = receiver input power
F = frequency in MHz

At 100 MHz, $dB\mu = dBf - 5$. Of course, this equation presumes perfect, lossless coupling between an ideal antenna and the receiver. Most installations of consumer receivers are far from ideal. Use of a 3 dB derating figure is considered appropriate to cover connector, cable mismatch, and antenna deficiencies.

Receiver Data

Good statistics on the performance of typical consumer receivers are hard to come by. The most readily available sources of information are the specification sheets of the buyer's guide editions of consumer audio magazines. This data is far from ideal, but, at the time of this paper's writing, was the only pertinent, large sample available.[1] Furthermore, the primary purpose of this paper is to illustrate the technique for defining antenna elevation maxima. The actual values may be recomputed readily when a better receiver data sample is available.

Our purpose here is to define a range of receiver performance. Dishonest "specsmanship" will tend to be balanced by overly conservative performance claims when a large data sample is considered. The overall statistics will still show performance trends that are likely to be useful, particularly if the application of that data yields results that are consistent with our general experience.

First, the performance ranges for 82 component-type tuners and 129 similar receivers were established from the data contained in a recent consumer audio magazine.[2] Receiver input sensitivity was defined in terms of 50 dB output quieting for monaural and stereophonic reception. A statistical analysis was performed to determine the trend of component-type receiver sensitivities. The results obtained are set forth in Table I.

Sensitivity data for automotive receivers, which account for a large proportion of FM listenership, especially along the fringes of station service areas, was defined in the pertinent buyer's guide specifications only for monophonic reception.[3] To obtain data useful for estimating stereophonic reception requirements in the automotive reception environment, a projection was made on the basis of the known component receiver and tuner data.

A regression analysis was performed to establish correlation between monophonic and stereophonic performance for the receivers and tuners. The resulting equation, which exhibited reasonable correlation consistency ($\pm$2.5 dB) within the receiver/tuner data set, was then applied to the automotive receiver monophonic data to approximate stereophonic performance. Statistics for the resulting projected car receiver performance were then computed, as shown in Table I.

A probability curve was fit to the stereophonic sensitivity projected for automotive receivers. Figure 1 illustrates the result obtained. Table II summarizes the resulting data, incorporating the 3 dB derating figure mentioned earlier when converting sensitivities to necessary field strengths.

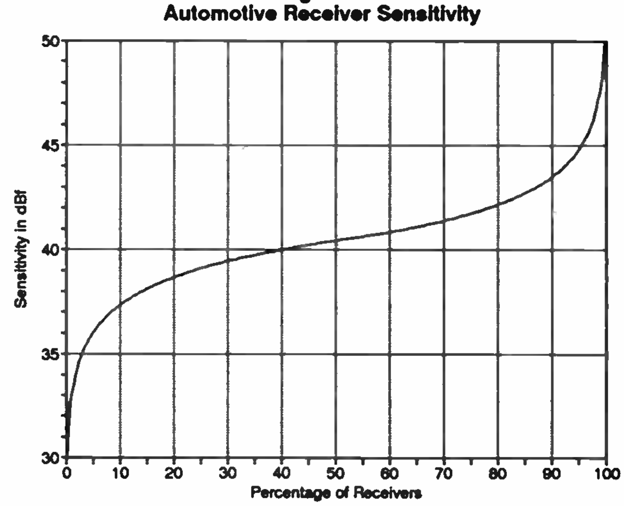
Table I: FM Receiver Sensitivity

Receiver Type	Mean Sensitivity (dBf)	Standard Deviation (dB)	Sample Size
Comp. Tuner	36.6	3.1	82
Comp. Receiver	37.8	2.3	129
Car Receiver	40.4	2.7	464

Table II: Projected Car Receiver Performance

Proportion of Receivers (percent)	Sensitivity (dBf)	Field Strength + 3 dB (dBμ)
95	44.8	42.8
90	43.5	41.5
85	42.7	40.7
80	42.2	40.2
75	41.7	39.7
70	41.4	39.4
65	41.1	39.1
60	40.8	38.8
55	40.6	38.6
50	40.4	38.4

Figure 1
Automotive Receiver Sensitivity



A field strength of 41½ dBμ at the receiving antenna is necessary for satisfactory stereophonic performance in 90 percent of the receivers. A minimum field strength of 38½ dBμ is projected as being necessary for such performance in 50 percent of the receivers evaluated.

Propagation Principles

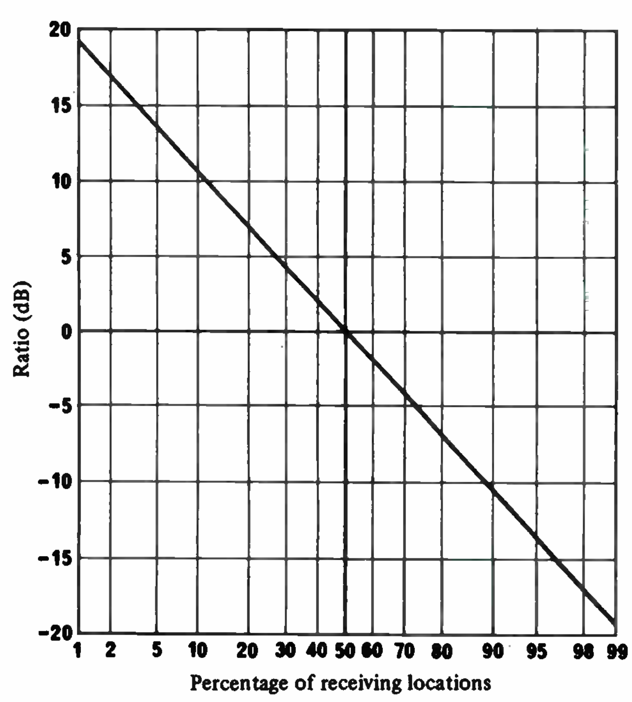
Before considering the details of finding practical limits for antenna height, it is useful to review some fundamental premises and principles of VHF propagation theory.

Terrestrial wave propagation at VHF and higher frequencies is expressed in statistical terms. This is because propagation is influenced by many random phenomena, which impact signal variability with respect to time and location for a given path from transmitter to receiver. An additional variability factor is described for prediction "confidence", which relates to the variability of the data sample from which the propagation theory was developed, for a given set of model parameters.

The two most basic statistical terms used in VHF propagation theory are location and time variability. The FCC's low-band VHF propagation curves, contained in §73.333 and §73.699 of its Rules, are based on these criteria. The service field strengths are defined by §73.333, Figure 1, and §73.699, Figure 9, for 50 percent location and 50 percent time variability. Given a specified distance from the transmitter site and a particular antenna height, the predicted service field strength will be exceeded at 50 percent of the locations and for 50 percent of the time. The location variability is presumed to occur over a short course (several hundred feet), at the specified distance from the transmitter. Interference field strengths are defined by §73.333, Figure 1a, and §73.699, Figure 9a, for 50 percent location and 10 percent time variability. In this latter case, the predicted field strength will be exceeded at least 10 percent of the time at 50 percent of the locations.

VHF propagation relationships are derived from empirical (measured) data. Wide variations in such results are caused by the different environments within which measurements were conducted, the collection and analysis techniques used, and the knowledge of the data collectors. The variation in results for a given set of parameters (e.g., height, distance, time variability, and location variability) may be viewed as the "experience" or "confidence" of the prediction. A field strength result that is supported by 50 percent or more of the data sets for those given parameters is said to be at the midpoint of experience, i.e., it involves 50 percent confidence.

Location variability introduces a complicating factor into the analysis process. The CCIR has defined the relationship between location variability and field strength by the following graph.[4]



Predicted field strengths should be reduced by the ratios shown when higher location variability factors are used. In other words, to achieve a 90 percent signal strength at a specified level, the 50 percent predicted field strength must be 10.5 dB greater. For example, a 50 percent signal strength of approximately 70.5 dBμ is necessary to achieve a 90 percent signal strength of 60 dBμ for the given location and height. For 70 percent location variability, a -4 dB ratio of 90 to 50 percent field strength is shown.

Time variability is a function of path length and antenna height. Within the radio horizon, it has much less of an effect than location variability. The following table gives an idea of the magnitude of time variability, based on the Rice 1990 propagation formulas.[5] For short distances, the time variability range is small. As the horizon is approached, it increases. However, the variability is not the same at every horizon distance. Greater distances involve greater variability, as there are more factors in propagation that might affect the signal strength with time over large distances.

Table III: Time Variability

Antenna Height (feet)	Path Length (miles)	50% - 90% Time Field Strength Difference (dB)
328	15	1.0
328	17½	1.3
328	25½	2.7
492	31½	3.3
492	32½	3.6
982	45	4.7
1968	57	5.2
1968	62½	6.0

The statistical quantities pertinent to VHF field strength prediction are, therefore, experience, location, and time variability. When all of these divergent factors are considered, one might quickly conclude that it is hopeless to make any sense out of field strength predictions.

For the sake of simplicity, especially in coverage prediction for the broadcast services, 50 percent experience variability is usually presumed. In the interest of keeping this study clear and straightforward, that same presumption is used here. This paper will largely concentrate on optimization of parameters to result in satisfactory coverage at the outer boundary of usable service. Therefore, it is appropriate to hold location variability at 50 percent. In considering service closer to the transmitter site or within urbanized areas, use of a higher location variability factor is appropriate. Time variability is the smallest of the foregoing statistical factors. To add some statistical perspective to the analysis set forth herein, without turning the data into a confusing muddle, time variabilities of 90% and 50% were used.

Another significant factor in propagation modeling is the presumed receiving antenna height. The FCC's propagation curves were derived for an assumed receiving antenna elevation of 30 feet, which was typical of residential television installations in the 1950s and 1960s, when all of the data underlying the FCC's propagation curves were gathered. That was a time when FM reception frequently involved attachment of the receiver to the home television antenna. Signal strengths at low receiving antenna elevations will necessarily be less, because of increased diffraction and reflection attenuation caused by clutter in the vicinity of the receiving antenna.

Propagation Modeling

The FCC's FM/TV field strength prediction curves do not facilitate propagation modeling where the receiving antenna elevation is less than 30 feet. A typical receiving antenna elevation is around 6 feet. There is no particularly reliable means of derating field strength based on receiving antenna height available to use with the FCC's prediction curves. The relevant CCIR documents provide little more guidance.[6]

In 1989 and 1990, well-known propagation physicist Philip L. Rice developed a mathematical model of generalized propagation characteristics from 30 to 1000 MHz.[5] This propagation prediction technique will be referred to as the "Rice 1990 formulas" hereafter. Mr. Rice's method permits specification of the receiving antenna height and frequency. It closely matches the FCC's low band VHF propagation curves for most distances at the low VHF band center frequency, 76 MHz.

The Rice 1990 formulas were used to predict field strengths in this study. A receiving elevation of 6 feet was presumed. The center frequency of the 88-108 MHz FM broadcast band, 98 MHz, was specified in applying this model to the situation at hand.

The distance to the radio horizon is given by the familiar equation:

$$d = \sqrt{2\,h_t} \qquad (2)$$

where: d = distance in miles
h_t = antenna height in feet

Received Field Strength Examples

Now that the pertinent theory and receiver performance parameters have been set forth, consider the field strength that is predicted at various distances of interest to FM stations. Field strength was computed using the Rice 1990 formulas, presuming a receiving antenna elevation of 6 feet. The primary distance used for each FM station category is the "class contour distance" specified in §73.211(b)(1) of the FCC's Rules (the 60 dBμ contour radius for the maximum permitted ERP/HAAT combination). This distance represents the limit of station service as the FCC defines it. For Class A and C stations, the radio horizon distance is also shown (noted by an *), because that range is substantially different than the "class contour distance". Use of maximum permitted power and height was presumed. The minimum values shown at the bottom of the table were obtained from Table II.

Table IV: Predicted Field Strengths

Station Class	Distance (miles)	Field Strength 50% Time (dBμ)	Field Strength 90% Time (dBμ)
A (3 kW)	15	54½	53½
	25½*	42	39½
A (6 kW)	17½	54½	53
	25½*	45	42½
B1 / C3	28	49	46
B / C2	32½	51½	48
C1	45	52	47½
C	57	52	46½
	62½*	49	43
(minimum)		38½	41½

Field strengths well exceed the minimum value deemed necessary for satisfactory receiver performance for all station classes and distances except in the case of 3 kW Class A service at the radio horizon. Therefore, it is not surprising that FM stations often have usable service well beyond their FCC-recognized 60 dBμ coverage contours. However, the limiting effect of field strength attenuation at greater distance can be seen for the Class A horizon examples.

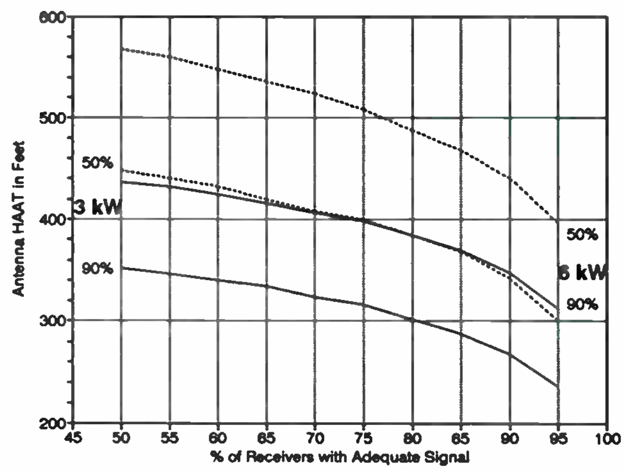
Height Limit Analysis

The distance to the radio horizon is mostly a function of transmitting antenna height. The received signal strength is a function of the latter quantity and the effective radiated power (ERP) of the station. As discussed in the Introduction to this paper, power must be reduced where the antenna height above average terrain (HAAT) exceeds the reference value for the pertinent station class.

The maximum recommended antenna HAATs were determined for each FM station class based on the simple criterion that the signal strength should not fall below the value necessary for the specified level of receiver performance at the radio horizon. This was implemented by a computer program which establishes an initial value of antenna HAAT, finds the maximum permitted ERP at that level, computes the horizon distance, and predicts the received signal strength at that distance based on those HAAT and ERP parameters. The HAAT is adjusted iteratively by the program until the received field strength agrees with the desired level within a tolerance of 0.1 dB. Computations were performed for 50 to 95 percent of the receivers, at time variabilities of 50 and 90 percent. Location variability was held constant at 50 percent. Receiving antenna height was set at 6 feet.

Figure 2 illustrates the transmitting antenna HAAT limit versus receiver percentage relationship for Class A stations. Graphs are shown for both 3 kilowatt and 6 kilowatt equivalent Class A operation. At 3 kilowatts or its equivalent and for 50 percent time variability, acceptable receiver performance results at the radio horizon for the class maximum HAAT value (328') and, for 50 to 90 percent of the receivers, at higher elevations. Above 450 feet, the trend of the relationship shown predicts that reception quality will be adequate in less than 50 percent of the receivers at the radio horizon. Any antenna HAAT increase above that level appears to waste tower cost, absent a compelling reason for increased height, such as the need to overcome terrain obstructions, because signal strength will not be adequate at the horizon distance achieved.

Figure 2
Maximum HAAT - 3 kW & 6 kW Class A



At 90 percent time variability, use of the class maximum antenna HAAT value (328') is predicted to result in adequate service to about 67 percent of the receivers. To adequately serve 90 percent of the receivers at the coverage fringe, a HAAT of about 270 feet appears optimum. For heights below 328 feet, power was automatically adjusted upward in accordance with the recently adopted provisions of §73.213(c)(1) of the FCC's Rules.

The foregoing analysis indicates that, for some 3 kW equivalent Class A situations, there may be advantages to operating at HAATs below 328 feet with ERPs in excess of 3 kW, as now permitted by the FCC's rules. Signal density closer to the transmitter site (within half the horizon distance, 11½ miles for the

90% situation cited above) is increased by the higher ERP permitted by operation at an HAAT below the class maximum. However, this option of reducing height and increasing power is not available to other classes of FM stations at this time, nor is it available to those Class A stations spaced sufficiently to operate with an ERP of 6 kilowatts.

For 6 kW equivalent Class A stations, the 50 percent of time predicted antenna HAAT limits range from 400 to 570 feet and the 90 percent of time predicted HAAT limits range from 310 to 440 feet. Because operation of Class A stations above 6 kilowatts is not permitted, use of HAATs below 328 feet offer no advantages at 6 kW equivalency.

The reason why the Class A height limits are predicted to be comparatively close to the nominal maximum antenna HAAT specified is that the horizon distance for that height is well beyond the 60 dBμ (30 foot receiving height) contour distance of the class. By the time the nominal horizon is reached, the signal is so attenuated that it approaches the range of minimum acceptable receiver performance.

Figure 3
Maximum HAAT - Classes B, B1, C2, C3

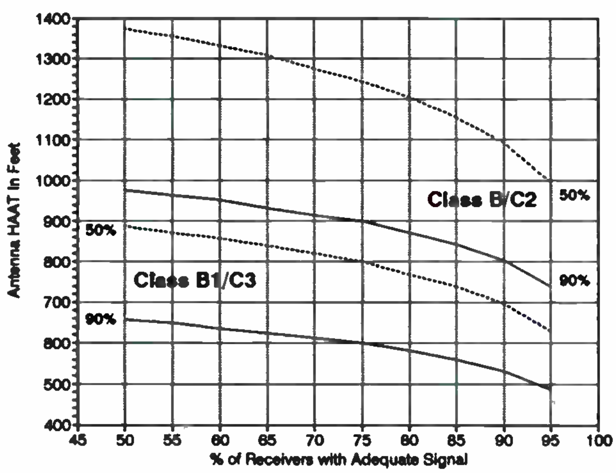
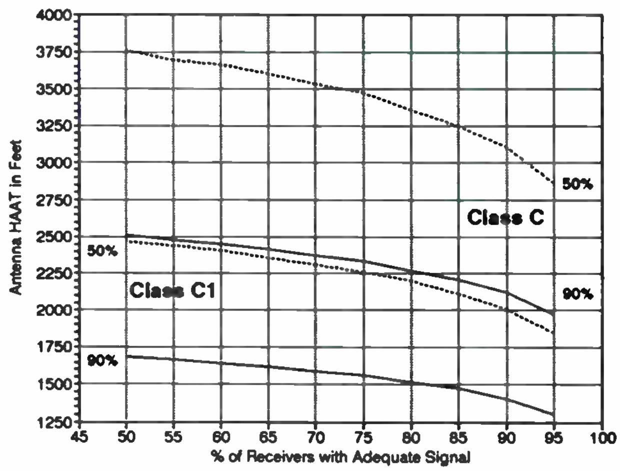


Figure 3 illustrates the predicted antenna HAAT limits for Class B1/C3 and Class B/C2 stations. Here the heights extend well above the nominal class maxima. This is because a strong, 60 dBμ (30 foot receiving height) signal is predicted at or near the horizon for nominal facilities.

Classes B1/C3 station reach is maximized when the antenna HAAT ranges from 500 to 900 feet, with the pertinent limit depending on the acceptable time and receiver statistics. For Classes B/C2, the maximum reach antenna HAAT limits are predicted to range from 750 to 1400 feet.

Figure 4 illustrates the maximum recommended antenna HAAT limits for Class C and C1 stations. The Class C1 height limit range is from 1300 to 2500 feet. As expected, Class C limits are higher, ranging from the nominal maximum for the class (1968 feet) up to nearly twice that height, 3750 feet. Again, the limit to be used is dependent upon the time and receiver statistics desired.

Figure 4
Maximum HAAT - Classes C & C1



Summarizing Predicted HAAT Limit Data

The foregoing data and its variations are interesting. However, further reduction of the data into a simpler form may be more useful to the reader.

Receiver variability ranges were predicted for car receivers. Many other types of receivers in the hands of consumers are expected to offer poorer performance. Others will be better. Without having a more representative quantity of statistical data on the receiver population and the performance of an adequate sample thereof, we are left to do some guesswork in projecting the data set forth herein to the general case. Accordingly, it is suggested that, for the purpose of establishing practical antenna HAAT limits to be used as "rules of thumb", the 90 percent car receiver figures are useful. The vast majority of component-type receivers/tuners will be included in those receiving an adequate signal. The better portable receivers are likely to be included, as well.

The time variability to be used is a function of the nature of the "fringe" area to be served. For lightly populated, rural areas, a 50 percent time variability is not unreasonable. But in an urbanized area, 90 percent is a more useful parametric statistic.

The following table presents the predicted antenna HAAT limits for the various station classes, at 50 and 90 percent time variability, sufficient to achieve an adequate signal in 90 percent of the car receivers.

Table V: HAAT Limit Summary

Station Class	50% of Time HAAT (feet)	50% of Time Horizon (miles)	90% of Time HAAT (feet)	90% of Time Horizon (miles)
A (3 kW)	340	26	270	23
A (6 kW)	440	29½	350	26½
B1 / C3	700	37½	530	32½
B / C2	1100	46½	800	40
C1	2000	63½	1400	53
C	3100	79	2100	65

The limits shown above are consistent with our experience. For example, the Detroit and Washington Class B facilities, with HAATs ranging from 700 to 900 feet, serve their markets well. A 25 mile practical service radius is not atypical of Class A stations with HAATs around 300 feet, in the absence of interference and/or terrain obstruction.

Field Strength Boundary for Practical Service

We know from experience that many stations have usable service beyond their FCC-recognized 60 dBμ contours. Station salesman love those old 34 dBμ contour maps that show the Class A station serving as big an area as that defined by the local Class C station's 60 dBμ map. But we also know that those 34 dBμ maps represent, at best, barely adequate monaural performance that only the most underserved listener will tolerate.

An interesting offshoot of this work is the 30 foot field strength that is predicted at the horizon when acceptable receiver performance is achieved at a 6 foot antenna elevation. This field strength value is believed to be a useful guideline in defining a station's practical service area, in the absence of interference. Field strengths were computed for the height and distance parameters set forth in Table V, presuming the use of the maximum permissible ERP at those heights for each station class.

The resulting data, set forth in Table VI, suggests that a conventionally predicted contour at 48 to 50 dBμ is a good measure of stereophonic service to outlying areas. For service to suburban regions, a conventionally predicted contour at 52 to 54 dBμ appears to be a good measure of practical coverage. In dense urban areas, the location variability factor, described by the CCIR graph cited earlier herein, should be considered and the necessary field strength increased accordingly.

Table VI: Usable 30' Field Strengths

Station Class	Field Strength 50% of Time (dBμ)	Field Strength 90% of Time (dBμ)
A (3 kW)	49½	52
A (6 kW)	49	52½
B1 / C3	49½	52½
B / C2	49	53
C1	49	54
C	48	54½

Of course, both of the foregoing field strength ranges presume that no interference is present. In the so-called "rust belt" and parts of California, "grandfathered" short spacing and resulting interference is more the rule than the exception, so practical service will be limited thereby.

Conclusion

There really is such a thing as "too much antenna HAAT". A practical technique has been described for determining maximum useful FM station antenna elevation values for average terrain conditions. A side benefit is some insight into what conventionally predicted field strength contour describes adequate stereophonic reception in the absence of interference. This method should be applicable to other VHF services if the pertinent receiver performance data is available.

Notes

1. It is hoped that more reliable and consistently measured data can be obtained by the time this paper is presented orally. Interested readers should contact the author, at (703) 591-0803, after mid-April 1992 for an updated copy of this paper.

2. Audio, "34th Annual Equipment Directory"; Volume 75, Number 10; October 1991; pp. 179-188

3. Audio, "16th Annual Car Stereo Directory"; Volume 74, Number 5; May 1990; pp. 110-124

4. International Radio Consultative Committee (CCIR); Recommendation 370-5, "VHF and UHF Propagation Curves for the Frequency Range from 30 MHz to 1000 MHz: Broadcasting Services", Recommendations and Reports of the CCIR, 1986, Volume V: Propagation in Non-Ionized Media, XVIth Plenary Assembly, Dubrovnik, 1986; published at Geneva, Switzerland, 1986

5. P.L. Rice, "Formulas for FCC Broadcast Curves"; Moffet, Larson & Johnson, Inc., Falls Church, VA; presented to the December 1989 meeting of the Association of Federal Communications Consulting Engineers, Washington, D.C.; updated March 1990

6. CCIR, *supra* n.4, ¶2.4

A NEW MULTI-CHANNEL COMMUNITY ANTENNA FOR FM BROADCAST

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Abstract- Until recently, the only available option for multi-channel FM transmitting antennas has been the well known panel antenna arrays. However, panel antenna arrays have inherent shortcomings that make them undesirable in many applications. In this paper a new design is introduced based on a multi-section helix antenna. This design provides a superior RF performance without the mechanical and structural disadvantages of panel arrays. A comparative analysis of this antenna is presented here.

INTRODUCTION

FM broadcasters are facing new challenges in this era of advanced technology, among which are fierce competition and environmental awareness. Advances in receiver sensitivity, selectivity and the introduction of new FCC dockets along with the strict restriction on RF exposures, provide a positive incentive for broadcasters to cooperate in efficient use of environmental assets while minimizing the environmental impact of RF radiation. These considerations will have great impact on the present as well as the future licensing of antenna fields and transmit sites.

There is no doubt that the location of a transmit site is an important element in providing maximum coverage in any market. Rarely, however, is there an ideal transmit site that fulfills all coverage requirements of a broadcast station, but in any market there are usually one or two locations that are decisively better than others. As a result, these locations become crowded antenna sites with complex RF and environmental problems. The type and the extent of problems associated with a site differ from one location to another. Problems usually range from unpredictability of radiation due to the presence of other antennas or towers, to the presence of high RF field at the ground level of the site thus creating a health hazard to the people in or around the site. The majority of the antennas in these sites are old in design and are built using yesterdays technology. This is especially true in the case of FM antennas.

Traditional FM antennas are

omni-directional, circularly polarized, and side mounted on a tower or a pole. Due to the complicated structure of towers and the complexity of electromagnetic coupling between the near field of the antenna array and the tower, there is no theoretical means by which the resulting radiation pattern of the antenna plus the tower could be fully understood. Unfortunately the size of antenna arrays and tower make it difficult and even sometimes impossible to run enough tests and measurement to provide a comprehensive representation of the radiation pattern. Even in those cases where there is such possibilities, the cost of conducting such measurements make the antenna unaffordable for a single station.

Presently the requirements related to the radiation characteristics of FM antennas are limited only to the azimuth pattern of the vertical and horizontal polarizations. Unfortunately these two patterns alone only give little information about the radiation profile of the antenna. Information such as the axial ratio, downward radiation, and even the true radiation pattern of the antenna plus tower cannot be obtained from these two patterns.

Figs.1&2 compare a free space pattern of a typical side-mount antenna vs. the pattern of the same antenna when side mounted on a tower as measured by a rotating dipole. Note that in the case of the side-mounted antenna, the maximum field at each heading is not in vertical or horizontal pole but somewhere in between. In fact, the actual field pattern (the envelope of the maximum field values at each heading) is totally different from the pattern of either polarization. It also indicates, clearly, that the resulting radiation is everything but circularly polarized. The same sort of anomalies exist in the elevation pattern of the side-mounted omni-directional CP antennas. However, the subject is beyond the scope of this paper. The point is that there are thousands of FM antennas in use, including the ones that are crowded in a single site, with actual radiation characteristics that may only remotely resemble what their documentation shows. The situation is even more sever in crowded sites where multiple reflection between towers and antennas totally deform the radiation characteristics of each and every one of the antennas.

Obviously the solution to this problem is consolidation of single station transmitting antennas into fewer, more powerful multi-channel antennas which use the present state of the art in antenna technology. By doing so, FM broadcasters can afford to broadcast from a WELL ENGINEERED antenna with known operating characteristics and at the same time eliminate environmental problems created by using inexpensive but inadequate antennas.

In this paper we present the performance data of the newest and the most advance antenna design for FM broadcast "The

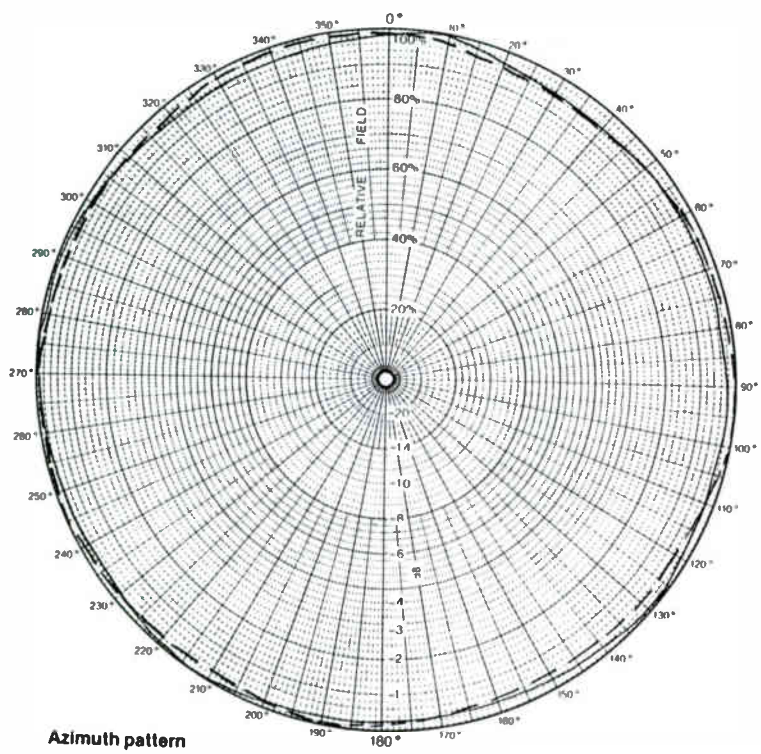


Fig. 1
Typical free space pattern
of a CP side mount antenna.

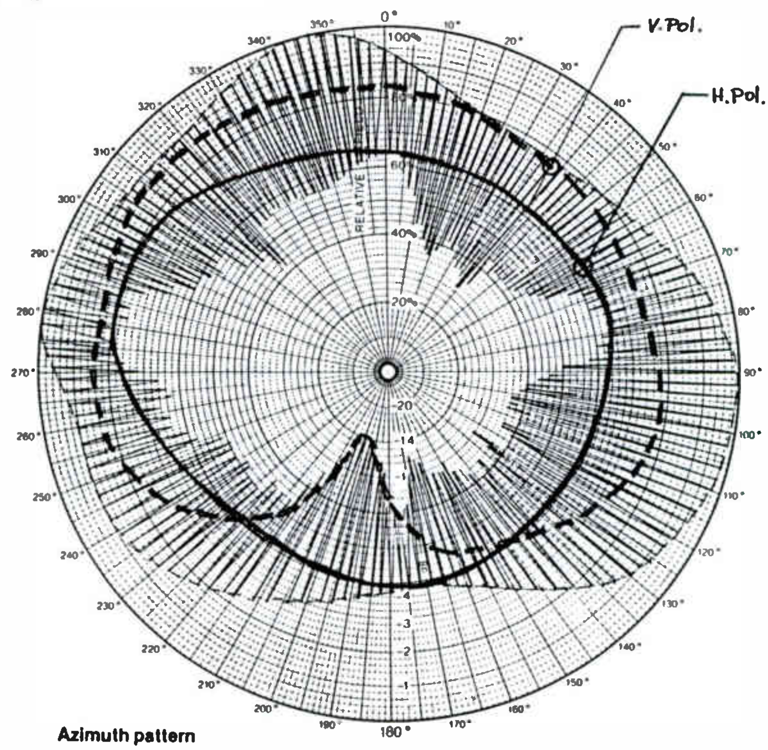


Fig. 2
Typical patterns of
side mounted
moni-directional
FM (CP) antenna.

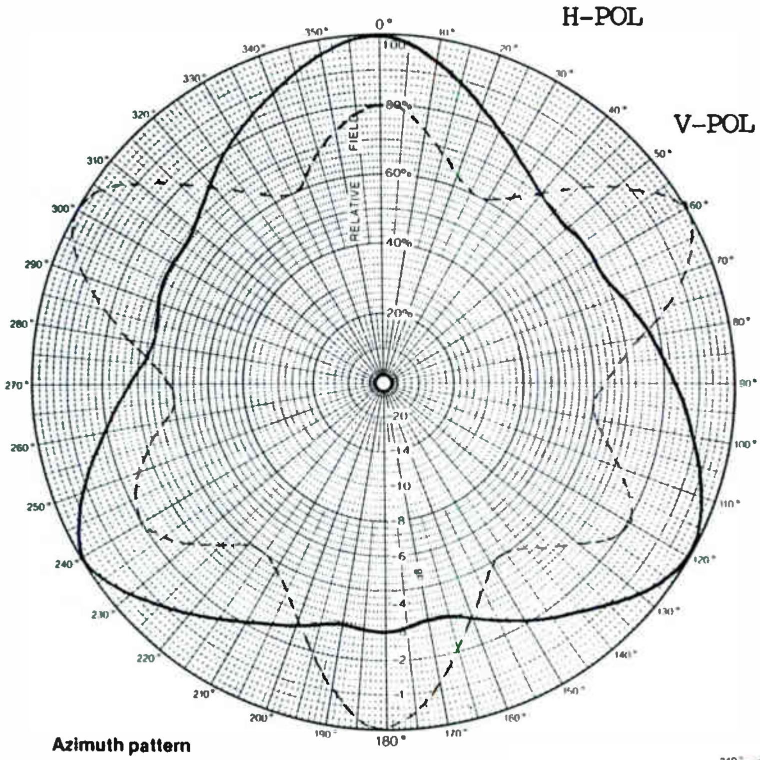


Fig. 3
Typical azimuth pattern of a flat panel antenna (3-around configuration)

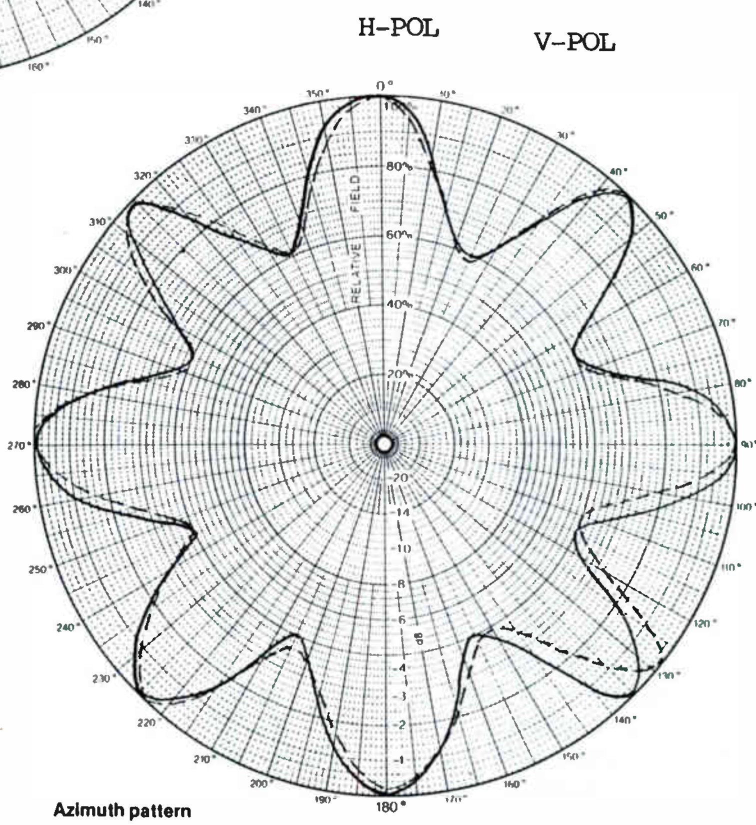


Fig. 4
Typical azimuth pattern of cavity back antenna array (3-around configuration)

multi-channel helix antenna".

ALTERNATIVES IN MULTI-CHANNEL ANTENNAS

The idea of multi-channel antennas is not new, in fact there are a few multi-channel FM antennas that are currently operating in the U.S. and around the world. These antennas are all, with the exception of two, panel type antennas. Some are horizontally polarized only, while others are circularly polarized. Some of these antennas are made of flat panels and some of cavity back radiators. Here we concern ourselves only with circularly polarized antennas.

FLAT PANEL ANTENNAS

A typical radiation pattern of this type of antenna is shown in Fig.3. It is well known that the H-POL and V-POL patterns in this type of antenna do not trace well. Consequently, the radiation does not have the same quality of circular polarization in all directions. Furthermore, unlike the rather circular H-POL pattern, the V-POL pattern has unacceptable circularity in most applications.

It should also be pointed out that typically the elements in flat panel array have narrow band pattern characteristics. In other words, the azimuth pattern changes with frequency in major ways. Some of the stations, as a result, end up with azimuth patterns that are inferior compared to others and broadcasters know the ramification of such situations very well!

CAVITY BACKED ANTENNAS

To eliminate some of the problems with flat panel antennas, antenna designers have come up with cavity back antennas. The major advantage of a cavity back over the flat panel antenna is the pattern control. Unlike the flat panel, the H-POL and V-POL patterns of cavity back antennas trace quite well. Furthermore, by appropriate design of the cavity, the circularity of the pattern is optimized. Nevertheless, circularity of better than +/- 2 dB is hardly ever achieved at midband and is deteriorated by 1 or 2 dB at the band edges. The uniformity of circular polarization is much better than the flat panel and remains good throughout the design band.

Fig.4 shows a typical pattern of a cavity back antenna. Major disadvantages of both types of antennas are the complex and cumbersome feed system, high windload and dead weight, and marginal circularity. Complexity of the feed system increases the probability of system failure dramatically, thus making the system less reliable. The high windload leads to the increase of the tower costs.

HELIX (SPIRAL) ANTENNAS

Here we present a new multi-channel FM antenna that has all of the advantages of a multi-

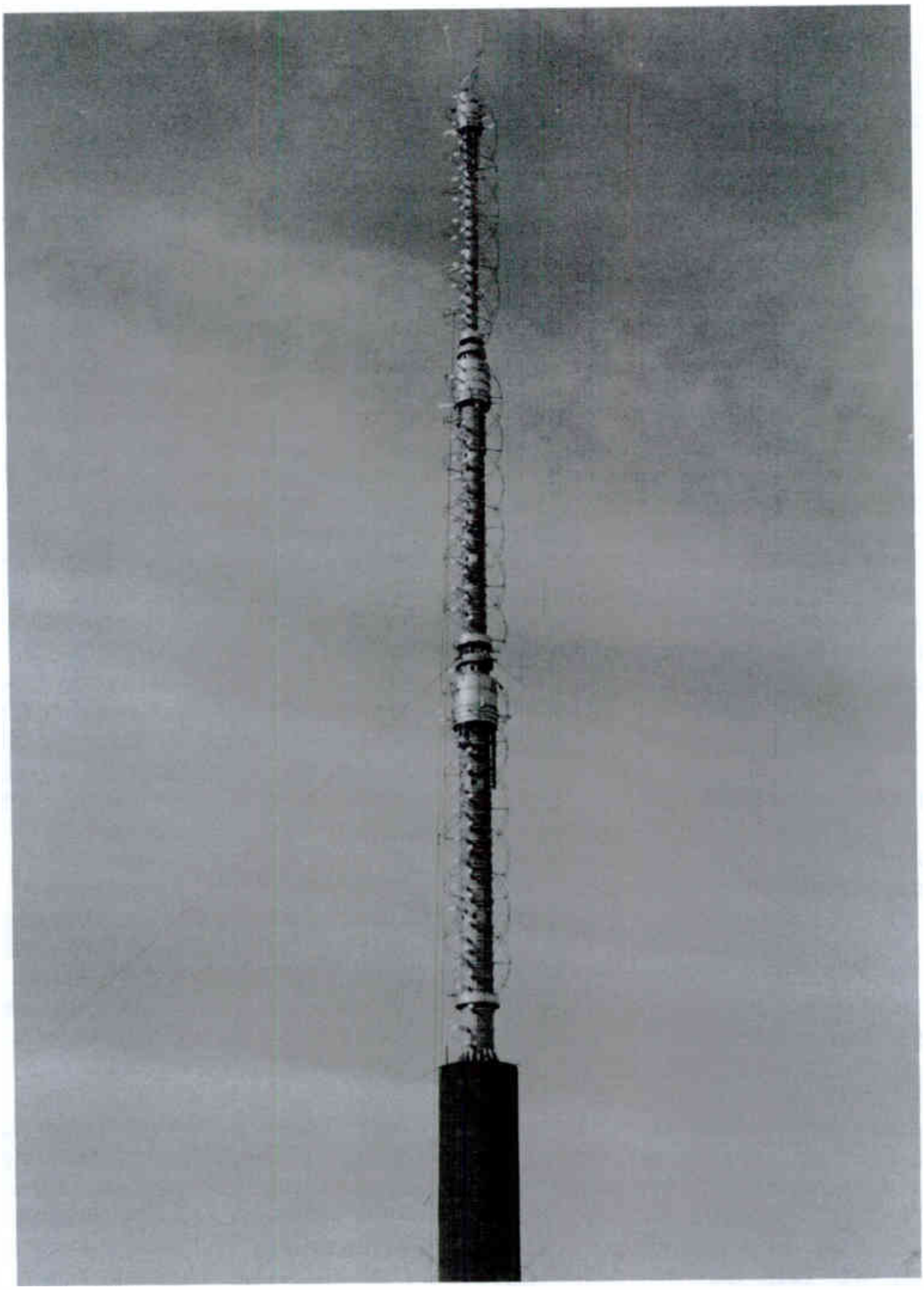


Fig. 5

FM Sprial (Helix) antenna at JAMPRO ANTENNAS, INC. test range.

cast community antenna and none of the above disadvantages. As we will see, the RF performance of this antenna surpasses those of the above antennas while its simple feed system makes it more reliable. Furthermore, the low windload and low dead weight of this antenna further simplifies the structural design of the tower.

The design of this antenna is based on our patented Multi-filar helix antenna. This type of antenna has been used for TV applications, successfully, for more than a decade. The present multi-channel FM antenna is an enhanced version of this antenna that is specifically developed for multi-channel broadcast at the FM band.

STRUCTURAL CONFIGURATION

The new FM antenna consists of three sections (or bays) of so called Tri-filar helix antenna.

Fig.5 Each section is approximately 30 feet long. Each bay consists of a central pole and three groups of helices that are wound, in the left hand screw sense, around the pole. The grouping of the helices is intended to improve the radiation efficiency of the antenna and to reduce the mismatch at the feed point of the helices. Pole diameters are, typically, 14" to 30", depending on the height of the section. The helices are made out of stainless steel tubular wire which also house the heating elements that perform the de-icing function.

ELECTRICAL/RF CONFIGURATION

Each group of helices is fed at the top of the bay via a three way power divider and flexible cables or rigid transmission lines. The entire feed system is housed inside the pole. Fig.6 compares the feed system of this antenna with that of an equivalent panel antenna.

Electrically, each section of the antenna has a nominal gain of 1.2 with respect to dipole. The design, however, is flexible enough to accommodate higher gain values per section. The active impedance throughout the antenna, from the feed point of each helix to the input of each section, is nominal 50 ohm over the entire band. This ensures the utmost stability of both elevation and azimuth pattern over the operational band of the antenna.

The first multi-channel antenna of this type was developed and built by Jampro Antennas, Inc. to replace at least six FM stations broadcasting from Healy Height Towers in Portland, Oregon.

Here we present the measured data taken during the development phase as well as the operational phase of the antenna.

RF PERFORMANCE

This particular antenna was designed to accommodate six stations in the frequency range of 89 MHz to 101 MHz. Total average power was specified to be 160 KW. Aside from excellent RF performance expected from

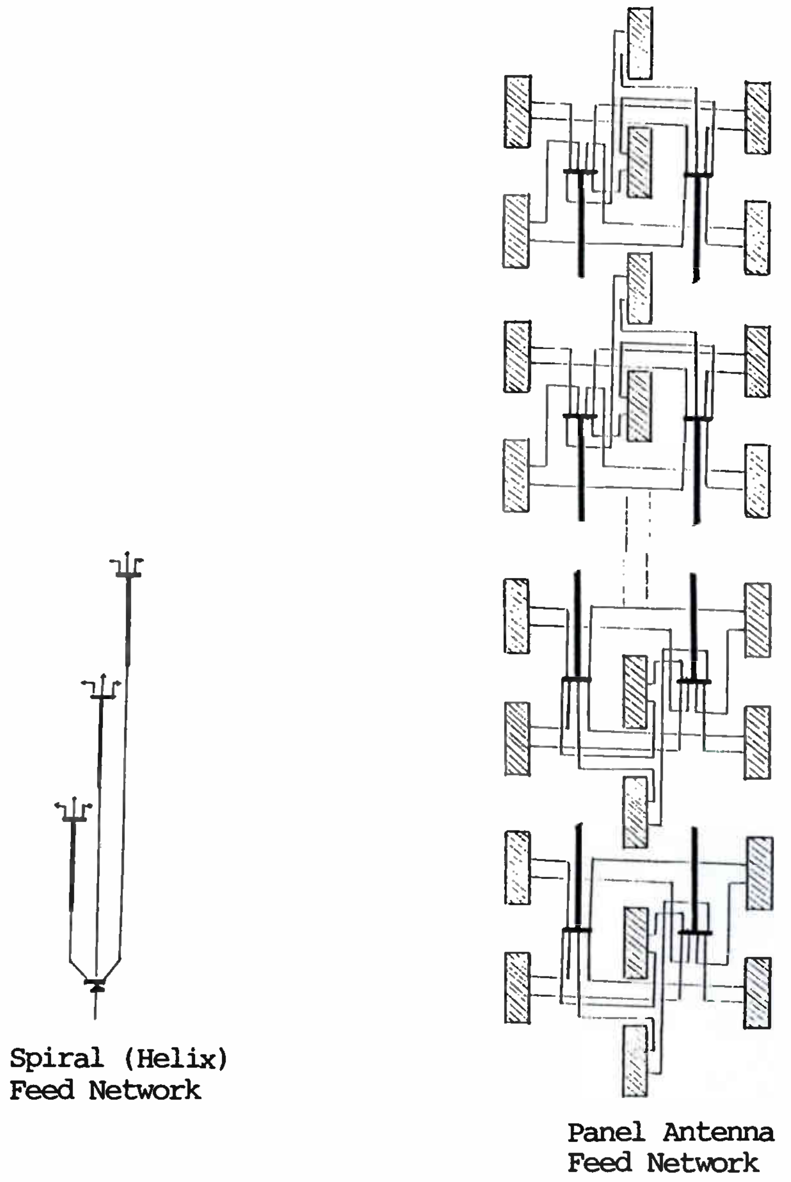
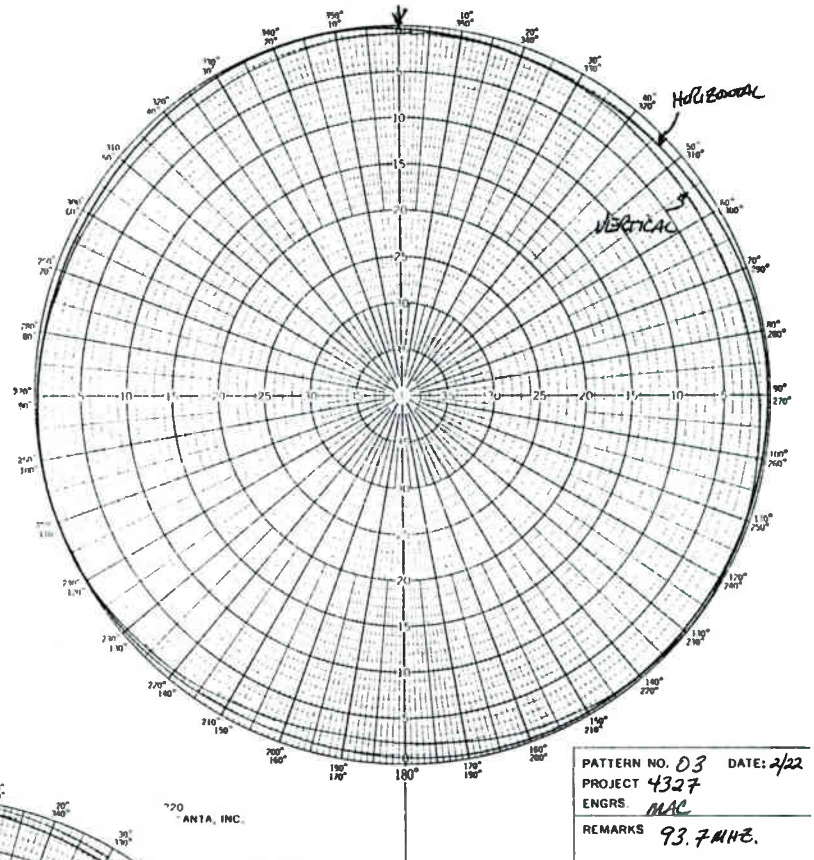


Fig. 6 Feed networks of Spiral (Helix) Antenna and an equivalent panel antenna array.



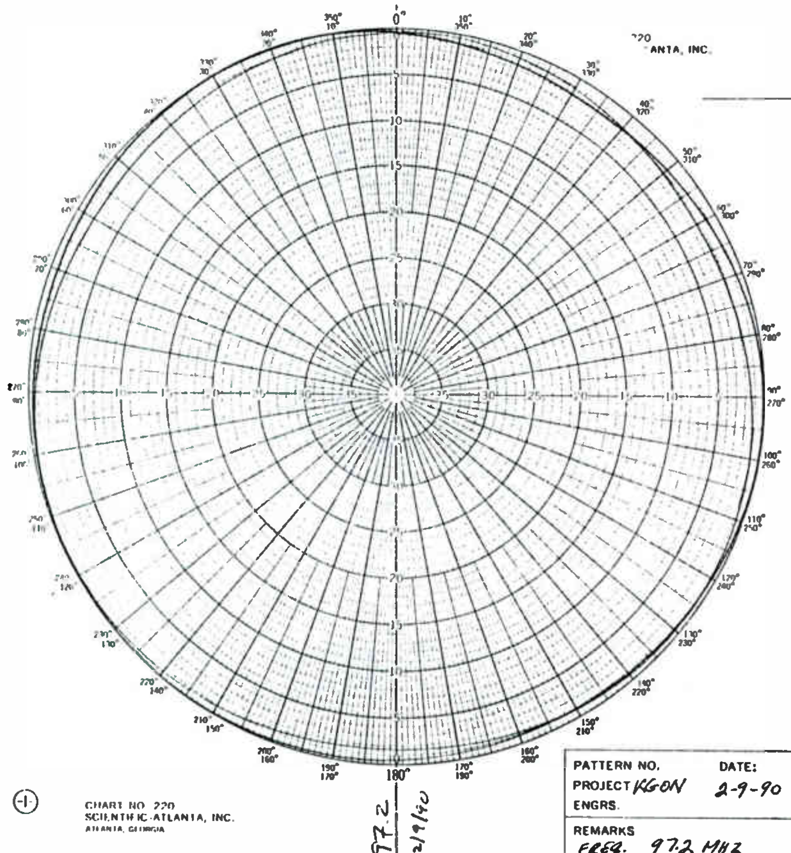


Fig. 7
Measured Azimuth Pattern
of Helix antenna at
two frequencies

the antenna, there were other tough requirements imposed by the local residence on the levels of radiation and the antennas aesthetics. For example, the RF level near and around the tower were supposed to stay below 50 Microwatts per centimeter-squared, i.e. one twentieth of the ANSI proposed safe level of 1 MilliWatt per centimeter-squared. As we will see later, this antenna surpassed all expectations in pattern circularity, VSWR across the band as well as the 50 microwatt requirement on the RF radiation in and around the tower area.

Azimuth pattern of the antenna at the design frequencies are shown in Fig.7 As you note, the tracking between the two polarizations is excellent. Furthermore, the circularity of the pattern is better than +/- .9 dB at all frequencies.

Fig.8 shows the return loss of each section as well as the complete antenna. Markers indicate the stations frequencies.

Due to the concerns about the RF exposure in the Healy Height area, the city of Portland requested Richard Tell & Associates to conduct a rigorous study of the RF levels in the area, before and after the replacement of the antennas by the new antenna. According to this reporty; with the new antenna in place "the highest spatially averaged value at any of the 171 measurement cells was 36.02 microwatt per centimeter squared!" compared to the 488 microwatt level measured before the switch. Fig. 9. gives a more accurate picture of the situation before and after the installation of the new antenna. Note that after switching to the new antenna, at only 5 percent of the points the RF level exceeds 20 microwatts and at no point it exceeds 100 microwatts per centimeter-squared.

CONCLUSION

The data that is presented here clearly indicates the superiority of the new Multi-channel FM broadcast antenna. The aerodynamic structure of this antenna makes it a great candidate for top mount applications. The flat response along with the excellent pattern characteristics in both polarization surpasses the performance of any panel antenna. The simple feed system and low level of radiation in and around the tower ensures reliability and safety of operation for long periods of time. Cleaning of an antenna site takes a state-of-the-art antenna and cooperation among the tenants. If the initiative and cooperation is there, this antenna will provide the best solution.

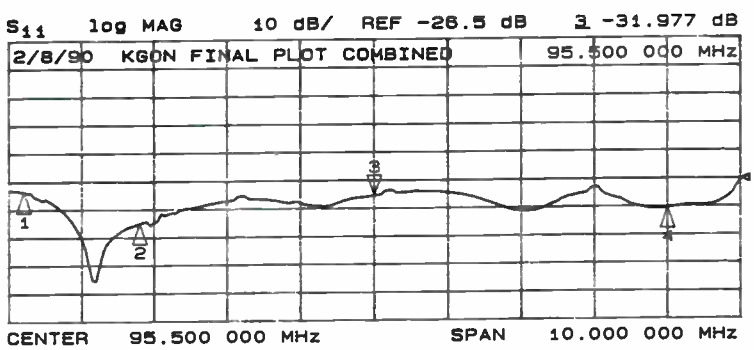


Fig. 8 Measured Return Loss of Helix (Spiral FM) Antenna

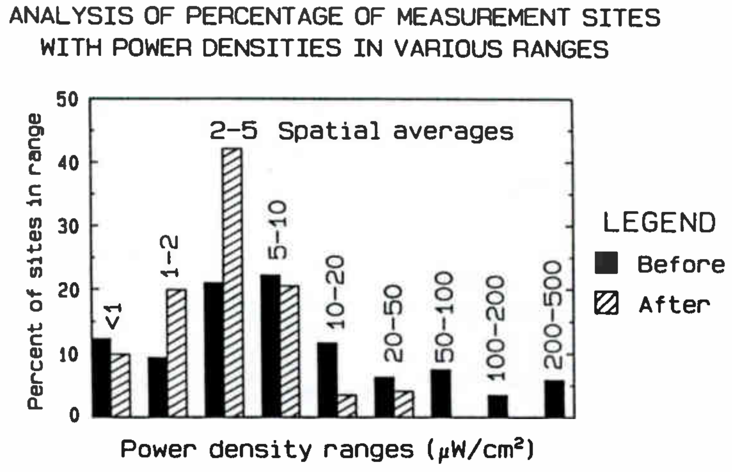


Fig. 9

A NEW HIGH POWERED SOLID STATE TRANSMITTER

Hilmer I. Swanson
Harris Corporation, Broadcast Division
Quincy, Illinois

Abstract - Vacuum tubes have dominated high powered broadcasting for many years. Solid state devices can now replace vacuum tubes at any power level in the medium wave band. This paper describes the world's first 300kw all solid state MW transmitter, the DX300, part of a family of high powered medium wave transmitters. These transmitters employ a patented digital modulator.

High Power Mw Transmitters

What is high power MW AM? For this paper it is 100kW and above in the frequency range of 530 kHz to 1.7 mHz.

First, a brief history of solid state transmitters. When I first started designing transmitters about 40 years ago, there were only tubes to work with, diodes, triodes, tetrodes, and pentodes. In the mid 50's silicon diodes started to replace vacuum and mercury vapor rectifiers. In the 60's transistors started replacing the small tubes in transmitters. In 1975 Harris came out with the MW1, a 1kW all solid state transmitter. This transmitter employed bipolar transistors. The transistors employed were ones originally designed for 100 kHz switching power supplies. These transistors were operating at their maximum frequency capability in the medium wave band. A faster switching type transistor was desirable. In about 1980 International Rectifier developed a family of large MOSFET transistors. We tested several of these new devices as soon as they were available and found that they were much better at producing RF in the medium wave band than switch mode bipolar transistors. However, we found that the early MOSFET transistors were not very rugged. They were efficient,but fragile. In the last few years almost all MOSFET manufacturers now make a very rugged device. Broadcasters now have an almost ideal RF generator. Because the new MOSFETS are efficient, rugged, and low cost it is now possible and practical to make MW transmitters of any power level, including up to several megawatts.

Harris has an architecture that permits building transmitters of any power level. In a system that has "soft" failures the more devices there are, the smaller the change in performance if one device fails. If there are only 2 devices in a circuit and one device fails, there may be a 50% change. If there are a thousand devices in a circuit and one device fails, there will only be a 0.1% change in performance.

Harris manufactures three 100kW transmitters models. The DX100 is a standard 100kW transmitter. The DX100FA is a 100kW transmitter that is built to be frequency agile. Lastly, the DXD100 is a dual transmitter. It consists of two DX50 transmitters that are combined to produce 100kW output. We have many customers that like the extra redundancy of combined transmitters. In some countries almost all transmitters are combined as a matter of policy. In a combined transmitter one transmitter can fail completely and the station will still be on the air.

Modern solid state transmitters are very reliable. However, there can still be failures that might put the station off the air. An example of a failure that could put the station off the air is a failure in the output filter of one of the transmitters. Figure 1 is a schematic of a 2 way combiner as employed in the DXD100 transmitter. In this combiner the reject load is switched so that it can also be a test load for one of the transmitters. If one transmitter should fail, the power out of the system will drop 6 db for about 1 second and then the failed transmitter is automatically switched to the reject load for testing. The other transmitter is switched into the antenna. Switching occurs in about 100 milliseconds so that there is virtually no program interruption. The reject load is forced air cooled with the fans operating only when required.

For powers above 100kW Harris employs 100kW power blocks and an "N" way combiner. A 100kW power block is similar to a DX100 transmitter except it has one less cabinet due to the removal of the high voltage power supply and part of the output network. The following is a brief explanation of the operation of "N" way combiners. The simplest "N" way combiner is a "2" way. Figure 2A is a simplified schematic of a "2" way combiner. Let us start by separating the two circuits. With S1 and S2 open, each circuit is adjusted to provide a 50 ohm to 100 ohm transformation. For transforming this impedance we could use 90 degree transmission lines with a Z0 of $\sqrt{50x100}$ = 70.7 ohms or we could employ lumped components with a reactance of 70.7 ohms. If E1=E2 and both circuits are identical, E3 will equal E4. Therefore, switch S1 and S2 can be closed with no affect on the circuit. The two 100 ohm resistors can be replaced by a single 50 ohm resistor. Now if E2 is turned off there will be current in R1 (see figure 2B). There will also be current brought back to E2 through the combiner network. The current through the combiner is shifted 90 degrees twice and therefore arrives at E2 180 degrees out phase from the current through R1. There two currents are equal and out of phase at E2 and therefore produce no signal at E2. These currents cause one half of the power out of the 100kW transmitter to be dissipated in the reject load (R1 & R2) and one half in RL. Typically we find about 40 db of isolation between the input ports. With 40 db of isolation, each transmitter behaves as if it was operating all by itself. Almost any number of Pi networks can be combined together to produce almost any power transmitter. For the protection of the combiner and antenna system, the combiner has several VSWR detectors and ultraviolet arc detectors.

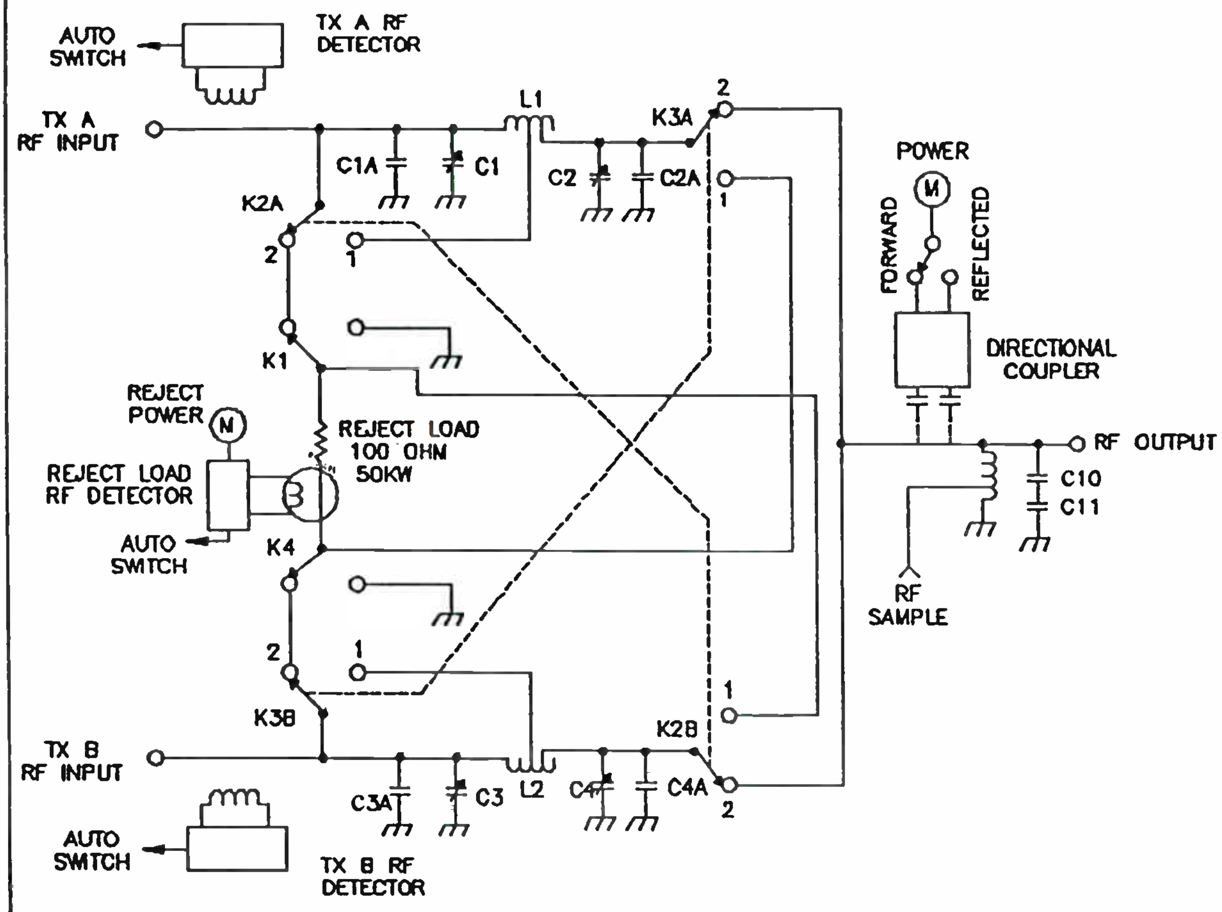
The current line of Harris TV transmitters employs combiners up to 17 way. Harris currently has MW combiner designs for up to 10 way. Figure 3 is a schematic of a 5 way combiner for a DX500 transmitter. Figure 4 is a photo of a DXD100 transmitter. Figure 5 is a photo of a DX300 transmitter.

It has been my experience in the past with tube transmitters, that as the power went up the audio performance went down. However, with the current solid state transmitters, we have found that the audio performance of our large transmitters is as good as if not better than our low power transmitters. The following is data that was measured on a DX300 transmitter: From figure 6 it can be seen that there is essentially zero tilt and overshoot on the transmitter. Figure 7 illustrates the good linearity and high positive peak capability of this transmitter. It can be seen from figure 8 and 9 that the distortion is less than 0.3% from 25 hz to 12,000 hz at 100kW and 300kW output.

The overall AC to RF efficiency of the DX300 is about 84%. The power required by this transmitter is low; however, it can be reduced an additional amount by the use of dynamic carrier control. Dynamic carrier control adjusts the carrier level according to the modulation. This techniquie is used internationally in certain countries, but not in the U.S.

In conclusion, we see that solid state transmitters are the future of High Power Medium Wave Broadcasting. The high power DX designs offer:

A. Solid state reliability (no tubes to wear out).
B. Soft failure modes by utilizing many devices instead of a few.
C. Highest quality audio performance.
D. Highest efficiency (most cost effective use of electricity).



	K1 POS	K2 POS	K3 POS	K4 POS
A AIR (B LOAD)	OFF	1	2	ON
A + B	OFF	2	2	OFF
B AIR (A LOAD)	ON	2	1	OFF

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Figure 1. Schematic, Simplified RF, DXD 100

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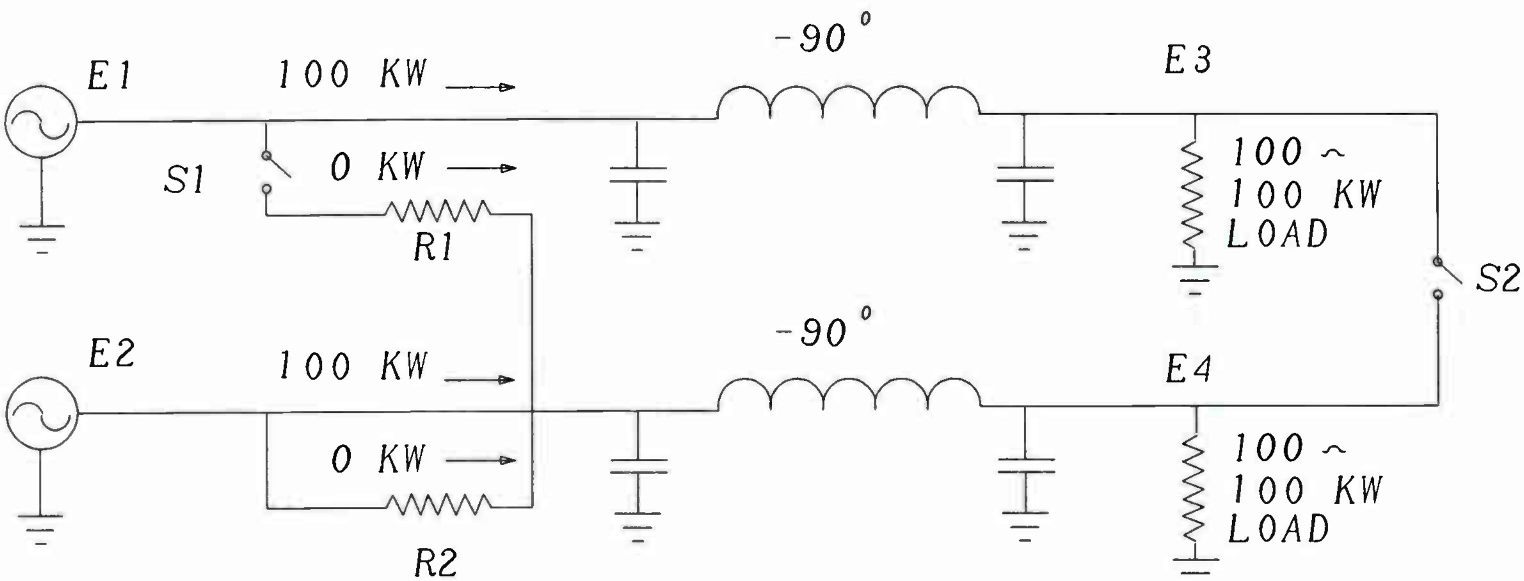

TITLE FIGURE 2A	SHEET OF	HARRIS-ALLIED SALES 3200 WISMANN LANE QUINCY, ILLINOIS 62301 UNITED STATES OF AMERICA
	DRAWN BY L. ARCHER	
	DATE 01/20/92	
DRAWING NUMBER Q000045C	THIS DOCUMENT CONTAINS PROPRIETARY DATA OF HARRIS CORPORATION. NO DISCLOSURE, REPRODUCTION OR USE OF ANY PART THEREOF MAY BE MADE EXCEPT BY WRITTEN PERMISSION.	

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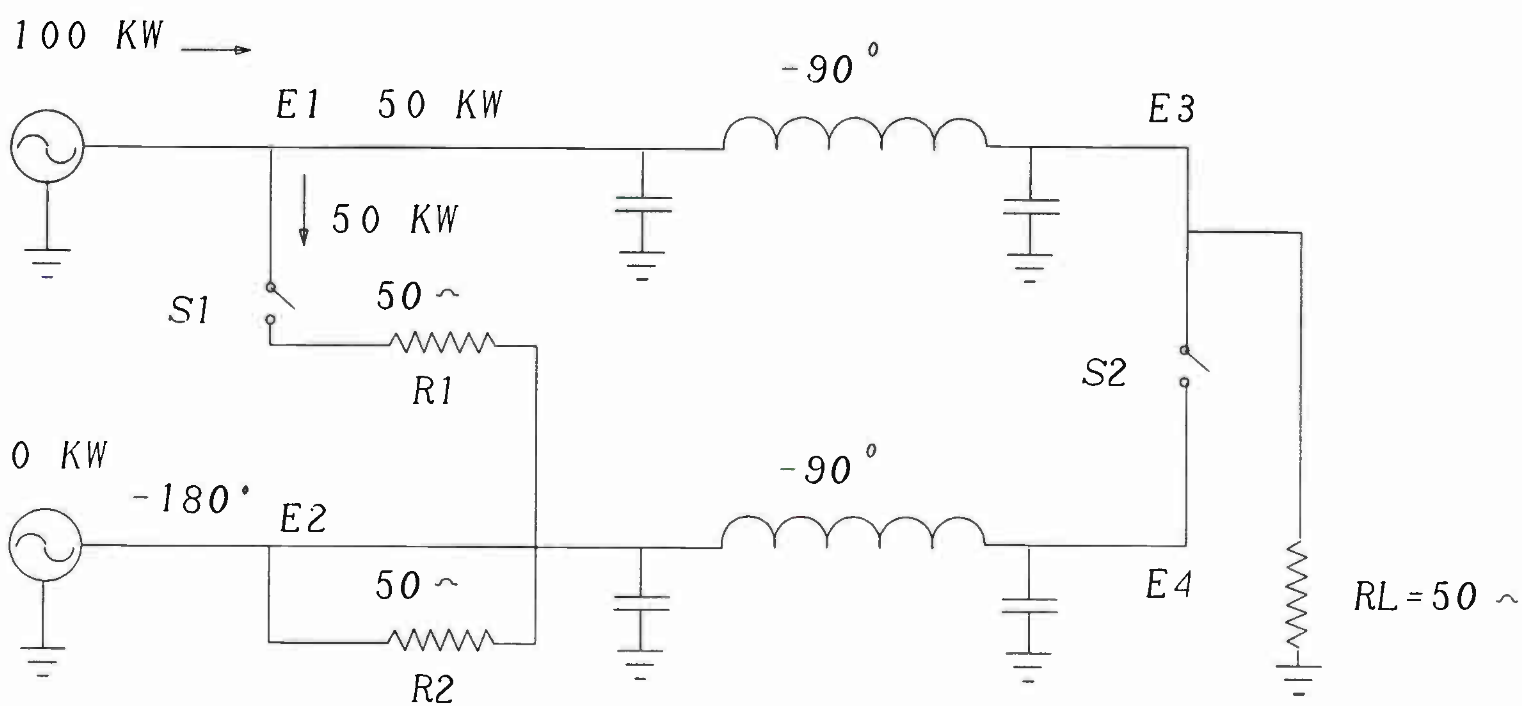

TITLE FIGURE 2B	SHEET OF	HARRIS-ALLIED SALES 3200 WISMANN LANE QUINCY, ILLINOIS 62301 UNITED STATES OF AMERICA
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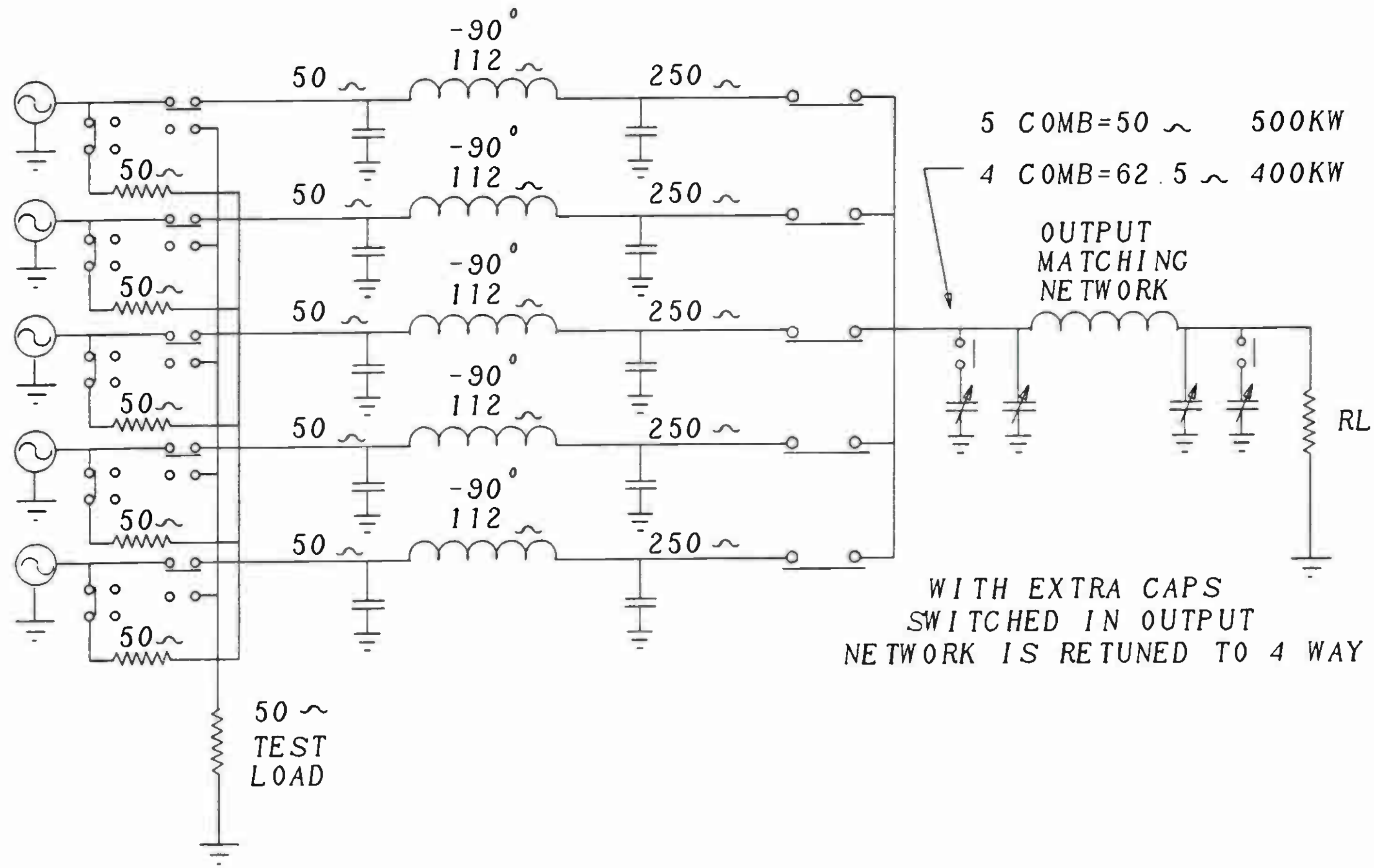

TITLE FIGURE 3	SHEET OF DRAWN BY L. ARCHER DATE 10/12/90	HARRIS-ALLIED SALES 3200 WISMANN LANE QUINCY, ILLINOIS 62301 UNITED STATES OF AMERICA
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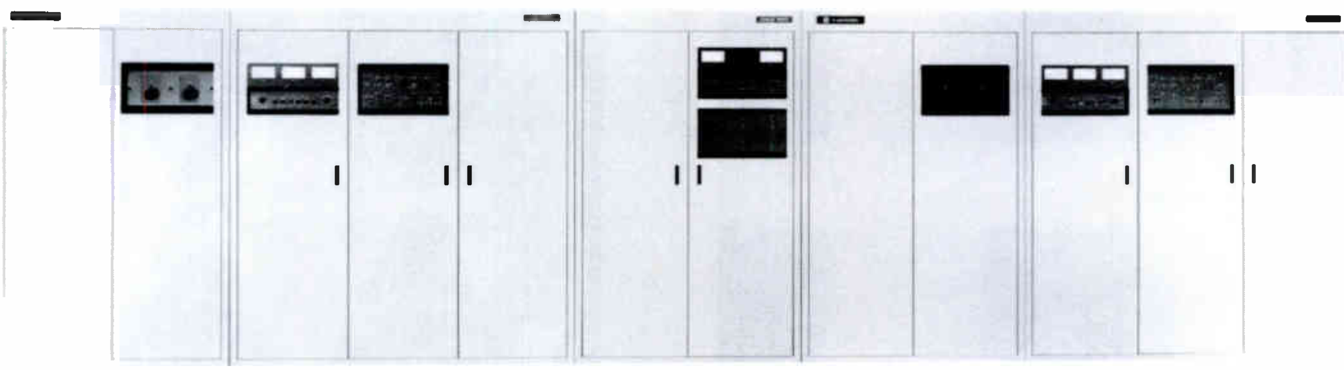

Figure 4. DXD100 Transmitter

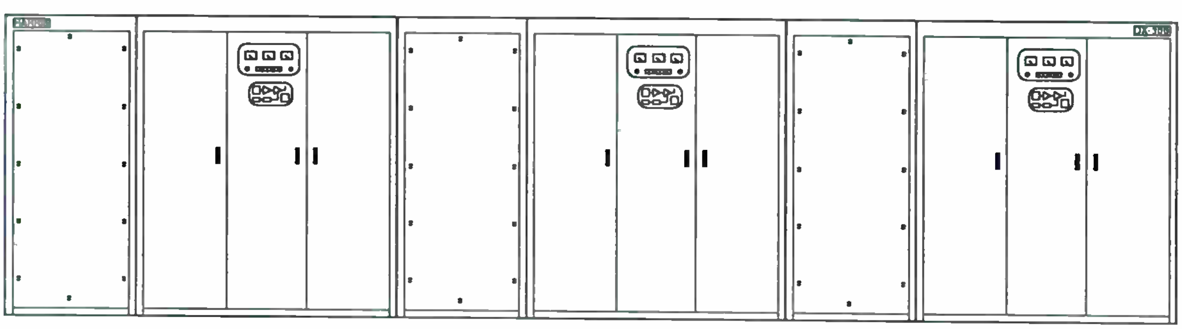

Figure 5. DX300

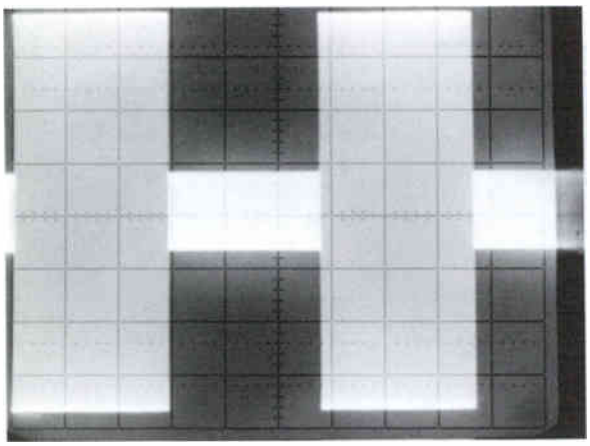

Figure 6
DX 300 40 Hz Square Wave

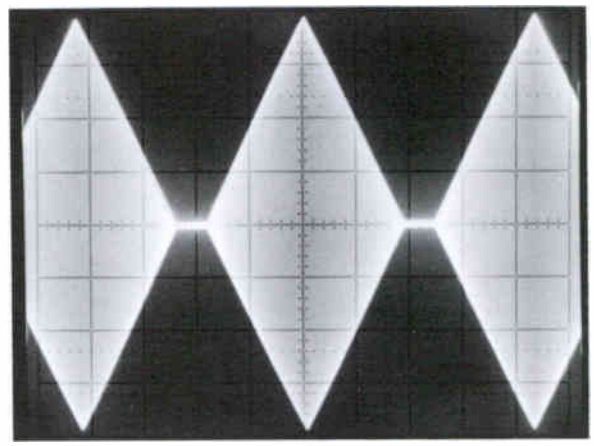

Figure 7
DX300 125% Positive Peak

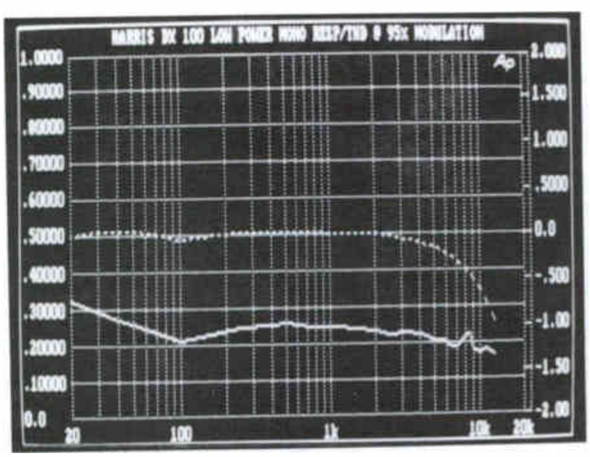

Figure 8
DX300 Distortion & Frequency
Response 100 Kw Output
(Upper curve frequency response)

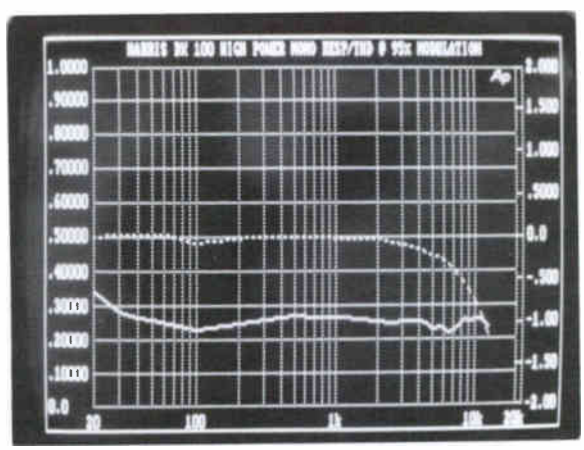

Figure 9
DX300 Distortion & Frequency
Response 300 Kw Output
(Lower curve distortion)

SBE DAY AT NAB '92

The Society of Broadcast Engineers (SBE) developed a full day program as part of the NAB Broadcast Engineering Conference. The program consisted of three technical sessions covering radio and television engineering in the morning and technical regulatory issues in the afternoon.

SBE is an international organization representing over 6,000 technical professionals in 33 countries. Based in Indiana, the Society has been serving the broadcast and related industries for 27 years.

SBE

RADIO: COPING WITH NEW TECHNOLOGY

Tuesday, April 14, 1992

Moderator:
John Battison, Consulting Engineer, Canton, Ohio

***RADIO IN THE 1990S: CHALLENGES AND OPPORTUNITIES**
Brad Dick
Broadcast Engineering magazine
Overland Park, Kansas

***DIGITAL CABLE AUDIO: WHEN AND WHERE**
Don Lockett
National Public Radio
Washington, District of Columbia

***THE EXPANDING ROLE OF DSP IN AUDIO TECHNOLOGY**
Michael Collins
Motorola
Austin, Texas

IMPROVING THE EFFICIENCY AND RELIABILITY OF AM BROADCAST TRANSMITTERS THROUGH CLASS-E POWER
David W. Cripe
Broadcast Electronics, Inc.
Quincy, Illinois

THE DEPENDENCE OF AM STEREO PERFORMANCE ON TRANSMITTER LOAD PHASE
Jerry Westberg
Westberg Consulting
Quincy, Illinois

*Paper not available at the time of publication.

IMPROVING THE EFFICIENCY AND RELIABILITY OF AM BROADCAST TRANSMITTERS THROUGH CLASS-E POWER

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Broadcast Electronics, Inc.
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Abstract: Class-E is a novel means of power amplification which overcomes many of the shortcomings of conventional Class-D power amplifiers. As compared to Class-D power amplifiers, Class-E circuits offer improved efficiency, reliability and performance into non-unity VSWR's. These features give Class-E power amplification the distinct advantage in broadcasting applications.

This paper covers the theory of operation of Class-E power amplifiers, and compares the operation and performance of a typical Class-D power amplifier with a working Class-E power amplifier circuit.

INTRODUCTION

The development of power MOSFET technology has made possible the creation of solid-state AM transmitters. It has been all too often apparent to the designers and operators of these transmitters that in comparison to the venerable vacuum tube, a transistor is much more sensitive to and less forgiving of thermal and electrical stresses. Consequently, in the interest of system reliability, the designer of a solid-state transmitter must reduce wherever possible the stresses on the power amplifier transistors.

This paper compares the operation of Class-D and Class-E power amplifiers and predicts how the performance and reliability of each is affected by different non-ideal conditions likely to occur in an AM broadcast transmitter.

MOSFET Reliability

Admittedly, the task of the designer of solid-state AM broadcast transmitters has been made much easier in recent years by the development of MOSFET devices which are much more rugged than those available a decade ago. This has not, however altered the fact that there are still operating conditions in which MOSFETs fail and operators of solid state broadcast transmitters continue to experience device failures.

The most readily identifiable enemy of the power MOSFET is heat. All other variables being equal, an elevation in MOSFET junction operating temperature of 26 C° will result in a ten-fold decrease of device lifespan.[1] This will be manifested as an increase in "random" failures otherwise unrelated to any identifiable stress. Furthermore, the "on" resistance of the MOSFET increases with junction temperature, causing an increase in MOSFET conduction losses and a decrease in power amplifier efficiency, further aggravating the over-temperature situation.

The following electrical stress factors have been identified as causing device failure:

a) Gate insulation breakdown, which occurs as the result of excessive voltage on the gate. The oxide gate insulation will be punctured, and instantaneous failure of the device will occur. With proper MOSFET driver design, this is a relatively uncommon occurrence.

b) Drain-source overvoltage, which causes avalanching within the device. This could be the result of either high reflected power or excessive levels of peak audio modulation within the transmitter. While in the earlier-generation MOSFETs, this would result in localized hot spots and rapid device destruction, in the new, rugged devices this has been largely eliminated as an "instantaneous" failure mode. Avalanche breakdown will however increase the junction temperature of the device, leading to an increase in the "random" failures as noted above.

c) Turn-off dV/dt stress. Intrinsic to the MOSFET device is a parasitic bipolar transistor, whose base-collector junction actually comprises the body diode of the MOSFET. It is possible, through the application of a rapid positive-going transient to the drain of the device, to induce current flow in this bipolar transistor. Current flow in the intrinsic bipolar will tend to localize in a single point of the device, destroying it. The MOSFET is particularly susceptible to this type of damage when a high dV/dt (rate-of-change of voltage per time) is applied to the drain while the body diode is undergoing reverse recovery.[2] Additionally, elevated junction temperatures increase the gain of the MOSFET's intrinsic bipolar transistor, further increasing its susceptibility to damage. The likelihood of the MOSFET encountering potentially damaging turn-off dV/dt is determined by the topology of the circuit in which it is applied, and the load into which it operates.

The Class-D Power Amplifier

The Class-D circuit, which currently finds near-universal application in solid-state AM broadcast transmitters, consists of an 'H' bridge of switching transistors (Figure 1) each driven at a nominal 50% duty in such a manner that at any time the diagonal opposite transistors conduct. Consequently, the voltage waveform impressed across the output is a square wave at the carrier frequency. Connected to the output nodes of the power amplifier is a load network possessing a finite, resistive impedance at the carrier frequency, and much higher impedances at the harmonics of the carrier frequency. Since the harmonics are rejected, the load current is sinusoidal, and the MOSFET drain-current waveforms resemble a half-wave rectified sine wave. Assuming that the switch is ideal, i. e. possessing no parasitic inductances or capacitances, the efficiency of the Class-D power amplifier can approach 100%. MOSFET's are, however not ideal switches, so there will exist in any MOSFET power amplifier conduction losses equal to the product of square of the RMS drain current of the MOSFET and its 'on' resistance. Additionally, the MOSFET possesses a substantial output capacitance C_{oss}, which in the Class-D power amplifier is alternately charged to B+ potential and discharged again through the MOSFET during each RF cycle. This results in a power dissipation in each MOSFET;

$$\text{Switching Loss} = (B+)^2 \cdot C_{oss} \cdot F \qquad (1)$$

where
B+ = power amplifier supply voltage
C_{oss} = parasitic MOSFET output capacitance, and
F = carrier frequency.

For example, consider a Class-D power amplifier comprised of IRF360 MOSFETs, which at 150 volts V_{ds} possess a nominal C_{oss} of 210 pF.[3] When operating at 1.6 MHz with a B+ voltage of 300 volts, the resultant switching loss per MOSFET will equal 31 watts. This loss usually equals or exceeds that attributable to conduction loss, and due to its frequency-dependence is the biggest limiting factor in the higher-frequency application of Class-D switching technology.

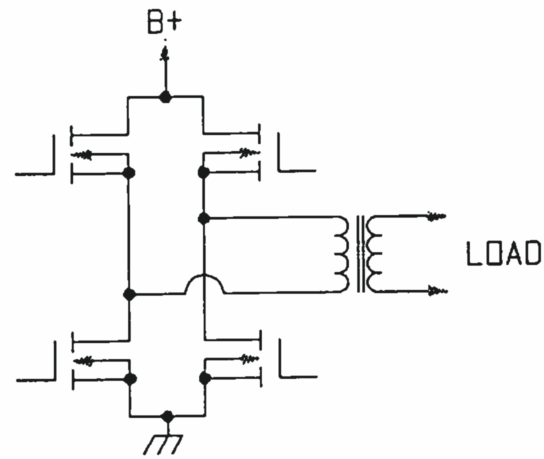

Figure 1. Class-D Power Amplifier

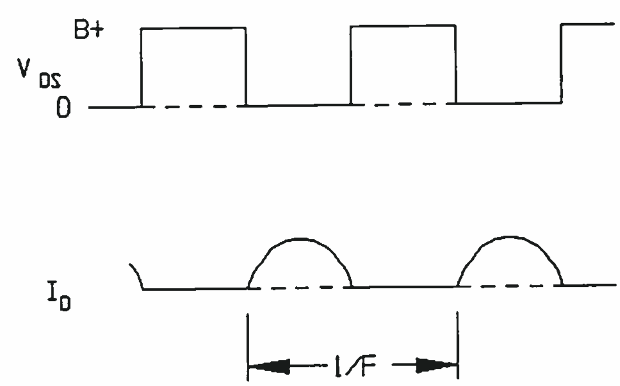

Figure 2. Class-D Voltage and Current Waveforms

In the ideal Class-D power amplifier, the drain current waveform resembles that of a half-wave rectified sine wave. However, if the transmitter utilizing Class-D power amplifiers operates into a load with non-unity VSWR, it is possible to shift the phase of the MOSFET drain current with respect to that of the square wave of drain-source voltage in such a way that the body diode

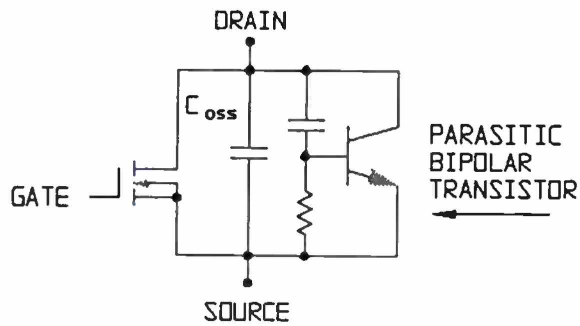

Figure 3. MOSFET Equivalent Circuit Showing Parasitic Components

of the MOSFET is induced to conduct. If the power amplifier is presented a load containing a capacitive reactive component, the MOSFET will then be turned off during the reverse recovery of its body diode, which is one of the known means of destroying the device. Some of the newer "rugged" MOSFETs specify the maximum safe limit for the reverse-recovery dV/dt which can be withstood. For example, the International Rectifier IRF360 is rated to withstand 4 V/ns reverse-recovery dV/dt.[3] A Class-D power amplifier will typically possess switching times roughly one-tenth that of the MOSFET's conduction period during each cycle. A Class-D power amplifier operating at 1.6 MHz with a B+ voltage of 300 volts could then have a switching slew rate of 9.6 V/ns, which if occurring during IRF360 MOSFET reverse recovery, could cause device failure.

It is noteworthy that the reverse recovery period of a typical MOSFET body diode is in the neighborhood of 500 ns.[3] This corresponds quite closely to the conduction period of the MOSFET when operated in the AM broadcast band. In a Class-D power amplifier operating in conditions of a capacitive load, during the period of the diode reverse recovery, the recovered charge flows from the B+ power supply, through the other MOSFET in the half-bridge, which conducts during this interval. Based on the specifications for the IRF360 MOSFET, the recovered charge following 25 amps of body diode forward current is typically 7 μC.[3] When operating a Class-D power amplifier from 300 volts B+ at 1.6 MHz, this would result in over 3300 watts of additional MOSFET dissipation and the near-immediate destruction of the device! This diode-recovery loss is roughly equal to

$$10^{-6} \cdot F \cdot (VSWR - 1) \cdot (\text{Operating Power-per-Device}) \quad (2)$$

Consequently, to obtain reliable operation, the manufacturers of broadcast transmitters utilizing Class-D power amplifiers must take great pains to ensure that the MOSFETs do not operate in any of these conditions which are known to cause device failure or significant increase in device dissipation. One technique used to accomplish this end is to strictly limit the VSWR conditions in which the transmitter is allowed to operate. During reflected power transients, such as those occurring during near-by lightning hits, or during high-frequency modulation into narrow-band antennas, the transmitter will mute itself, to the annoyance of listeners. One means which has been used to prevent MOSFET body diode conduction, and its related problems, has been to parallel each MOSFET by a Schottky diode, which by virtue of its lower junction voltage drop, will conduct the reverse current instead. Since high-voltage Schottky diodes are of limited availability, the MOSFETs used in such a power amplifier circuit must be limited to 100-volt devices. While operation of a Class-D power amplifier from a low-voltage supply will also reduce the $(B+)^2 \cdot C_{oss} \cdot F$ losses, the added junction capacitance of the Schottky diode parallelling the MOSFET's C_{oss} tends to offset this. Furthermore, Schottky diodes themselves possess a failure mode associated with high dV/dt, which for Schottky diodes typical of this application, must be limited to less than 1 volt-per nanosecond.[4] Consequently, the Schottky diodes intended to protect the MOSFETs may themselves be the least-reliable component in the Class-D power amplifier.

The Class-E Power Amplifier

In order to avoid the shortcomings inherent in the Class-D power amplifier, a new approach is required. This has been realized in the form of the Class-E power amplifier, invented by Alan and Nathan Sokal.[5] A variation of their basic patent is used in the Broadcast Electronics AM-1, one-kilowatt AM broadcast transmitter. Depicted in Figure 4, the circuit comprises two MOSFETs driven push-pull at a nominal 50% duty. The circuit load is coupled to the drains of the MOSFETs via an output transformer, the primary of which is center-tapped. The transformer center-tap connects to the power supply voltage B+ through an RF choke. Each MOSFET is shunted by an external capacitance into which the MOSFET parasitic capacitance, C_{OSS}, is absorbed. The circuit load appearing at the secondary of the output transformer presents a high impedance to the harmonics of the carrier frequency, and at the fundamental contains an inductive reactive component, resonating with the capacitors to produce the characteristic 'Class-E' waveforms. The equations required to calculate the proper component values to achieve these waveforms has been treated extensively in the literature.[5,6] Software is also available to greatly ease the design process.[7]

In operation, the Class-E power amplifier has features of both Class-D and Class-C circuits. As in a Class-D power amplifier, the Class-E circuit utilizes MOSFETs as 50% duty switches. The load circuit of the Class-E circuit comprises a resonant tank, as in a Class-C circuit, and the Class-E MOSFETs are switched "on" as the load voltage on the MOSFET's drain approaches zero. However, unlike the Class-C circuit, the Class-E tank circuit is relatively low-Q, tuned in such a manner that immediately prior to the MOSFET entering its conduction period, the MOSFET drain voltage waveform is nominally zero, and the time-derivative of drain voltage (dV/dt) at this instant is also nominally zero. As a result, the C_{OSS} energy of the MOSFETs are zero immediately prior to the turn-on of the devices, *so there are no* $(B+)^2 \cdot C_{OSS} \cdot F$ *losses as there are in a Class-D power amplifier.* Due to the elimination of these frequency-proportional switching losses, the Class-E power amplifier is unique as it may operate as efficiently at 1600 KHz as it does at 530 KHz. A highly-efficient Class-E power amplifier utilizing conventional power MOSFETs has been demonstrated at frequencies as high as 6 MHz.[6]

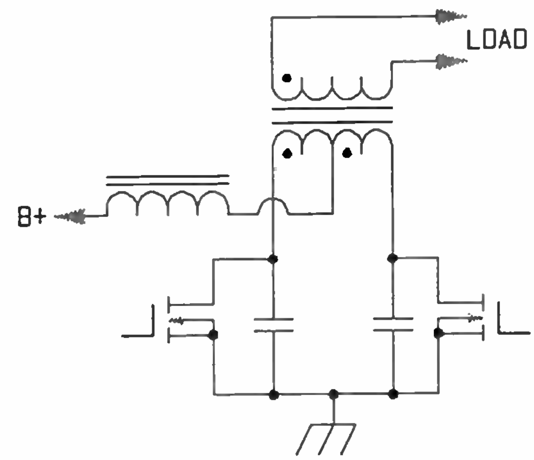

Figure 4. Class-E Power Amplifier

COMPARISON OF CLASS-D AND CLASS-E POWER AMPLIFIER PERFORMANCE

The efficiency of a power amplifier is a determining factor both of the electrical expense of operating a transmitter, and of system reliability. With MOSFET junction temperature a strong determinant in device failure rate, the efficiency and junction temperature rise of hypothetical Class-D and Class-E power amplifier circuits operating under similar conditions may be calculated and compared as criteria for judging relative merit of each design. Consider Class-D and Class E power amplifier circuits utilizing IRF360s generating 300 watts of RF per device, with 300 volts peak drain voltage, operating at 1.6 MHz:

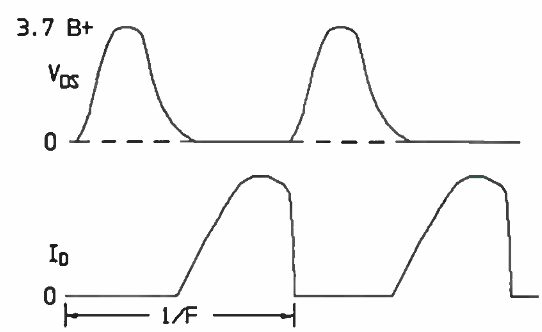

Figure 5. Class-E Voltage and Current Waveforms

Since the DC power consumption of each power amplifier is equal to the product of B+ voltage and average drain current, the Class-D power amplifier generating 300 watts per device, or 600 watts-per-half bridge, operating from 300 volts B+, consequently possesses 2 amps average drain current. The current waveform of a Class-D power amplifier has an RMS-to-average ratio of $\pi/2$, therefore, in this example, the Class-D power amplifier circuit has an RMS drain current of 3.14 amps. The conduction losses of each MOSFET will equal the product of the square of the RMS drain current and the "on" resistance (R_{ds}) of the MOSFET. An IRF360 has an R_{ds} of 0.2 Ω, so a 3.14 amp RMS drain current will result in 1.97 watts of conduction loss.[3] The $(B+)^2 \cdot C_{oss} \cdot F$ switching loss for these conditions has already been calculated to be 31 watts. The total loss in the Class-D power amplifier is therefore 33 watts, which at 300 watts-per device yields an efficiency of 89%. If the MOSFET junction-to ambient thermal resistance for the circuit is 2 C°/W (a value not atypical for most solid-state AM broadcast transmitters), the junction temperature of this Class-D circuit will be 66 C° above ambient.

In the case of the Class-E power amplifier, the peak drain voltage is approximately 3.7 times the B+ voltage. Consequently, to operate with 300 volts peak drain voltage as does the Class-D power amplifier in this example, the Class-E power amplifier circuit in this example must operate with a B+ of 81 volts. Operating at 300 watts-per-MOSFET, the average drain current per MOSFET is then 3.7 amps. The drain current waveforms of the Class-E power amplifier have an RMS-to average ratio of approximately 1.5, so the RMS drain current per MOSFET for this circuit is 5.6 amps. The conduction loss of 5.6 amps RMS through the 0.2 Ω R_{ds} of the IRF360 in the Class-E circuit equals 6 watts. On the basis of MOSFET conduction losses alone, the Class-D power amplifier is superior to the Class-E circuit. However, unlike the Class-D power amplifier, the Class-E power amplifier possesses no $(B+)^2 \cdot C_{oss} \cdot F$ losses. Therefore, the Class-E circuit in this example, possessing only 6 watts loss while producing 300 watts of RF power is therefore 98% efficient! Assuming, as in the Class-D example, a MOSFET junction-to-ambient thermal impedance of 2 C°/W, the junction temperature rise above ambient of the Class-E MOSFET will only be 12 C°, 54 C° cooler than that of the MOSFETs in the Class-D circuit of this example. Solely on the basis of this decrease in MOSFET junction temperature, compared to those of the Class-D circuit, the devices in the Class-E circuit of this example will experience an approximate one-hundred-twenty-fold improvement in reliability.[3]

VSWR Performance

It has been shown that the Class-E power amplifier offers improved efficiency and reliability over a Class-D circuit under ideal conditions of the comparison above. However, from a cursory examination of the circuit and its waveforms it is possible to attain the misconception that the Class-E circuit, being a "tuned" power amplifier would be more succeptable than a Class-D circuit to damage when operated into non-unity VSWR conditions. An examination of the behavior of the Class-E circuit under these non-ideal loads can dispel this perception:

When comparing the drain current waveforms of Class-D and Class-E power amplifiers, the most striking difference present within the Class-E waveforms is that the value of drain current in the Class-E power amplifier is non-zero immediately prior to the end of the MOSFET conduction interval. This contrasts with the drain-current waveform of the Class-D power amplifier which is zero at the beginning and end of the MOSFET conduction interval. In any Class-D power amplifier where the MOSFETs are unprotected by shunt Schottky diodes, any capacitive VSWR condition will cause a negative phase shift of the drain current waveform, resulting in MOSFET body diode current prior to MOSFET turn-off. This will lead to a further increase in MOSFET dissipation from the reverse recovery current of this diode, and can lead to turn-off dV/dt failures of the MOSFET. The Class-E power amplifier, on the other hand, would require a capacitive VSWR at carrier frequency of approximately 20:1 before such an occurrence of body diode current. Even then, a comparison of the respective circuit waveforms reveals that the turn-off dV/dt of the Class-E MOSFET drain voltage, as well as the likelihood of the resultant failure, is much lower than that occurring within a comparable Class-D power amplifier. The dV/dt of a Class-E circuit operating at 300 volts peak drain voltage at 1.6 MHz is only 3 V/ns, compared with 9.6 V/ns for a comparable Class-D circuit.

The Class-E power amplifier does exhibit an increase in losses in the presence of a capacitive VSWR. In these conditions, the drain voltage waveform does not return to zero before the MOSFET is turned on. As a result, the stored energy in the capacitor parallelling the MOSFET is dissipated in the MOSFET. Regardless of operating voltage or frequency, this loss is approximately equal to

$$.05 \cdot (VSWR - 1)^2 \cdot \text{Operating Power-Per-Device} \qquad (3)$$

A Class-E power amplifier delivering 600 watts (300 watts per device) into a 1.5:1 capacitive VSWR will experience an additional 3 watts dissipation per device. When delivering 600 watts into a 2:1 VSWR, a Class-E power amplifier will dissipate an additional 15 watts per device. It is not until this Class-E power amplifier is operated into a 2.5:1 capacitive VSWR that the additional MOSFET losses exceed the 31 watts of $(B+)^2 \cdot C_{oss} \cdot F$ losses per MOSFET previously calculated to occur within the 300 volt, 1.6 MHz Class-D power amplifier *operating at unity VSWR.* Comparing equation (2), that of body diode reverse recovery loss as a function of VSWR in a Class-D power amplifier, with equation (3), that of non-zero drain voltage switching loss as a function of VSWR in a Class-E power amplifier, it is not until an 30:1 capacitive VSWR is reached in both circuits that the VSWR-related losses of a Class-E power amplifier exceed those of the Class-D.

To allow reliable transmitter operation into any kind of "real world" VSWRs, a Class-D power amplifier must therefore be designed with a much higher derating factor than for a comparable Class-E power amplifier to withstand the increased stresses it would encounter.

Optimum MOSFET Voltage Rating

Since the $(B+)^2 \cdot C_{oss} \cdot F$ switching losses of the Class-D power amplifier are proportional to the power amplifier operating voltage squared, the efficiency, and consequently, the reliability of the Class-D circuit can be improved by operating it from lower B+ voltages. Additionally, Class-D power amplifiers which have been designed to avoid the problems associated with MOSFET body diode conduction incorporate Schottky diodes in parallel with each MOSFET. These MOSFETs must then be 100 volt rated parts to correspond with the maximum voltage rating of available Schottky diodes. The Class-E power amplifier, on the other hand, has much less susceptibility to body-diode conduction problems and no $(B+)^2 \cdot C_{oss} \cdot F$ losses, so there is no similar incentive to utilize lower-voltage MOSFETs. When only conduction losses are considered, what then is the optimum voltage rating of MOSFET devices for maximum power-per-device?

When considering conduction losses as the sole limiting factor for the MOSFET power handling capability as is the case for the Class-E power amplifier, there exists a figure of merit for silicon utilization, in terms of maximum power-output capacity per MOSFET die area as a function of device drain voltage rating. This Silicon Utilization factor is inversely proportional to the ratio of MOSFET conduction loss to output power. It is given by the following equation:

$$SU = \frac{V_{ds} \cdot \Theta_{jc}^{1/2}}{R_{ds}^{1/2}} \qquad (4)$$

where
SU = silicon utilization factor
V_{ds} = peak drain voltage for part
R_{ds} = 'on' resistance for part, Ω
Θ_{jc} = thermal resistance, junction-to case, C°/W

The Silicon Utilization factors for selected MOSFETs are listed in Table 1. From this it can be seen that the power-per-device of a given die area is maximum for devices in the 500 to 600 volt range. The values here are nearly twice that of 100 volt devices. Consequently, the conduction losses of a power amplifier utilizing 100 volt MOSFETs can be nearly halved by utilizing 500 volt devices. Consequently, unlike the Class-D power amplifier, the Class-E circuit can most reliably utilize the higher-voltage-rated MOSFETs and exploit the higher operating efficiencies inherent in their use.

Combining of Power Amplifier Outputs

The waveforms present at the output of the Class-E power amplifier are roughly sinusoidal in nature. Compared to the 'square-wave' voltage waveforms generated in a Class-D power amplifier, those of the Class-E circuit have their high frequency harmonics attenuated by roughly 6 dB-per-octave. This means that the Class-E power amplifier is less likely to excite any high frequency resonances which may be present in the power amplifier output combining transformer. Furthermore, the task of harmonic filtering a Class-E power amplifier is eased, as well.

Failure Performance

When MOSFETs do fail, the failure mode is typically a short circuit from drain to source, or from drain to gate, which by virtue of the MOSFET gate drive circuitry will also present a low-impedance path from drain to source. In a Class-D power amplifier, after the failure of one MOSFET, the MOSFET occupying the other leg of the half-bridge will then see a low impedance path between B+ and its return during its conduction period. Consequently, its drain current will only be limited by its "on" resistance, and the source impedance of the B+ supply. This will result in the rapid over-heating of this other MOSFET, and its destruction also. As a result, MOSFET failures in a Class-D power amplifier tend to

Table 1. Silicon Utilization Factor for Selected Devices

Part Number	R_{ds}	Θ_{jc}	V_{ds}	Silicon Utilization Factor
IRF150 [8]	0.055	0.83	100	388
IRF250	0.085	0.83	200	624
IRF350	0.3	0.83	400	665
IRF450	0.4	0.83	500	720
IXTM10N60A	0.55	0.83	600	737
IXTM6N80A	1.4	0.83	800	615
IXTM5N100A	2	0.83	1000	644

occur in pairs, doubling the necessary repair effort. On the other hand, a MOSFET failure in a Class-E power amplifier causes no similar stress in the opposite device, allowing protection circuitry time to act before damage to other components occurs. Failures in a Class-E power amplifier, when they occur, therefore tend to be isolated to a single device.

SUMMARY

For applications within the AM broadcast band, when compared to a Class-D power amplifier circuit, a Class-E power amplifier of similar power output per number of MOSFETs will exhibit improved efficiency due to the elimination of $(B+)^2 \cdot C_{oss} \cdot F$ switching losses. The Class-E circuit also shows a less rapid decrease in efficiency when operated into non-unity VSWRs, and is virtually immune to turn-off dV/dt failures. These factors combine to make the Class-E power amplifier a more reliable and efficient choice for application in AM broadcast transmitters.

Acknowledgements

The author wishes to extend thanks to Nat Sokal of Design Automation for assistance in understanding the operation of Class-E circuits, and to Chris Renard and Sandy Blickhan of Broadcast Electronics for assistance in preparing the illustrations used in this paper.

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[3] International Rectifier. HEXFET Designer's Manual HDB-4. International Rectifer Corporation, 1987: G-136, G-137

[4] Motorola. Rectifiers and Zener Diodes Data, Rev. 3. Motorola, Inc. 1987: 3-84 to 3-157.

[5] Sokal, Nathan O. and Alan D. High Efficiency Tuned Power Amplifier. United States Patent 3 919 656, Nov. 11, 1975.

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THE DEPENDENCE OF AM STEREO PERFORMANCE ON TRANSMITTER LOAD PHASE

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ABSTRACT

It has been shown that in regard to stereo performance, AM transmitters perform well into a load which is 50 ohms over a 20 kHz span about the carrier frequency. When the load varies as a function of frequency, the transmitter performance is degraded. It was demonstrated that by orienting a non-ideal load with the use of a phase shifter (line stretcher) that the stereo performance would approach that of an ideal 50 ohm load.

This paper gives results of stereo performance done at 590 kHz where a non-ideal load was rotated by the use of a line stretcher.

INTRODUCTION

It is well known that the addition of a line stretcher located between the transmitter and the load can improve the monaural performance of the transmitter. The phase shift required out of this network, usually a tee network, is dependent on the transmitter used. That is, one transmitter will require a different phase shift than a transmitter of a different design. The ideal load is a function of the power amplifiers of the transmitter and the output network. In general, the load presented to the power amplifiers of the transmitter should be symmetrical at the sideband frequencies. For example the load presented to the power amplifiers at plus and minus 10 kHz should have the same resistive component and equal and opposite reactive components.

This has not been proven for stereo performance. It has been said that if a transmitter performs well in monaural mode that it will perform well in stereo. It is the purpose of this investigation to determine if good monaural performance will result in good stereo performance and to determine the optimum load impedance orientation for a Broadcast Electronics, Inc. (BEI) model AM-1 stereo transmitter.

PROCEDURE

The transmitter used for this investigation is the BEI AM-1 stereo transmitter with a built-in stereo exciter. This transmitter eliminated many of the variables in aligning the stereo exciter. The stereo exciter contains circuitry that eliminates the need to equalize for the transmitter. The only variable that would result in additional stereo equalization is sideband impedances of the load presented to the transmitter.

The transmitter was loaded with an artificial antenna. The artificial antenna design is shown in Figure 1. The design consists of a series resonant inductor and capacitor in series with a test load. The reactance of the inductor and capacitor was adjusted to produce approximately a 1.6:1 VSWR at the 10kHz sideband frequencies. Between the transmitter and the series resonant load is an adjustable tee network. The tee network was used to change the phase orientation of the load impedances that are presented to the transmitter.

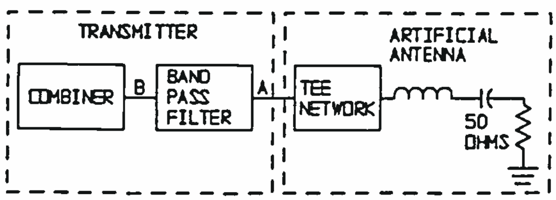


Figure 1. Artificial Antenna

The tee network was adjusted to 7 different phase shifts at 22.5 degree increments. The tee network was removed for a zero degree phase shift.

Impedance sweeps were taken at the output of the transmitter (Point A on Figure 1) and the input to the bandpass filter network of the transmitter (Point B on Figure 1) using a vector impedance meter. These impedances for the two cases discussed in this paper can be found in Tables 1 and 2.

Table 1
Transmitter Load Impedance

Tee Network Phase	Frequency (kHz)	Impedance R±jx (Ohms)
Case 2	580	42.0-j18.7
	590	58.0+j0.0
-22.5 Degrees	600	84.1+j20.2
Case 3	580	41.1+j15.8
	590	58.0+j0.0
-75 Degrees	600	53.0-j27.0

Table 2
Normalized
Bandpass Filter Input Impedance

Tee Network Phase	Frequency (kHz)	Impedance R±jx (Ohms)
Case 2	580	.69+j.17
	590	1.0+j0.0
-22.5 Degrees	600	.88-j.39
Case 3	580	1.43+j.36
	590	1.0+j0.0
-75 Degrees	600	.61+j.18

The procedure for aligning the stereo exciter was done the same way for all setups. A 1 kHz sine wave at 50% modulation was put into the left audio input. A sample was received and demodulated with an AS-10 modulation monitor. An x-y plot of L versus R was viewed on an oscilloscope. The delay and level potentiometers on the stereo exciter were adjusted for maximum separation.

The modulating audio frequency was then increased to 7 kHz. The peak and cutoff potentiometers, high frequency equalization controls, on the stereo exciter were adjusted for maximum separation.

The right channel was aligned in a similar fashion.

A proof of performance was then obtained using an Audio Precision System One, and a BEI model AS-10 AM stereo modulation monitor. A procedure was written to obtain Incidental Phase Modulation (IPM), which is measured in dB below 100% L-R C-QUAM[R] modulation (1.57 radians max). Monaural distortion, stereo distortion, and stereo separation were also measured.

RESULTS

A wide range in stereo performance could be obtained depending on the phase shift of the tee network. For the purpose of simplicity, only three cases will be compared. Case 1, which is used as a reference, will be with the transmitter loaded with 50 ohms non-reactive, Case 2, which exhibited the worst performance, the transmitter will be loaded with the artificial antenna with a tee network phase shift of -22.5 degrees. For Case 3, which showed the best performance, the transmitter will be loaded with the artificial antenna with a tee network phase shift of -75 degrees. Below are the results of these three cases.

C-QUAM[R] is a registered trademark of Motorola,Inc.

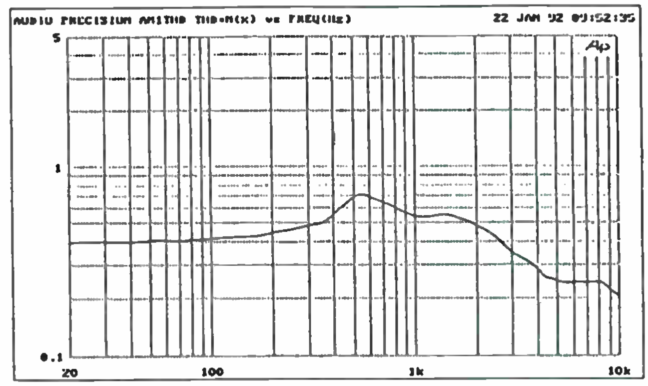


Figure 2. Case 1 Mono THD+N vs. Frequency, 90% modulation, 50 ohm load

For Case 1, the total harmonic distortion for the frequency sweep does not exceed 0.7%.

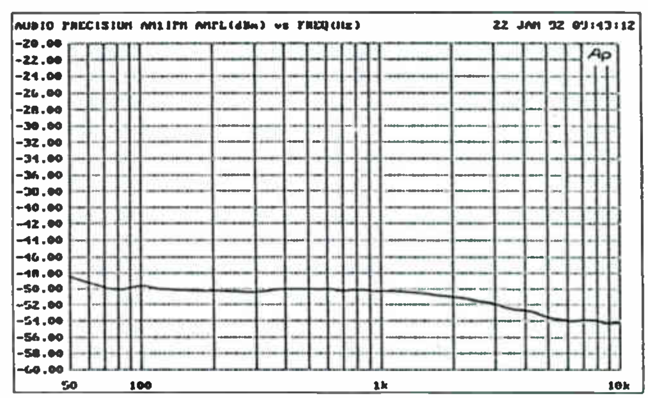
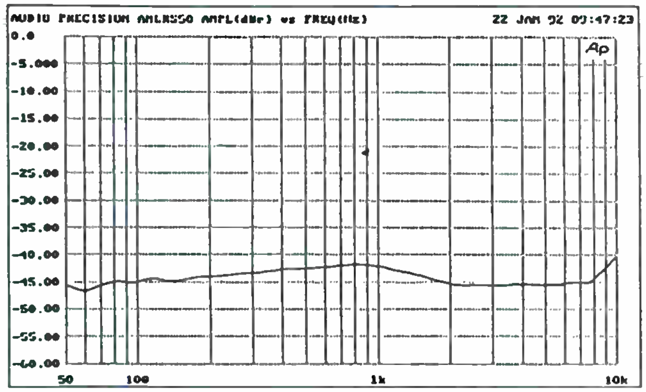
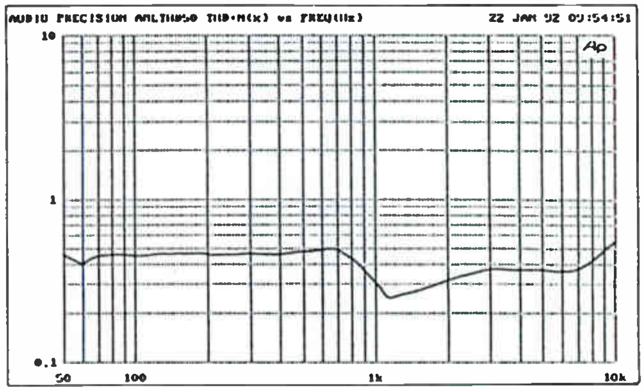


Figure 3. Case 1 IPM, 95% Modulation, 50 ohm load

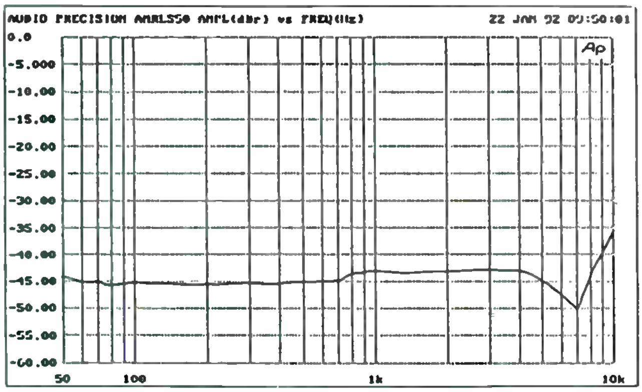
For Case 1 the IPM does not exceed -48 dB.



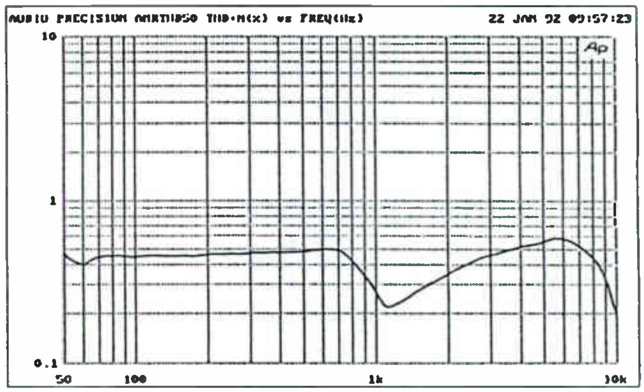
Left to Right Separation



Left Channel



Right to Left Separation



Right Channel

Figure 4. Case 1 Channel Separation, 50% Modulation, 50 ohm load

For Case 1, the left to right channel separation is better than 40 dB. The right to left channel separation is better than 35 dB.

Figure 5. Case 1 THD+N, 50% Modulation, 50 ohm load

For Case 1, the left and right channel THD is less than .6%.

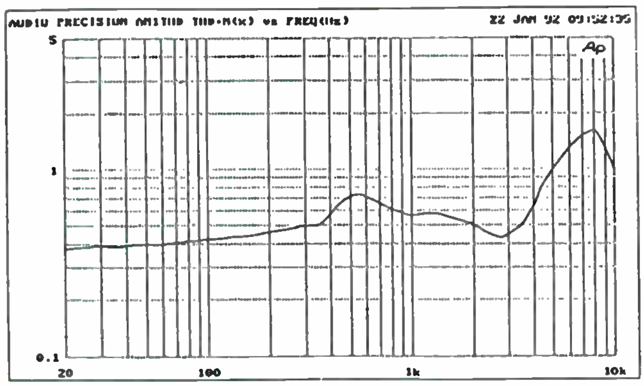


Figure 6. Case 2 Mono THD+N vs. Frequency, 90% modulation, Artificial Antenna, -22.5 Degree

For Case 2, the total harmonic distortion for the frequency sweep went as high as 1.7% at 10kHz.

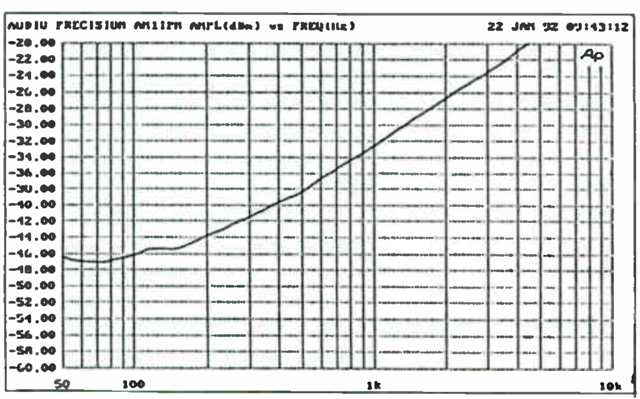
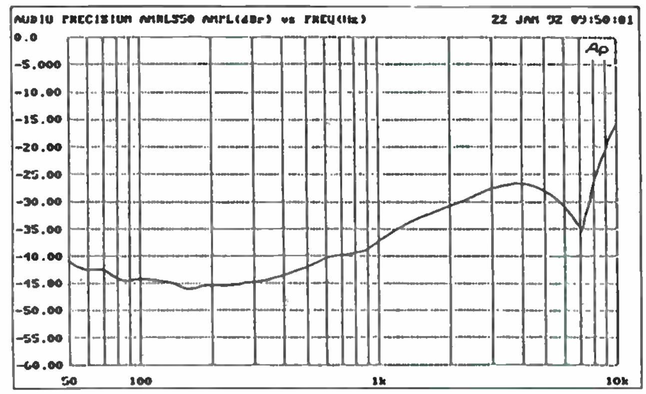
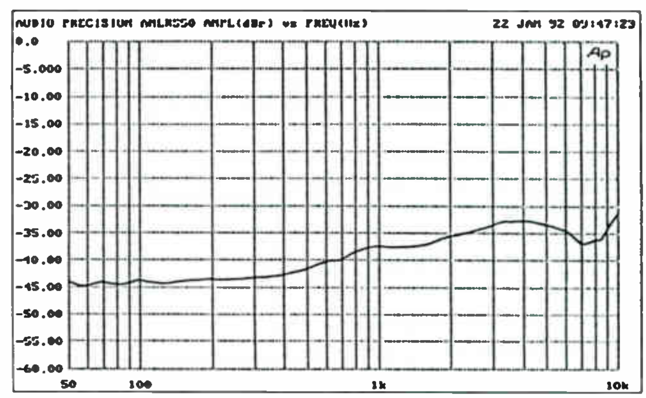


Figure 7. Case 2 IPM, 95% Modulation, Artificial Antenna, -22.5 Degrees

For Case 2, the IPM exceeds -17 dB at 10kHz.



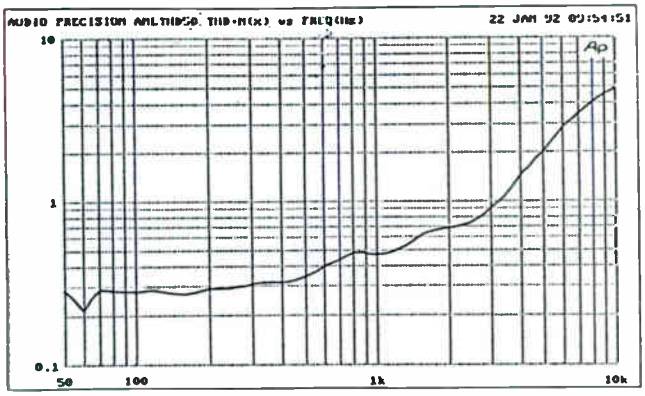
Left to Right



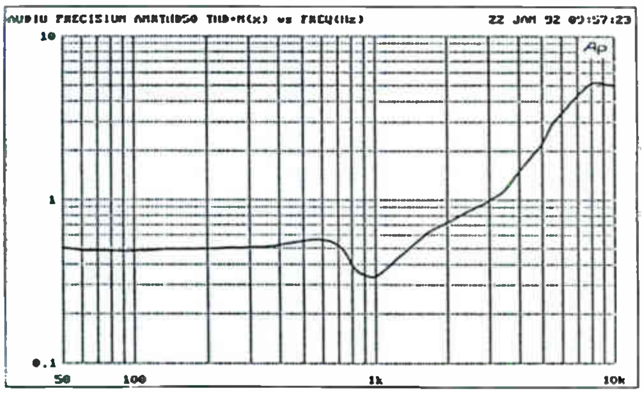
Right to Left

Figure 8. Case 2 Channel Separation, 50% Modulation, Artificial Antenna, -22.5 Degrees

For Case 2, the left to right channel separation degrades to 17 dB at 10 kHz while the right to left channel separation is still better than 30 dB.



Left Channel



Right Channel

Figure 9. Case 2 THD+N, 50% Modulation, Artificial Antenna, -22.5 Degrees

For Case 2, the left and right channel THD rose as high as 5% at 10kHz.

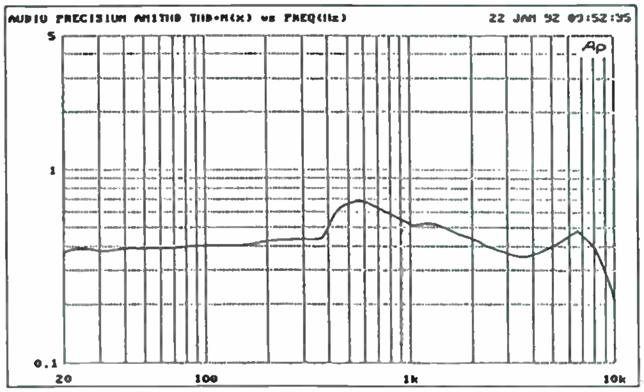


Figure 10. Case 3 Mono THD+N vs. Frequency, 90% modulation, Artificial Antenna, -75 Degrees

For Case 3, the total harmonic distortion for the frequency sweep was less than .6%.

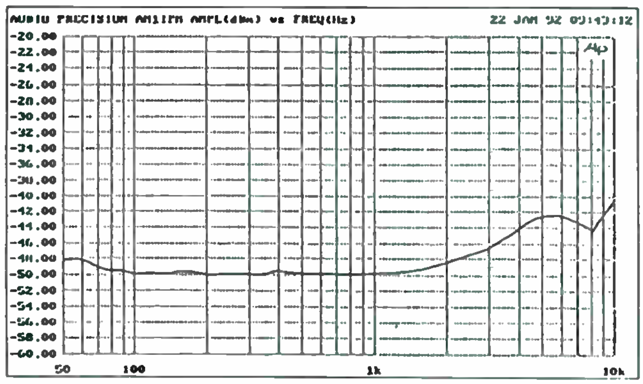
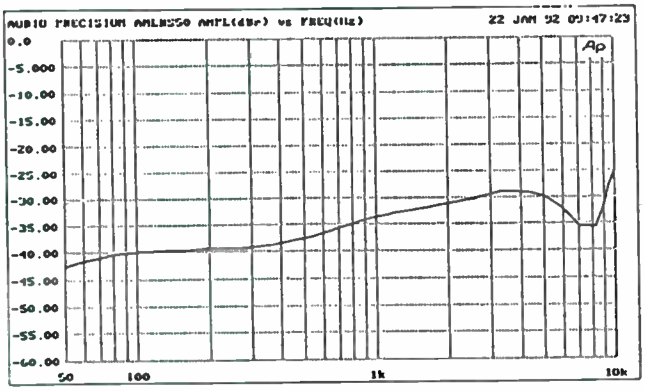
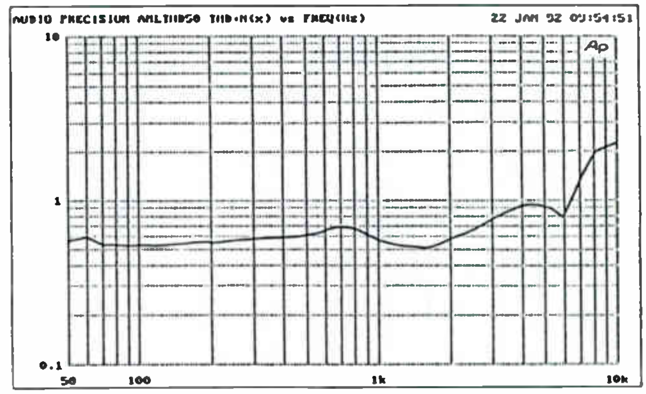


Figure 11. Case 3 IPM, 95% Modulation, Artificial Antenna, -75 Degrees

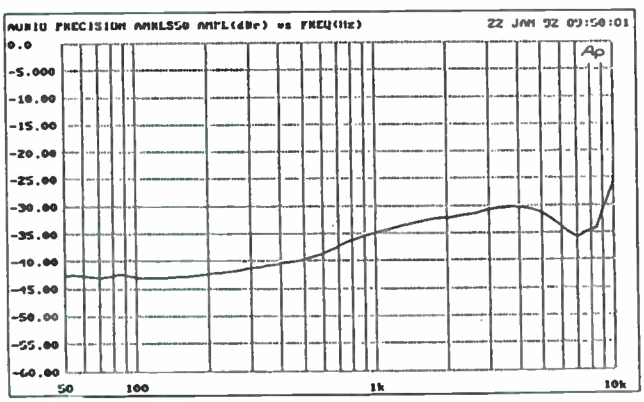
For Case 3, the IPM does not exceed -40 dB.



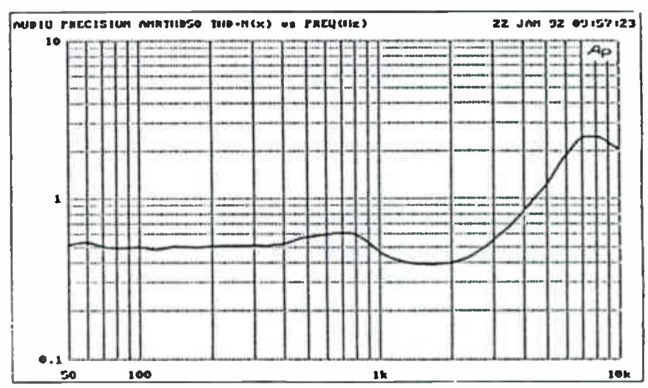
Left to Right



Left Channel



Right to Left



Right Channel

Figure 12. Case 3 Channel Separation, 50% Modulation, Artificial Antenna, -75 Degrees

For Case 3, the left to right and the right to left channel separation is better than 25 dB.

Figure 13. THD+N, 50% Modulation, Artificial Antenna, -75 Degrees

For Case 3, the left and right channel THD has been reduced to 2.5% or better.

CONCLUSION

It can be seen by comparing the results of case 1 to case 2 that the mono and stereo performance of the AM-1 stereo transmitter was degraded by a load that varies with frequency.

The Mono THD for case 1 is .25% at 8 kHz, where Case 2 is 1.7%. The IPM at 10 kHz for Case 1 is -54 dB, where Case 2 is -17 dB. The channel separation for Case 1 is up to 20 dB better than for Case 2 depending on the audio modulating frequency. The left and right channel THD is over 4% higher at 10 kHz for Case 2 than for Case 1.

The only difference between the load in Case 2 and Case 3 is the phase shift of the tee network. The VSWR of each load is the same. This difference in phase shift will present a changed load to the power amplifiers at the sideband frequencies causing significant changes in transmitter performance. The IPM of Case 3 is better than 40 dB which approaches that of Case 1. The channel separation for Case 3 is better than 25 dB which is about a 10 dB improvement over Case 2. The worst channel distortion for Case 3 is 2.5% which is half that of Case 3. Even though the load VSWR for Case 3 is the same as Case 2, the performance is considerable better.

To be able to predict the best load impedance orientation, further investigation needs to be done at different carrier frequencies. Once this is done the impedance sweeps for each frequency tested can be compared to determine the optimum load at any carrier frequency. At 590 kHz, a Smith chart plot of the desired transmitter impedance for the BEI AM-1 stereo transmitter is shown in Figure 14. The resulting impedance at the bandpass filter input terminals (Point B on Figure 1) is shown in Figure 15.

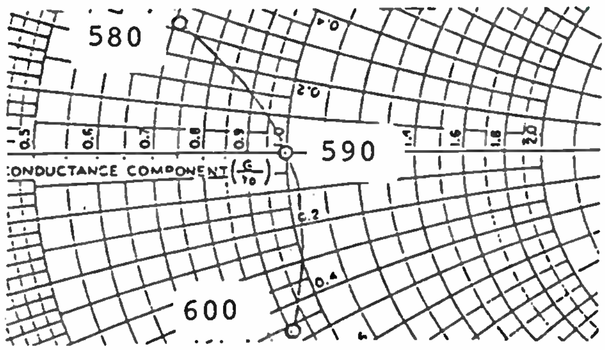


Figure 14. Normalized Transmitter Load Impedance (Point A, Fig.1)

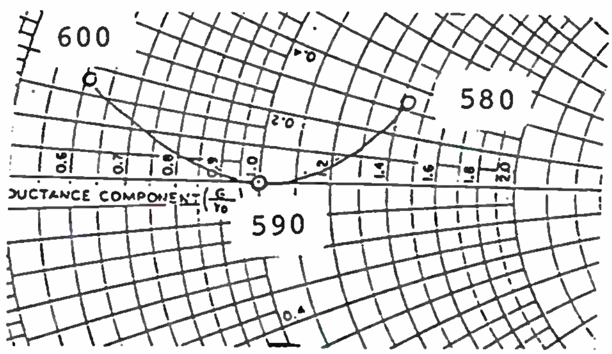


Figure 15. Normalized Impedance at the BPF (Point B, Fig. 1)

SUMMARY

For a load with a given VSWR, the stereo performance may be optimized by presenting this load with the proper phase orientation. This can be accomplished by inserting a tee network between the load and the transmitter. Although the stereo performance will not be as good as the transmitter operating into a 50 ohm non-reactive load, it will be significantly better than with other load impedance orientations.

For the case of 590 kHz using a BEI AM-1 stereo transmitter, the plot of the best load phase orientation is given in Figures 14.

TELEVISION: COPING WITH NEW TECHNOLOGY

Tuesday, April 14, 1992

Moderator:
Richard Farquhar, SBE President, Canal Winchester, Ohio

***TELEVISION: WHERE HAS ALL THE MONEY GONE?**
Jerry Whitaker
Technical writer
Beaverton, Oregon

ENTERPRISE-WIDE AUTOMATION
THE LOCAL BROADCASTER'S CLAIM STAKE ON THE FUTURE
C. Robert Paulson
AVP Communication
Westborough, Massachusetts

***A CASE HISTORY: MASTER CONTROL AUTOMATION**
Marvin Born
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*Paper not available at the time of publication.

ENTERPRISE-WIDE AUTOMATION
THE LOCAL BROADCASTER'S CLAIM STAKE ON THE FUTURE

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ABSTRACT

Television network operations automation in 1992 is in exactly the same state as was AT&T's long distance network operations automation in the 1950s. That era's technology had created a network of automation islands (pulse-dial local exchanges) in a national ocean of operator-initiated long-distance circuits.

Similar automation islands exist everywhere in the 1992 television networks, enabling automated and manual remote control of nearby machines and dedicated systems (editing suites, master control, station break, news copy preparation). However, keeping a network or local station operating on time, airing what the programming log calls for, and sending it to the right destinations, requires hordes of ("local and long distance operator") skilled people operating around the clock. "Bottom line"-oriented managements have reduced the numbers of these people to the absolute minimum needed to operate on time and recover from catastrophes.

In the face of constantly shrinking national and local audiences for national and local programming offerings, networks and stations are therefore facing a 'Hobson's choice": Develop new ways to generate income and profits with the facilities, equipment and people assets you still have, or get off the air.

Enterprise-wide operations automation offers the opportunity to convert this choice (The tired old horse nearest the stable door -- Ie, traditional network and local over-the-air broadcast operations) into a winner.

The victory can be won! Compare the ease and low cost of making a phone call, sending a fax, dumping a computer file, from anywhere to anywhere in the country, to the time and difficulty and costs of telephoning, telexing and telegraphing in the 1950s!

PROLOG: 70 YEARS OF PROFIT PURSUIT

Radio's and TV's Golden Eras: A Parable of the Audience Farmers

During the first four decades of television broadcasting, local ***commercial*** television stations have operated as one of two types of "produce" producers. The produce is called "programming," created and scheduled for consumption by residents (the viewing audience) in the station's coverage area. The producers were and are either the indentured field hands (stations) of one of four and later three and now again four television networks headquartered in New York, or they are independent small farmers.

In both activities, local stations owe the acquisition and continued operation of a precisely defined local piece of farm land to the licensing and regulatory beneficence of the FCC. They don't own it, but can expect to continue to till it as long as they don't break any laws or regulations and operate "in the public interest."

This operating environment is a "crazy-quilt" welter of operating rules, regulations and traditions which evolved during ***radio*** network broadcasting's first four decades of existence. ***Traditions*** created by national, regional and local entrepreneurs exploited the "magnetic" attraction of "wireless" sound entertainment. ***Rules and regulations*** created by legislative and executive branch actions attempted to keep the entrepreneurs operating ***mostly*** in the public interest.

Commercial stations were generally allowed to make money without limit by selling their produce as free enterprisers. Income was generated first by selling time slots to sponsors who could create their own programs to fill them, and later by just selling "spots" on programs created by the stations. Network affiliates were also compensated by the networks for their relinquishing of "day parts" which the networks filled with their own sponsored programs.

In the first three decades of the history first of radio and then of television, "Having a commercial license," was synonymous with "owning a gold mine" or "milking a cash cow."

"Educational TV stations" were another unique category of local farmer. In contrast to FM educational stations, which were all bunched together at one end of the spectrum territory, "ETV" stations (at least the early 1950s startups) could stake out large and fallow plots (VHF channels) anywhere they found an open space. They could till the land and grow crops (create programs)in competition with their abutting commercial TV neighbors, but couldn't sell them to sponsors who were attracted to them.. However, ETV stations could legally beg for money on the air as they gave away their produce.

To give themselves better growing (audience creation) conditions, ETV stations banded together and created their own network (née NET, now PBS) to supply them fertilizer (audience-attracting programming).

And so at least the commercial dirt farmers (local broadcasters) became gentlefolk estate holders (group owners), and lived fat and happy -- until satellites started raining programming into every city and village and farm in the whole United States. In retrospect, it's easy to see that they were really fat, ***dumb*** and happy.

Changing objectives of "broadcasting:" Selling "air time" versus selling "sets"

In the early days of both radio and television broadcasting, receiver sales were the source of profit "fat," not program sponsorship. Radio broadcasting quickly became a profit-making industry after World War I, when Westinghouse and GE took out licenses for stations in the cities where they had radio receiver manufacturing plants.

In the early days of black and white television broadcasting after WW II, RCA and Dumont started television stations and established program distribution networks for the same reason. RCA's General Sarnoff then began a strenuous and bitter several years of battle with CBS's Chairman Paley to establish monochrome-receiver-compatible, RCA-developed color as the US color broadcasting standard. CBS's field-sequential color system had been announced and demonstrated in the late 1940s, when NBC was was beginning to build its black and white network, and RCA was selling receivers at a gratifying profit margin. CBS's color system was never intended to be compatible with the 525/60 black and white format, and the color broadcasting service was proposed to be located in the untapped Channel 14 through 83 UHF spectrum.

What has never been widely known (because of RCA imputations?) is that Goldmark's engineers were working on an all-electronic receiver to replace the mechanical color wheel monstrosity ASAP. If the Peter Goldmark CBS Technology Center troops had won that battle (which they did, temporarily, in 1950), RCA's flourishing VHF B/W receiver market faced near-future annihilation. Chairman Paley had made an eventually financially disastrous purchase of a receiver manufacturer to equip CBS to make money from CBS color receiver sales.

CATV (***Community antenna*** TV) was similarly birthed by "boondocks" radio stores as the only solution to their desire to make money on TV set sales.

The sagas of the profitable growths of both over-the-air radio and television broadcasting and cablecasting are all identical. Within a decade of the first local broadcasts "in the public interest" and the start of receiver sales, generating profits by selling sponsorship of air time was the primary objective for getting into the business. Programming was judged by its ability to create large viewing audiences, a definition of "public interest" not intended by the legislative and executive branches of government.

Jaundiced viewers of the industry call it

"programming to the lowest common denominator."

PRESENT PLIGHTS

"Serving the public interest" in 1992

Merging wired and wireless transmission technologies (enabling program delivery via satellite and coax cable) have created a 1992 definition of "public" which totally compromises the traditional broadcasting goal of making money by creating large viewing audiences. The public is now many small publics, each with unpredictable desires to "watch what they want when they want it," not to watch only what the networks send them.

Merged satellite and coax transmission technologies initially gave CATV systems operators a unique opportunity to make money by renting boxes which connected viewers to many program sources. That started the never-to-end network slide in prime time share. Cablecasters then began to "bite the hands that fed them" (local stations) by setting up local entertainment program, public affairs, news and sports delivery channels indiscernible from FCC-licensed stations. Some of them even moved stations to channels other than their own over-the-air assignments, or dropped them entirely.

None of this ferocious competition for viewing audience among traditional broadcasters and cablecasters can be ameliorated by judicial decisions based on half-century old traditional legislation. Nor can new legislation be written to reestablish an environment in which which traditional broadcasters can survive by doing business in traditional ways.

"Serving the public interest" now and forevermore means

providing access to what the public wants, wherever it is,when they want it.

You can't legislate that kind of need!

The broadcast industry's 1992 plights

The launching of the first geostationary television relay satellites in the mid 1970s was the death sentence pronouncement on traditional broadcasting. Could the current death throes of traditional "television network broadcasting" which started thereafter have been averted or slowed? NO!

The television product applications of digital signal processing technology which appeared in the late 1970s were the first rejection of sentence appeal. [1, 2] The appearance of one or more broadband digital fiber drops from the nearest "telephone pole" to set-top tuners will be supreme final appeal denial, sometime in the next ten to twenty years. [3]

Who is the real killer whom the broadcaster must confront to stay alive? Every individual in that uncontrollable, self-satisfaction-driven, demographically proliferating viewing audience, relishing that array of program selection options powers first made available when the "rampaging technology" revolution began in the late 1970s.

What is their death weapon? The *Zapper* -- whose invisible rays emanating from remote controls for tuners, VCRs, video games, et al, are operated by free-thinking human brains. They react instantly to "bad" (disliked or unappealing for any reason including whim) programs and commercial messages.

Network broadcasters' 1992 survival efforts

In recognition of the impact of constantly shrinking audience size on sales and profits, the originally programming-oriented commercial networks are now rather being run by financial entrepreneurs with "Produce - or else -" operating philosophies. [4] But simply cutting costs does not fit the networks for competition with their former and present program suppliers distributing their products via their own satellites to cable system head ends.

Networks are therefore entering into distribution alliances with cable MSOs, the mortal enemies of the very affiliates who originally delivered the nationwide audiences that made net-work broadcasting profitable.

Ie, networks and affiliates will stay friends only as long as more money can't be made by forming other alliances.

Local broadcasters' 1992 plight

FCC-licensed "local stations" operating in the good old days tradition of the golden decades are a death-approaching anachronism here in the 1990s.

Reprising:

Program networks, including local stations' former monogamous "soul mates," now number well over 50;

More than 60 percent of viewers' television sets are connected to coax cables delivering 35 to 50 channels or more;

When there's trouble in the cable system, very few viewers have any means for receiving VHF/UHF channels "off the air;"

Each VHF and UHF affiliate licensee is a single signal deliverer "on the cable," and may be competing for local audience with another affiliate carrying the same programming;

FCC-approved call letters and channel IDs are meaningless in this new "broadcasting" environment, since cable system operators can arbitrarily assign stations to channel locations unrelated to their over-the-air IDs;

In constantly increasing numbers, cable systems are producing and "airing" their own locally-sponsored local news, sports and entertainment programs on VHF channels with self-assigned call letters, the first of which was "WGRC" (Greater Rochester Cable-Channel 5) in Rochester NY.

FIGHTING BACK TO THE FUTURE

The local broadcaster's claim stake: Metamorphose into a regional network

Street-smart opinion is that at least one of the four nationwide commercial television broadcasting networks will vanish before the end of this century. The survivors will cease compensation payments as the first step toward the dissolution of traditional networking operations and financing.

To survive into the next century, local stations have no alternative but to become networks themselves. They must metamorphose into new-look business enterprises operating as ***regional*** television networks. Metamorphosis is the right term to use as a descriptor of this change process. The process does not require abandonment of current assets in facilities and people and expensive acquisition of new ones.

Here are the steps in a several-year metamorphosis plan:

Promote your script-writing, graphics, studio and field production and postproduction facilities and people into a "production central" -- for any additional capabilities you need, contract for them as needed with local production houses and freelancers;

Negotiate with ***all*** the active and planned programming producers and networks as sources of audience-building programs for local viewers, and establish yourself as the exclusive regional importer for the product you can sell;

Set up your existing master control facility as the regional headquarters for diverse types of interconnect networks, with local telephone companies and alternative transmission services providers as your distribution partners;

Structure and promote these networks to attract new production and distribution business in government, business, education, retailing, health care, et al;

Sell your new-look organization to all the cable systems as ***the*** supplier for locally-produced and sponsored news, sports and entertainment programs;

Look at regional and local advertisers as prospects for advertising message production services that only begin with traditional 30-second cable spots.

Sounds simple! But is it? How do you change from a single over-the-air broadcast channel to a multiple input channel/multiple output channel/regional communications network,***without*** spending a lot of money on bigger facilities and adding lots of people?

Enterprise-wide automation: The metamorphosis key

Tradition and experience tell you that the proposed metamorphosis is difficult, impractical and/or impossible. To operate a facility with many input feeds rather than one, many production and postproduction jobs going on simultaneously, and multiple output feeds to the transmitter, cable head ends and new "non-broadcast" customers, means ***adding lots*** of skilled operations people.

That's true, if you propose to continue operating at, or add piecemeal to, your current rudimentary levels of automation -- ***machine control*** and ***special purpose systems*** (editing, master control, station break, news production, et al).

Two higher hierarchical levels of automation are the key to the metamorphosis process. 5

"Facility-wide automation" This connotes all the equipment and people under the control of a single organization, not necessarily in one physical location, but all in close proximity to and more or less permanently interconnected.

"Enterprise-wide automation" "Enterprise" is computerese for businesses established and operated by entrepreneurs to carry out a stated operating objective. (In broadcasting that objective is to make money.) The English word derives from the French "entreprendre." In the computer industry, enterprises are businesses whose activities are carried out independently, simultaneously and in parallel at widely scattered locations. Access to data bases, whose exact locations may not be precisely known, is a random but constant need of the people at work in the enterprise.

Computer "platforms" (hardware ranging from microprocessor chips to mainframes and the software that "turns them on") are the local access points (communications network "nodes").

"Common carriers," the interlocked yet competing hierarchy of local, regional, national and international "telephone companies" provide the circuits.

> ***Doesn't that description suggest to you that the functions you have to carry out constantly to keep your station on the air are exactly those of a "business enterprise"?***

Facility-wide automation status

No radio or television station or network anywhere in the world is currently "facility-wide" automated. Skilled people are at work in every one making patches, pushing buttons on cue, typing commands on keyboards, monitoring system performance prepared to take manual actions to fend off troubles, etc.

True, those actions are seldom directly mechanically coupled to the machine being controlled. Computers interconnected to scattered microprocessors interpret the actions and exchange their own digital communications over "buses" to carry out human decisions. BUT -- that only happens in systems where ***all*** the controlling computer components "speak the same language" ie, their hardware is all from the same family "platform," and all the operating system software is the same "rev" -- revision.

Facility-wide automation requires that any command issued by any "boss" computer, in any language and dialect, will be accepted, understood, and acted upon by any subservient computer addressed in the command's "header." That's impossible in the 1992 television industry.

Regrettably, moreover, you currently can't look to the computer industry for automated communications system design solutions to that need. Terminal- and board-level interconnection bus "standards" outnumber those in the television industry 10:1 or better. Standards-setting groups are legion compared to the television industry's one -- SMPTE. Interconnection protocols range from EIA 232C to the mind-dizzying, both loved and reviled, ISO/OSI (Open Systems Interconnection).

In any event, if you agree that your local broadcast station operation is already the headquarters of a network, it makes sense to include facility-wide automation planning as a subset of a plan to implement enterprise-wide automation. Doing so, you have begun to plan a geographically limitless "facility without walls". 6

Enterprise-wide automation status

Brief reflection on the ease of making phone calls and setting up audio conferencing networks, sending faxes to multiple destinations, and allowing computers to make their own data-swapping connections, establishes that enterprise-wide automation already exists in the world around us. Those networks include both permanent and ad hoc dialup circuits, interconnecting mainframes and their huge data bases and remote dumb, smart and intelligent terminals, to interchange information between and among people as needed, and among data bases as pre-established in protocols.

The system sounds like "just what the doctor ordered" as a prescription for television industry enterprise-wide automation. Again, regrettably, that's not a valid assumption, on three counts.

(1) The telecommunications industry long distance, regional, and even some metropolitan area (intra-city) networks all operate in the digital domain. All the widely-used television, radio and sound recording industry interface standards are analog. Digital program audio and NTSC video signal form standards are only accidentally compatible with telecommunications industry transmission standards.

(2) The telecommunications industry is ruled and regulated by the FCC and fifty state-level Public Utilities Commissions (PUCs). The tariffs set for use of transmission networks are based on the assumption that the digital bit streams are digitized voice (300 to 3000 Hz analog signals encoded into a 56 kbps bit stream) or computer outputs at that throughput level or less.

The starting point for promulgating a tariff for transmitting the television industry's "D2" or D3" signals (a 140 Mbps bit stream carrying full bandwidth digitized NTSC video and four digitized 20 Hz to 20 kHz audio channels) is to multiply the 56 kbps tariff by 2500 (140 M/56 k bps).

That's why "DS-3" 45 Mbps transmission service (video compressed 2.45:1 plus two audio channels), offered nationwide by VYVX, is now deemed by broadcasters to be "broadcast quality." The transmission cost is an hourly rate based on usage and the number of dropoff nodes. There is no milage charge.

(3) Television industry standards and protocols for routing and distributing signals within a facility are totally different from the telecommunications industry's single-circuit facility-to-facility transmission standards and protocols.

<u>Operations and transmission automation commands</u> (transmission sources and destinations and SMPTE time-coded In/Out points) are outputted from computer terminals generally at 38.4 kilobaud or slower throughput rates. These are carried on unshielded stranded copper twisted pairs. A few much higher-speed protocols, still proprietary to their developers, require "thin" or standard Ethernet cables.

<u>Ancillary data</u> (histograms about the signals being transmitted, Closed Captioning text, VITS and VIRS signals, source IDs, et al) are integrated into the blanking intervals of the video signal before transmission.

<u>Video signals</u>, traditionally NTSC but now including digital NTSC and serial multiplexed analog and digital components, are carried on 75-ohm copper coaxial cables.

<u>Audio signals</u> are each carried separately on shielded stranded copper twisted pairs.

To be compatible with telecommunications industry single-circuit transmission standards and protocols, the automation commands and ancillary data must be integrated into a digital domain "Header and Descriptor."

This information packet must precede the video and/or audio signal packets which follow it on the same circuit.

A draft of a "Headers and Descriptors" standard for the transmission of a hierarchy of digitized video (electronic image) signal forms has recently been publicized. [7] It was produced by a joint task force of standards experts from the SMPTE, the ATSC (Advanced Television Systems Center) and the IEEE/CCIP (Institute of Electrical Engineers Committee on Communications and Information Policy). A second Digital Image Architecture task force effort will define the hierarchy and create a protocol to facilitate interoperation of high resolution display systems.

Both task force efforts, which began in late 1990, were directed by Stanley N, Baron, then SMPTE's Vice President of Engineering, and Managing Director of Technical Development at NBC. Industry-wide ratified standards document issuance is planned for late 1992.

AN ENTERPRISE AUTOMATION PRIMER

Network types and capabilities

Network configurations can be any one of four "architectures" (general definitions in parenthesis):

Point to point (private line)

Point to multipoint (narrowcasting),

Multipoint to multipoint (teleconferencing),

Multipoint to point (information collecting).

Computer networking in all these configurations began as expensive, dedicated leased-line monolith systems. As the common carrier networks became more versatile and user-responsive, ad hoc networks became feasible to establish by destination dialup. These were initially comprised of 300 to 3000 Hz analog "telephone circuits" over which digital data could be transmitted.

In the 1970s and early 1980s maximum reliable transmission rate increased from 300 "baud" (essentially the same throughput rate as but still different from 300 "bits per second") to 19.2 and even 38.4 kilobaud over these circuits.

These circuits could accommodate far higher digital transmission speeds, even full bandwidth NTSC video for short distances. The terminal equipment and central office switches adequate only for 3000 Hz voice signals were the bottleneck.

Virtual networks

These are grids of any of the above architectures of interconnecting circuits. They are set up by software responding to commands initiated by a user. [8] The user is in control of all network functioning variables: configuration, bandwidth, time duration, interactivity, information access, etc.

In a television operations enterprise, the variables extend to the specification of scores of signal parameters.

System signal capacity variables [9]

Video - Aspect ratio, Horizontal resolution, Vertical resolution, Frame/field repetition rate, Line scanning mode, and ***Bandwidth compression ratio.***

Synchronous audio - One to eight channels, subject to compression both independently and as part of a compressed TV (video/multiple audio)channel.

Ancillary data - Independent channels without limit, carried as time- and/or frequency-division multiplexed components of video and audio signals, as headers/descriptors, and/or as separated signals, if for technical reasons they cannot be compressed.

Receive terminal capability - If it does not have digital signal processing (DSP) capabilities at least the equal of those in the source terminal, only the signal bandwidths and signal channels and ancillary data it can process should be transmitted.

Enterprise infrastructure and applications **10, 11, 12, 13, 14**

Signal path configurations - Point-to-point, point to multipoint, multipoint-to-point, and multipoint-to-multipoint, with constantly changing originating nodes.

Transmission circuit types - Circuit "legs" in any given end-to-end signal path may include multiple satellite and terrestrial microwave hops, copper coax, fiber, and even "thin Ethernet" unshielded twisted pair "wire" circuits.

Network traffic - Diverse television signals listed above, digital radio channels, multiple channels of audio only, images and optional voice/ data in medical, scientific, educational and industrial systems, computer-generated still and animated graphics, full duplex voice and ancillary data, and -- ? Any signal may or may not have been digitally compressed never, once or many times.

X-Y-Z versus time matrix plots of those totally independent variables starkly prove that enterprise operations management is rapidly leaving the realm in which humans can be the operations managers. The system controller must be able to deal in third-order equations for "next assignment" switching as the "normal" level of network management. The EXECUTE command can occur at straight up in any second of any minute of any hour around the clock. Each execute may independently change both the configuration of the network and the internal setup of each node..

SMPTE's Next Standardization Challenge

Now it should be apparent why SMPTE's 1991 standards-setting effort must be viewed simply as "another step" in an endless standardization journey. This step only enables different standards of products in the ***electronic imaging*** part of a video/audio/data ***system*** to work with each other. Further, the protocols must still be viable when those products are separated by walls and floors in buildings, or walls, floors and buildings in a campus ***facility***, or many miles of public right of way distance tied together via telecommunications industry standard common carrier circuits.

A portion of any new header/descriptor standard must be reserved for protocols which automate the setup and activation of dialup long distance circuits, including a once technically unattainable transmission network feature -- ***bandwidth on demand*** (BOD). This must be automatically adjustable both before transmission start and as bandwidth needs change in the course of transmission.

Implementing this capability calls for a new level of team effort including the television and computer industries, the telecommunications industry, ***and*** Federal and state legislatures, the FCC and state PUCs (cf FN 9). Otherwise it is unthinkable but entirely possible that the birth of technically feasible, economically mandatory ***BOD*** services could be aborted by political infighting or legislative fiat.

Coincident with that effort, header/descriptor and digital hierarchy protocols for ***sound*** format translation must be developed for system, facility and enterprise operations automation. And existing protocols and header/descriptor standards for packetized data transmission must be integrated into those protocols.

SUMMARY

When these next steps have been completed, hardware manufacturers, software developers and transmission services providers can begin to develop and sell products which create facility-wide and enterprise-wide automation systems. Until they are available, however, the "bottom lines" of local VHF/UHF stations are going to vary in color from light black to saturated red. Business plans for converting them to regional television services enterprises are going to have terribly suspect break-even projections.

When will the next steps start?

Who will take the first step?

Will SMPTE again take the lead role?

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DIGITAL VIDEO COMPRESSION—THE BASIC CONCEPTS

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Abstract- The technology of digital video compression is rapidly becoming the most dominant factor in the development of new digital video products and services. While compression creates a substantial risk of picture quality impairment, the continued improvement of compression technology promises to provide very good quality pictures with impressive compression ratios. This paper will examine the basic concepts employed by most compression algorithms and how they relate to broadcast applications.

INTRODUCTION

When digital video signals were introduced in the 1970's the primary focus was on quality pictures. Any digital representation of an analog signal was subject to a number of distortions. These distortions were of extreme concern to the broadcast and post production marketplace, an industry driven by product features and picture quality. This focus on quality provided for a number of very good digital products involving effects, graphics, videotape, disc recorders, and signal processors. However, the technology utilized to produce these products was expensive to implement, providing for few affordable products.

During this same period of time, the telecommunications industry was also pursuing the development of digital video products and services for their use. However, cost, not quality, was the issue of concern here. The target market for these products and services was extremely cost conscious. The goal was to provide an affordable product with a marginal, yet acceptable, quality of image for applications such as videoconferencing, distance learning, picturephone, etc..

While the broadcast industry was refining the A to D process and developing the technology of handling the large data capacity necessary for their applications, the telecommunications industry was developing ways to reduce the necessary data capacity to allow for more cost effective transmission and storage of video signals.

Digital Video Compression

This technology of data capacity reduction is called data compression. In the video domain it is called digital video compression.

Digital video compression is defined as any process that creates a reduction in data rate from some specific video image reference data rate. The compression ratio is the relationship between the reference data rate and the processed, or compressed, data rate. These rates can be given in either bits per second (bs) or bits per frame of video (b/frame). Direct 8 bit digital conversion of an analog video signal yields a reference data rate of 140Mbs (4.7Mb/frame) for composite NTSC and 270Mbs (9.0Mb/frame) for 4:2:2 component video. These rates include audio and error correction codes.

While there is a method of data compression that introduces no loss in signal quality (lossless compression), the data reduction of such methods is not great enough to be useful for video applications. Therefore, it is assumed that all video compression techniques will create the risk of some picture impairment. The key word here is risk. Digital video compression is more sensitive to some picture characteristics than to others. For this reason, it is entirely possible for some images to undergo substantial compression and exhibit minimal, if any, loss of quality. It is extremely important to match the compression algorithm with the video application such that the compression criteria is most sensitive to the image characteristics that are least likely to occur.

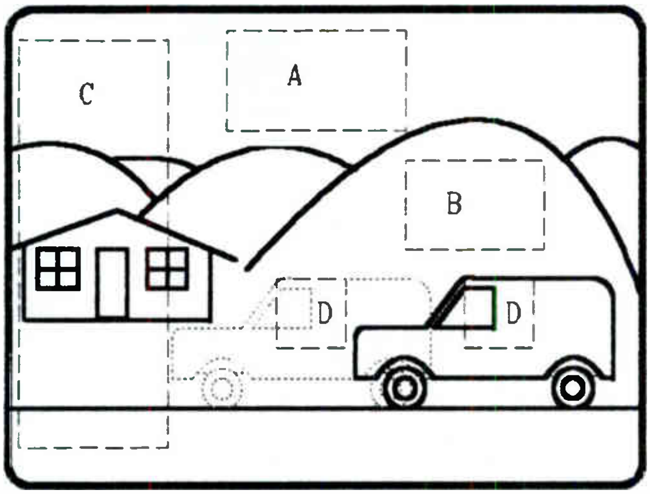
A good compression algorithm has two primary quality goals. The first goal is to provide a more efficient method of representing the total image. The second goal is to sacrifice those characteristics of the picture that are least perceptible and least likely to occur.

Signal Efficiency

One way to improve signal efficiency is to reduce the use of repetitive or redundant information. Most images have a large amount of spatially redundant information -- adjacent pixels that have the same or similar luminance and color values. Providing a way to send a large block of similar pixels as a single value could greatly improve signal efficiency. In Figure 1 the sky, hills, house walls, and truck sides are all spatially redundant information. In reality, all of the pixels in these areas probably will not have identical values. However, each adjacent pixel will be extremely close in value to its neighbors, suggesting a compression scheme that rewards pixel similarity.

The fact that most picture information does not change from frame to frame provides another opportunity for improved efficiency. This interframe similarity is known as temporal redundancy. The improvements offered by temporal redundancy involve sending only those portions of the picture that change between frames. In most applications enough memory to store an entire frame of video is necessary at both the transmit and receive locations. This adds a significant cost factor to the codec (A to D encoder and decoder).

A further enhancement of interframe efficiency is the use of motion compensation vectors.



A & B - Spatial Redundant Picture Data

A, B, & C - Temporal Redundant Picture Data

D - Temporal Prediction Picture Data

Figure 1 - Picture of Moving Truck

This concept allows specific areas of one frame that move to another location in the following frame to be represented by the data from the previous location and a displacement vector. In this case the picture data does not have to be transmitted. The truck in Figure 1 is such a case. As the truck moves along the road, each succeeding frame does not need to recreate the truck; but merely move the data necessary to represent the truck to a different location in the image plane. The only new data that needs to be transmitted is the scenery behind the truck that becomes visible as the truck moves forward. While this method would save a moderate amount of signal capacity, a tremendous savings would occur if the camera were to follow the truck and pan against the background.

There is a further adaptation of this concept, called motion prediction, used in some algorithms. A prediction algorithm calculates the apparent direction and velocity of a moving object and stores that vector in the receive codec. In this case only a deviation from the calculated course now need be sent, further reducing the necessary transmission data.

Compression Architecture

While there are a large number of digital video compression algorithms either in current operation or under development, one common compression architecture has emerged as the most popular. This basic format architecture was developed a number of years ago by the telecommunications industry for image compression. With continual modifications and enhancements, it has proved to provide the most flexibility and efficiency.

The common architecture calls for the analog video signal to be sampled in luminance and color difference components. Luminance can be sampled at either 6.75 MHz or 13.5 MHz depending on resolution requirements. Color components are subsampled at half the rate of the luminance in both the vertical and horizontal direction. This two dimensional use of 4:2:2 sampling allows a

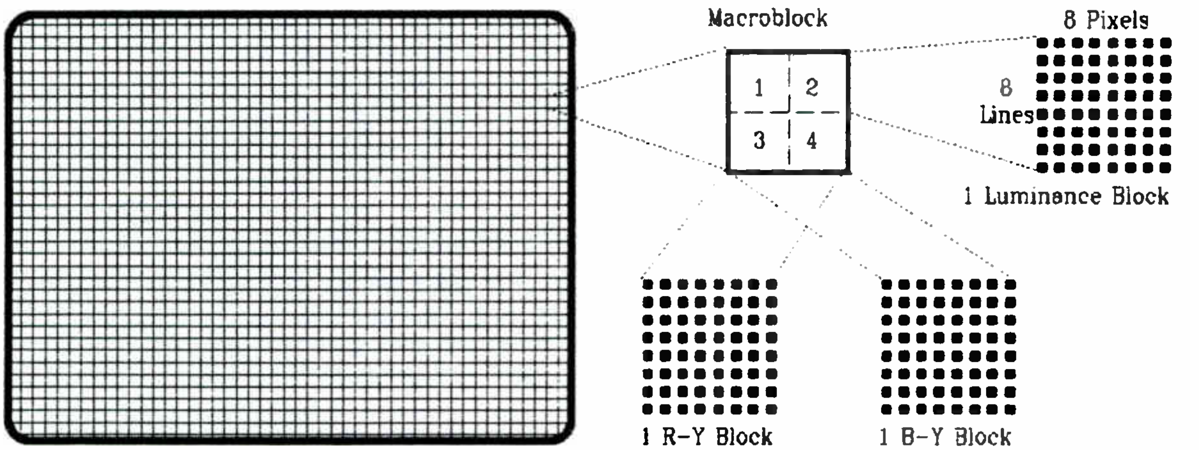


Figure 2 - CIF Digital Video Sample Block Format Structure

Each Macroblock contains 4 luminance blocks and one each of the R-Y and B-Y blocks of 64 pixels each

Each pixel is defined by a digital sample point.

4:2:2 quality picture to exist in a 4:1:1 space (reducing both the horizontal and vertical bandwidth by a factor of two reduces the total bandwidth by a factor of four).

The digitally sampled frame is now divided into square blocks of 64 pixels (8 pixels high by 8 pixels wide). Four of these luminance blocks in a 2x2 square make up a macroblock. Each macroblock also contains one each of the two 8x8 pixel color difference blocks. The video frame is made up of an array of these macroblocks.

A Discrete Cosine Transform (DCT) is performed on each of the 8x8 pixel blocks. The DCT performs a time domain (discrete value) to frequency domain (rate of change) conversion on the entire block of 64 values. Each of the 64 values now represents the equivalent of a frequency coefficient in a Fourier Series. This is not a Fourier Series, but the concept is the same. In the event of adjacent pixels with similar values, only a few of the lower frequency coefficients would have values. The higher frequency values would all be zero and could be lumped into one "run of

64	65	66	67	68	69	71	73
64	65	66	67	68	69	71	74
65	66	67	68	69	71	73	75
66	67	68	69	70	72	74	76
67	68	69	71	72	74	76	78
68	69	70	72	74	76	78	80
69	70	72	74	76	78	80	83
70	71	74	76	78	80	82	85

→ Descrete Cosine Transform

310	20	1	0	0	0	1	0
38	10	0	0	2	0	0	1
15	2	0	0	0	1	0	0
1	0	0	1	0	0	1	0
0	0	0	0	0	0	0	0
1	0	0	0	1	0	0	0
0	0	1	0	0	0	0	0
0	1	0	0	2	1	0	0

→ Quantization

300	20	0	0	0	0	0	0
40	10	0	0	0	0	0	0
14	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0

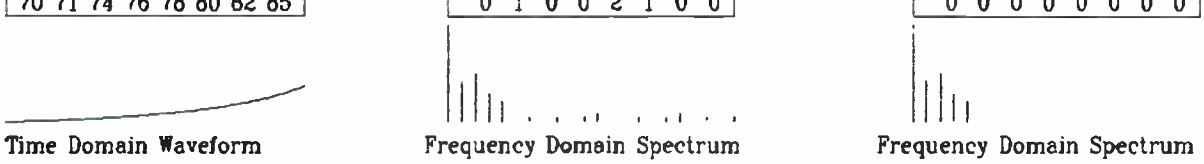


Figure 3 - Data Compression of Digital Video Sample Block

zeros" value. In other words, a DC level byte, four frequency coefficient bytes, and one run of zeros byte (6 bytes) could replace a 64 byte block of data -- a compression of 10:1. The run of zeros can also be at the front of the block as in an application with a large amount of high frequency content (striped or checked clothes of newsperson). Rapid light/dark transitions would contain a limited number of higher frequencies with no low frequency content and could be sent as one DC byte, one run of zeros byte, five high frequency coefficient bytes and one run of zeroes byte (8 bytes). In this application the available image compression is not determined by the presence of either high or low frequencies but by the complexity or number of frequency coefficients necessary to recreate the image.

In the quest for greater compression, a process called quantization is often implemented. Quantization provides a method to reduce the total number of values each frequency can have. Instead of 256 values (8 bit coding), each coefficient might have 50-100 available values based on a rounding table. This table can be variable in nature such that values of 90-95=93 and 95-100=98 while 0-1=0, 2-3=2, and 20-22=21, allowing consideration of the perceivable effect of each range.

The DCT processing of digital video creates a condition whereby the occurrence of some specific data values is significantly higher than others. Under these conditions a variable bit coding process can create further data compression. In variable bit rate coding schemes, such as a Huffman Code, the data values that occur most often are defined with the fewest bits while the data values that occur the least are defined with the most bits. Since it is important to keep any combination of two bytes from looking like any other single byte, the large bytes can have 15-20 bits. The quantity of available bytes with few bits will be small in comparison to the quantity of bytes with large bit size. Unlike all of the other factors previously discussed, this process has the capability to create reverse compression if a sufficient number of large bit bytes is needed. It is imperative that any algorithm using this process carefully define the new code such that this will be an extremely rare occurrence.

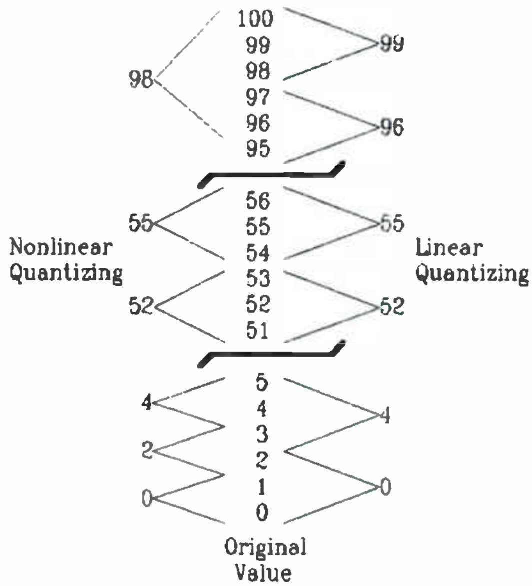


Figure 4 - Data Quantization

All of the aspects of the DCT compression format discussed to this point occur within a single frame of video. Compression algorithms that use these techniques exclusively are called intraframe (within the frame) coding schemes. Intraframe coding is limited to modest amounts of data compression. However, the entire picture is contained in each frame creating an acceptable product for frame accurate video editing and eliminating any motion distortion.

Compression techniques that seek to take advantage of temporal redundant image data and motion compensation can achieve a greater amount of data compression. These compression algorithms are called interframe (between the frame) coding techniques. While interframe coding algorithms use the above intraframe format concept, only those blocks and macroblocks that contain new data are sent for each new frame of video. The rest of the picture will be stored from the previous frame in a video frame buffer in the receive codec. Blocks that change position in the new frame will be sent a new displacement or motion vectors to allow for movement of image content between frames without retransmission.

Picture Quality

It should be obvious to anyone still awake at this point that all these techniques are based on assumptions regarding picture content. These assumptions, when valid, allow for significant data compression without picture impairment. However, when the picture content cannot support the necessary data compression some picture quality must be sacrificed. As stated earlier, a good compression algorithm sacrifices those aspects of the image which are least noticeable or disturbing to the target application. Table 1 lists most of the image quality variables.

Table 1 - Quality Variables in Digital Video Compression

- A. Motion
  - 1. Frame Rate
  - 2. Random / Predictable
- B. Luminance Resolution
  - 1. Contrast
  - 2. Shading
  - 3. Sharpness
  - 4. Location
- C. Color Resolution
  - 1. Contrast
  - 2. Hue Accuracy
  - 3. Sharpness
  - 4. Location
- D. Noise

When an image is too complex to compress without quality loss, there is a large risk of data overrun. In the case of data overrun only a portion of the picture is encoded before data capacity is spent. The frame buffer fills up and defines each remaining 64 pixel block with a single value, thus creating a mosaic image distortion called "tiling". To avoid tiling, each frame is analyzed for complexity, and a graduated algorithm adjustment is implemented to provide an evenly distributed quality sacrifice over the entire video frame.

Intraframe coding schemes usually sacrifices color detail and luminance detail first. They might also adjust the quantizing scale to create more noise and transition distortion (short time distortion). In low quality applications, the original analog signal can be subsampled to produce a basic picture with low H and V resolution.

While interframe coding algorithms are subject to the above problems as well, they are most frequently plagued by motion artifact. Motion artifact is defined as a perceptible error in moving video images. These errors are usually perceived as image blurring, jerkiness (strobing), and aliasing. Some algorithms slow down the frame rate to compensate for data overrun, creating a jerky motion. Many low bit rate video-conferencing products look like strobed stop-action photography. Other coding methods use a frame averaging technique that causes the image to appear as a blur during motion. Obviously these artifacts are not desirable and only appear in situations where the picture content creates a data overrun for the desired compression.

Data compression ratios for digital video have very little meaning at this point in time. Since no true reference standard is accepted by all those involved in video compression, the compression figures are hard to analyze and compare. It is the earnest desire of this author that all compression be referenced to the 4:2:2 serial component digital data rate of 270 Mbs.

Video Compression Standards

A number of standards have emerged in the last few years to help provide a common framework for algorithm development. The most notable formats that follow the above DCT concept are JPEG, MPEG, DVI, and H.261.

JPEG is an acronym for Joint Photographic Experts Group. This format is an intraframe coding scheme designed primarily for still frame image applications. A number of products for motion video have been developed using JPEG in a 30 Fps application.

MPEG and DVI are both interframe coding techniques. MPEG stands for the Motion Picture Experts Group and DVI comes from Digital Video Interactive. While MPEG is a true ANSI standard, DVI is a proprietary algorithm of INTEL. Both of these standards are different adaptations of the DCT based compression using inter-frame algorithms. The target application for both standards is in computer multimedia. However, both formats could be upgraded to provide picture quality suitable for industrial and entertainment applications.

H.261 is a DCT based interframe standard developed for the video-conferencing industry. While it is a low quality based algorithm structure, it is also a variable bit rate compatible standard. This is the only video compression standard that defines a compatible format, and in addition allows for multiple bit rates to be used. These rates vary from 56Kbs to 2.0Mbs. While this is serious compression, the picture content and quality requirements of the video-conferencing industry make the results perfectly acceptable.

None of the above standards were developed for broadcast video applications. There are DS-3 (45 Mbs) broadcast quality codecs in use today, but no standard has been developed to provide any incentive for inter-product compatibility.

Table 2 - Speculative Quality Comparison of Future Digital Products

BIT RATE	4:2:2 COMPRESSION	APPROXIMATE PRESENT QUALITY EQUIVALENT
1.5 Mb	180:1	VHS Tape Viewed in NTSC
3.0 Mb	90:1	U-Matic Tape Viewed in NTSC
6.0 Mb	45:1	AM-VSB Transmission
12 Mb	23:1	S-VHS/Hi-8 Tape Viewed in NTSC
23 Mb	12:1	NTSC
45 Mb	6:1	Beta-SP/M-II & D-2/D-3 Tape
90 Mb	3:1	D-1 Tape

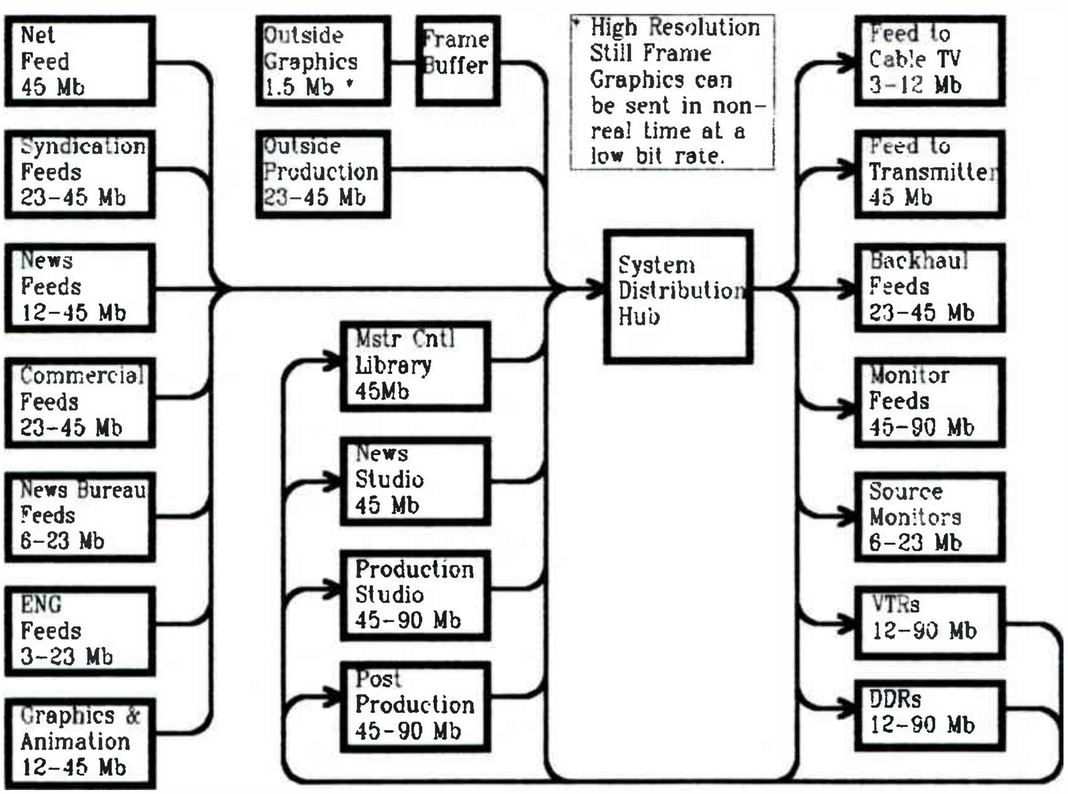


Figure 5 - Possible Digital Broadcast Facility

Applications

Digital video compression is still an emerging technology. Specific information regarding applications of various compression ratios and methods is, at present, an exercise in prophecy. Table 2 and Figure 5 are given under the First Amendment right of personal speculation.

Table 2 is an attempt to forecast a quality comparison of present analog and digital formats with possible future compressed digital video data rates.

Figure 5 is a speculative look at how a broadcast facility might be impacted by compressed digital video products and services.

The present implementation of video compression technology exhibits noticeable image quality reduction. However, the rapid pace of continued improvement in this area promises a short wait for products and services of acceptable quality and value.

CONCLUSION

The technology is definitely in place to provide an avalanche of compressed digital video products and services into the broadcast marketplace. It is possible to provide quality pictures with digital video compression. The next frontier is to provide a variable bit rate compatible video compression standard that will allow all of the various broadcast video application segments to be able to communicate directly with each other.

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SBE

BROADCASTER'S RULES OF THE ROAD

Tuesday, April 14, 1992

Moderator:
Dane Ericksen, P.E., Hammett & Edison, Inc.,
San Francisco, California

***FCC ENFORCEMENT EFFORTS: NOT BUSINESS AS USUAL ANYMORE**
Richard Smith
Chief, Field Operations Bureau
Federal Communications Commission
Washington, District of Columbia

THE FCC AM BROADCAST SELF-INSPECTION PROGRAM
James R. Zoulek
Federal Communications Commission
Cerritos, California

***CHANGES IN STRUCTURAL STANDARDS FOR COMMUNICATIONS TOWERS**
John Windle
Stainless, Inc.
North Wales, Pennsylvania

EMERGING TRENDS FOR BROADCAST AUXILIARY
Richard A. Rudman
Chair, SBE National Frequency Coordinating Committee
Indianapolis, Indiana

A HIGH SPEED DIGITAL SOLUTION TO THE EMERGENCY BROADCAST SYSTEM
Gerald M. LeBow
Sage Alerting Systems, Inc.
Stamford, Connecticut

***THE FUTURE FOR EBS**
William F. Ruck, Jr.
KFOG/KNBR Radio
San Francisco, California

***WARC-92: WHAT IS IT AND WHY SHOULD I CARE?**
John Reiser
Federal Communications Commission
Washington, District of Columbia

Roundtable Discussion: Dealing with a Changing World
All presenters

*Paper not available at the time of publication.

THE FCC AM BROADCAST SELF-INSPECTION PROGRAM

James R. Zoulek
Federal Communications Commission
Cerritos, California

The views expressed are those of the author and do not necessarily reflect the views of the Commission.

Abstract- Faced with the reality of dwindling resources, the FCC has begun to seek ways of improving compliance in the broadcast services by methods other than the traditional field inspection. The AM Broadcast Self-inspection program is one such method being tested. The program allows the broadcaster to inspect his station and then report the findings to the FCC. The hope is that, given an opportunity to pre-inspect his station, the broadcaster will reduce the likelihood of violations found during the traditional random inspection.

I. Purpose of the Program

The Broadcast Self-inspection Program was conceived to accomplish two important goals. First and foremost, the program seeks to establish and maintain the highest level of compliance possible in the broadcast services.

This program seeks not only to increase compliance but also to educate licensees about the current FCC Rules and Regulations. Dramatic changes in 47 CFR 73 over the last decade have left stations unsure of what the FCC currently looks for during a station inspection. This program identifies those expectations in hopes of attaining 100% compliance.

In addition, the self-inspection format, which allows the broadcaster to conduct the inspection, was developed to be "user friendly" in the broadcast industry while conserving precious government resources. In this way, the broadcaster, the Commission and the general public will benefit from the program.

II. Description of the Program

The experimental stage of this program was conducted in the FCC's San Francisco Region during the Spring of 1991. Nine AM stations were approached with the idea of voluntarily participating in the experiment and all agreed.

Each participating station was mailed two copies of the Broadcast Self-inspection (Report Booklet). One booklet was to be completed and returned within 20 days to the FCC Los Angeles Office. After the completed booklets were returned, a follow-up questionnaire was sent to each participant requesting comments and tabulations of the time spent completing the self-inspection.

In addition, unannounced visits were made to two participating stations. The purpose of the visits was to verify the information reported by the station and to personally discuss with the stations staff the merits and faults of the program.

III. Findings

The returned Report Booklets were carefully reviewed for compliance and verification. A high level of compliance was indicated even though some discrepancies were noted by the participants. Those stations that discovered areas of non-compliance reported them and either took immediate action or promised to comply.

During the unannounced follow-up visits, a further check was made for discrepancies in the reported information. No such discrepancies were found in the information that could be verified. Comments and suggestions on the particulars of the program were sought and received. Among the most popular of the comments was the need for a format which would allow easier access to reporting pages so that they could be removed from the booklet and typed responses could be made. Many suggestions for clarification of questions were made as well.

Above all, however, the project was well-accepted by all participants. The news of this program spread quickly and dozens of requests for sample Report Booklets have been received. Most of those receiving samples expressed their gratitude for revealing FCC expectations for broadcasters as a result of this program.

IV. The Present Time

Given the number and quality of suggestions arising from the experimental program, a revision of the Report Booklet was in order. Nearly all the suggestions received were incorporated into the revised version. The most significant change was to the format for reporting information from the self-inspection. The Report Booklet has been scaled down and the new version will include separate reporting pages. Only the reporting pages will be returned for verification by the FCC.

Several minor changes were made to clarify ambiguities in the Report Booklet's questions. These also reflect valid criticisms made by the participants in the experimental stage. Such improvements will benefit the broadcasters and the Commission.

The Commission is now reviewing whether to proceed with the program on a full scale basis.

EMERGING TRENDS FOR BROADCAST AUXILIARY

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Abstract- Broadcast Auxiliary Service (BAS) bands are highly congested in many markets. Local volunteer groups, many under SBE sponsorship, have tried to make the best of a situation that is growing steadily worse. More stations are using Remote Pickup to bypass their local telephone company. More stations are using airborne traffic reporting. Yet, some manufacturers and distributors still market solid state transmitters without proper output filtering. These and other trends have set the stage for unprecedented levels of interference. The FCC has not only been reluctant to increase exclusive BAS spectrum, but has now set in motion a plan to reallocate the 2 GHz band for new non-broadcast technologies. This paper will outline how this spectral resource can be conserved through cooperation and education, outline consequences of current FCC actions, and attempt to predict the future of BAS.

INTRODUCTION

The SBE's National Frequency Coordination Committee is aware that the local coordination process has not gone smoothly of late; especially in m[illegible]s where Part 74 use is heavy. Several reasons for this will be discussed:

- Widespread misunderstanding of the process
- More users than channels
- Excessive transmitter power levels
- "Line-of-sight" interference
- System technical deficiencies
- Illegal operation
- Lack of cooperation
- International interference

The FCC announced in ET Docket 92-9 that it wishes to reallocate BAS channels between 1.85 GHz and 2.20 GHz for "emerging communications technologies". This paper will also outline the serious consequences of this effort as viewed by the All-Industry National Frequency Coordination Council. It will then review recent FCC Rules changes and industry trends that are important to achieve a thorough understanding of the present and future Part 74 world.

Widespread Misunderstanding of the Process

The underlying premise of BAS coordination is often misunderstood. Local groups are not "Frequency Police". They do not assign frequencies. Contrary to what some people think, the FCC still has these responsibilities under the Communications Act.

Local coordination groups are the custodians of accurate database information and facilitators of the coordination process. Potential licensees are responsible for the actual contact and coordination with established co and adjacent channel licensees. Once parties agree that a proposed use will work, the local coordination entity updates the database with this information as a "proposed" use and awaits notification of final FCC disposition of the licensee's application. If agreement cannot be reached, the local coordination entity acts as the first level mediator. The local group strives to be impartial.

The Southern California Frequency Coordinating Committee, Inc. (SCFCC) created a statement that appears on its newsletter masthead and on key correspondence that addresses what it feels is a serious educational issue regarding Part 74 coordination. The SBE's National Frequency Coordination Committee has endorsed this statement after review by its legal counsel:

> *The SCFCC does not assign frequencies. Our purpose is to facilitate licensee to licensee communications. The Committee will recommend possible frequencies for compatible usage and provide a list of existing licensed users on co and adjacent channels to the proposed channel. It is the responsibility of the new user to ascertain that its usage will not cause problems to existing systems. The Committee will attempt to mediate any conflicts that cannot be solved between licensees. It is the goal of the SCFCC to accommodate all legal users.*

The SCFCC's Board of Directors recently met to draft a clear "Mission Statement" that is now also printed in its monthly newsletter, to further clarify what its role is (and is not) regarding the BAS coordination process:

SCFCC Mission Statement

❑ *To provide database information leading to coordination for all legally authorized users of Part 74 and other shared spectrum*
❑ *To provide assistance to members in efficient spectrum utilization*
❑ *To facilitate and mediate differences between members*
❑ *It is the goal of the SCFCC, Inc. to accommodate all legal users*

These statements should be the core of an education campaign to alleviate the current high level of misunderstanding concerning the BAS coordination process.

More Users Than Channels
Early BAS spectrum users licensed as many channels as they needed. Some of these users managed to retain these channels despite today's congested conditions. Many local coordination groups are now wrestling with this issue as they are approached by new users who are sometimes surprised when even simplex channels are not readily available. There are alternatives that local groups have suggested to both new and existing licensees to help alleviate this problem:

❑ **Dispatch Options**
Pure dispatch communications that are not intended for rebroadcast can be licensed in the Land Mobile band. Some users have had good luck with trunked systems employing the latest technology. These new systems work much better than the trunked radios that were tried by some broadcasters ten years ago. Trunked systems have the added advantage of being harder for the competition to tune in on scanners.
❑ **Cellular Telephones**
Some stations use cellular telephones in place of RPU's. Frequency extenders with multi-line cellular phones achieve excellent transmission quality. Caution: Cellular transmissions from aircraft are illegal.
❑ **Time Sharing**
Coordination groups should suggest this option when compatible sharing partners can be found. Much like the "Home Channel Plan" pioneered for the 1984 Olympics in Los Angeles for the 2 GHz band, and now in use in a growing number of markets. Even some airborne traffic services have experimented with RPU channel sharing.
❑ **Channel Re-use**
RPU channels can often be re-used since directional antennas, antenna polarity, terrain, and distance can be coordinated.

Line of Sight Interference
VHF, UHF and microwave signals travel great distances under line of sight conditions. It has long been good practice to take geography and topography into consideration when a coordination group constructs or revises their region's band plan. With more users, and uses like airborne traffic reporting, interference has caused the loss of program elements, endless hours of interference tracking, as well as chronic loss of sense of humor. Interference is still on the rise.

Excessive Transmitter Power
We must accept a challenge to cut back power for Part 74 operations. The challenge is in the same spirit as that accepted by "nuclear club" nations to pull back from the brink of mutually assured destruction. We must start using only enough transmitter power for reliable communication. Section 74.461(b) states "the authorized transmitter power for a remote pickup broadcast station shall be limited to that necessary for satisfactory service..."

The author recently sent a letter to the Chairman of the Southern California Frequency Coordination Committee:

> *...is time for Part 74 licensees in Southern California using the 450 band to pay more than lip service to the technical and operational realities of our region. Line of sight conditions from major repeater sites must be taken into account if our band plan is to continue to work. There has been a significant rise in airborne operations that can tie up a channel from the Mexican border to well north of Mt. Wilson. Finally, we have more users than available channels, and more are on the way.*
>
> *Some results of our failure to fully acknowledge these realities in our region are:*
> • *Many licensees use excessive power for repeater transmitters on mountain tops.*
> • *The current FCC 15 watt limit for airborne RPU is excessive and is violated in some cases.*
> • *IFB transmitters sometimes exceed the 2.5 watt FCC limit for "vehicular repeaters" in 74.431 (e).*
>
> *We suggest a voluntary "roll back" for repeater transmitter power to levels adequate to maintain proper communications. Our initial proposal is that a 30 watt power level is more than adequate to cover operations from repeaters located on Mt. Lukens, Mt. Wilson, Saddle Peak, Santiago Peak, and Rolling Hills. We further propose all 450 band airborne activity be conducted at a power level not to exceed five watts. Section 74.461 (b) specifies a 15 watt limit that is unrealistic for our region...*

We also suggest that TV IFB operations be conducted at power levels not to exceed one watt.

If we all agree to turn down repeater power to reasonable levels, lower airborne transmitter power to 5 watts or less, and lower power on overpowered vehicular transmitters, including those used for cuing by TV stations, the world will become a better environment for overworked receivers that are unable to cope with overpowered co and adjacent channel systems using excessive power.

The author's station has already:

- ❑ lowered power at all four of our hilltop repeaters to 30 watts antenna input power
- ❑ taken over licensing of the UHF channel used for our traffic service reports. All airborne transmitters are now set to 5 watts maximum transmitter power output (TPO)
- ❑ begun experiments using less power for these systems and for our mobile units.

Will systems work better if we do this? The answer depends on all of us. We might be able to reduce airborne power ultimately to one watt at the antenna and still have greater than 99.99% reliability. Airborne is the best kind of line of sight transmission.

Land Mobile systems operating between 451-455 Mhz, and from 456 to 470 MHz, could do well to follow our example. Operators of paging transmitters below the 455- 456 MHz Part 74 channels say they need high power (typically 250 watts or more ERP) for building penetration.

A recent survey of several New York region two-way sites atop tall buildings uncovered many improperly installed Land Mobile band base stations. These have helped raise the noise floor and intermodulation levels to the point that these sites are almost useless as receiver locations. We must demand that all services comply with standards of good engineering practice.

System Technical Deficiencies

Solid state transmitters, regardless of class of service, must be installed with ferrite circulators with reject loads and cavity filters. Some manufacturers and distributors still market systems without them. Without a properly installed ferrite circulator and reject load, RF from other systems can enter non-linear solid state finals through the transmitting antenna. Mixing products appear. Such a system is a ticking time bomb for potential "intermod".

Cavity filters help receivers deal with out-of-channel signals. A popular three cavity filter can attenuate a signal 1 MHz from the desired channel by 50 dB with only three dB insertion loss. Insertion loss can be made up (often with net gain) by using a properly installed low noise gasFET preamplifier.

We should demand that all high power Land Mobile radio systems install transmitter output filters that exceed the current rule that emissions more than 250 KHz from the carrier be attenuated $43 + 10\log_{10}$(mean transmitter power in watts) dB. The reduction in wide band noise for our 455 - 456 MHz channel group would promote at least three highly desirable effects:

- ❑ reduction of receiver desensitization
- ❑ reduction of receiver overloading
- ❑ reduction of intermodulation products

Part 74 systems are not immune from design and installation shortcomings. Some broadcast engineers do not have enough training in two-way radio engineering, or gained their experience when there were fewer systems to interfere with. Our industry must subscribe to the same standards we demand of other radio services.

Another technique quickly gaining favor is employing additional receivers located at "quiet" receive sites away from "noisy" repeater sites. The best so-called "satellite receiver" sites have no transmitters at all. The downside aspects of this approach are:

- ❑ additional site rental costs
- ❑ backhaul charges (either wireless or wired)
- ❑ additional installation / maintenance costs

The liabilities must be weighed against the benefits:

- ❑ No receiver desensitization if the site is selected carefully
- ❑ Enhanced system reliability and redundancy
- ❑ Enhanced performance from low power portables, hand-helds, and airborne transmitters
- ❑ Better immunity from local oscillator radiation and "flying transmitters"; common problems at popular hilltop and high-rise radio sites

Illegal Operation

Many broadcast engineers are aware of parties who openly flaunt FCC Part 74 rules:

- ❑ A recent check of two airborne transmitters in the same market showed one running almost twice legal power in violation of Section 74.461(b) and the other deviating in excess of 5 KHz in a 25 KHz channel, in violation of Sections 74.462 and 74.463(a)

- Some broadcasters employ non-type accepted equipment including RF power amplifiers and illegally modified amateur radio gear.
- Frequency stability measurements and other routine preventive maintenance necessary for proper compliance with FCC Rules are frequently ignored, in violation of Sections 74.464 and 74.465
- Remote pickup stations rarely comply with Section 74.482 regarding station identification, so necessary to interference tracking
- Willful and malicious interference takes place in a number of markets from various sources
- Serious violations of the FCC's power, deviation, and frequency tolerance rules occur frequently in bands adjacent to those covered by Section 74, including the 440 MHz amateur band
- Some Part 73 licensees and non-broadcast entities fail to coordinate properly, supply wrong information on 313 forms, or conduct "bootleg" operations without any attempt at licensing.

All of these illustrations are drawn directly from the author's personal knowledge. These instances of illegal operation lead to a conclusion that something must be done to increase compliance with Part 74 Rules. We can start now by using peer pressure to encourage compliance. If this approach does not work, we must seek more stringent enforcement of FCC rules so legal BAS operations can take place with higher reliability. It's a jungle out there.

Lack of cooperation

Most broadcast engineers have a high degree of respect for both the Rules of the FCC and the laws of physics. This appreciation does not extend to some non-technical people in our industry who have decision making power over BAS operations. This is not to say that all general managers and news directors look upon coordination as a liability. However, there are enough who do resent or fail to take the time to learn about our technical concerns to make our lives more difficult.

A basic tenet of management is that employees will pay attention to those things that are important to the boss. The opposite is true as well. The NAB, the SBE, and each of us need to make sure all of our managers have respect for all the rules that hold BAS operations together. Better cooperation based on clear management directives and clear management cooperation with coordination groups will lead to a reduction of interference.

International interference

Interference to Part 74 operations in the 450 MHz band along the Mexican border has become an issue within the last few years. The Mexican government has allocated 450 MHz spectrum for Land Mobile use, in conflict with the allocation in the U.S.A. Efforts are underway to address this issue through diplomatic and other channels. The SBE has even contacted its counterpart in Mexico, AMITRA, to seek help in creating a dialog. Unless these efforts bear fruit, this source of interference will continue and probably get worse.

Evolution of the Home Channel Plan

The band of choice for TV ENG is 2 GHz. While Ku band satellite may be the most direct way to bring programming from the field to the studio, 2 GHz terrestrial microwave still offers the most cost effective overall transmission system for day-to-day ENG operations. The 2 GHz band offers significantly longer path lengths than other BAS bands. It also supports the need to "bounce" signals off of buildings and shoot through foliage to achieve transmission paths from difficult locations in city "canyons".

The 1984 Olympic Games held in Los Angeles became the vehicle that forced many broadcasters to seriously consider ways to share the 2 GHz TV Auxiliary channels. The ABC Network, as the "Host Broadcaster", proposed what became known as **The Home Channel Plan.** This plan allowed ABC to cover a large number of sports venues while allowing almost normal news operations to continue for the rest of the market. All Los Angeles TV broadcasters subsequently agreed to this plan for day-to-day operations.

The plan is simple. Each TV entity operates on what is called a "Home Channel". If their operations require using another channel to avert interference, or for a special feed, a request is made of the entity who normally uses that channel. The agreement exists for the duration of the special feed. Each user agrees to operate "narrow band" and to use the 4.8 MHz audio subcarrier to further reduce occupied bandwidth.

A list of telephone numbers for ENG control points is published along with the "Home Channel" assignments to make real-time coordination possible. A special radio system also links the ENG control points of several stations.

Many technical and operational advances have helped make these plans work by cutting down the number of times a station has to ask for another channel:

- Multiple ENG receive sites make channel reuse much easier
- New 2 GHz TV ENG transmitters allow reduced bandwidth and TPO power control
- Highly selective receiver filters and better designed receivers improve interference rejection.
- Highly directive "Silhouette" type antennas that replace inefficient "rod" antennas on ENG trucks dramatically reduce adjacent channel interference

- More trucks are now equipped with tall pneumatic masts. These help get signals through under adverse conditions.

Home Channel Plans are being implemented in more markets according to information gathered by the SBE. The two chief reasons given for this are more users than available channels and a high level of network activity that is in conflict with local news operations.

FCC regulatory issues concerning TV ENG

The FCC announced almost ten years ago a Notice of Proposed Rulemaking (NPRM) that resulted in adoption of Docket 82-334. Broadcasters were successful during the Comment and Reply period in making the FCC understand that Category A antennas and path length restrictions proposed in the NPRM were not compatible with real-world TV ENG operation. The SBE, NAB, group broadcasters and some individual stations worked hard to explain the critical "back stage" support relationship of TV ENG to the "on stage" news, sports, and special programs. TV news, as we now know it, could not take place in a cost effective and timely way if antenna size and path restrictions had been placed on the 2 GHz band.

Several coordination groups had already begun working with licensees to move fixed links out of the 2 GHz band to 7 GHz and 13 GHz. This strategy reduces risk to STL's from mobile operations and makes more 2 GHz channels available for more day-to-day users. Technical enhancements mentioned above were being rapidly introduced. Peer pressure forced many TV news operations to invest in costly but necessary receive site upgrades and new equipment. Although the net effect raised 2 GHz TV ENG reliability, it dramatically increased dependence of broadcasters on 2 GHz TV ENG transmission as a key element of their day-to-day operations.

ET Docket 92-9 and beyond

One year ago, the FCC announced its intent to make spectrum available for "emerging technologies" such as wireless local area networks (LAN's) and so-called personal radio services. They identified the 2 GHz spectrum now in use for TV ENG, Public Safety, and Common Carrier as the home for these new services.

The Steering Committee for the National Frequency Coordinating Council (NFCC) has agreed to the following facts, positions and conclusions regarding the FCC's efforts to displace Broadcast Auxiliary users from the 2 GHz band. The current Council members are the Society of Broadcast Engineers (Secretariat), the National Association of Broadcasters, ABC/Capitol Cities, CBS, NBC, Group W, Turner Broadcasting, and the National Cable Television Association. The NFCC exists to evaluate FCC policy regarding Part 74 and Cable Antenna Relay Service (CARS) spectrum, facilitate the coordination process for network entities, and act as an all-industry contact point for the FCC.

Because of the perceived impact of ET Docket 92-9 on TV Auxiliary operations in the future, the current conclusions of the NFCC, as of mid-January, 1992, are presented in this paper.

Are There Realistic Alternatives to 2 GHz?

The capacity of Channels A1 - A9 has been exceeded in major markets. Channel A10 has already been reallocated by the FCC, adding more congestion to operations in several markets. Sharing with Public Safety and Common Carrier licensees had already been in place in the 2.5 GHz band; making this spectrum non-exclusive for BAS.

Real-time sharing and coordination for 2 GHz, as pioneered during the 1984 Los Angeles Olympics, is becoming a way of life in more and more markets. Delivery of news programming from the field to the studio in most markets is utterly dependent on 2 GHz. Spectrum efficiency innovations such as split channel operation, enhanced receiver filtering, multiple receive points and Home Channel Plans for ENG have mitigated channel shortages in markets where they are in use.

There is a great danger that moving BAS ENG to any band shared with non-broadcasters will not work for either party. Real-time sharing depends on linking microwave control points so operators can perform "instant coordination". Methods to accomplish "instant coordination" with other services are theoretically possible but do not yet exist. Forcing such shared use would be perceived by our industry as a tactic that would seriously compromise public access to breaking news events and to national sports coverage.

The alternative of broadcasters using the existing 7 GHz band is not practical. The industry has already moved most fixed links out of the 2 GHz to the 7 GHz band in major markets over the last ten years to avoid the serious problem of conflicts between fixed and mobile operations.

Replacing wireless links with fiber or cable as the FCC has suggested on several occasions does not begin to address the issue of "instant" remotes; the very essence of local television news operations. Further, the NFCC does not understand how cable or fiber could be of serious value to our industry for any but a few very narrow uses. Even these narrow uses could have a deleterious effect on operational reliability. Reliance on cable or fiber, even for fixed links, leaves systems at greater risk from sabotage and infrastructure failures, not to mention loss of control of the transmission system . Just one example of the negative consequences of the FCC's plan is that the vital earthquake recovery information role of television in California would be crippled.

Current broadcast industry investment in multiple receive sites and electronics, ENG trucks, and other equipment related to the 2 GHz band, is estimated to exceed 75 million dollars according to a recent estimate by the NFCC. Aside from dollars, relocation to a higher frequency microwave band would require more intermediate relay points (with their attendant added microwave channel use for backhaul) due to shorter path "reach", and more receive sites with ongoing rental costs. The NFCC estimates that migration to a new, shared band would cost at least another 75 million, and not work nearly as well as the systems now in place.

Timely news coverage in certain downtown and deep fringe areas would become impossible, causing a return to the "film at eleven" era; a giant step backward from immediate news coverage. The NFCC is investigating the chilling effect of this decision would have on the entire news gathering process. This issue clearly crosses purely technical lines.

FCC NPRM 91-337 seeks comment on policies and rules for ATV service in the U.S.A. Section C deals with Broadcast Auxiliary Services. The NFCC takes issue with the following points in Section C.33 of the NRPM:

The FCC believes that BAS spectrum for ATV will be "limited". We believe it is not just limited, but would become almost extinct should ET Docket 92-9 become a reality. The NFCC believes ATV transmissions will not be compatible with existing band users and will present a serious impediment to the Industry's coordination effort. Successful ATV operations may not be possible without debilitating coordination problems, even in markets well below the top 50.

While the FCC does not mention Ku band satellite transmission to alleviate BAS spectrum shortage, we believe that cost, coordination problems, and limited space segment availability are all factors that do not make it a viable and cost-effective alternative for most stations.

BAS coordination rests on all users not exceeding a stated maximum occupied bandwidth. The FCC presumes that a BAS channel carrying an ATV signal would not exceed this maximum value. The NFCC believes this assumption is not technically sound. It postulates use of compression technology that does not yet exist.

2 GHz Band conclusions and actions

I. The NFCC vigorously disagrees with the basic premise of ET Docket 92-9 to displace broadcasters from 2 GHz band. **This action offers no viable spectrum alternatives.**

II. The NFCC asks that additional spectrum be allocated for ENG and STL/ICR use to cope with ATV's new needs and alleviate spectrum congestion in major markets.

III. The NFCC is reviewing information obtained from local coordination groups such as the 2 GHz database of the Southern California Frequency Committee and the 13 GHz CARS database being assembled by the NCTA for data to support this position.

IV. The NFCC is strongly opposed to any attempt by the FCC to displace broadcast, public safety and common carrier licensees from the 2 GHz band until technically viable, reliable and cost-effective alternatives are found.

V. The NFCC does not understand how the unique propagation characteristics of the 2 GHz band that are essential to day-to-day ENG operations can be duplicated elsewhere in the spectrum within the framework of known technology.

VI. If ET Docket 92-9 unfolds according to plan, within 10 to 15 years the broadcast industry will have to hope for a series of technical and metaphysical miracles. Even FCC Commissioner Duggan in his separate statement on ET Docket 92-9 expressed his:

> *...strong concern that when there is any danger of displacing proven communications services in favor of unproven or speculative services, a heavy burden of proof rests upon us. I believe that the Commission must always demonstrate maximum sensitivity to the needs of incumbent users....*

VII. Commissioner Duggan's call for "ample transition periods" and "generous substitute spectrum" does not go far enough. The FCC must provide spectrum with similar propagation characteristics as well.

VIII. The inevitable comparison to the migration plan being developed for the AM band will be made by those interested in seeing broadcasters displaced from the 2 GHz band. The NFCCbelieves that the AM expansion band does not represent prime and desirable spectrum for most AM broadcasters for the very same reason that most Part 74 users are wary of moves to higher frequencies: The top part of the AM band has vastly inferior propagation characteristics than the rest of the band.

Category A Compliance

October 1, 1991 was supposed to be the date for compliance with Category A antenna standards for fixed links accepted for filing prior to October 1, 1981 in the 2, 7, 13, 18 and 31 GHz bands. This date was established in Docket 82-334, mentioned earlier in this paper. The FCC stepped backward from Category A compliance when some licensees complained that installation costs for the larger Category A antennas would be burdensome. The date for compliance with minimum Category B standards was delayed to April 1, 1992. Although most existing fixed links already comply with Category B, many coordination experts strongly recommend full "Cat A" compliance to achieve the highest level of spectrum utilization efficiency. The FCC reserved the right to insist on a "Cat A" antenna installation to correct an interference complaint upon a convincing showing by a affected party according to Section 74.641(b).

Some of the "weight" of this section is countered by 74.641 (d), which says the Commission may approve any antenna system where a "persuasive showing" is made that indicates "in detail why an antenna system complying with the requirements (above) cannot be installed", and "includes a statement that frequency coordination as required in 74.604(a) was accomplished".

Category A antennas for the 7 GHz band have a beam width of 1.5° degrees or less compared to Category B antennas which have a beam width up to 2.0°. Maximum beam widths for the 13 GHz band are 1.0 ° and 2.0° respectively. The greater side lobe radiation suppression of Category A antennas and narrower beam width are both excellent tools to assist designers of fixed link systems achieve greater system integrity in congested markets. A complete table showing all compliance factors appears in Section 74.641(a)(1) of the FCC's Rules.

Category A compliance in the 2 GHz band is moot in congested markets for two reasons:

- ❑ It does not apply to mobile transmitters used for TV ENG (Section 74.641(a)(5)
- ❑ The long established pattern to move fixed links <u>out of the 2 GHz band</u> in highly congested markets to avoid conflicts between fixed and mobile operations and pave the way for successful Home Channel Plans for TV ENG real-time coordination.

Section 74.641(a)(3) explains that the choice of receiver antennas is not regulated by the FCC. However, licensees who document interference from a properly designed system using "Cat B" antennas will not be "protected" if they use a receive antenna with "poorer performance than identified in the table in this section".

While the economic health of the broadcast industry has become much more a factor that everyone, including the FCC, should take into account, it is unfortunate <u>from a coordination standpoint</u> that the FCC took this action. If all fixed link transmit antennas comply with Category A standards, a higher level of channel sharing and protection for fixed links becomes possible. This will ultimately benefit all users.

<u>The Aural STL Band</u>

New users are placing added strain on the limited number of aural STL channels in congested markets. While not yet recognized officially by the FCC, a new class of temporary high quality aural service is being tested by several broadcasters on a waiver basis with the support of local coordination groups. High quality mono or stereo transmission for program length remotes cannot be accomplished in the 450 MHz UHF or VHF band Part 74 channels in many markets. Interference from Land Mobile sources (as outlined earlier in this paper) has brought the noise floor to a point that is dangerous to the health of remote broadcasts.

While the question of "Cat A" antennas for the Aural STL band was addressed by the FCC in 1991, with the FCC incredibly concluding that there was no need to extend minimum antenna standards to the extremely congested 2 GHz band, many coordination groups already recommend them in order to accommodate new users. In a few cases, a new user upgraded an adjacent channel licensee's older system at their own expense to get their new system on the air.

<u>The FCC's View of BAS Coordination</u>

Local Coordination is mentioned in Part 74 several times:

❑ **Section 74.24**

Section 74.24(g) calls for potential short term operators to "notify the appropriate frequency coordinating committee... Information on active (committees) may be obtained by contacting the FCC's Auxiliary Branch at 202-634-6307."

Unfortunately the last sentence in (g) negates much of the impact of this statement by stating "...this notification provision shall not apply where an unanticipated need for immediate short-term mobile station operation would render compliance with the provisions of this paragraph impractical."

The lesson here is that local committees should go the extra mile to provide 24 hour service. A 24 hour telephone or pager number should be listed with SBE so it can be included in the National Listing of all coordinators.

❑ **Form 313**

The November, 1989 FCC's new Form 313 ushered in a new era of awareness concerning coordination. Question 16 now asks if a coordinator has been contacted. The SBE believes that the heightened awareness of the coordination process among licensees resulting from this question is a very constructive step in the process to educate all of our industry about the necessity of prior coordination. Contacting a local coordinator prior to sending in a Form 313 is also a way for licensees to get added assurance that their new systems and modifications to older systems will neither be the cause or the victim of harmful interference.

❑ **Other References**

Part 74 makes indirect reference to coordination with several statements that licensees should make sure no harmful interference is caused to other licensees. The lesson for local coordinators here is to make the process as "user friendly" as possible. If licensees find the process is easy to with , they will want to use it before things go wrong.

The Future

Broadcast engineers who wish to remain in the discipline must be willing to accept change. Our motto might well be, *"Adapt or Perish"*, inscribed under a picture of a dinosaur with a line drawn through it. The changes to be wrought on our society by so-called "emerging technologies" (that may ultimately succeed in displacing 2 GHz ENG) must lead to (or force us to create) new ways to do remotes that we can only dream about now. If politics and economics do not make these alternatives too costly for us, Low Earth Orbit Satellites (LEO's), ISDN, advanced compression algorithms, spread spectrum technology, and other advances all might be important facets of future Broadcast Auxiliary operations.

Economics will play a key role as the future of BAS unfolds, whether we like it or not. The "downsizing" of radio and TV staffs and station budgets has not yet run its course. Overall U.S. economic health (or lack of health) is only part of the story. We should expect to hear the term "zero based budget" much more often as we try to improve existing systems or build new ones along the lines described in this paper. We will also have to provide more studies to show there will be a solid return on investment before dollars will be released.

The high cost of multiple receive sites for TV ENG, or even properly designed UHF RPU sites, may take some players out of the game or keep some new players from entering it. It will undoubtedly result in more "cheating" as some try to enter the game and not play by either the "rules" as outlined in this paper, or by the FCC's "Rules" with a capital "R".

Whatever the FCC says in their rules, more "Cat A" antennas for transmitters and receivers for fixed links from the 950 MHz aural STL band to the TV Auxiliary 31 GHz band are in our future if more users are to be accommodated through greater spectrum efficiency. Who will pay to upgrade older systems, or what can be done if an antenna structure will not support a heavier "Cat A" antenna needed to accommodate a new fixed link, are issues that will have to be solved on a case-by-case basis; hopefully without FCC involvement.

Politics and economics will continue to play important roles in the evolution (or revolution) of BAS policy. Members of the Commission are appointed by the Executive branch. We have every right to work within the system to challenge the FCC/Executive Branch strategy that some observers say is still driven by a marketplace rather than a technical philosophy. We should not forget that the Congress has oversight and may be sympathetic to a request for a hearing on this topic.

The FCC has made what many people in our industry view as serious technical mistakes based on *perceived* marketplace pressures such as:

❑ The AM stereo decision
❑ The DBS allocation that caused costly evictions from the 12 GHz band (and the resultant 13 GHz band sharing with CARS) and has so far shown what some see as scant benefit to the public
❑ The early attempt to promote Amplitude Compandored Sideband (ACSB) technology for two-way was an idea born inside the FCC that was not able to achieve manufacturer support

The NFCC has so far viewed the FCC's decision to evict BAS from 2 GHz without offering a viable alternative as a serious mistake, but one that they hope can be averted.

The SBE became very active during the 80's in frequency coordination. Many Chapters look upon their coordination efforts as yet another way to give something tangible back to our industry. SBE will to continue to support both Chapter level coordination and non-SBE local coordination groups. Assistance will continue to include:

❑ Some free assistance from SBE's legal counsel
❑ Updating the National Coordinator's List mailed quarterly to the FCC, networks, broadcast and cable coordination groups and interested individuals
❑ Support for the SBE compiled Database Program offered free to bona fide coordination entities
❑ Distribution of the SBE "Coordinator's Cookbook", a how-to manual
❑ Support of an Industry educational campaign on the coordination process
❑ Support for greater Industry recognition of the vital role played by volunteer local coordinators
❑ Liaison with other industry groups through the All-Industry Council

Postscript

If a comparison to President Bush's *1,000 Points of Light* is to be found broadcast engineering, it is in the ranks of the more than 115 volunteer frequency coordinators in the SBE's National Listing. It has been almost ten years since the FCC asked our industry to compile and submit to it a list of volunteer coordinators. Coordination is a thankless task at best, and an example of the adage *No Good Deed Goes Unpunished* at worst. The efforts of these volunteers are so much a part of day-to-day life that Part 74 activity would crumble of its own weight without them. Our industry owes these people a great debt of gratitude.

Coordination is an exercise in the art of the possible. If we hold this thought in mind for the future, we will find the resolve to prevail when confronted with the critical challenges we face.

Acknowledgments

The author wishes to thank in addition to all of the local volunteers we depend on, the following individuals:

- Mr. Dane Ericksen, P.E., of Hammett & Edison, Inc., Consulting Engineers, Chairman of the SBE's FCC Liaison Committee, for his work on behalf of our industry on numerous Part 74 issues and for his prompt and thorough assistance in reviewing and improving this paper.
- Mr. Paul Lentz of the SBE for his work compiling and updating the National Listing.
- Mr. Chris Imlay, SBE's Counsel, for ten years of Part 74 filings and hard work.
- Mr. Gerry Dalton, Chief Engineer of KKDA, who developed and maintains the SBE's Database Program.
- Messrs. Louis Liben (NBC) and Ken Brown (ABC) of the National Frequency Coordination Council for their help on this paper.

A HIGH SPEED DIGITAL SOLUTION TO THE EMERGENCY BROADCAST SYSTEM

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Abstract

The Present Emergency Broadcast System (EBS) has its origin in the 1950's when Harry Truman was President. The two-tone analog EBS technology, developed in the mid 70's, was designed to improve the dissemination of emergency information on broadcast facilities via the CPCS-1 daisy chain concept. Little has been done since the 1970's to augment or improve EBS technology or its operational procedures. The SAGE I System developed as the NATO Alerting System for World War III is a major step towards a more effective EBS System.

Introduction

In the last few years numerous failures of the EBS system can be identified. The reasons for these failures are rooted in the underlying technology and operational procedures of EBS. Hurricane Hugo, the San Fransico earthquakes, the Oakland fires, and, most recently, Hurricane Bob, were times when EBS failed to fulfill its mission.

New high speed digital data technology with error detection and error correction is now available to augment or replace the EBS System. This new technology was originally developed in Europe to deal with military threats. Known as SAGE I, or WARI in Europe, it provides rapid, secure highly selective alerting and warning notification to the public. by utilizing <u>all mass media - AM, FM, TV, cable and closed caption</u>. This technology is presently being proposed to the Federal Communications Commission as an augmentation or replacement for the existing EBS system.

Originally designed as a national system, EBS over the years has evolved into regional and local activations for severe weather, and alerts for chemical releases and nuclear power plant accidents. Though the day-to-day mission of EBS has changed, the technology and operational procedures have not.

Present Alerting Weaknesses

One of the major weakness of EBS is the use of the daisy chain concept. Under current rules certain radio stations are designated as CPCS-1, or Common Program Control Stations. They receive warning and requests for EBS actuation usually by telephone. These stations transmit the two tone EBS alert and broadcast the emergency message. Secondary stations should retransmit the information from the CPCS-1 and by doing so, create a daisy chain. The weakness here is the need for station operators, usually disk jockeys, to switch from providing entertainment to providing alerting and warning.In addition, if the primary CPCS-1 fails or opts not to transmit the alert, as happened with WCBS in New York during Hurricane Bob, the entire daisy chain concept collapses.

The need for an improved emergency alerting and warning system grows daily. With new weather forecasting techniques such as Doppler radar, it is possible to detect tornados minutes before their arrival. New earthquake sensing technology can detect earthquakes before they occur. Chemical and nuclear facilities are required to have sensors around their premises to quickly detect leakage of radioactive or toxic substances. Having created these new detection techniques, it is imperative that an alerting and warning system have the capability of delivering this information to the public in a rapid, accurate, effective, selective and reliable manner.

The typical life of a tornado is five to seven minutes. Under the best scenario, the actuation of the current EBS system can easily take ten minutes before alerts are transmitted. Chemical and radiation leaks such as Bhopal, Chernobyl, Charleston, Baytown, Three Mile Island, and others highlight the need for a new high speed alerting and warning technology.

History of Alerting in Europe

In the early 1980's, Western Europe was threatened by the possibility of military attack. The accepted scenario would have been a nerve and mustard gas attack by the East Germans on West Germany followed by a land and air assault into Western Europe. The NATO nations recognized that such an attack would be devastating within a very few minutes and that reaction time to a gas attack needed to be virtually instantaneous.

Numerous government agencies and companies studied the problem of public alerting and warning to determine the best way to create a common system for use by the NATO nations. A comprehensive network of broadcast stations already existed which could reach virtually every person with a very strong and reliable signal.

Studies were made of both AM and FM stations as primary alerting facilities. AM stations were found to be significantly inferior to FM as primary alerting and warning stations because of their susceptibility to propagation anomalies, such as sky wave and ground wave, co-channel and adjacent channel interference, their susceptibility to high noise levels during severe weather or nuclear attacks and their very limited data transmission capacity. FM radio stations were found to provide reliable and stable coverage over a defined area, and were much less prone to interference because of FM modulation and the capture effect. FM stations have the capability of transmitting relatively large amounts of digital data securely and transparently.

The RDS Connection

The Sage 1 System makes use of the RDS (Radio Data System) 57 kHz digital data subcarrier. In the early 1980's RDS was developed by the European Broadcasting Union (EBU), the British Broadcasting Corporation (BBC), and the Swedish Post Office (SPO) as a way of enhancing the use of radios in cars and homes. Extensive engineering and testing went into the design of the RDS digital data subcarrier which operates at approximately 1200 baud (1187.5 bits per second) with full error detection and error correction. RDS provides robust and reliable reception in a moving vehicle under adverse conditions of multi-path, low signal level, co-channel interference, and other signal corruptions.

After years of testing RDS technology was adopted and standardized by the European Broadcasting Union in 1986 and by Cenelec in 1990. RDS is now being standardized in North America by the RDS subcommittee of the National Radio System Committee (NRSC) sponsored by the Electronics Industries Association (EIA) and the National Association of Broadcasters (NAB).

Contained in the Cenelec, EBU and North American RDS Standards, is an alerting and warning system called EWS, Emergency Warning System. The SAGE I System uses a significant portion of the EWS transmission capability for emergency warning and alerting. For security reasons, details of the EWS are not printed in the Standards.

Within the SAGE I System, RDS carries command and control information to a variety of addressable and frequency agile emergency alerting and warning devices such as alphanumeric pagers, receivers in schools, homes and hospitals, hotels, and airports, electronic road signs, electronic voice capable sirens. It also carries control and closed circuit information to AM, FM, TV, cable and closed caption facilities.

EUROPEAN STANDARD — **EN 50067**
NORME EUROPÉENNE
EUROPÄISCHE NORM — December 1990

UDC 621.396 61:621.396 99RDS

Descriptors Broadcasting, sound broadcasting, data transmission, frequency modulation, message, specification

English version

Specification of the radio data system (RDS)

Spécifications du système de radiodiffusion de données (RDS)

Spezifikation des Radio-Daten-Systems (RDS)

CENELEC

European Committee for Electrotechnical Standardization
Comité Européen de Normalisation Electrotechnique
Europäisches Komitee für Elektrotechnische Normung

Central Secretariat: rue de Stassart 35, B - 1050 Brussels

UNITED STATES RDS STANDARD FOR BROADCAST DATA SYSTEMS NRSC DOCUMENT September 11, 1991

Descriptors: Broadcasting, sound broadcasting, data transmission, frequency modulation, message, specification

Specification of the radio broadcasting data system

The SAGE I System

In Emergencies, the SAGE I System uses all mass media to carry emergency alerting and warning information on their main audio and video channels. This capability is required to reach the hearing and hearing impaired public with emergency information on their existing car and home radios and televisions.

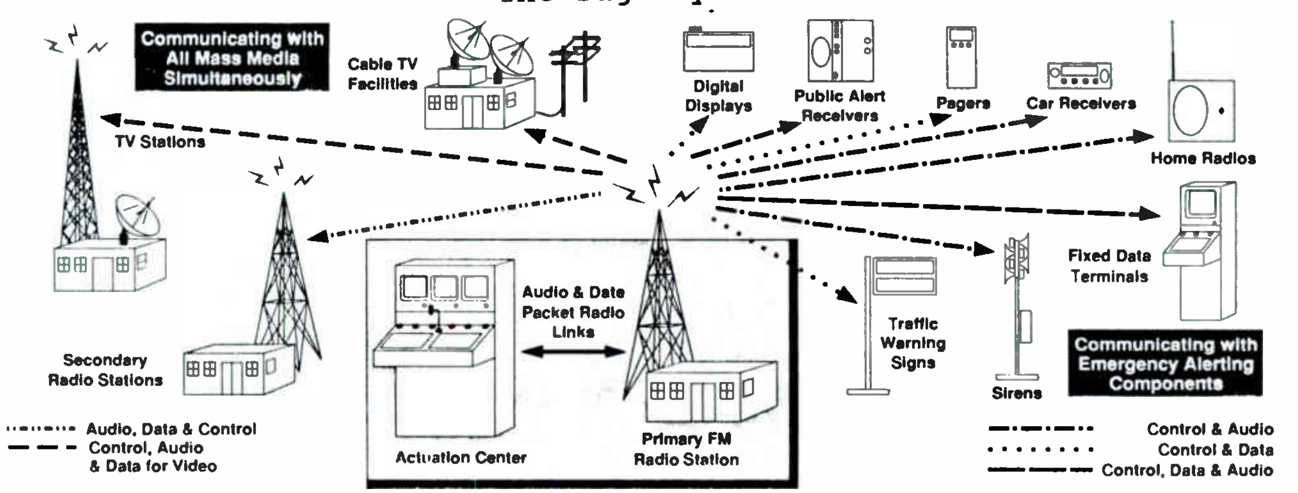
The SAGE I system departs from the single CPCS-1 station daisy chain concept to a multiple station concept whereby many stations are directly connected by radio links (both digital data and audio) to any number of actuation centers. If a primary alert station cannot or chooses not to carry the emergency message, the SAGE I System, automatically re-tunes all receivers to the back up stations within 6 seconds. This design concept would eliminates the possibility of another WCBS/Hurricane Bob situation.

Securely encrypted RDS-EWS data in the 9A and 1A groups addresses and activates specific receivers and devises in affected locations. The system can selectively "turn on" addressable receivers, activate sirens, put digital messages onto electronic road signs, and page emergency workers. The system provides digital data for full screen text transmission to TV and Cable facilities.

Required Equipment and Costs

At the primary radio station(s) an RDS encoder is installed between the stereo generator and the exciter or STL. The encoder controls the RDS data stream on the 57 kHz subcarrier. On stereo stations, the 57 kHz is phase and amplitude locked to the 19kHz pilot to ensure transparency and reliability even during severe multipath. Most encoders mix the baseband signal with the RDS signal and output a composite plus RDS signal at 1-10 volts peak to peak. These encoders carry digital information from the actuation center(s) which is sent on a radio link or RPU or other frequencies. In addition, audio from the actuation center(s) is received by the primary radio stations on a second radio

The Sage System



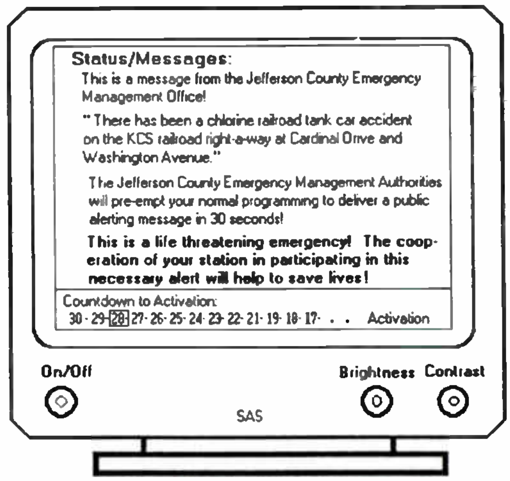
channel. For the most part, the audio messages are pre-stored on hard discs in the actuation center to ensure consistency of presentation, appropriateness of information, and that the message is read in a calm but forceful manner. No telephone connections exist in a SAGE I system. Stations can actuate the SAGE I System directly from their facilities.

RDS encoders cost between \$1,800-\$4,300 depending on features. Such encoders also give stations all of the other capabilities of RDS, such as display of call letters, alternate frequency, program type identification, clock time, radio text, etc.

Radio stations also have a digital display bringing closed circuit information from the actuation centers. These displays consist of an 80286 computer and monitor plus an RDS data receiver. This system should cost approximately \$1,300. The closed circuit messages indicate (1) the nature of the emergency, (2) the affected area, (3) the proposed response, (4) the remaining time to alert transmission.

Before and alert in a radio studio, master control of a TV station or at a cable headend, an operator would see a flashing closed circuit message indicating a "<u>tornado warning for Chicago in 30 seconds</u>." An attention getting sounder would also be activated. A countdown clock is started on the display showing the remaining time before the emergency message is to be transmitted. If an operator takes no action, the station's audio and/or video channels will be automatically brought into the emergency alerting network when the count down display indicates "zero."

The closed circuit data channel also serves as a continuing information stream to broadcasters providing updates on the status of the emergency.



If a station does not want to carry the emergency alert it can opt out by pushing the "inhibit" switch. Pushing the "inhibit" switch not only keeps the station out of the emergency alerting network, but it also sends a digital data message back to the emergency actuation centers indicating that the station is not carrying the emergency message. This way emergency management personnel have a clear indication of which stations are or are not carrying emergency messages. Pushing the "inhibit" button also sends a digital data code to the RDS encoder at that station so that the frequency agile alert receivers tuned to that station "switch" frequency to a back-up station which will carry the emergency message. The SAGE I system is the only frequency agile alerting system for reliability and redundancy. It does not rely on a single radio frequency but numerous stations on many frequencies at different locations to carry messages and alerts. If a receiver tunes away from a primary station, it periodically checks back to see if that station has returned to its normal status. If it has, the receiver returns to the primary station.

Comparison of In-Band and Out of-Band Alerting Systems

The Federal Communication Commission's Notice of Inquiry of June 26, 1991 describes two types of emergency alerting and warning systems, those which send control and data signals in-band (audible, and on the main audio channel) and out-of-band systems which operate transparently on subcarriers. The use of an in-band system imposes many limitations. First, transmission of data and signaling are audible to the public. The disruption and "tune out" caused by the existing EBS tones will continue to be a problem for all in band systems. The public has been desensitized to the Emergency Broadcast System by mandatory weekly tests. When listeners hear the two tone signals, or any data signal, their reaction is to reach for the tuning scan or seek button and find a station that is not broadcasting that signal. All in-band systems will have this problem as their testing will be audible.

Out-of-band systems continuously self test and are totally transparent to the listening public. The SAGE I System tests 10 times per second to ensure system integrity. The only time the public will hear any alerting tones is during a real emergency. This will ensure their recognition and response to every emergency message.

The data rate of in-band systems is quite limited. Systems such as WRSAME and ICEBS can transmit only a few hundred baud on the audio channel. This limits their ability to rapidly and selectively activate alerting and warning components of the system. In addition, in-band systems lack error detection and error correction. The only error checking system WRSAME incorporates is the transmitting of the same codes three time to see if they can be matched. By contrast, RDS incorporates a robust error detection and error correction system. Data bits that are lost because of propagation anomalies or reception problems such as going under a bridge can, for the most part, be fully corrected by the RDS system to produce at 100% accuracy. This is vital for an effective emergency alerting system.

The use of AM radio stations imposes other limitatins and problems as the primary alerting and warning facilities with in-band systems. During lightning storms and other electrical disturbances such as nuclear attack, aurora borealis, computers, TV synch and RF lighting, reception of audio or digital data on AM is virtually impossible. It is critical that an alerting system be able to operate effectively during severe weather such as thunder storms. Only FM stations as primary facilities using an out-of-band system can ensure accurate, reliable data and audio transmission.

Security is of the utmost importance for alerting and warning systems. Hackers, saboteurs, and others can easily record and decode in-band data transmission systems. It is relatively easy to use a digital audio tape (DAT) recorder to copy and rebroadcast transmissions from in-band systems. These can be played back to falsely activate sirens, and create alarms where none exist. Out-of-band encrypted digital data systems virtually eliminate the possibility of saboteurs and hackers corrupting the system. The SAGE I System uses a sophisticated encryption algorithm, as well as secure synchronization techniques to virtually eliminate the possibility of false or malicious triggering.

Many emergencies do not occur 7 A.M. to 10 P.M. when people are normally listening to radio or watching television. There is a need for an automatic system which will turn radio and TV's on any time of the day or night. It is unlikely that any consumer electronics manufacturer will build any in-band alerting system into their car or home radio exclusively for the purpose of emergency alerting. The perceived consumer benefit versus cost will not lead to in-band systems being incorporated into car and home radios.

RDS, however, provides many consumer features with a definite cost/benefit relationship. For this reason, RDS will be built into car and home radios at all price ranges including alarm clock radios. Once RDS is built in, you automatically have EWS on board.

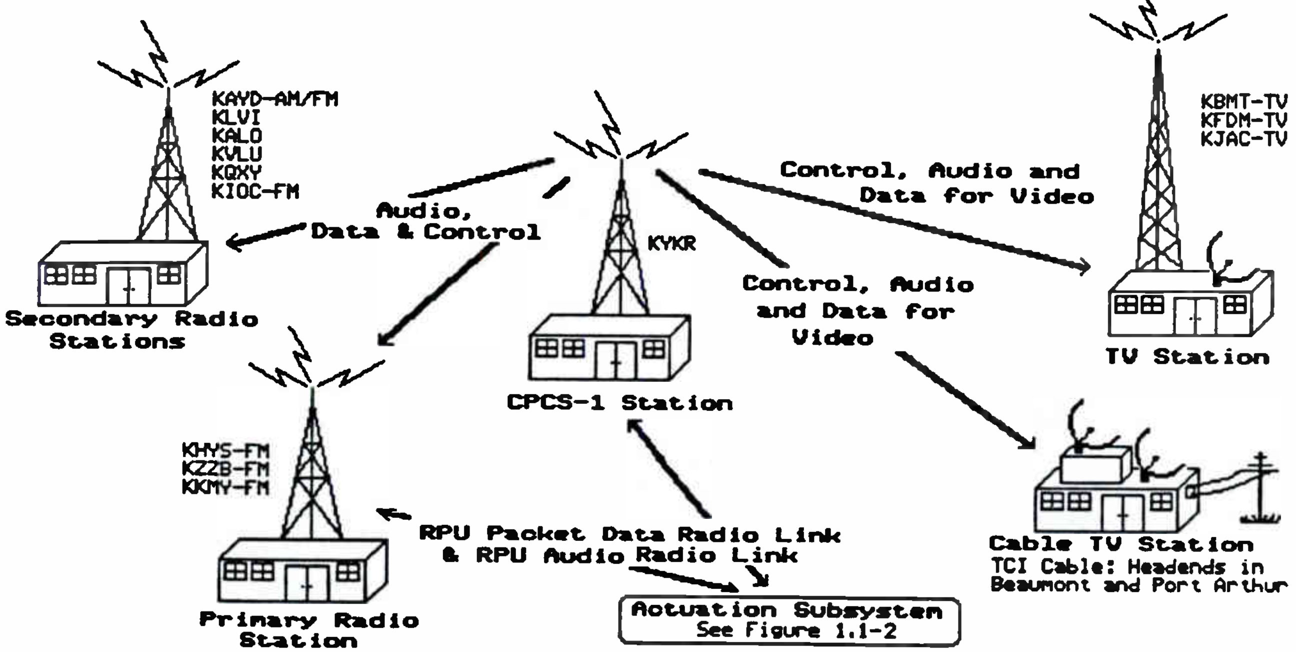
Already over 130 models of RDS equipped radios exist from such manufacturers as Sony, Pioneer, Ford, JVC, Blaupunkt, Alpine, Denon, etc.

Delco Electronics, which produces 7.5 million car radios per year, has produced an RDS model which can be turned on automatically by the SAGE I system even if the radio is switched off or listening to a cassette or compact disc. Similar U.S. models are expected from many other car and home receiver manufacturers in 1992.

The SAGE I System In The United States

The first fully implemented SAGE I System in the United States will be operational in Jefferson County, Texas (Port Arthur, and Beaumont) by September 1992. That system will incorporate 12 radio stations (7 FM & 5 AM) three TV stations (network affiliates) and two 41 channel cable head-ends. The system will automatically activate sirens, turn on receivers in schools, hotels, airports, hospitals, nursing homes, shopping stores, movie theaters, page emergency workers, and provide digital data communication between emergency medical service, police, fire, and the emergency operation center. The system will be operated by the Jefferson County Emergency Management Office and will have six actuation centers one of which is in a mobil van which can operate up to 60 miles outside of the County.

SAGE I Alerting System



The Jefferson County System

The FCC Notice of Inquiry on EBS

The Federal Communications Commission released a Notice of Inquiry about the Emergency Broadcast System on June 26, 1991 looking for an improved and automated EBS system to solve many of the shortcomings of the existing system. The Commission originally requested comments to be filed by December 31, 1991 however, it has extended that deadline to March 31, 1992. Reply comments are due by June 30, 1992. Comments should be sent to the FCC letting them know how broadcasters feel about the existing EBS system and which new technology will provide maximum safety and security while improving reliability and operational ease.

Comments should be sent to the Federal Communications Commission, 1919 M Street, Washington, DC 20554 RE: FO Docket 91-171.

BEFORE THE
FEDERAL COMMUNICATIONS COMMISSION
WASHINGTON, DC 20554

FCC 91-189
38189

13676

In the Matter of)
)
Inquiry into Possible)
Technical Improvements in) FO Docket NO. 91-171
the Emergency Broadcasting)
System)

NOTICE OF INQUIRY

Adopted: June 13, 1991; Released: June 26, 1991

Comment Date: December 31, 1991
Reply Comment Date: February 14, 1992

By the Commission

I. INTRODUCTION

1. This commences an inquiry into possible changes in the Emergency Broadcasting System (EBS), the basic structure of which was last revised in 1979[1]. We seek information in two basic areas: Whether there is a need for an updated automatic alerting system in EBS to notify the public of the existence of an emergency condition, and if so, what type of equipment is most appropriate for this application. The Commission also seeks information on applications of technology that might reduce net burdens on licensees in connection with our EBS Rules consistent with our objectives of ensuring reliable service to the public and meeting our statutory mandates with respect to national defense and "promoting safety of life and property through the use of ... radio communications".[2]

Conclusion

The need for an improved EBS System in the United States is apparent. An acceptable system should be:

1 - Operable with AM, FM, TV, Cable, Closed Caption.
2 - Self Testing,
3 - Uneffected by propogation anomalies,
4 - Rapid,
5 - Robust,
6 - Secure
7 - Selective,
8 - User Friendly,
9 - Transparent Except In Emergencies,
10 - Cost Effective,
11 - Built Into New Home and Car Radios,
12 - Frequency Agile,
13 - In Contact With Multiple CPCS Stations,
14 - Able To Turn "ON"-"OFF" Radios,
15 - Independent Of Telephone Circuits,
16 - Capable Of Activating Sirens, Pagers, Road Signs, Special Addressable Receivers.
17 - Battery activated.

The SAGE RDS System fulfills all of the above requirements, and should be the augmentation or the replacememt for the current EBS System.

AM AND FM IMPROVEMENT

Wednesday, April 15, 1992

Moderator:
William Ruck, KNBR-AM/KFOG-FM, San Francisco, California

***THE DENON/NAB SUPERRADIO**
Robert Heiblim
Denon USA
Parsippany, New Jersey

***FM TECHNICAL STUDY**
Karl Lahm
Lahm, Suffa and Cavell, Inc.,
Fairfax, Virginia

UPDATE ON RDS IN EUROPE
Dietmar Kopitz
European Broadcasting Union
Geneva, Switzerland

IMPROVING THE INTERMODULATION DISTORTION CHARACTERISTIC OF YOUR PRESENT AM TRANSMITTER
Timothy C. Cutforth, P.E.
Vir James Engineers
Denver, Colorado

IMPROVING FM MODULATION PERFORMANCE BY TUNING FOR SYMMETRICAL GROUP DELAY
Geoffrey N. Mendenhall, P.E.
Broadcast Electronics, Inc.
Quincy, Illinois

THE TOWERS INDUSTRIAL PARK PROJECT AT KTNQ
Ogden Prestholdt
Consultant
Nokomis, Florida
Ronald Rackley
du Treil, Lundin & Rackley, Inc.
Washington, District of Columbia

*Paper not available at the time of publication.

UPDATE ON RDS IN EUROPE

Dietmar Kopitz
European Broadcasting Union
Geneva, Switzerland

ABSTRACT

The aim of this presentation is to describe the development and implementation of the Radio Data System RDS for FM radio. This activity was coordinated by the European Broadcasting Union, a professional association of major broadcasting organizations with its headquarters in Geneva, Switzerland.

RDS started to be used in Europe in 1987 and reached the stage where it is supported by the large majority of receiver manufacturers. The development of the system has been largely completed, and the system is now also increasingly implemented outside Europe.

HISTORY OF MORE THAN 15 YEARS OF RDS DEVELOPMENT

The development of the Radio Data System started in Europe in the mid 1970's[1], and it was since then coordinated by the European Broadcasting Union (EBU). During the early development phase, broadcasters and manufacturers were contacted about their requirements, and before a unified European solution was finally adopted in 1983, several competing systems were tested. The system selected was based on the modulation parameters used in the Swedish PI - system which, at that time, was already in use in Sweden for a radio paging system called MBS. However, for RDS, the baseband coding parameters had to be completely redefined by the EBU expert group, which used the experience gained in testing the systems proposed under extreme multipath propagation conditions in the Alps, and mobile receiving conditions which were typical for car radios.

After the RDS specification[2] was published by the EBU in 1984, contacts with car radio manufacturers were intensified, and first measures were taken by the broadcasters to start RDS transmissions. In 1987, when Volvo presented the first RDS car radio at the IFA exhibit in Berlin, broadcasters in many European countries were ready for the RDS implementation. While the first RDS car radio was still priced at about $1200, prices already started to come down during 1988, and all major European car-radio manufacturers (Blaupunkt, Grunding, Philips) were then introducing RDS products on the market. Major Japanese car-radio manufacturers (Alpine, Clarion, Panasonic, Pioneer) were also ready with RDS receivers for the European market. For several years then, the average prices of the RDS radios remained relatively high ($800), and only during the IFA exhibition at Berlin in August 1991, it became clear that RDS had now finally reached the European mid-range price class of receivers ($350-500).

A few years later, after the EBU had already issued the RDS specification, CENELEC, the official European standardization body in the electrotechnical sector, started at the request of European car radio manufacturers, and with the support of the EBU, the transcription of the original specification into a European Standard. After formal voting by all the countries concerned, that standard was first issued in its final form in early 1991[3].

At that time, RDS was already largely implemented in Europe, practically in almost all West European countries. Also, outside Europe several presentations and on-air demonstrations of RDS had already been made. The CCIR had adopted RDS in the form of a Recommendation, as the data system to be used in FM broadcasting for identification and automated tuning purposes[4]. In the meantime, the EBU continued to promote RDS, especially by means of its RDS Newsletter, and also published Guidelines to assist broadcasters and manufacturers in the implementation of RDS[5]. The other broadcasting unions were also informed through the EBU about RDS, and at one of their World Conferences held in 1986 in Prague, RDS was recommended to be studied for implementation by broadcasters all over the world[6].

The EBU gave considerable assistance to broadcasters and their sister regional associations, such as the NAB in the US, to carry out such studies. However in a case, like the US, where the broadcasting practices differ significantly from Europe, certain adaptations of the specification to meet the particular requirements were necessary. As in Europe, the NAB preferred to undertake this standardization process jointly with the industry (EIA), and in 1992 the NRSC, dealing with this standardization task, will most likely present a proposal that should, on the one hand, meet the requirements identified, and also, on the other hand, maintain to a large extent compatibility with RDS as standardized in Europe, and recommended by the CCIR.

A REVIEW OF THE MOST INTERESTING RDS FUNCTIONS AND THEIR IMPLEMENTATION

The system's main objective is to offer listeners simplified tuning of receivers for FM broadcasting. This is achieved by displaying call letters or station slogans on an eight character display, the same that is normally used to display the frequency of the station, but now also able to give alphanumerical information. The basic idea is thus very simple: Listeners can "see" what they hear; but more is of course achieved, since RDS permits manufacturers to implement car radios that now can literally "surf" the radio waves. By this expression the following is meant: Most RDS radios have an RDS function-key "on/off". If RDS is "on", these radios speed up search tuning and can step tune from one RDS radio station to another, always displaying of course the respective call letters or station slogans, formally called in the RDS Specification the "Program service name".

In Europe, in contrast to the US, almost all major radio programs are broadcast over large transmitter networks so that they can be received over large areas which is particularly useful for the drivers on highways.

Then, RDS enables the listeners to remain tuned to the radio program on the network, because with the help of a Program identification (PI) and Alternate Frequencies (AFs), an RDS radio will automatically re-tune itself to that frequency which gives the best reception quality that would be technically feasible. For example, multipath distortions could be minimized, if the broadcaster has implemented low power transposers in areas where such a technique permits a better mobile reception quality to be achieved.

Users of highways also require traffic information, and there are many broadcasters in Europe that provide such a service. Formerly special traffic sign posts were installed on European highways to assist the user of a car radio to tune to the appropriate frequency.

RDS has now simplified tuning significantly in this respect, also because it includes the ARI functions that Germany already started to use around 1975. The ARI system is still in use in a few European countries. However, since RDS is fully compatible with ARI, and includes exactly the same functions (TP - Traffic Program identification and TA - Traffic Announcement identification), ARI will now be phased out and RDS will take over the same functionality.

Although RDS is a relatively slow data rate transmission system, it should not be overlooked that it was optimized for mobile reception in cars. It uses the 57 kHz sub-carrier within an FM multiplex signal. A differentially coded bi-phase data-signal is AM modulated on the 57 kHz sub-carrier with a data-rate of 1187.5 bit/s (57 kHz : 48), and the carrier is then suppressed. The overall spectral width is limited through appropriate filtering so that no components occur outside the range 57 ± 2.4 kHz.

The low signal level of RDS, and the spectrum range occupied, make RDS compatible with SCA, generally not used in Europe. The choice of the 57 kHz sub-carrier range is also the result of an optimization with respect to minimizing cross-talk of the data to the stereo channel, during mobile reception under multipath conditions. The recommended injection level for RDS signal is as low as ± 2 kHz. Only if radio paging is implemented would a higher value be necessary and typically ± 4.0 kHz would be required[7]. The criterion chosen for fixing the injection level is that of equivalent coverage of the data service and the stereophonic radio program. The upper limit tolerated by the CCIR[8], and also the receivers, is ± 7.5 kHz.

Other possibilities of functions that can be and, in fact, are already implemented in Europe with RDS are shown in Table 1[9]. These include Programme-type identification, Radiotext, Radio paging and a range of other possibilities.

Car radios on the European market (more than 100 than different models) have in the past implemented only PI, PS, AF, TP/TA, but the most recent models have increasingly also the EON (Enhanced other networks) feature, and some have also PTY.

All car manufacturers in Europe are increasingly in favor of equipping cars with RDS radios, and recently a new car model (GM/OPEL-ASTRA) was released where the PS display for the car radio is already part of the dashboard. This design makes use of the same display area that is also used for displaying the information from the car's board-computer.

RDS encoder equipment for broadcasters is also available from a large number of manufacturers in Europe. Prices differ in the range between $3000 and $12000, depending of whether these decoders would be used at local radio stations or within broadcasting networks where it is necessary that they be equipped with more complex data communication facilities. In the past, all manufacturers of these encoders implemented their own data communication protocols, but more recently the EBU has defined a new harmonized protocol[10] that several manufacturers are likely to use in addition to their own.

REFERENCES

1. K. Ilmonen (1972), "Transmission of additional Programme information signals for improving the broadcast listener satisfaction". Proceedings of the International Broadcasting Convention 1972, pp. 45-49.

2. EBU doc. Tech 3244 (March 1984), "Specifications of the radio-data system RDS for VHF/FM sound broadcasting", published by the European Broadcasting Union, Ancienne Route 17A, CH-1218 Grand-Saconnex, Switzerland.

3. European Standard EN 50067 (1990) "Specification of the radio data system (RDS)", published by CENELEC Central Secretariat, rue de Stassart 35, B-1050 Brussels, Belgium. This document is also available from the EBU.

4. CCIR-Recommendation 643-1 (1986) "System for automatic tuning and other applications in FM radio receivers for use with the pilot-tone system". Published by the ITU in Geneva, Switzerland.

5. EBU doc. Tech. 3260 (1990) "Guidelines for the implementation of the RDS system".

6. Sixth World Conference of Broadcasting Unions, Washington 1989, Recommendation WASH.T9. This is a re-enforcement of a similar recommendation adopted at the 5th Conference of Broadcasting Unions, held at Prague in 1986.

7. D. Frossard (1991), "RDS Radio paging in France". EBU Review (Technical) No. 245 pp. 22-28.

8. CCIR-Recommendation 450-1 (1982) "Transmission standards for FM sound broadcasting at VHF". Published by the ITU Geneva, Switzerland.

9. RDS Newsletter No. 12 (1991), published by the EBU, Geneva, Switzerland. This Table was also published by Radio World (International Edition) Vol. 15, No. 14, July 1991.

10. EBU doc. SPB 490 "Universal encoder communication protocol (RDS)". To be published in 1992.

RDS Features:	PI	PS	AF	TP[1]	TA[1]	PTY	DI	MS	PIN	EON	CT	RT	TDC[2]	IH[2]	RP	Subcarrier level
Austria - ORF	I	I	I	I	I	-	-	-	-	L	-	-	-	-	-	1.2 kHz
Belgium - BRT/RTBF	I	I	I	I	I	-	-	-	-	A	I	A	-	-	-	
Denmark - DR	I	I	I	I	L	-	-	-	-	L	-	-	-	-	-	
Finland - YLE	I	I	I	L	L	L	L	L	L	L	I	L	-	-	T	2 kHz
France - Radio France	I	I	I	I	I	-	-	-	-	A	P	-	-	I	P	4 kHz
Germany, FR - ARD	I	I	I	I	I	-	-	-	-	-	-	T	-	T	-	1.25 kHz
Ireland - RTE	I	I	I	I	I	T	-	-	L	A_{92}	I	T	-	I	I[4]	2.5 kHz
Italy - RAI	I	I	I	I	I	-	-	-	-	-	-	T	T	-	-	
Luxemburg - RTL	T	T	-	-	-	-	-	-	-	-	-	-	-	-	-	
Netherlands - NOS	I	I	I	I	I	T	-	-	-	-	-	T	-	-	-	
Norway - NRK	I	I	I	T	T_{92}	-	-	-	-	-	I	-	L	A_{92}		
Portugal - RDP	I	I	I	I	I	L	T	T	-	T	T	T	-	-	-	2 kHz
Spain - RNE[7]	T	T	T	L	L	-	-	-	-	-	T	T	-	-	-	2 kHz
C. Iberica[8]	I	I	I	I	I	T	T	-	T	-	I	T	-	A_{92}	I	
Catalunya R.	I	I	I	T	T	T	-	-	-	I	I	T	-	-	-	
Sweden - RR/SLR	I	I	I	I	I	P	P	P	P	T	P	P	L	L	I	2.5 kHz
Switzerland - SSR	I	I	I	I	I	-	L	-	-	L	-	T	-	T	-	
United Kingdom - BBC	I	I	I	I	I	P	-	-	P	I	I	P	-	I	-	2 kHz[9]
ILR	I	I	I	I	I	T	-	-	-	L	L	T	-	L	-	2 kHz
Yugoslavia[5] - JRT	I	I	I	I	I[10]	L	T_{91}	T_{91}	-	I_{91}	I	T_{91}	-	-	I	4[11] kHz

Notes:
1. - *Usually only on one network*
2. - *TDC and IH may be used for special applications internally be broadcasters, or as part of other applications*
3. - *On France Inter and France Musique*
4. - *RDS paging on Radio 1 service. Time shared, MBS paging with RDS, on Radio 2 service*
5. - *For the time being only the south part of the country is not covered*
7. - *Only Madrid area for testing*
8. - *Operational in 12 major cities*
9. - *Except 1.2 kHz on Radio 3 (classical channel)*
10. - *Not on all networks*
11. - *In the case when ARI is impplemented 1.2 kHz/RDS and 3.5 kHz/ARI*

Codes:
A = Announced intention
I = Implemented
L = Likely feature
P = Preoperational phase
T = Test transmission
A_{90} = Announced for 1990, etc.

Source: EBU RDS Newsletter

Terms for the applications:
PI: Program identification
PS: Program service name
AF: Alternative frequencies
TP: Traffic-program identification
TA: Traffic-announcement identification
PTY: program type
DI: Decoder identification
MS: Music/speech switch
PIN: Program item number
EON: Enhanced Other Networks information
CT: Clock-time and date
RT: Radiotext
TDC: Transparent data channel
IH: In-house applications
RP: Radio paging

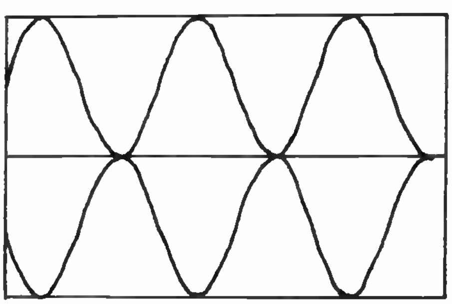
Table 1: RDS features as implemented now in Europe

IMPROVING THE INTERMODULATION DISTORTION CHARACTERISTIC OF YOUR PRESENT AM TRANSMITTER

Timothy C. Cutforth, P.E.
Vir James Engineers
Denver, Colorado

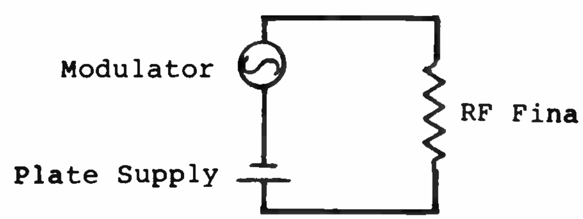
ABSTRACT

The high level plate modulated AM transmitter suffers from some inherent nonlineartities. Several practical modifications and adjustments are described which can dramatically reduce the audible intermodulation distortion and thereby increase both the clarity and loudness of transmitters presently in service.



INTERMODULATION
- Shows on the modulation monitor the same as program modulation
- Robs you of peak audio program modulation
- Contributes to splatter
- Makes your station sound muddy and unclear

High level Plate modulation is inherently nonlinear. The power supply, high level modulator and final output load stage can be shown as below in simplified form.



The modulator parameters over one cycle of the modulated RF envelope are as follows:

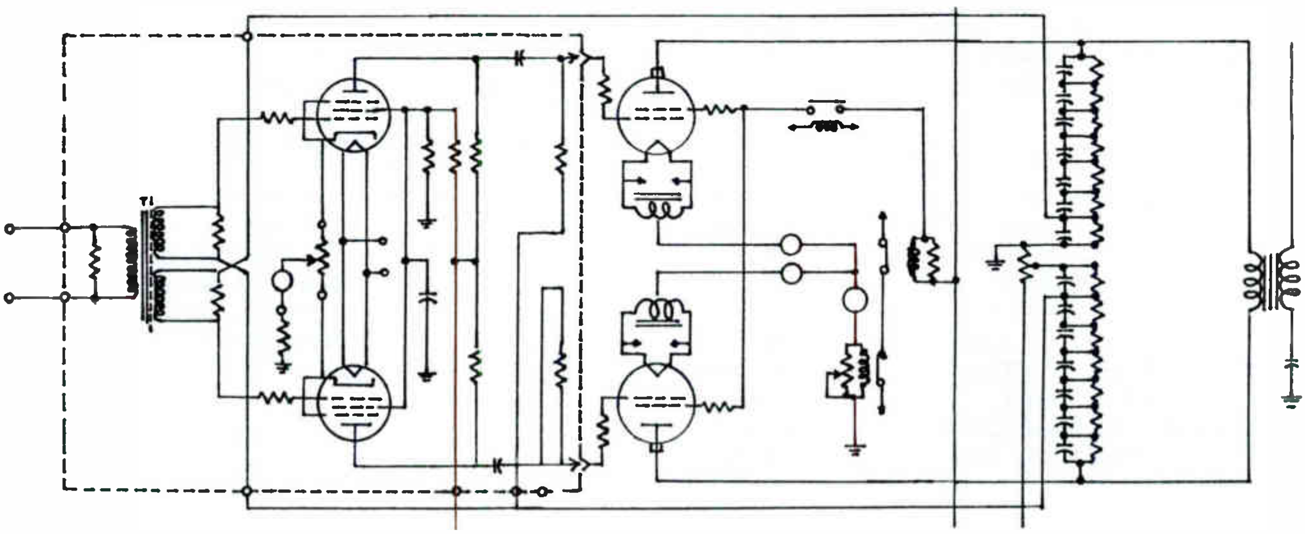
Modulation Percent	Mod Voltage Volts	Mod Current Amperes	Mod Power Watts	Load Z
+100	1.0	2.0	2.0	0.5
+ 75	0.75	1.75	1.31	0.43
+ 50	0.50	1.50	0.75	0.33
+ 25	0.25	1.25	0.32	0.2
0	0	1	0	0
- 25	-0.25	0.75	0.19	0.33
- 50	-0.50	0.50	0.25	1.0
- 75	-0.75	0.25	0.19	3.0
-100	-1.0	0	0	0

Neither the power or the load impedance are linear functions. Therefore, a standard linear amplifier does not produce linear modulation. For truly linear modulation to occur the amplifier must correct for this inherent problem. Luckily most Hi level plate modulated transmitters have an adaptable circuit.

The most common circuit is

The first and easiest adjustment is to offset the gain between the positive and negative side of the modulator so that the heavily loaded positive peak side pushes harder about 10 to 20% extra gain On the positive side 1 1/2 db seems to be needed on most transmitters and reduces the IMD by about 50%.

An offset in the feedback loop also seems to linearize the modulation process. Most transmitters respond to a 1 db offset in the feedback.

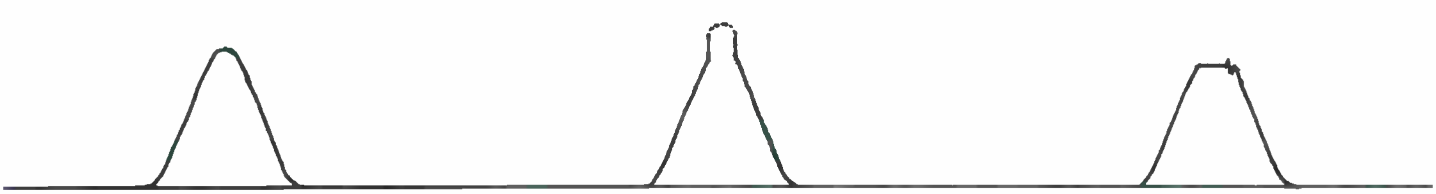


which is in fact two class B amplifiers linked by the feedback loop.

The easiest linearity test is a two tone high frequency IMD test. Pick two high frequency tones - say 8 and 8.1 kHz and set each one for 40% modulation. The IMD product is a 100 Hz tone clearly audible on headphones or speaker. No test analyzer is needed - just your ears. The less beat note you hear the lower the IMD.

Ideally the feedback should run out just as the tube clips. Excessive feedback results in out of band splatter products as the driver kicks in additional gain to "force" the final beyond saturation.

For most transmitters the best performance is with 5 or 6 db feedback.



Under sine wave modulation conditions the "feather" on the falling waveform can be seen on an oscilloscope but more than that it can be heard all around the dial. As the feedback is reduced the splatter is reduced and so is the IMD.

Tube linearity also greatly affects IMD. Adjusting the bias on the modulator tubes can improve linearity as well. Tube gain matching also can be helpful to lower IMD.

Each of the above modulator adjustments should be repeated in order until the best IMD is reached.

In one case a Bauer FB50000J transmitter was adjusted as described and the IMD product was reduced by 12 db from 6% to 1%. A very notable difference!

The comparative loudness increased dramatically! The music became clear and very bright with excellent definition! There was no splatter even at saturation of the modulator!

However there are cases where the modulator does not even see a resistive load. In some installations bandwidth limitations at the antenna can cause the final RF stage to present a complex reactive load to the modulator stage. I have seen one case where despite having a perfect 50 Ohm match at the operating frequency, the Intermodulation Distortion measured over 50 percent in the night pattern and above 30 percent in the day pattern. The station did not operate nondirectional at all but tests in the nondirectional mode showed Intermodulation Distortion reduced to 6 percent. Needless to say adjustment of the modulator stage could not correct for this extremely poor antenna bandwidth. If you find that the final plate tuning has a significant effect on the Intermodulation Distortion it is likely that the bandwidth is marginal and that broadbanding would improve performance.

IMPROVING FM MODULATION PERFORMANCE BY TUNING FOR SYMMETRICAL GROUP DELAY

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ABSTRACT

FM broadcast transmitters are often tuned for minimum synchronous AM resulting in a symmetrical amplitude response which is *assumed* to minimize FM distortion. This would be true if the RF power amplifier circuit topology resulted in simultaneous symmetry of the amplitude and group delay responses.

Recent research using computer modeling and empirical tests have verified that simply tuning for minimum synchronous AM does *not* result in best FM modulation performance, because the symmetry of the group delay response has a much greater effect on distortion of the FM modulation than the amplitude response.

This paper describes a simple procedure to optimize the FM modulation performance of any FM transmitter by tuning for symmetrical group delay using a broadcast modulation monitor and standard test equipment found in most radio stations.

INTRODUCTION

FM Broadcast Engineers need a simple and effective procedure to tune FM transmitter power amplifier(s) for best FM modulation performance. Over the past several years, procedures for tuning the amplifier(s) for minimum synchronous AM (ICAM) resulting from FM modulation have been used with some success to improve FM modulation performance. This paper explains an alternative tuning procedure that offers further improvements in FM modulation performance. The new procedure involves tuning the amplifier(s) for symmetrical group delay instead of minimum synchronous AM (ICAM). Before getting into this procedure, the terminology needs to be defined.

DEFINITION OF TERMS

Synchronous AM

The perfect FM transmitter would have an absolutely constant output amplitude, regardless of FM modulation or power supply variations. A practical FM transmitter produces an output which varies in amplitude as well as frequency. The portion of the amplitude variation resulting from the FM modulation is called "Synchronous AM".

Synchronous AM, also referred to as Synchronous AM Noise or Incidental Carrier AM (ICAM), is a measure of the amount of incidental amplitude modulation introduced onto the carrier by the presence of FM modulation. Since all transmitters have limited bandwidth, there will be a slight change in power output as the carrier frequency is swept to either side of the center frequency. This change in RF output level follows the waveform of the audio frequency being applied to the FM modulator causing AM modulation in synchronization with the FM modulation. This concept is similar to the slope detection of FM by an AM detector used in conjunction with a tuned circuit. The technical paper entitled "Optimum Bandwidth for FM Transmission"[1] provides further insight into the overall bandwidth required to achieve a given level of FM modulation performance.

Synchronous AM measurements give the station engineer a rough idea of the overall system bandwidth and whether the transmitter is tuned to position the amplitude passband correctly.

Symmetrical Group Delay

Another way in which practical transmitters deviate from the ideal, is the group delay response. Group delay refers to the propagation time delay variations between a group of several different frequency components (FM sidebands) propagating through the transmitter power amplifier(s). Each different frequency component (FM sideband) will pass through the input and output network(s) at a slightly different rate, so that the FM sidebands in the group are not all delayed equally in time, hence the term Group Delay refers to the total time delay variation of the group of FM sidebands. Symmetrical Group Delay refers to a tuning condition that causes the Group Delay (time) variations to be equal above and below the center frequency of the transmitter.

Another way of viewing Group Delay, is to observe the phase shift of each FM sideband passing through the output amplifier(s). If each FM sideband is phase shifted linearly with frequency, the Group Delay (time) will be constant. Constant Group Delay is therefore equivalent to linear phase shift with frequency. A properly terminated piece of transmission line provides constant group delay and linear phase shift. All components of the signal are delayed equally in time and no phase distortion occurs.

FM Modulation Performance

For the purpose of this presentation, FM modulation performance is defined as the overall quality of the FM baseband information transmitted to the consumer. Quality factors include the amount of harmonic distortion, intermodulation distortion, stereo separation, and crosstalk between subcarriers. Factors which are not part of the demodulated baseband include AM signal to noise ratios that do not directly affect FM modulation performance.

LIMITATIONS OF SYNCHRONOUS AM MEASUREMENTS

Synchronous AM measurements are an indirect way of evaluating and optimizing FM performance. Even though synchronous AM measurements are a helpful aid to begin tuning an FM transmitter, these measurements tell only the amplitude response half of the total story. Transmitter tuning also affects the group delay (time) response which changes the relative time delays of the higher order FM sidebands.

FM broadcast transmitter RF power amplifiers are typically adjusted for minimum synchronous AM. This results in a symmetrical amplitude response by centering the transmitter's amplitude passband on the FM channel. The upper and lower sidebands will be attenuated equally or symmetrically which is *assumed* to result in optimum FM modulation performance. This would be true if the RF power amplifier circuit topology resulted in simultaneous or coincidental symmetry of the amplitude and group delay responses.

The tuning points for symmetrical amplitude response and symmetrical group delay response normally do *not* coincide, depending on the circuit topology of the RF power amplifier. Therefore, simply tuning for minimum synchronous AM (symmetrical amplitude response) normally does not result in best FM modulation performance. The technical paper entitled "The Significance of Power Amplifier Circuit Topology on FM Modulation Performance"[2] provides detailed information about various power amplifier circuit topologies.

Symmetrical Amplitude (Synchronous AM) versus Symmetrical Group Delay Response

A computer simulation called FMSIM[3] was jointly developed by Broadcast Electronics and Quantics Software to explore the effects of the transmitter output network(s) and FM filterplexing systems on FM modulation performance. FMSIM allows the effects of the Amplitude Response to be independently compared with the effects of Group Delay response. The independent evaluation of amplitude versus group delay effects are difficult or impossible to do empirically since the amplitude and group delay responses are inseparable in a real network. The results of these simulations showed that amplitude differences between the upper and lower sidebands of the FM signal have little direct effect on FM modulation performance, while *Group Delay (time)* differences between the upper and lower sidebands have a much more profound effect on FM modulation performance.

Furthermore, the analysis revealed that when the group delay is symmetrical above and below the carrier frequency, the total FM distortion is minimized. In particular, the even order (2nd, 4th, 6th, ...) harmonics of the audio modulating frequency drop out.

Figure-1 shows a spectrogram of baseband distortion products from a power amplifier that is tuned for a symmetrical amplitude response (minimum synchronous AM) with an asymmetrical group delay response. Note that both even and odd order distortion products are visible in the demodulated baseband.

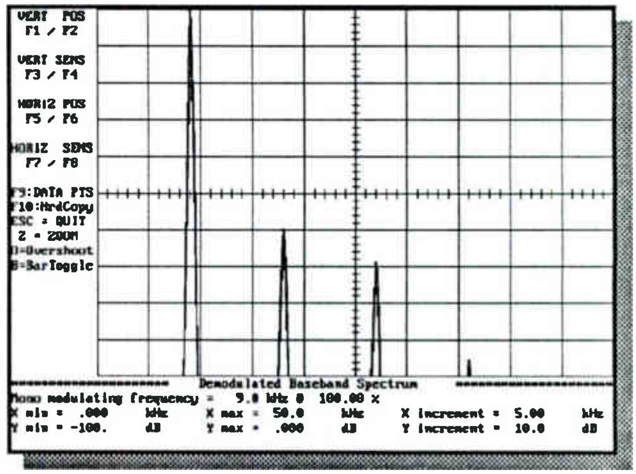

FIGURE-1

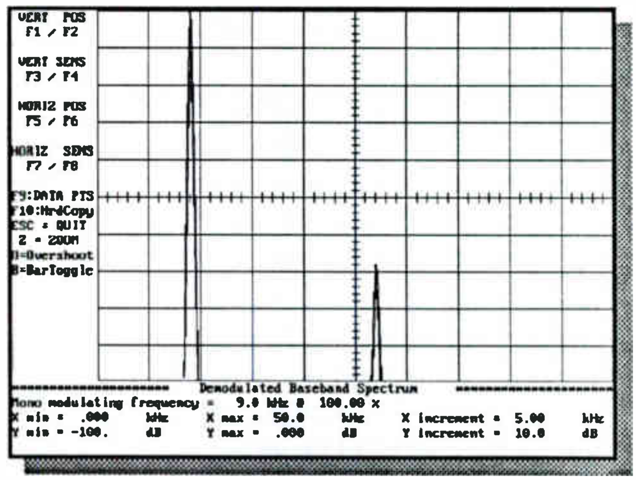

FIGURE-2

Figure-2 shows a similar spectrogram from the same power amplifier that has been re-tuned for a symmetrical group delay response centered on the carrier frequency. Note that only odd order distortion products are now visible. This effect leads to a simple, low cost procedure for tuning the amplifier(s) to this condition without the use of network analyzers or other complicated test equipment.

Tuning for Best FM Modulation Performance

Tuning for minimum synchronous AM is a good starting point, but it is desirable to finish tuning at the symmetrical group delay point. Fine tuning the input and output for minimum even order harmonic distortion will optimize the group delay (time) response.

Tuning the transmitter for minimum even order harmonic distortion will result in a symmetrical group delay response and optimum FM modulation performance. This can be accomplished by: (1) observing the even order harmonics in the demodulated baseband with a spectrum analyzer or by (2) placing an audio bandpass filter (tuned to the second harmonic of the audio modulating frequency) on the input of the audio distortion analyzer.

Most FM stations have an FM stereo modulation monitor with a 19kHz bandpass filter and metering circuitry that is normally used to measure the 19kHz pilot tone injection level. This monitor function can also be used to tune for symmetrical group delay if the transmitter is 100% modulated with a single 9.5kHz monaural tone *without 19 kHz pilot*. The second harmonic distortion produced by transmitter amplifier(s) mistuning will fall within the 19kHz bandpass of the monitor's pilot injection level metering and will appear as if there was a pilot tone present. Tuning the transmitter power amplifier(s) for a minimum pilot injection level indication will null the second and other even order harmonics of the 9.5 kHz modulating tone resulting in symmetrical group delay of the sidebands.

If the FM station does not have a suitable stereo modulation monitor, a simple 19kHz bandpass filter can be inserted between the composite output of the RF to baseband demodulator and the audio voltmeter. The transmitter is then tuned for minimum audio voltage produced by second harmonic distortion.

The passive 19kHz bandpass filter can be purchased from one of the companies shown in the references[4] or it can be easily constructed out of readily available inductors and capacitors[5] according to the schematic shown in Figure-3.

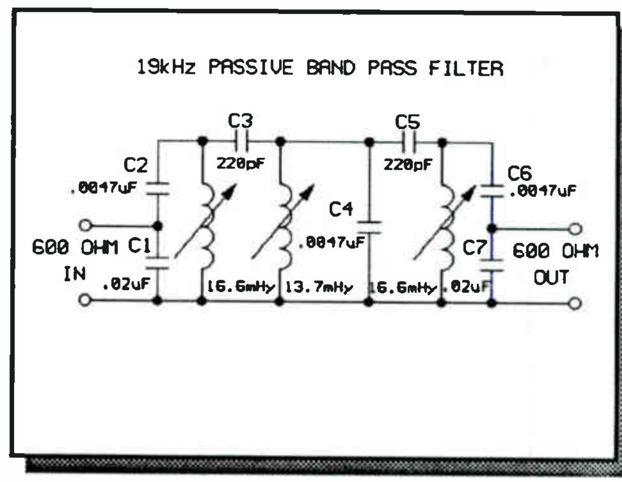

FIGURE-3

Figure-4 shows the amplitude response and insertion loss of the 19kHz bandpass filter illustrated in Figure-3.

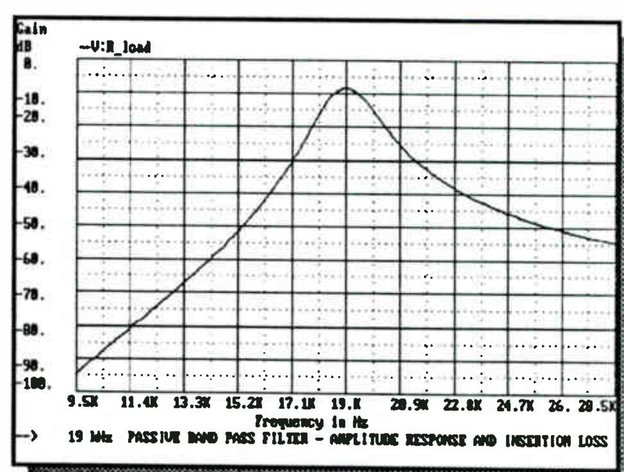

FIGURE-4

The 19kHz filter should have at least 80 dB rejection of the 9.5kHz modulating tone and at least 50 dB rejection of the third and higher harmonics of 9.5kHz.

Be certain that the FM demodulator has good linearity and does not introduce distortion products that would cause the broadcast engineer to mistune the transmitter to compensate for the distortion introduced by the demodulator. Modulation monitors that utilize a pulse-counting discriminator are usually the most dependable for this measurement.

CONCLUSIONS

1. The transmitter should be tuned for symmetrical group delay response which results in best FM modulation performance rather than symmetrical amplitude response which results in minimum synchronous AM. Depending on the circuit topology, the tuning conditions for symmetrical group delay response may not coincide with the symmetrical amplitude response.

2. Simply tuning for minimum synchronous AM (symmetrical amplitude response) does not necessarily result in best FM modulation performance. Best FM modulation performance is always obtained when the system is tuned for symmetrical group delay (time) response.

3. Most FM transmitters will exhibit a significant increase in synchronous AM when tuned for symmetrical group delay response even though this condition results in best FM modulation performance.

4. The symmetrical group delay tuning point usually does not coincide exactly with the symmetrical amplitude tuning point and generally falls between the point of minimum synchronous AM and the point of maximum RF power amplifier efficiency.

5. RF power amplifier circuit topologies that exhibit coincidence of symmetrical amplitude and group delay responses will result in a better overall FM modulation performance.

6. Tests on several FM broadcast transmitters verified that tuning for minimum even order harmonic distortion provided the best FM modulation performance with minimum distortion to the demodulated FM baseband and resulted in symmetrical group delay through the transmitter as measured with a network analyzer. These tests also confirmed the FMSIM prediction that group delay response asymmetry causes higher FM modulation distortion and crosstalk than amplitude response asymmetry.

ACKNOWLEDGEMENT

The author wishes to thank Sandy Blickhan, Quantics, and the Broadcast Electronics engineering team for their assistance in preparing and reviewing this paper.

REFERENCES

1. "Optimum Bandwidth for FM Transmission". Anthony, Edward J. Broadcast Electronics, Inc. 1989.

2. "The Significance of RF Power Amplifier Circuit Topology on FM Modulation Performance". Shrestha, Mukunda B. Broadcast Electronics, Inc. 1990.

3. "FMSIM", the FM Simulation Program is available from Broadcast Electronics Inc., P.O. Box 3606, Quincy, IL 62301-3606 or Quantics, P.O. Box 2163, Nevada City, CA 95959-2163

4. SOURCES OF 19kHz BPF:

Filtronetics Inc., 6010 Parretta Dr.,
Kansas City MO. 64120,
(816) 231-7375

Torotel Products Inc., 13402 S. 71
Highway, Grandview, MO. 64030,
(816) 761-6314

Belar Electronics Lab Inc.,
119 Lancaster Ave., Devon, PA 19333,
(215) 687-5550

5. 19kHz BPF BILL OF MATERIALS:

C1,C7 = .02uF   C3,C5 = 220pF
C2,C4,C6 = .0047uF
All capacitors are disc ceramic or dipped mica and are available at Radio Shack.
L1,L2,L3 = J.W. Miller #9061 or VLS153

THE TOWERS INDUSTRIAL PARK PROJECT AT KTNQ

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INTRODUCTION

In the fall of 1987, H & G Communication of California, Inc., approached the firm of du Treil, Lundin & Rackley, Inc., to have them and Ogden Prestholdt act as their radio engineers to consider alternate uses of their KTNQ transmitter site. A basic concept was proposed which would permit approximately 50% coverage of the site with buildings and still maintain an effective five-tower directional antenna for KTNQ. Developer, architect, supporting consultants and a construction contractor were selected. The project was designed and a construction schedule of approximately one year was made. KTNQ operated at reduced power with temporary pairs of antenna towers during construction. The project was successfully completed in less than the scheduled time and resulted in an excellent building development and a radio plant that was more efficient and stable than before the construction project.

PRELIMINARY PLANNING

Figure 1 is an aerial photograph of the site as construction began. The items considered in the preliminary planning included, but were

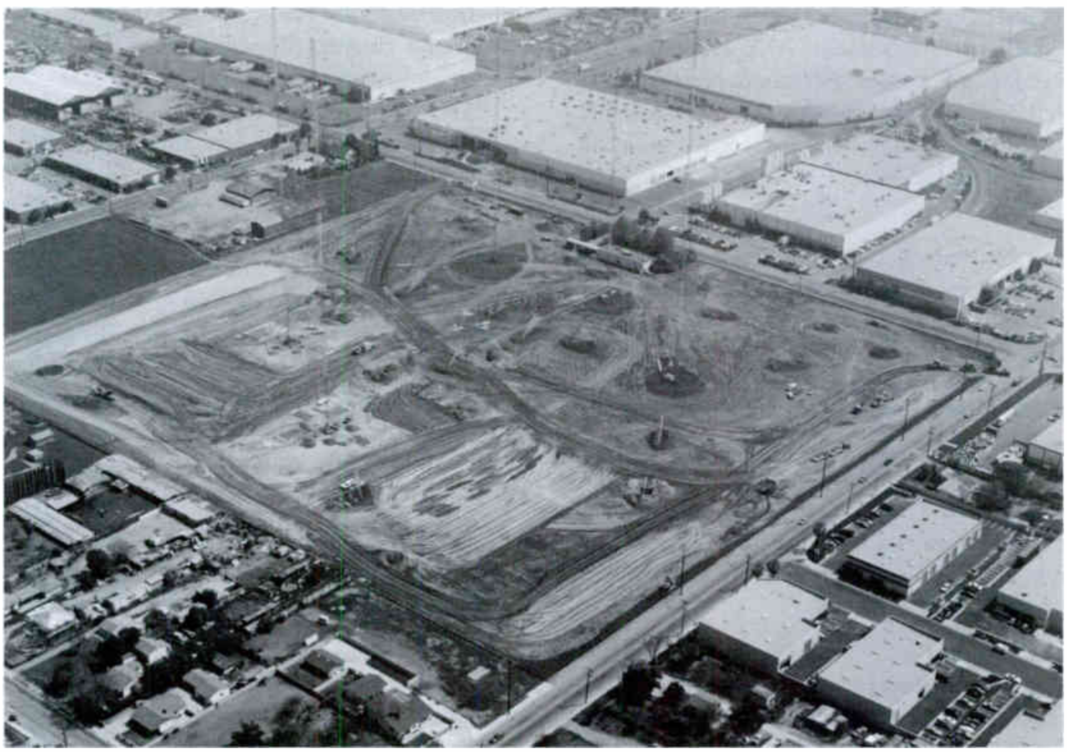


Figure 1. AERIAL PHOTOGRAPH OF SITE

not limited to, the following: layout of the building relative to the towers on the site; calculation of directions and magnitudes of ground currents between and around the antennas; prediction of near fields in and around the antennas; design of shielding required in the buildings for personnel and electronic equipment protection; study of possible effects of conducting vertical walls between the antennas; design of methods for operating temporarily on sub-groups of towers with appropriate reduction of power while affording protection to other radio station assignments so that large land areas could be released for construction work without undue exposure of construction personnel to non-ionizing radiation; and consideration of levels of non-ionizing radiation in work areas throughout the construction.

The Trammell Crow Company was selected as the developer. The architectural firm of Lee & Sakahara AIA, Inc., was selected along with a group of support engineering firms. Also the company of Prizio & Prizio General Contractors was selected as the general contractor. The team consisting of representatives from Trammell Crow, Lee & Sakahara Associates AIA, Inc., Ajit Randhava & Assoc., Holmes & Narver, Inc., Prizio & Prizio, Prestholdt and du Treil, Lundin & Rackley, Inc., proceeded with the detailed design and scheduling of the project.

The concept of two buildings as shown in Figure 2, with a total of 522,000 square feet and with tail-gate height floors, was adopted. The design concept included placing the buildings so that the five 500-foot towers would be located in wells within the buildings and at least 75 feet from the nearest building perimeter, complete shielding of the buildings, and covering the driveway between the buildings with a wire grid to make a complete elevated ground plane for the antenna system.

Since the buildings were to have truck tail gate height floors, significant fill grading and compacting was needed for the building areas. The construction schedule was established to permit working on either half (area of one building) of the site while operating from two towers on the other half of the property.

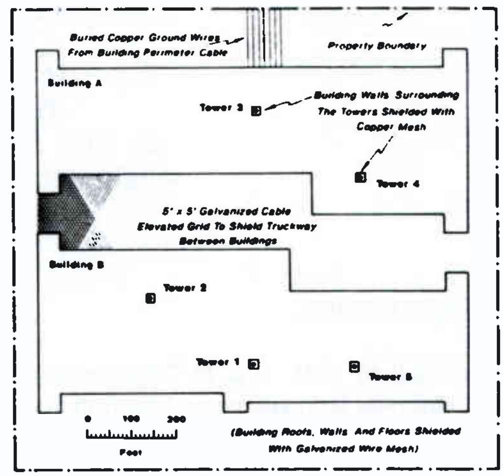


Figure 2. SITE PLAN

The split of the area into two buildings would have made an unusual gap in the ground plane for the antenna system. Using the work of Prestholdt's paper "THE DESIGN OF NON-RADIAL GROUND SYSTEMS FOR MEDIUM-WAVE DIRECTIONAL ANTENNAS", presented at the 1978 NAB Broadcast Engineering Conference, the ground currents flowing on the KTNQ site for both antenna patterns were examined. Much as expected, the basic currents in the truck-way area between the two proposed buildings were found to vary in direction during the R.F. cycle. In much of the area the magnitude of the ground current did not vary much in amplitude during the cycle and therefore required a ground system with high conductivity in all directions. Because of this, a wire grid between the two buildings was designed. The grid wires were designed to run at 45 degrees with respect to the building walls, with two crossing sets used and five-foot spacing between adjacent wires. At all crossing points the design called for the wires to be connected together with standard cable clamps. Further, each wire at each building end would be connected to the wall roof shielding junction. A large level ground plane for the antenna system, conducting in all directions and consisting of the two buildings and the wire grid covered truck-way between them, was the end result.

The buildings were to be constructed with concrete floors, 30 foot high concrete tilt-up walls and a lumber-supported layered roof. The towers were to be in 16 foot square wells in the buildings. Telephone service and electric power were to come in underground and consequentially would be adequately shielded.

The use of chicken fence wire was selected for the shielding material in the concrete walls, floor and roof. This was after a thorough consideration of using heavy reinforcing wire mesh, copper screen (which was much too expensive and not necessary) and other wire meshes. A galvanized wire mesh with a multi turn twist between openings was selected. The contractors superintendent received some interesting comments when he was ordering 1.25 million square feet of chicken wire fencing!

The wells for the five towers were to be lined with expanded copper screening such as used in our normal ground systems. This was to be joined to both the floor and roof shielding so as to make a complete shielded box for the buildings while keeping the antenna fields outside of the buildings.

CONSTRUCTION SCHEDULE

The construction sequence plan called for the following:

1. Operate KTNQ from towers 1 & 2 while doing the following:
   a. Fill and grade the area for building A (north).
   b. Pour the floor with its shielding for building A.
   c. Erect the wells around towers 3 & 4.
   d. Connect the shielding in the wells to that in the floor.
   e. Start exterior work on site around building A.
2. Operate KTNQ from towers 3 & 4 while doing the following:
   a. Fill and grade the area for building B (south).
   b. Pour the floor with its shielding for building B.
   c. Erect the wells around towers 1, 2 & 5.
   d. Connect the shielding in the wells to that in the floor.
   e. Start exterior work on site around building B.
3. Operate KTNQ from towers 1 & 2 while doing the following:
   a. Pour and erect the walls for building A.
   b. Connect the shielding between the wall sections and the floor.
   c. Install and complete the roof and its shielding.
   d. Continue exterior work.
4. Operate KTNQ from towers 3 & 4 while doing the following.
   a. Pour and erect the walls for building B.
   b. Connect the shielding between the wall sections and the floor.
   c. Install and complete the roof and its shielding.
   d. Build the new KTNQ transmitting plant in building A.
   e. Continue exterior work.
5. Operate KTNQ from towers 1 & 2 while doing the following.
   a. Demolition of old transmission lines to towers 3 & 4.
   b. Install new feed system to towers 3 & 4.

c. Continue construction of new KTNQ plant.

d. Continue exterior work.

6. Operate KTNQ from new plant and towers 3 & 4.

   a. Install wire grid at roof level between buildings A & B.

   b. Install new feed system to towers 1, 2 & 5.

   c. Complete exterior work.

7. Tune up KTNQ on all towers and both patterns and conduct the proof-of-performance.

CONSTRUCTION PROCEDURES AND SAFETY

Even with KTNQ operating at a reduced power of 25 kW in the daytime there would be significant areas on the site where high electromagnetic fields would exist. Extensive studies were made of these fields so that working areas could be safely delimited. Those studies included calculation of E and H fields at various heights so that workers on large machines would be considered and prediction of contact currents so that possible problems with workers touching large machines either from the ground or while sitting on them could be understood.

A safety memo was prepared to advise the workers about the potential hazards of working on the site and to delineate areas where personnel would be permitted to perform specific kinds of tasks. Since the antennas would be used in two modes (normal base fed and with the base enclosed in a 30 ft. high grounded shield well) during different phases of the construction, two sets of data were needed. Workers would be walking on the ground, riding on graders and compactors, working with cranes, etc. Fields at numerous heights and as effected by various machines had to be considered. Also, since these towers are approximately one-half wavelength in height and have high base voltages and low base currents, the E field is the significant limitation near ground level. This leads to a concept where the pickup of energy by machines (or any object) is capacity driven which leads to simple modeling concepts.

First studies were made of the E fields at various heights from two to five meters and at distances of 2 to 150 meters from the base of one of the antennas with a power of 1 kW for both modes of operation. Simple graphs of these data were prepared and, by the use of appropriate scaling techniques, the exposure levels at any distance and for any antenna current or power could be determined. This would establish the safe areas, based on dielectric heating, for workers at any ground location and height desired and could be based on any choice of safety factor desired. This, of course, was then available for both antenna modes.

Second, studies were made of possible contact currents. Since the excitation of machines by the radiated fields is capacity driven, we approximated trucks, graders, cranes, etc., by simple cylinders which would have capacitances similar to the selected machine when it was insulated from the ground by rubber tires. This would permit calculation of conduction current to a model person (again a simple cylinder) when making the ground contact to the machine. Capacitive coupling to the machine was also modeled to simulate the wearing of gloves. Of course, all of these tests with simulated machines and persons were only rough guides to establish verification of concepts and limits on the exposure values. Although only rough guides, they may well be maximum possible values since they are based on infinite conductivity of the models and hence should yield maximum currents. An adequately safe working environment was planned with these results and the addition of significant extra tolerances.

In the process of tilting up the antenna well walls and all other tilt up walls (with shielding these become current conducting areas), procedures had to be established for handling them with due safety to the workers.

Here the problem was one of contact current between wall sections and rigging and between wall sections in their final positions. Arcing at such contact points would be a possibility prior to completion of all electrical bonding. Calculations had shown that the E and H fields near such parasitically excited conductors would not be at hazardous levels. Sequence procedures were established for each such type of construction activity.

The laying of the shielding screen in the roof structure required care in making the first connections across the roof. Even on the roof of the building where the towers were not excited there would be significant currents flowing and the first connections needed to be made carefully and early in the roof installation.

Many details with respect to connections in the shielding required specific procedures and careful quality control. Although these were established as part of the design phase, continuous surveillance was required. Harvey Rees was hired by KTNQ to provide on-site construction assistance and to lay out plans for the new transmitter plant.

CONSTRUCTION

The logistics of bringing in more than 25,000 yards of fill for each building and then grading and compacting it was no small feat, the lines of trucks in the neighborhood were real eye catchers. Then, of course, there were large graders, compactors and watering trucks. All of these had to be kept away from the 30 guy anchors, five tower bases, and existing buried transmission lines. The routing for all of these functions had to be coordinated with personnel protection from high radiated fields.

But that is only one phase of the major construction project. Imagine paving, no not just one football field, but four side by side to cover an area of more than 900 by 300 feet. The typical ready mix concrete truck carries 9 to 12 yards of concrete. The lines were not as long as for the fill -- but they were long! The project benefitted from the excellent work done by Jerry Frank, the construction superintendent for Prizio & Prizio.

Note that the towers each have guys at six elevation levels which terminate in a set of six anchors for each tower, a total of thirty guy anchors for the five towers. Eleven of these anchors had to be replaced by elevated stanchions to minimize obstructions to roadways. Thirty-six guys had to be modified for roof attachment. Nineteen guys had to have insulator spacing adjusted to limit coupling into the center truck-way.

Guy anchor relocation called for moving the anchor closer to the tower and making a vertical stanchion for the new anchor. This kept the guy lines in the same paths and consequently did not alter the basic structural loads in the tower and its guys. This structural design was done by Holmes & Narver, Inc., and the field erection work by a very competent antenna erection company. Since this effected the lower guy insulator position in many guys, proper relocation of those insulators was important. These insulator reactions fell into two categories, those in the truck-way between buildings and those in the external roadways. For those in the external roadways there were no problems, it was necessary just to break the guy line above the stanchion connection to minimize current in the stanchion. However, for those in the truck-way, additional insulators were required to prevent coupling significant KTNQ energy through the wire grid and into the truck-way. The guys were electrically connected to the wire grid at the point of penetration. The insulators had to be placed to isolate the guys from the stanchion and to limit the lengths of the segments penetrating the wire grid.

SHIELDING

During construction and particularly as the buildings were being finished, numerous measurements of the fields within and around the buildings were made. It was soon found that within the nearly completed buildings the fields were too low for reading by the standard meters for non-ionizing fields. A standard Potomac Instruments FIM-41 field strength meter was used. This meter actually measures the H field but is calibrated to read the equivalent E field based on a plane wave without reflections. In most of the area, the variations in observed field levels were sufficiently uniform inside the buildings to indicate that the standing wave pattern in the building is not very strong.

The first factor observed was that, before the walls were connected to the floor (electrically), a significant amount of energy leaked into the buildings at the bases of the walls. When the connections were completed the field was more uniform out to the walls except where large windows or doors were present. The antenna wells had access doors during the construction. Until they were closed and the screening added inside of the door area, the leakage through them was very significant. In the completed state there is only a modest increase in field near the wells.

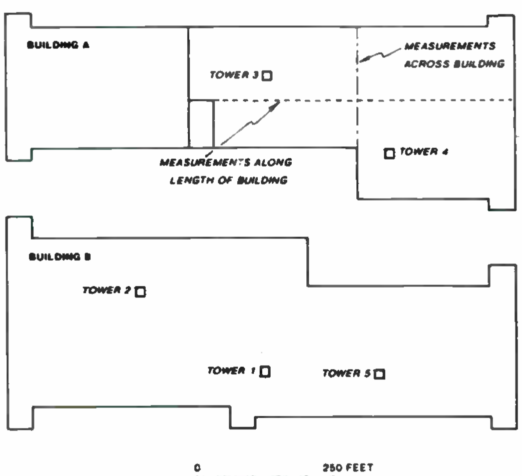


Figure 3. MEASUREMENT LOCATIONS TO DETERMINE EFFECTIVENESS OF SHIELDING

Here are some typical values of measured equivalent E fields inside of the buildings following completion. Operation was with the night pattern at 25 kW so the full-power values will be the square root of two times these values. Figure 3 shows the buildings with the locations of the measurements reported.

1. The field off the property at 0.6 miles in the major lobe - 4800 mV/m.

2. The field across the building between the two antenna wells varied from 45 to 110 mV/m.

3. The field along the length of the building from the KTNQ space to the end except near the tower wells varied from 30 to 47 mV/m.

4. The field near the tower wells varied from 450 mV/m at 50 feet to 1100 mV/m at 2 feet.

5. Similar results were found at the second building except that the highest fields were about 760 mV/m at 2 feet from an antenna well.

6. Measurements on the KNX signal on 1070 kHz showed an attenuation of approximately 36 dB within the buildings.

Figure 4 is an aerial view of the completed facility.

ANTENNA SYSTEM CONSIDERATIONS

New phasing and coupling equipment was installed in the transmitter facility. Its design had to consider the shunt effects of the tower wells on base operating impedances. Because of the half-wavelength height of all five of the towers, accurate prediction of the shunt capacitance effects was of the utmost importance.

Using electrostatic field theory for the geometry of a tower enclosed by the copper screen lining the walls of each well, a shunt capacitance of 330 pF was predicted. Base impedance predictions made by method-of-moments techniques were modified by the addition of the predicted shunt capacitance along with other shunt effects, such as the sampling line isolation coils and tower lighting transformers, to determine operating impedances at the ATU output ports.

Figure 4. AERIAL VIEW OF THE COMPLETED FACILITY

Because of previous adjustment difficulties with the old phasing and coupling equipment, great care was taken in the new design to optimize both front-panel adjust-ability and bandwidth by employing extensive nodal analysis modeling in the network selection process. The directional antenna patterns presented interesting design challenges because two towers each in both day and night modes must be fed less than 1000 watts, (for 50,000 watts transmitter power) to produce the proper patterns.

The "flexible power flow" power divider concept, which was described in Rackley's paper "Modern Methods in Mediumwave Directional Antenna Feeder System Design" presented at the 1991 NAB Engineering Conference, was employed for the low-power towers. System phase shifts were selected to allow the phase controls to operate rather smoothly through the positive-to-negative power flow transition region.

ANTENNA SYSTEM ADJUSTMENTS

Following all installation work, adjustment to the antenna monitor loop current parameters predicted by moment-method modeling and line matching to allow full power operation were accomplished in approximately two days. Since field strength measurements showed very close agreement with the theoretical radiation patterns, operation with full power was restored at that time.

Because of potential scattering from the very rugged terrain traversed by certain critical measurement radials within 20 miles of the KTNQ site, numerous points on each critical radial were "talked down" to virtually zero field. Antenna monitor parameters were recorded for each point used in the "talk down" process. Complex-plane mapping of field strength versus distance for each radial was used to evaluate the scatter in field strengths introduced by the terrain.

Complex-plane mapping makes it possible to view the resultant field strength vector, from addition of the individual tower fields and fields scattered from sources other than the towers, for each point on a radial. No scattering effects were found attributable to anything other than the intervening terrain features within a few miles of the site. The new buildings seemed to have no significant effect on the far-field directional antenna radiation vectors.

Using the complex-plane mapping information, fine adjustments were made to the antenna parameters to ready both day and night patterns for proof-of-performance field strength measurements. The proof-of-performance measurements showed both patterns to be properly formed and within their standard pattern radiation requirements. Figures 5 and 6 show the standard and measured nighttime radiation patterns, respectively.

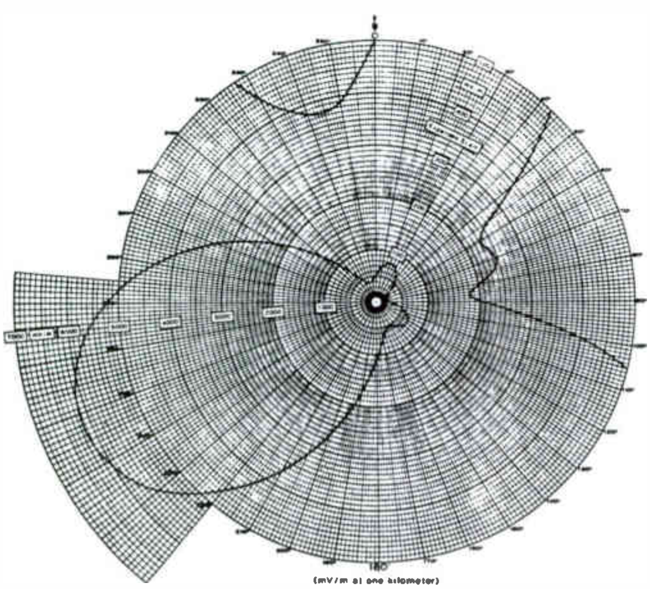
ANTENNA SYSTEM EFFICIENCY

The theoretical RMS radiation values for the day and night patterns, calculated assuming one ohm loss per element, are 2605 mV/m at one kilometer and 2546 mV/m at one kilometer, respectively. The measured RMS radiation values were 2649 mV/m at one kilometer and 2598 mV/m at one kilometer for the day and night patterns, respectively. The day pattern RMS radiation is 1.7 percent higher than theoretical, while the night pattern RMS radiation is 2.0 percent higher than theoretical.

It is obvious that the shielding of the buildings not only works to effectively reduce field strength levels within them to well below acceptable levels, but also serves to conduct induced currents very efficiently. No losses in radiated field due to the buildings is evident. Indeed, the RMS efficiencies of the two patterns are several percentage points higher than they were at the time of the original proof-of-performance, when the land was bare and copper wire ground radials were underground in the area where shielded buildings now exist.

CONCLUSION

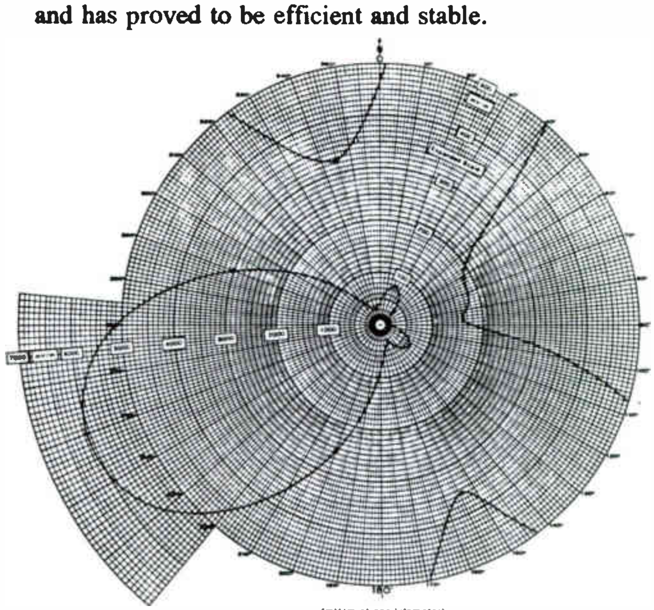
The construction was completed within the time schedule, the shielding met the design concept, the antenna system adjusted easily and has proved to be efficient and stable.



DIRECTIONAL PATTERN PARAMETERS

TOWER	FIELD RATIO	PHASE (DEGREES)	SPACING (DEGREES)	ORIENTATION (DEGREES T)	HEIGHT (DEGREES)
1	1.000	0.0	0.0	0.0	180.0
2	0.921	-83.6	100.0	252.0	180.0
3	1.066	-77.0	203.2	310.5	180.0
4	1.124	3.6	173.4	340.0	180.0

Figure 5. KTNQ NIGHTTIME STANDARD RADIATION PATTERN



DIRECTIONAL PATTERN PARAMETERS

TOWER	FIELD RATIO	PHASE (DEGREES)	SPACING (DEGREES)	ORIENTATION (DEGREES T)	HEIGHT (DEGREES)
1	1.000	0.0	0.0	0.0	180.0
2	0.921	-83.6	100.0	252.0	180.0
3	1.066	-77.0	203.2	310.5	180.0
4	1.124	3.6	173.4	340.0	180.0

Figure 6. KTNQ NIGHTTIME MEASURED RADIATION PATTERN

DIGITAL TELEVISION

Wednesday, April 15, 1992

Moderator:
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Providence, Rhode Island

***SIGNAL DISTRIBUTION AND PROCESSING IN A SERIAL DIGITAL WORLD**
Marc Walker
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A TOTALLY DIGITIZED IN-HOUSE NTSC ROUTING SWITCHER SYSTEM
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COMPRESSED DIGITAL VIDEO: A TECHNOLOGY OVERVIEW
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A NETWORKING SOLUTION FOR STILLS AND GRAPHICS
Bob Pank
Quantel Ltd
Newberry, Berkshire, England

A STILL-ANIMATION FILE SYSTEM EMPLOYING A VIDEO SOLID RECORDER
Takayuki Tanaka, Toshiyuki Sakamoto and Hisashi Fujimura
NHK (Japan Broadcasting Corporation)
Tokyo, Japan

*Paper not available at the time of publication.

A TOTALLY DIGITIZED IN-HOUSE NTSC ROUTING SWITCHER SYSTEM

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NHK (Japan Broadcasting Corporation)
Tokyo, Japan

ABSTRACT

The digitization of the broadcasting facilities has been progressed from the stage of device to system. NHK recently achieved update its in-house transmission facilities, including signal distributors and routing switchers, aiming at a complete digitization of its transmission systems. These new systems are now put into operation and proving to be highly stable, reliable and manpower saving besides they offer excellent quality picture and sound.

NHK has newly developed the Parallel Processing 10B1C coding for the use of its in-house signal transmissions, which has already been applied to digital communications networks. We are now planning to develop the gate arrays which adapt to the signal transmission using both 10B1C and Scrambled NRZI coding format.

1. INTRODUCTION

NHK broadcasts two terrestrial television channels, two direct broadcasting satellite television channels and three radio channels. In 1968, NHK introduced its TOPICS (Total On-line Program and Information System) and ABCS (Automatic Broadcasting Control System) to streamline its operations. The nuclei of these systems have been routing switcher systems which distribute program material for program production and send out programs to air and domestic networks.

NHK recently renewed it's core facilities at the head office in Tokyo. The old analog production routing switcher and on-air routing switcher systems were replaced with totally digitized counterparts which utilizes the state of art in digital technology.

A newly developed serial transmission interface plays an important role in the new systems which ensure various features of the systems. Several LSIs were also developed for video and audio matrices, serial formatters and a 157 Mbps signal synchronizer. All new developed digital technologies are combined to realize the streamlined operation almost maintained automatically.

2. System Design and Features

The following design objectives were considered for the new routing switcher system.

- high quality signal transmission.
- high reliability and high maintainability.
- streamlined, flexible, 24 hour operation.
- ability to cope with an ever-changing program schedule.
- laborsaving, space-saving and energy-saving.
- expandable architecture.

To achieve these objectives, NHK's facilities have been totally digitized. Optical fibers are used for video signal transmission.

Both the production and the on-air systems use separate control computers to ensure reliability. LANs (Local Area Networks) have been adopted to control the production routing switcher and the on-air routing switcher systems. This isolation of these systems enables us to improve the maintainability of the system and flexibility in operations.

Wide screen touch panel controllers, a large message board, synthesized audio alarms and a logging system are all provided to reinforce the functions of the man-machine interface.

The conceptual depiction of the systems is shown in Figure 1.

3. Digitization

3.1 Parameters of Digital Signal

The parameters of digitization of video and audio signal are shown in Table 1.

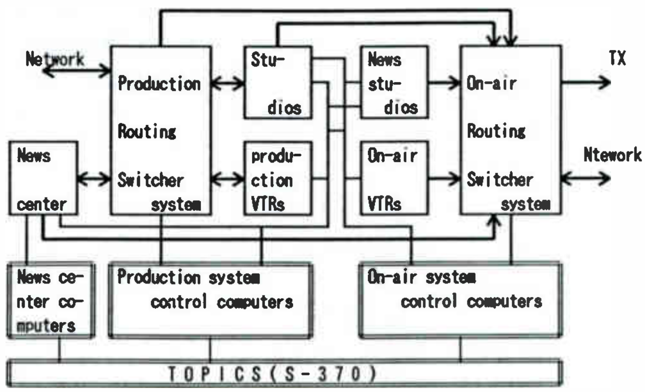

Fig. 1 NHK's routing switcher systems

3.2 Coding Format

Composite video signals are sampled at a frequency of 14.3 MHz, which is four times the NTSC color sub-carrier frequency, and are quantized linearly into 10 bits. The quantization levels are the same as D-2 VTR's, if the least significant two bits are set to zero, in consideration of the direct interface with composite NTSC VTRs.

The sampling frequency of audio signal is 48 KHz and quantization is linear 16 bits.

Item	Video Signal	Audio Signal
1. Sampling specification	14.3 MHz (4*fsc), I/Q axis	48.0 kHz
2. Quantization	Linear 10 bits	Linear 16 bits
3. ID multiplex position	Horizontal blanking period	User's bit
4. Transmission format	Parallel Processing 10B1C	AES/EBU
5. Transmission rate	157.49 Mbps	3.072 Mbps

Table 1 Parameters of Digitization

3.3 Serial Transmission Format

3.3.1 Video Signal

As the composite serial transmission format was not recommended internationally, we decided to adopt 10B1C coding format which had already been applied in common carrier digital networks, according to the evaluation of existing coding methods.

NHK newly developed "Parallel Processing 10B1C" coding method to generate 10B1C bit stream for the use of its in-house signal transmission.

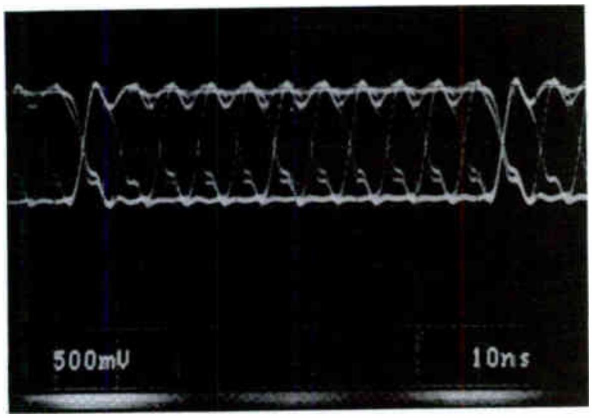

Fig.2 Wave form of 10B1C Signal

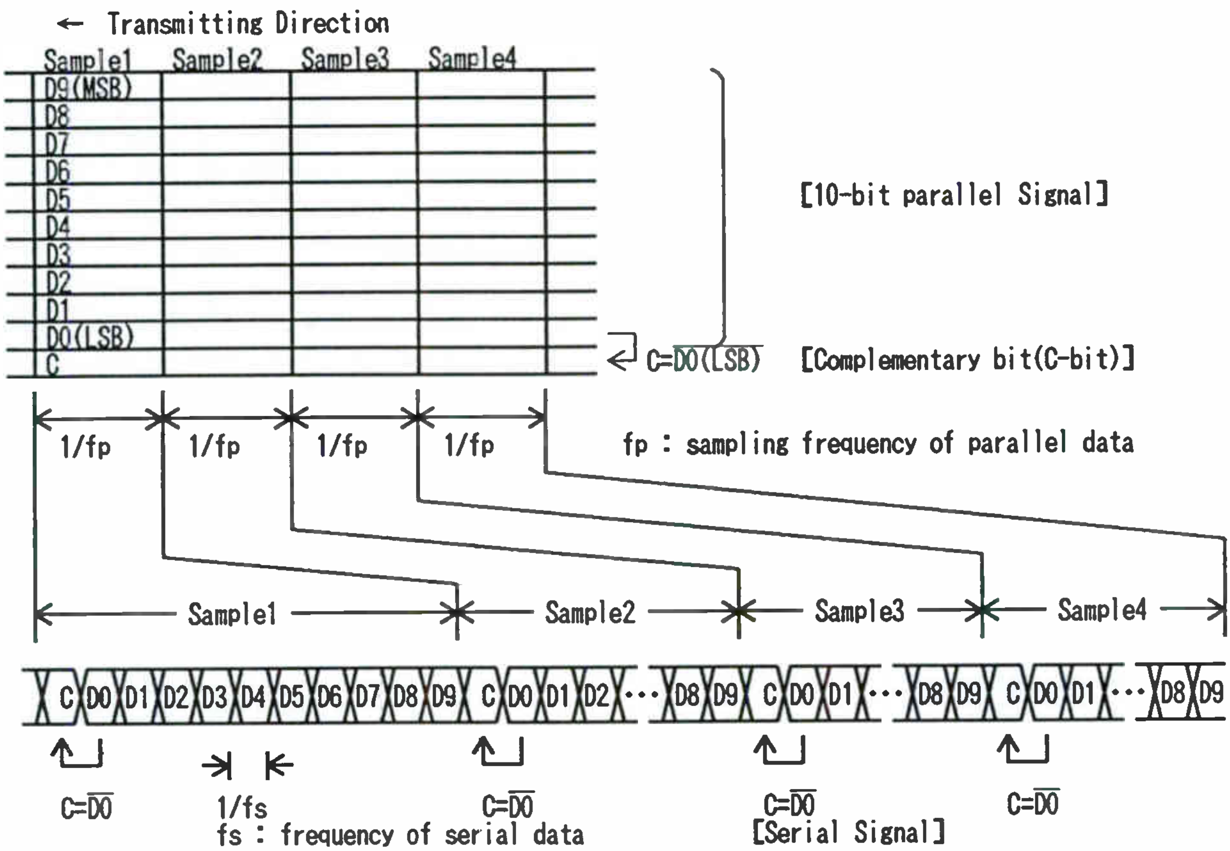

Fig. 3 Concept of Parallel Process 10B1C Coding

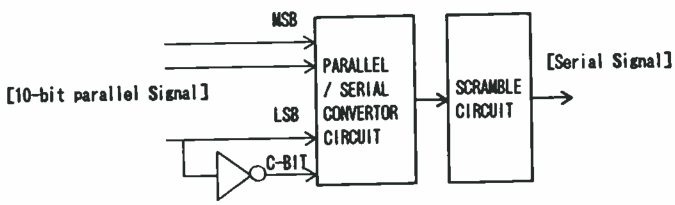

Fig. 4 Parallel Processing Circuit Configuration

A wave-form of 10B1C signals is depicted in Figure 2. Its conceptual diagram and circuit configuration are shown in Figure 3 and Figure 4. The least significant bit (LSB) of parallel digital data is inverted and added to the 11th bit position as a complementary bit (C-bit). This 11-bit parallel signal is converted into a serial signal, and scrambled maintaining the inverted relationship between the LSB and the C-bit.

The aim of the self-synchronizing scramble is to suppress continuation of the same bit polarity, which facilitates the accurate clock recovery at the receiving side as well as the reduction of the jitter caused by the repetition of the same bit patterns. Taking into account the effect of the bit inversion by the C-bit, scramble polynomial order is reduced to a 5th, which minimizes the bit error permeation. The generator polynomial is $(X^5 + X^2 + 1)(X + 1)$.

The main reasons for the choice of 10B1C are as follows.

(1) Stability in transmission

The stability of digital transmission is dependent upon the contents of signals transmitted. Artificially generated computer graphic image, for example, is likely deteriorated in quality because of the continuation of the same bit polarity and the repetition of same bit patterns. To realize this characteristic, 10B1C has a polarity inversion at every 11th bits and low order scramble which greatly reduces the permeation of impairment caused by bit error.

(2) Easy monitoring of signals

To monitor a digital signal under operation, special test signal should be added to the original signal in general. There is no need to add such a test signal in case of 10B1C because of the presence of a C-bit at every pixel. The bit error rate of a virtually error free transmission path is easily measured in a comparatively short time. The periodical presence of a C-bit in a 10B1C signal makes it easy to facilitate the monitoring of the wave-forms using an oscilloscope.

(3) Fast synchronization

A C-bit in every pixel of 10B1C signal works as a timing reference so that synchronization is establish in a very short period of time. In terms of the bit synchronization, the periodical presence of the inversion at the every 11th bit allows to accomplish a very stable bit synchronization.

3.3.2 Audio Signal

The AES/EBU transmission format was chosen to convey digital audio signals.

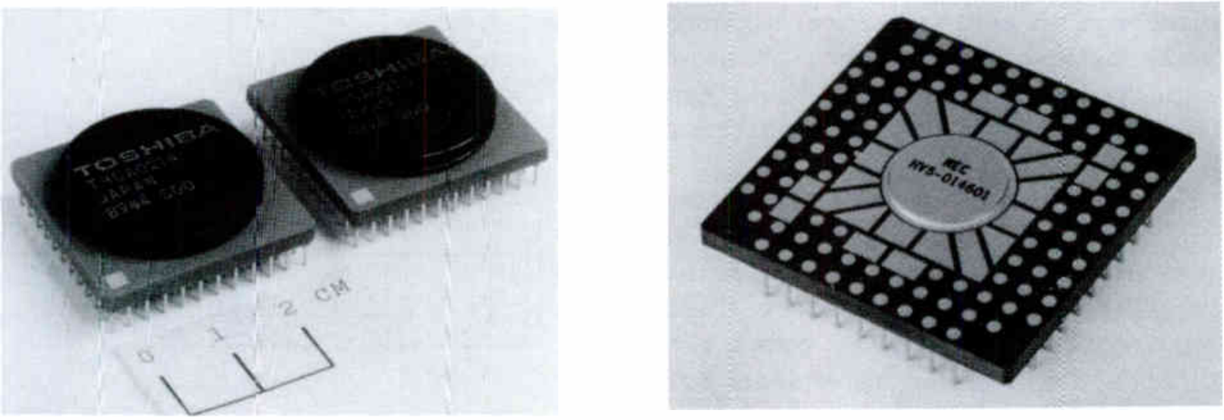

Fig. 5 Newly Developed LSIs

3.4 Ancillary data

Both digital video and audio signals have ID signals, which indicate the sources of the signals and display the resource name. They are also used to check whether the signal switching is done correctly.

4. Whole System Construction

4.1 Newly Developed LSIs

Special purpose gate arrays were developed for these systems. New LSIs are shown in Figure 5.

4.2 Production Routing Switcher System

The production routing switcher system provides arbitrary connections among studios, VTRs, network lines and other program resources for video, audio, telephone and control signals. The conceptual depiction of the production routing switcher system is shown in Fig. 6.

The new system uses an electronic switcher incorporating LSIs instead of the relays used in the old system. This electronic switcher reduces space requirements and permits point to multi-point connections, while the old system only allowed point to point connections.

The bit error rate of signal transmission paths is continuously measured by the complementary bit of 10B1C. Alarms are issued at bit error rates of 10^{-6} and 10^{-8}. A 10^{-6} alarm means that the error is at the threshold of being perceptible on the screen.

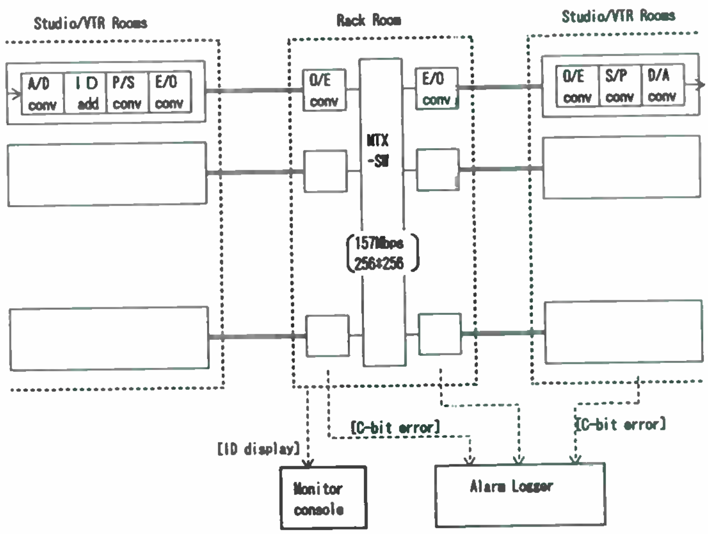

Fig. 6 Production Routing Switcher System

Telephone signals are switched and transmitted in analogue form. Control signals between studios and VTRs are transmitted through the Local Area Network in stead of a matrix switcher.

In the production system, the operation control consoles offer various functions to support efficient program production in studios. These functions include reading tape codes from VTR tapes and putting them into TOPICS, setting tapes at starting points, and checking video and audio signal levels. A manual control console is available for emergency modification of the production schedule, and for the case of breakdown of the automatic control computers. The operation of the main console is supported by a large screen touch panel to input orders, and graphical displays to show connected routes and displays to indicate signal source names for video and audio signals.

The Alarm Logger gathers operating information through LAN and issues alarms both on the message board and by a synthesized human voice. The status of VTRs and studios is also displayed on large screens in front of the operation control consoles.

4.3 On-air Routing Switcher System

The conceptual depiction of the on-air routing switcher system is shown in Figure 7. The on-air routing switcher accepts unsynchronized signals. It then switches them within a relatively short recovery time, during which the system establishes the bit and word synchronization. The 10B1C ensures the short recovery time and high stability.

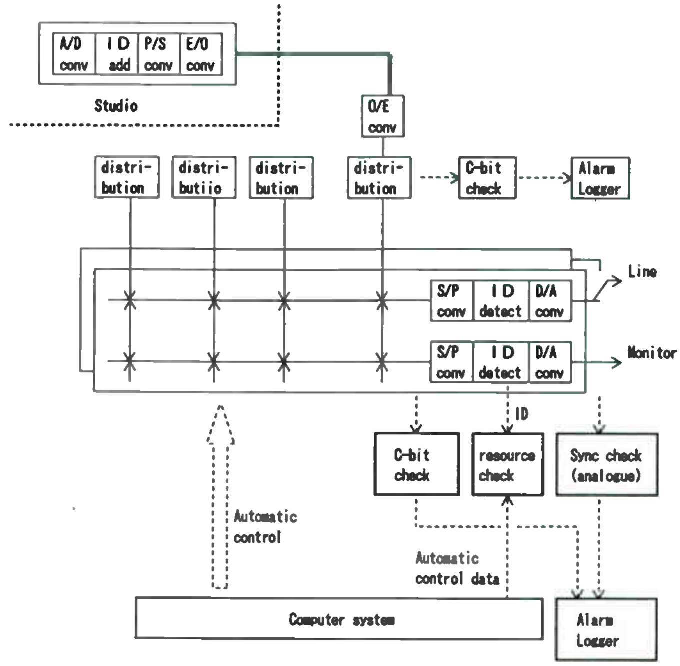

Fig. 7 On-air Routing Switcher System

This switcher has 96 inputs, a maximum of 3 outputs and 2 monitor ranges per channel. One monitor range is used for the signal check by operators, and the other is used for the automatic check of the signal level and the resource ID before it is sent out.

The video switcher includes mixing and keying amplifiers to superimpose news text onto program pictures and to insert the program transmission control signals. The test signals and teletext signals are also inserted into vertical interval lines by these amplifiers. The audio compressor-limiter amplifiers were newly developed using digital signal processors.

This on-air routing switcher is constructed with duplicated circuits. Auto-monitors for video and audio signals can detect the accidental breakdown of circuits and switch over the circuits to continue programs automatically.

The on-air system was designed to aim at more automatic operation than ever. The program signal and the condition of the facilities are checked by the automatic monitoring system which is improved by the C-bit and resource ID check. It is not necessary for operators to monitor the program signals always but only to operate in the case of the modification of program schedule and the breakdown of the facilities.

In the case of urgent news or sports programs, the predetermined broadcast schedule will be often changed by the emergency operation. The urgent news controller and the program scheduler were improved in their functions. There are three operating consoles, and each console can control two channels. These two channels must be selected before each console is used. This method makes the operation of the consoles flexible and robust against malfunctions.

4.4 Automatic Control System

The automatic control system consists of host computers which are the core of TOPICS, data transaction computers (Production Schedule Processor, On-air Schedule Processor), control computers (Production Control Processor, On-air Control Processor) and controllers (Production Switcher Controller, On-air Switcher Controller, Device Controller etc.). These computers are under dual or duplex operation, and are connected to the controlled devices via LANs.

5. Conclusion

NHK's new production routing switcher system was put into operation in July 1990, and the similarly updated on-air routing switcher system in September 1991. Both systems are totally digitized and allow streamlined operations thanks to automatic supervisory functions made possible by the use of the 10B1C coding format. However, we still have to solve the problem of increased power consumption by large scale digital facilities. An optical routing switcher system is the best candidate to overcome the weakness.

NHK has finished the development of a new 1/2 inch digital VTR and plans to introduce them this year. After the replacement of 1 inch VTR with the cart type digital VTR (where tape is loaded and removed automatically), manpower will be saved more. A totally digitalized broadcast system, from the studio to the transmission site, will thus be in operation in the near future.

COMPRESSED DIGITAL VIDEO: A TECHNOLOGY OVERVIEW

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ABSTRACT

Advances in digital signal processing and digital communication would by themselves make digital video inevitable. Compression has accelerated the transition, perhaps by decades. This paper reviews the major processing components in state-of-the-art digital compression, as well as the critical question of how compression systems are evaluated.

INTRODUCTION

Since the 1930's, electronic communication has allowed us to view moving images that were captured at some distant location. Visual imagery is one of the most powerful media for human communication, so the popularity of electronic transmission of moving images (video) is easy to understand. Unfortunately, video is also expensive to move around. The adage "a picture is worth a thousand words" has a surprising technical analogue: standard (analog) transmission of video requires almost exactly 1000 times the bandwidth of a telephone conversation (4.2 MHz vs. 4 kHz).

A series of technical advances has changed the picture, however. Digital electronics and its VLSI based implementations, information theory (a branch of mathematics devoted to representing and transmitting information), digital communications and networking, and the study of how the human eye and brain process visual information have made possible new economies in existing video applications and a whole host of new applications. The key to this emerging revolution in video has been the marriage of digital transmission with digital compression.

This paper provides an introduction to the technical aspects of compressed digital video. In particular, I will describe the underpinnings of digital compression, the basic techniques that can be used for video compression, and how to evaluate compressed video. I will also examine some of the non-video components that are vitally important in compressed digital video applications. Finally, I will look in detail at the evolution of standards in compressed digital video and some existing and emerging applications as well as some of the products that are available.

WHY COMPRESSION?

Compression is justified by economics. If the cost of the communication or storage is decreased by more than the cost of the compression and decompression equipment, yet the quality of the decompressed video meets the requirements of the application, then compression is justified. As this criterion is met in more and more applications, compressed digital video is becoming more widespread and indeed enabling some applications that would otherwise have been inconceivable.

The analog video signal most widely used in the United States is the NTSC signal, with a bandwidth of 4.2 MHz. Following the Nyquist theorem would require sampling at a frequency of 8.4 MHz, and with eight bits per sample, this would lead to a bit rate of 67.2 Mbps. Multiplying by three for the three color components (red, green, and blue) yields a bit rate of 201 Mbps. The difficulty of moving and storing information at such high data rates is easy to see. For instance, current floppy disks hold about 1.44 MByte of data, or 11.5 Mbits. A typical 100 minute movie would require over a 100,000 floppy disks. Transfer rates between hard disks and computer CPUs are on the order of 2 MByte/sec or 16 Mbps, more than 10 times too slow for the required data rate. CD-ROMs provide a transfer rate of around 1.5 Mbps, over a factor of 100 times too slow. The highest speed modems for voice-grade channels in the public telephone network operate at less than 20 kbps, so we would need 10,000 such channels to carry the signal. Many of the most interesting applications of video would continue to be inconceivable without compression. Emerging compression applications are already displaying video from computer hard disks and CD-ROMs, and CLI has a videophone under development that will allow voice and video communication over a single voice-grade telephone line.

VIDEO COMPRESSION TECHNIQUES

Significant advances in digital compression research became possible with widespread availability of digital computers starting in the late 1960's. The 1970's and early 1980's were marked by advances in algorithms and in the components required to implement them. The late 1980's have seen the beginning of widespread application of compression to video and a drive towards standardization.

In the many different approaches to video compression studied by researchers, several key elements appear over and over again. Although their are different algorithmic approaches and many trade-offs associated with each of these elements, they appear in almost all video compression systems in one guise or another. Figure 1 shows a block diagram of a basic video compression encoding algorithm. The various components are described in the following sections.

It is important to understand that not all video compression techniques employ all the processing steps described below. It is quite possible to simply avoid motion compensation, or not use a frequency domain decomposition, in order to reduce implementation complexity. However, it is also true that each of these components is well within the capability of contemporary technology, even in very simple, consumer oriented applications. Systems that choose not to employ some of these techniques suffer significant performance disadvantages, yet show little cost benefit. This trend is expected to continue as implementation costs fall with improving chip based technology.

Pre-processing

Pre-processing is generally an attempt to remove information from the video signal that is most difficult to code yet relatively unimportant to the visual quality. Hence, pre-processing is closely tied to both human perception and to identifying signals which will be hard to code. Typically a combination of spatial and temporal non-linear filtering is used. Judicious use of pre-processing according to application is as much an art as science and is most effective when solid experience is married with some careful experimentation using the particular system and application being developed.

Temporal Prediction and Motion Compensation

Video sequences are usually highly correlated in time. In other words, each frame of the sequence is quite similar to the preceding and following frames. So, taking the difference between frames and coding the difference instead of the entire original frame means a lot less information needs to be sent. Simple frame differencing can be improved by noting that many of the changes that occur from frame to frame can be approximated as translations of small regions of the image. By breaking a frame into small blocks (typically 16 x 16 pixels) and searching at nearby positions in the previous frame, it may be possible to find a very good predictor block, so that only the position of the predictor block and the relatively small differences between the predictor block and the current block need be sent. Because this process helps greatly reduce the bit rate when portions of a scene are in motion, it is typically called "motion compensation" (Figure 2).

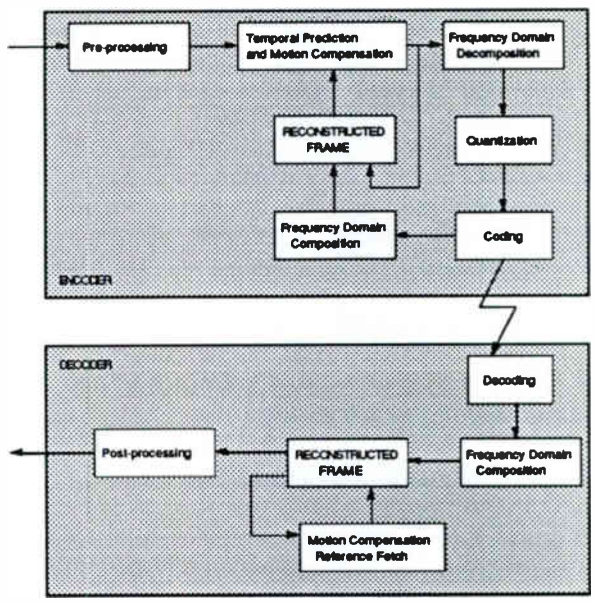

Figure 1. Compression encoder and decompression decoder for compressed digital video applications.

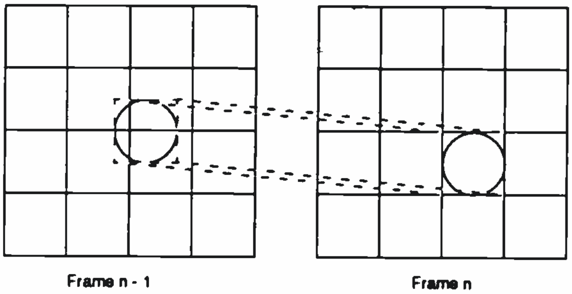

Figure 2. Motion Compensation. A block in frame n is best predicted by a small offset block in frame n - 1. The encoder must send the offset and information about the difference between the predictor block and the current block.

The use of motion compensation is sometimes blamed for the introduction of "motion artifacts". This is a misconception. Motion compensation *does* yield more efficient compression, since much smaller differences need to be communicated. However, it introduces no distortion by itself. Indeed, if the differences are communicated perfectly to the receiver, then the video sequence can be perfectly reconstructed. So-called "motion artifacts" can arise when insufficient bits are available to describe detailed and/or rapidly changing scenes, but can also occur for a number of other reasons, such as conversion from film to video, or use of a single "merged" frame to represent two fields of video, or when a non motion-compensated system suffers varying quality due to varying scene detail, and hence are not specific to motion-compensated systems.

Frequency Domain Decomposition

The next step is to find a frequency representation for the signal representing the difference between the best offset in the previous frame and the current frame (the "motion compensation residual"). The signal is analyzed into two-dimensional frequency components, much as a spectrum analyzer determines the frequency components of a one-dimensional signal. This has two advantages. First, the signal often has most of its energy concentrated in a small range of frequencies (typically the lower frequencies), so that very few bits need be used to describe unimportant frequencies. Second, this frequency domain decomposition mirrors the processing of the human visual system, and allows the following quantization step to be tailored to the sensitivity of the human visual system to frequency content.

By far the most popular frequency domain decomposition technique is the discrete cosine transform (DCT) (Figure 3). This relative of the discrete Fourier transform takes a block of the motion compensation residual (typically 8 x 8 or 16 x 16 pixels) and converts it to a corresponding set of coefficients, representing different frequency components.

$$\begin{pmatrix} 22 & 24 & 27 & 28 \\ 23 & 24 & 28 & 29 \\ 24 & 25 & 29 & 30 \\ 25 & 27 & 30 & 32 \end{pmatrix} \xrightarrow{\text{DCT}} \begin{pmatrix} 106.7500 & -10.0604 & -0.2500 & 8.8566 \\ -4.7877 & 0.4268 & -0.3266 & -0.1652 \\ 0.7500 & -0.0560 & -0.2500 & -0.6533 \\ 7.5193 & -0.5901 & -0.1353 & 0.7647 \end{pmatrix}$$

Figure 3. A 4x4 DCT Example. Note that the relatively flat input block produces a transform block with most energy in the upper left hand corner. Most of the lower right hand corner coefficients can be set to zero with little effect on picture quality.

Quantization

After motion compensation has reduced the signal to be transmitted and the frequency domain decomposition has concentrated the power into a few frequency coefficients and arranged the energy in a way similar to the human visual system, quantization is the process of irreversibly reducing the amount of information that will actually be provided to the decoder about the signal (Figure 4). Because of the previous steps, we can discard many bits without significantly degrading visual quality.

Fundamentally, quantization takes each coefficient from the frequency domain decomposition and reduces the precision with which it is described. For instance, if a coefficient can range between -127 and 127 as an integer, requiring 8 bits to send it as is, we can instead group the numbers -1, 0, 1 together and reproduce them as 0 at the decoder, group the numbers from 2 to 4 together and reproduce them as 3 at the decoder, and so on. By doing this, we identify 85 different groups of numbers, and all the decoder has to know is which group we have selected. That only requires 7 bits to send (actually, we can group quantizations together and reduce the bit rate to the true minimum of 6.4 bits per sample). The compression effect is obvious.

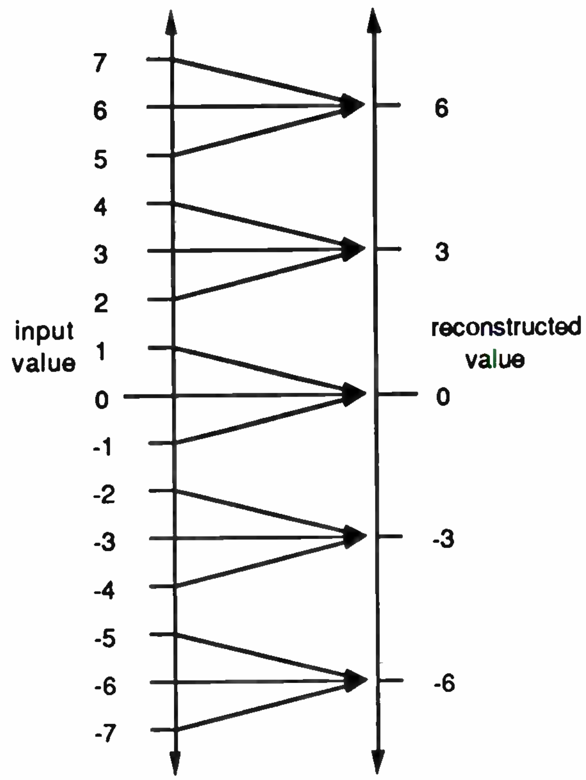

Figure 4. Diagram of a uniform quantizer. The step size is three since the range of possible inputs is divided into bins of width three with a single reconstruction for each bin. The step size controls the trade-off between bit rate and visual quality and can be adjusted dynamically to respond to system loading, for instance, during an extended period of difficult-to-compress source material.

Vector quantization (VQ) applies the quantization process to more than one coefficient simultaneously (Figure 5). A set of coefficients are considered as a vector, and a search takes place to find one of a small set of representative vectors that is close to this input vector. The collection of possible representative vectors is called the codebook and is available at both the encoder and decoder. Once the best representative is found, its index in the codebook is transmitted. Since the size of the codebook is much smaller than the number of possible input vectors, the number of bits required to transmit this index is much less than the number of bits that would be required for direct transmission of the input vector. VQ has a strong theoretical advantage: Shannon showed that in the absence of complexity constraints, VQ can come close to the theoretical limit on data compression performance. Unfortunately, VQ has a strong practical disadvantage: when complexity constraints are included, the performance advantage of VQ by itself (i.e., without frequency domain decomposition) can be overwhelmed by lower complexity systems using good frequency domain decompositions combined with good scalar (non vector) quantization and coding.

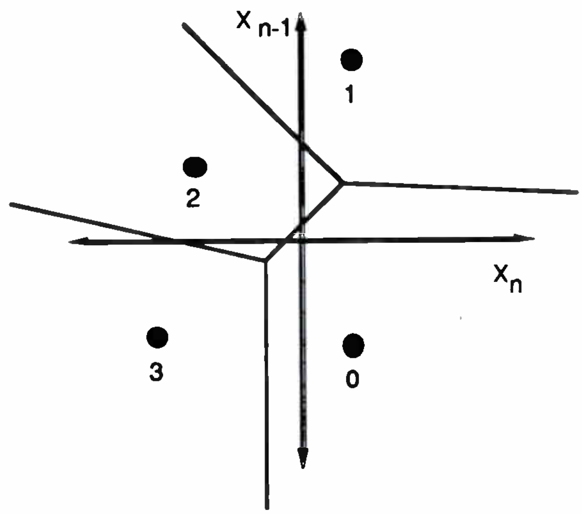

Figure 5. A two-dimensional vector quantizer with four codewords. Each pair of inputs (x_n, x_{n-1}) is quantized simultaneously by determining which bin the vector falls in and sending the index of that bin to the receiver. The receiver looks up the reproduction vector for that bin. Complexity of the search to determine which bin a vector lies in scales exponentially with codebook dimension at a fixed rate.

There is nothing that says VQ can't be combined with frequency domain decomposition such as the DCT or a pyramid decomposition. Indeed there has been a great deal of research into such combinations. Again, however, relatively few such combinations have been successful relative to combinations of the DCT with standard quantization, and none have been selected as the basis for standardization. In particular, the combination of run-length coding (next section) with scalar quantization exploits most of the inter-coefficient redundancy that can be exploited by a combined DCT/VQ system in a much simpler fashion. VQ systems which are not coupled with any of the other contributing techniques, such as no motion composition or no frequency domain decomposition, are at a dramatic disadvantage in terms of compression efficiency versus complexity, and can be expected to be successful in very few applications.

Coding

The quantized frequency domain coefficients tend to have the value zero at many different frequencies, and large groups of such zeros are clustered together at the higher frequencies in many cases. Hence, additional compression is often achieved by coding the number of zeros in a row rather than coding each individual zero by itself. Such coding is called run-length coding. Non-zero coefficients are coded by individually.

The non-zero quantized coefficients and the number of zeros in a row can each yield to further compression since some possibilities occur much more frequently than others. By assigning a short codeword to frequently occurring possibilities and a long codeword to infrequently occurring possibilities, a net savings in bit-rate can be realized. The most popular technique in video compression for achieving this additional compression is Huffman coding (Figure 9), although arithmetic coding and Lempel-Ziv coding (more commonly found in compression of computer files) can also be used.

Symbols	Probability	Fixed Rate Code	Huffman Code
A	0.5	00	0
B	0.25	01	10
C	0.125	10	110
D	0.125	11	111

Fixed Rate Code Rate = 2 bits/sample
Huffman Code Average Rate = 0.5(1) + 0.25(2) + 0.125(3) + 0.125(3) = 1.75 bits/sample

Figure 9. A Huffman Code Example. Huffman codes provide their most dramatic savings on sources with large probability on a relatively small number of symbols.

Decoding

Decoding undoes the operations of the encoder in a predictable way. The run-lengths of zeros and coefficient values are looked up and the reproductions that the quantizer in the encoder assumed are used to reconstruct the coefficients. The coefficients are applied to a frequency combining process which undoes the frequency decomposition and are added to the prediction that the encoder used during motion compensation, resulting in an output frame.

Decoding typically has significantly lower complexity than the encoding. This asymmetry is partially offset by the fact that error-correction, if performed, typically requires more work at the receiver than at the transmitter.

Post-processing

One additional step that is sometimes performed at the decoder is post-processing. This is typically an attempt to recognize the most annoying artifacts that may occur and reduce their visibility directly. Post-processing is generally limited to situations with fairly high distortion levels, since in low distortion situations it is difficult to successfully reduce visibility of artifacts without simultaneously distorting the original signal in some other annoying manner.

EVALUATING COMPRESSED DIGITAL VIDEO SYSTEMS

Evaluating and comparing compressed digital video systems differs in significant respects from conventional analog video systems. The objective measurements that have evolved for analyzing analog video systems are intended to predict the subjective quality level of a received video signal. They measure the types of signal impairments that were observed empirically to correlate well with picture quality degradation in an analog transmission environment. Unfortunately, these measures and associated techniques do not extend well to effective evaluation of compressed digital video systems. Instead, we will have to return to the basic goals of our evaluations and derive new measurements and procedures that will accurately determine the appropriateness of a given system in a given application.

The three principal areas of evaluation for compressed digital video systems are rate, distortion, and cost. Cost needs no special elaboration. Other important attributes are flexibility, robustness, and compatibility.

Rate

The capacity of a digital communication system is measured by its rate in bits per second. An efficient compressed digital video system consumes the minimum rate possible commensurate with its quality objectives. For the most part, a compressed digital video system can be considered separately from the communication system that will be used to deliver the bit stream. Consequently, a compression system should usually be quoted in terms of its required rate in bits per second unless the compression system and its associated communication system are being delivered as a package. In this latter case, resources consumed by the digital communication system (e.g., bandwidth and power) are the appropriate measurements.

One of the most widely misused statistics in the video industry is the compression ratio (e.g., a "100:1

reduction in data"). Unfortunately, compression ratios are often taken out of context and become confusing and misleading. The difficulty is that the ratio is highly dependent on what the supposed raw (uncompressed) data rate was. In a previous section, I used a data rate of 201 Mbps as the raw data rate for NTSC. It would be equally valid, however, to consider sampling the NTSC signal in the color space that is used for transmission (YIQ) rather than the color space used for display (RGB). NTSC specifies that the I and Q components have less bandwidth than the Y component, 1.2 and 0.6 MHz respectively, so they may be sampled at a lower rate. Nyquist sampling in this color space yields a total rate of 96 Mbps. So, a compression ratio based on the NTSC YIQ color space can be instantly doubled by shifting to an assumption that the raw signal is carried in RGB. Yet another compression ratio would result if the signal was sampled according to the CCIR-601 specification.

This is only one example of any number of assumptions that go into compression ratio calculations. The picture is further clouded when compression ratios are quoted without concern for loss of visual fidelity (next section). Is a 1000:1 compressed picture that is unrecognizable more impressive than a 100:1 compressed picture that looks pretty good?

Distortion

Distortion measures the degradation in a reconstructed video signal relative to the original.

Lossless Compression

A critical design decision in developing a compressed digital video system is whether the compression should be lossless or lossy. Lossless compression guarantees (in the absence of channel errors) that the digital input to the compression system and the digital reconstruction at the output of the decompression system will be identical down to the bit level. Lossy compression does not provide this guarantee.

On the face of it, lossless compression would seem to have a clear advantage. A well developed class of compression techniques is available for lossless coding. Such techniques have become quite popular in compressing data files on computers, where the lossless criterion is vital, and include Huffman coding, Lempel-Ziv coding, and arithmetic coding.

Two important factors make lossy coding better for most compressed digital video applications. First is that limitations of the human visual system mean that a reconstruction that differs from an original may appear the same. Lossy coding which results in a picture which is different at the bit level but nonetheless visually indistinguishable from the original is called transparent. Transparent lossy coding can typically be done at around one fourth the bit rate of lossless coding. Beyond transparent coding, the tremendous economies of compression make it acceptable to have some visual distortion if a low enough bit rate can be achieved. Once distortion is above the threshold of human perception, a careful tradeoff must be made in system design in order to minimize the overall distortion yet achieve the bit rate at which the application becomes economical.

The argument for lossless coding in compressed digital video applications is weakened by a second factor. From the system perspective, it is impossible to deliver the image as viewed by the camera losslessly to the eye of the viewer even if the digital compression and transmission is itself lossless. The camera, analog video processing, and digitization are inherently lossy, as is the digital/analog conversion and display processing at the receiver. The notion that lossless digital compression is somehow preserving a perfect system is misleading. In fact, every other component in the system is introducing distortion, and the system designer has already undertaken a tradeoff in cost vs. performance in selecting them. Similarly it is reasonable to make the same tradeoff in the compression system. If the distortion introduced is low relative to the overall system distortion budget, yet the transmission cost savings are significant, then lossy compression is the right choice.

Simple Distortion Measures

Standard "simple distortion measures" in signal processing are mean square error and the associated signal to noise ratios (which differ on how "signal power" is defined). Such measures are useful in comparing systems of essentially the same type where only minor parameters are being varied. They are quite unreliable in determining perceived quality differences for systems which take different approaches to compression (e.g., a simple DPCM system and a full-blown DCT based system with pre- and post-processing).

A larger class of simple distortion measures are those that have been developed through years of engineering experience in designing and maintaining facilities for production and transmission of the analog NTSC, PAL, or SECAM broadcast TV signal. Unfortunately, these measures tend to be inappropriate and even misleading when applied to compressed digital video systems. They may prove useful in checking the performance of the input and output analog circuitry, but provide little help in evaluating the core compression performance. For instance, differences in delay between luminance and chrominance only arise in the analog circuitry. Once digitized, the luminance and chrominance delay does not change, regardless of transmission length or characteristics. On the other hand, standard analog television signal to noise ratios can be very misleading. On a black input field, a digital compression system is likely to produce a nearly perfect black output field (again only limited by input and output circuitry) resulting in very high signal-to-noise ratios. On regular video, however, the SNR can vary dynamically and dramatically. However, such variations may occur with no perceived impact on visual quality, since the compression designer has intentionally limited the distortion to areas of the video where it cannot be seen by a viewer.

Perceptual Distortion

The level of sophistication employed in compressed digital video system design is such that no simple yet accurate objective engineering measurements are available for uniformly evaluating performance. This is inconvenient and perhaps uncomfortable for many practicing video engineers, but the truth of the matter is that compressed digital video applications will probably have to be evaluated by eyeballs for some time to come, i.e., if the system looks good to the customer, then it *is* good.

Such tests can vary from small groups of engineers and managers making the decision by themselves to full blown statistical subjective testing with large groups of subjects and careful experimental control. The latter is generally expensive and time consuming, and hence will likely be used in only a limited number of cases (for instance, the FCC HDTV tests).

Artifacts

In spite of the difficulty in attaching numerical measurements to perceived distortions in compressed digital video systems, several classes of distortions are common to many systems operating at high compression ratios. Typical among these are: (1) loss of sharpness, (2) granular noise ("dirty window effect"), (3) motion judder, (4) busyness around edges ("mosquito noise"), (5) appearance of block boundaries, (6) chroma bleeding. Systems can be informally evaluated be the amount to which such artifacts are visible for a given input video sequence.

Research Directions

Everybody will be happier if we can find a small set of easily measured values which will accurately predict perceived video quality for compressed digital video systems. In all honesty, though, this is going to take some time and a good deal of effort to develop. Two basic approaches are (1) to use our increasing knowledge of the human visual system to develop direct tests for visual quality and (2) to take a very large set of possibly useful measurements and attempt brute force correlation against perceived quality measured using statistical subjective testing. Both of these approaches will take some time to bear fruit and it is very likely that the first generation of video compression equipment will be employed in many applications without their benefit.

Other Aspects

While visual quality is a challenge to evaluate, other aspects of compressed digital video are easier than the analog video counterparts. Since the compressed bitstream is transmitted digitally, the quality of the transmission can be completely described by the bit error statistics. In well designed digital communication systems, integrity of the transmitted bit stream is relatively easy to maintain. This contrasts to analog transmission systems, in which many linear and non-linear distortions as well as additive noise can cumulatively degrade the signal.

Robustness relates to the behavior of the compression algorithm in the presence of uncorrected channel errors. Error concealment techniques can be incorporated into decoders to reduce the visibility of errors. Flexibility captures the ability of a compression system to extend to new applications. Compatibility captures the extent to which a system can interoperate with other systems.

STANDARDS AND PRODUCTS

Standards efforts are underway in several application areas.

Multimedia: MPEG 1

Multimedia is the mixture of a variety of media types on the personal computer. The data types that a user might typically wish to manipulate are text, graphics, pictures, speech, audio, and video. Of these, video is the most demanding in terms of storage resources and processing power. Fortunately, the CD-ROM is a relatively inexpensive way to provide very large storage capacity. The CD-ROM is a data storage device built from the same basic components as the audio CD player. Since the device shares volume with this highly popular product, CD-ROM drives and disks are inexpensive. However, the CD-ROM also inherits the audio CD's data transfer rate of approximately 1.5 Mbps. This has led to the desire for a video compression algorithm suited to storage and multimedia applications with this kind of transfer rate. The International Standards Organization (ISO) formed a working group, the Motion Pictures Experts Group (MPEG) to develop such an algorithm. The specification of the standard is basically complete, and several chip manufacturers have committed to development of MPEG chips or chip sets. The MPEG 1 algorithm can be applied to larger frame sizes and data rates, and hence can also form the basis for compression systems aimed at broadcast quality video.

Entertainment: Cable TV

Cable Television Laboratories (Cable Labs), in conjunction with cable operators and programmers TCI, Viacom, and PBS, has asked for proposals to develop compressed digital video systems for both delivery from studios to cable head-ends and from cable head-ends to consumers. Although Cable Labs does not set standards, it makes recommendations to the cable operators which fund it. The development of the studio to head-end system is expected to result in deployment of the studio to head-end system in 1992, with the head-end to consumer system following in 1993 or 1994.

Entertainment: MPEG 2

An extension of MPEG 1 is now underway to address higher quality considerations. This standards effort, MPEG 2, is in the nascent stages, with some continuing debate about the exact target of the standard. Currently, two main applications are envisaged: good quality entertainment video at 3 - 5 Mbps and studio quality video at 7 - 10 Mbps. The level of compatibility with MPEG 1 is not yet settled, and may range from using the same algorithm, to an entirely different algorithm of which no part can be decoded by an MPEG 1 decoder.

HDTV: The FCC Trials

Currently, four different groups are proposing a total of six systems for standardization in the United States. Four of these systems are based on digital compression. The FCC is overseeing a testing procedure funded by the broadcasters, television receiver manufacturers, and cable companies in the US and Canada. Each of the six proposed systems will be tested for several months, with tests ending in mid-1992 and a final decision due in 1993. So far the process has been full of surprises, so that it is not prudent to expect progress towards this final decision to be orderly.

The FCC is standardizing an emission standard, i.e., a standard for use by current over-the-air broadcasters. The cable industry has significant input through the Cable Labs consortium, but other interested industrial groups, such as the computer industry, have less input. Once HDTV is standardized, the penetration rate into consumer homes will depend to a large extent on the cost and availability of HDTV receivers with screens large enough to exploit HDTV's higher resolution.

Products

A number of products are likely to emerge to address full resolution television transmission needs in the next few years. As of the writing of this paper, the only available product is CLI's SpectrumSaver, a motion compensated DCT product for delivery of video over satellite. Other first generation products are likely to emerge as traditional suppliers to the cable companies prepare product offerings in response to the Cable Labs RFP. Beyond this, product availability in the broadcast industry will be strongly effected by regulatory developments.

THE BROADCAST INDUSTRY

While the development of HDTV will have a profound and much discussed effect on the traditional broadcast industry, the application of digital compression to standard resolution NTSC has received relatively less consideration. The wild card in digital compressed broadcast NTSC is the terrestrial broadcast environment. No widespread, cost-effective system for broadcasting in strongly multipath environments as yet been demonstrated. However, this is one of the problems being solved by the FCC HDTV proponents. Given a transmission solution, it is quite conceivable that the same scheme could be used to transmit multiple compressed NTSC channels in 6 MHz of spectrum.

The FCC proponents proposals suggest that the terrestrial broadcast channel can support between 10 and 20 Mbps for reasonable coverage areas. With this bit rate, we can expect to provide 2 - 3 super quality channels, 4 - 5 standard quality channels, and up to 10 VHS quality or movie only channels. Whether broadcast spectrum will be made available for these types of services remains to be seen.

TO PROBE FURTHER

Textbooks:

Netravali and Haskell, *Digital Pictures: Representation and Compression*, Plenum Press, 1988.

Lee and Messerchmitt, *Digital Communication*, Kluwer Academic Press, 1988.

Journals:

Signal Processing: Image Communication, Elsevier Science Publishers B.V., P.O. Box 1991, 1000 BZ, Amsterdam, Netherlands.

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Conferences:

IEEE International Conference on Acoustics, Speech, and Signal Processing. Held in April - May each year.

SPIE Conference on Image Processing and Visual Communications. Held in November each year.

EIA Workshop on Digital Video Processing. Held in October or November each year.

A NETWORKING SOLUTION FOR STILLS AND GRAPHICS

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ABSTRACT

The requirements for graphics networks varies widely. The numbers of machines to be connected and the types to be used together rule out a straight forward single networking solution; rather a set of solutions that can co-exist are chosen to piece together systems that can grow from a small, local facility a large arrangement operating on a global scale.

INTRODUCTION

Stills stores and graphics systems are often required to work in conjunction with each other, together forming a graphics preparation, storage and presentation system. As systems have evolved over the last decade their popularity has increased and the part they play in TV production become all the more important. Networking is an obvious way of combining the machines involved but there are a number of possible approaches which can be used to achieve this. The correct one is the one that works best for any particular required situation, fitting the size and type of operation ... and the budget.

Involvement with still stores and graphics systems such as the Paintbox has naturally led to the connecting of these individual systems together. Rather as the configuration and mix of equipment in a TV facility is custom designed to meet the requirements of each user, so it is with stills and graphics; each whole system is used in a different way and the users' priorities will vary. Any network design must be able to accommodate a wide range of requirements. The solution described addresses all these factors to produce the best results possible with currently available technology.

The discussion concentrates on networking and deliberately leaves aside other peripheral topics such as data compression and the use of component or coded video. Throughout the paper it is assumed that pictures are not compressed and are coded in the CCIR 601 4:2:2 component format at 8 bits resolution: the form most appropriate and commonly used in TV graphics and still store systems. A 525 line picture then comprises 700 KBytes of data and some 100 KBytes more can be allowed for other data for titling and management functions including browse images.

REQUIREMENTS FOR A SUCCESSFUL NETWORK

Although the individual needs of each user may vary there are requirements which all will look for. These give a measure of the quality of the network.

Expandability

The number of machines to be connected to a network will vary from user to user. The ability to start with an economic small system and grow to any size is an obvious advantage. At the same time there should be no redundancy as parts are added to make up a larger system. Expansion may be in the number of machines or the storage capacity connected to the system. As it grows so there must be the ability to manage the larger numbers of stills and the connections between units. Finally it should be possible to join whole networks together.

In some cases it may be important to adapt the system to changing demands, including the removal of some units for temporary use in another area. In such cases the ease of adding or removing units is crucial.

Speed

There is, quite rightly, much discussion about the speed of network systems. What is important is not to look at just one component but the system as a whole coupled with its configuration and the proposed method of operation. Only then will the true speed specification emerge.

It is always an advantage to have a fast, responsive network so that stills can be available in a required area in the minimum of time. Exactly how long this will take depends principally the times taken for a series of operations. The usual method for making a picture available is to copy it from a disk in one system to the disk in another. The total time will, at least, depend on all of the following:

The time to find the required picture.
The time to read the picture from disk.
The time to transmit it over the network.
The time to write it to disk.

The management system used to find the picture must be capable of efficiently supporting both the large number of stills to be stored and accommodating all the connected units. Certainly a first level priority should be the ability to find one picture from 10,000 within a second. This requires that keyword searches of full title lines must be provided rather than using the exact letter codes required by DOS or Unix. This may be supported by a browse facility for a quick view of the pictures of the found titles.

Disk read and write times will vary according to the type of disks being used but speeds of 1-2 MBytes/sec are commonplace. This implies that a picture could be written or read from disk in around ½ sec. Optical disks, which are now popular, are slower with typical picture read times of around 2 seconds and writes taking around twice that time.

The time taken to transmit the picture over the network will depend on the method used. This is discussed is more detail later but current methods provide times ranging from 0.033 to over 20 seconds or more.

Reliability

A network is a large system and it may be responsible for the entire handling and presentation of stills in any location. With such dependence in one system reliability is of prime importance. Certainly each connected machine should be able to continue operation even when the network has failed. Even better, it should still be possible to move pictures around between machines - possible with compatible removable media. Such autonomy can only be achieved if each connected station is an independent unit with its own disk drive and computer system. In this way the network exists as an addition to the unit, not as a fundamental part of it.

Distance

It is increasingly the case that equipment is spread out around a station rather than being solely concentrated in a central apparatus room. Being able to connect a network over some distance is an obvious advantage. Being able to connect over standard communications links gives the potential for extending the network to a global scale.

Ease of Use

The principle requirement of operation is to pass a picture from one location to another. It should be possible to present the network as just another source and destination for pictures. The complexities of exactly how this is done should be hidden from the operator. Ideally there should not be the need to add another control station as the network control should be integrated with the local operation.

Range of Equipment

Graphics and pictures are today generated and handled by a wide range of machines including computer graphics and 3D equipment. It would be advantageous to include all such machines on the network but the fact is that not such a high degree of standardisation of formats exists with computers as we enjoy in the TV industry. There are many formats and interfaces to design for if these are all to be included. They may render the whole system too expensive to set up and test, or slow, by having to use low speed interfaces.

THE MEANS OF CONNECTION

As with other video equipment there has been a shift from analog towards totally digital interfaces. Indeed since graphics machines and still stores were born of the digital age, digital connections have always been a priority and only these are now considered for networking. The computer and video worlds both offer a number of methods for making the necessary connections but making the network work well can involve rather more than just taking an available off the shelf solution.

The link should be easy to implement, work over long distances, be fast, reliable, simple to connect up and operate with every known type of relevant machine. Considering that each picture 525 line TV represents around 700 KBytes of data and that there is a very wide variety of equipment in use, this is a tall order.

Digital Video

There are parallel and serial interfaces for CCIR 601 component digital video. Clearly these are fast, transferring a picture every 0.033 seconds. The parallel connection carries the penalty of expensive cable with range limited to 50 or 100 yards and an overhead in video switching. The serial solution offers improvements for distance and cable costs. Using either, as the link is strictly uni-directional, for point to point operation, and a video switching matrix will still be required. In addition, control data must also be passed around the network, therefore another network will need to exist on top of the video.

SCSI

This is a widely used interface for small computers and is usually applied for connection to the disk system. It is reasonably fast, operating at up to 4 MBytes/sec, but is limited by its short length and a maximum of only eight addresses. The fact is that there are few manufacturers offering SCSI interfaces for third party machines so further restricting its use in a network situation. Even so, despite its limitations, it can be used very cost effectively where only a small number of units are required to be connected in a close area. In such cases it has been appropriate to use a custom made protocol, optimised for this application, which enables the full speed of the link to be realised. Picture transfers of ½ a second or less can be achieved using the faster SCSI disks. Such networks, or clusters, can fulfil the requirements of many small to medium sized installations.

Ethernet

The computer industry already uses a number of types of network. Among them ethernet is of particular interest as it offers the advantages of a high-speed, bi-directional link using a single coax cable with no insurmountable problems for operation over any distance. Even at its basic level thin wire ethernet has a range of some 600 feet. This can be extended with the repeaters and bridges which are all a part of the wide range of support items which are readily available. In fact it is possible to communicate over the whole globe. It is multi-drop and so can be distributed to numerous machines without the need for a switching matrix or distribution amplifiers. Being a data network it can carry all the control commands, as well as the picture information, on a single wire.

At first glance ethernet may seem to provide the ideal connection especially when you realise that many of the computer based graphics products interface to it. But it does have its drawbacks. Operating at 10 Mbps is usefully fast as a computer link, but can be slow for the transmission of TV pictures. The industry standard protocols such as TCP/IP and Decnet slow the effective data rate so that sending a picture can take of the order of 20-30 seconds. That's usually too slow for a TV environment. Specially developed protocols have reduced the time to less than three seconds bringing the speed back into line with the requirements for networking TV pictures. The cost has been that the protocol is not used by a wide variety of graphics equipment so the range of machines that can be connected is limited. However, if the design of this special protocol is implemented with thought, it is possible to let picture and computer file transfers co-exist on the same network.

THE NETWORKING SOLUTION

The logical conclusion is that there is no single ideal solution for networking. Looking for all the qualities of expansion, speed, reliability, distance, ease of use and the range of equipment that can be used, no one method of connection offers all the right answers. The best results are gained by taking advantage of the optimum features of each method and applying them where most appropriate. As a result the system devised employs a mix of several networking solutions, each dedicated to the most appropriate sector. In this way the total operation is optimised.

The Dual Bus System

Before describing the chosen solution there needs to be some attention given to the design of the individual machines used and the configuration of the network. Primarily there will be a number of Picturebox still stores and some Paintbox graphics systems, and it is paramount that the network operates most efficiently with these. Each contains their own computer system, disks and library management making them autonomous units. Fundamental to their design is the provision of two SCSI busses to provide both a public and private bus for disks and other interfaces. This dual bus structure allows the locally used pictures to be held quite separately from those provided for use in the public domain. The two buses cannot interfere with each other. The public bus has been supplied specifically for networking so leaving the local bus, with its own disks, quite separate from the network activity.

The network operates by adding a central store with which all units exchange pictures. In this configuration all traffic passes though the central store, which is effectively a file server, and not directly from machine to machine. Although the store appears as an overhead it does offer a number of important advantages which have made it the basis of the chosen system:

1. It allows each machine to continue work without the danger of any interruption from a third party. No one can interrupt operation on the private bus.

2. Only the pictures copied to the central store are available to others. In this way each operator can exactly control what pictures are used outside his machine.

3. The central store is separate from the others and so can be built into a large library without burdening any of the connected machines. Expansion beyond the capabilities of the individual machines is possible for all can share its facilities.

Small Networks- Shared User Bus

Using the public SCSI bus it is possible to create a small network. In hardware terms this only requires the addition of a shared disk with which all connected units can then

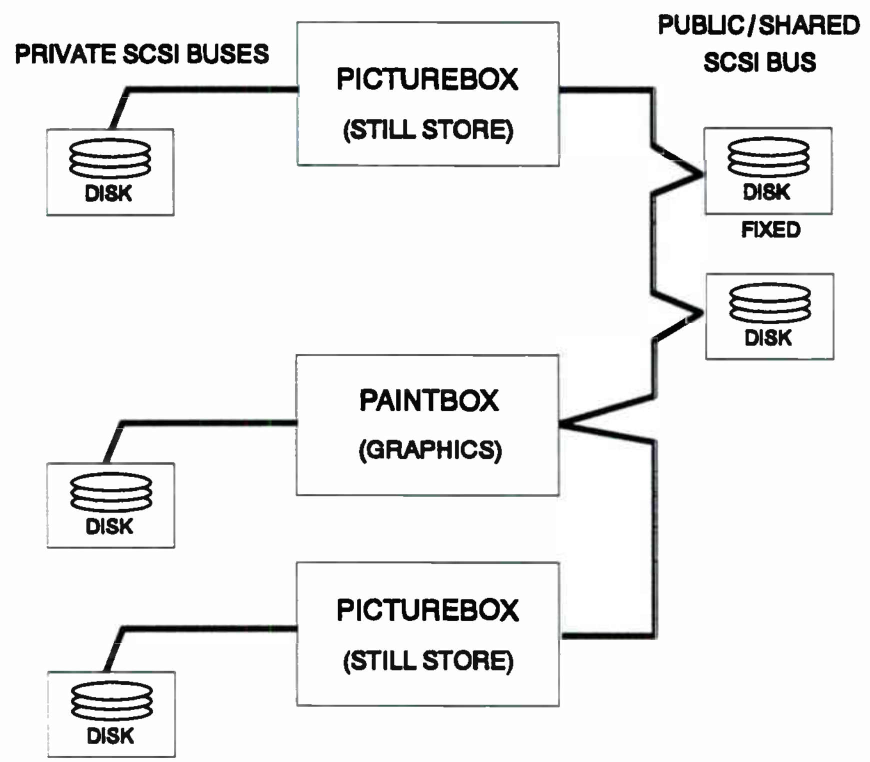

Fig. 1 A SHARED USER BUS (SUB) SYSTEM

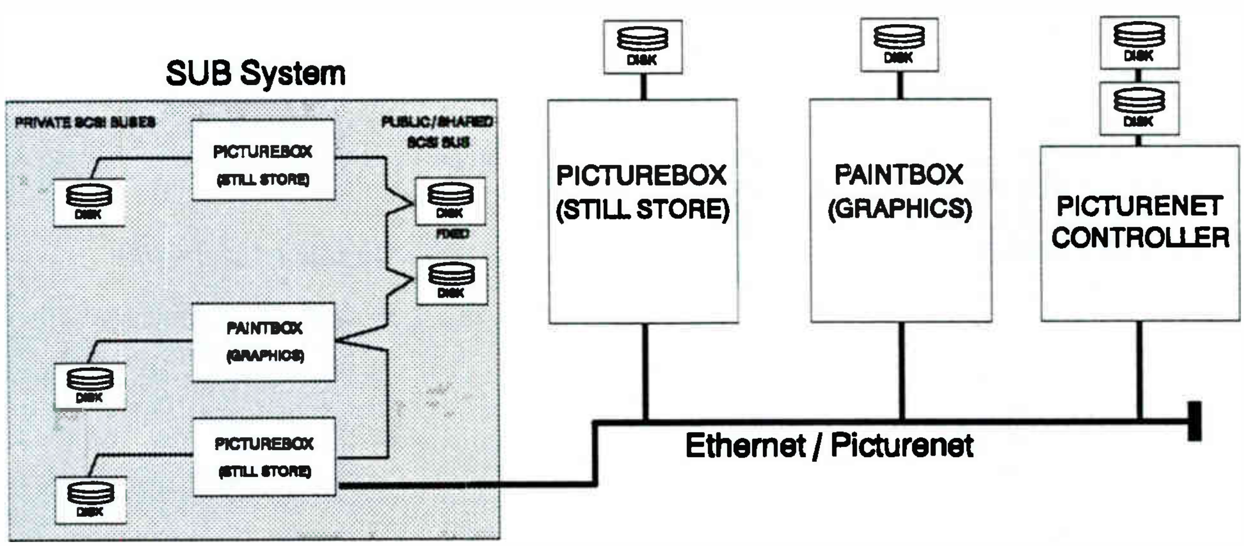

Fig. 2 A PICTURENET SYSTEM (With SUB)

exchange pictures. This is fast, with picture transfers taking around ½ second, and creates a useful local cluster of up to three machines (see Figure 1). The scheme is limited by the bus itself in that there is a limit of eight addresses available and a maximum total bus length of only six metres. Even so it is possible to add disks to provide a shared capacity of 10,000 pictures.

This configuration operates as a star with each connected machine able to buy and sell pictures, or groups of pictures, with the shared disk store. The store can be used either as a post box where pictures can be picked up by any machine on the bus, or as an archive. Both fixed and removable drives can be used. Operation only need involve the selection of the shared facility when reading or writing pictures, the remainder of the operation can stay the same as for local disk use. The intention is that the normal requirement of pictures is held on the local disks. As far as possible any additions and transfers would be made during preparation time. Working this way reduces any chance of the picture not being available at the last moment and gives each machine total independency.

Larger Networks

For connecting with larger numbers of machines, or over a greater distance, ethernet is used (see Figure 2). Thin wire ethernet can interface with up to 30 units over a distance of 600 feet. Further extensions to this are easily possible using industry standard ethernet devices such as repeaters and bridges, even making a world wide network.

The first thought may be to use an industry standard protocol, such as TCP/IP, which could connect with a useful range of computer products. However the time taken to transfer pictures can be unacceptabley slow at around 20-30 seconds on a 10Mbps - the special ethernet link. This was one reason that Picturenet protocol was developed, reducing the time to less than 3 seconds. Whilst providing a sufficiently fast hi-way for the connection of the larger numbers of machines it can also co-exist with other protocols using the same network, if required.

Although the ethernet cable actually connects the machines in a daisy chain formation operationally the configuration is the same as for the shared bus system. There is a file server, the Picturenet Controller, which includes the disk store for the network. This may hold up to 10,000 pictures which can be added to or accessed by any of the machines on the network. The theme is very much the same as for the smaller network as the ethernet actually connects to the public bus of the machines, so continuing the philosophy of the dual bus. Again individual machines use their own computer systems to operate with the network store. They retain autonomy. For operation the network can be presented as another available picture store on each machine. As with the smaller system the ideal method of operation calls for picture transfers to be completed prior to program time. In this way on-air use only accesses the local disks so the network would only be needed for any last minute requirements. network may be grown by extending the ethernet to more machines. As shown in Figure 3, these may be close enough to be connected on the same piece of coax, or any distance away, connected via an ethernet bridge and a suitable communications system. Times are quoted for image transfers over links of various speeds (they exclude any overheads which may be introduced by the communications system).

Connecting with other Machines

So far the most important machines for the network, still stores and Paintboxes, have been accommodated but no account taken of other graphics machines. The Picturenet Protocol was adopted for the best performance with the main elements of the network but does not interface with other machines. For this the concept of a Gateway has been developed. This allows the Picturebox still stores or Paintbox to act as a client to a file server on industry standard computer networks - as included in Figure 3. By the very nature of image

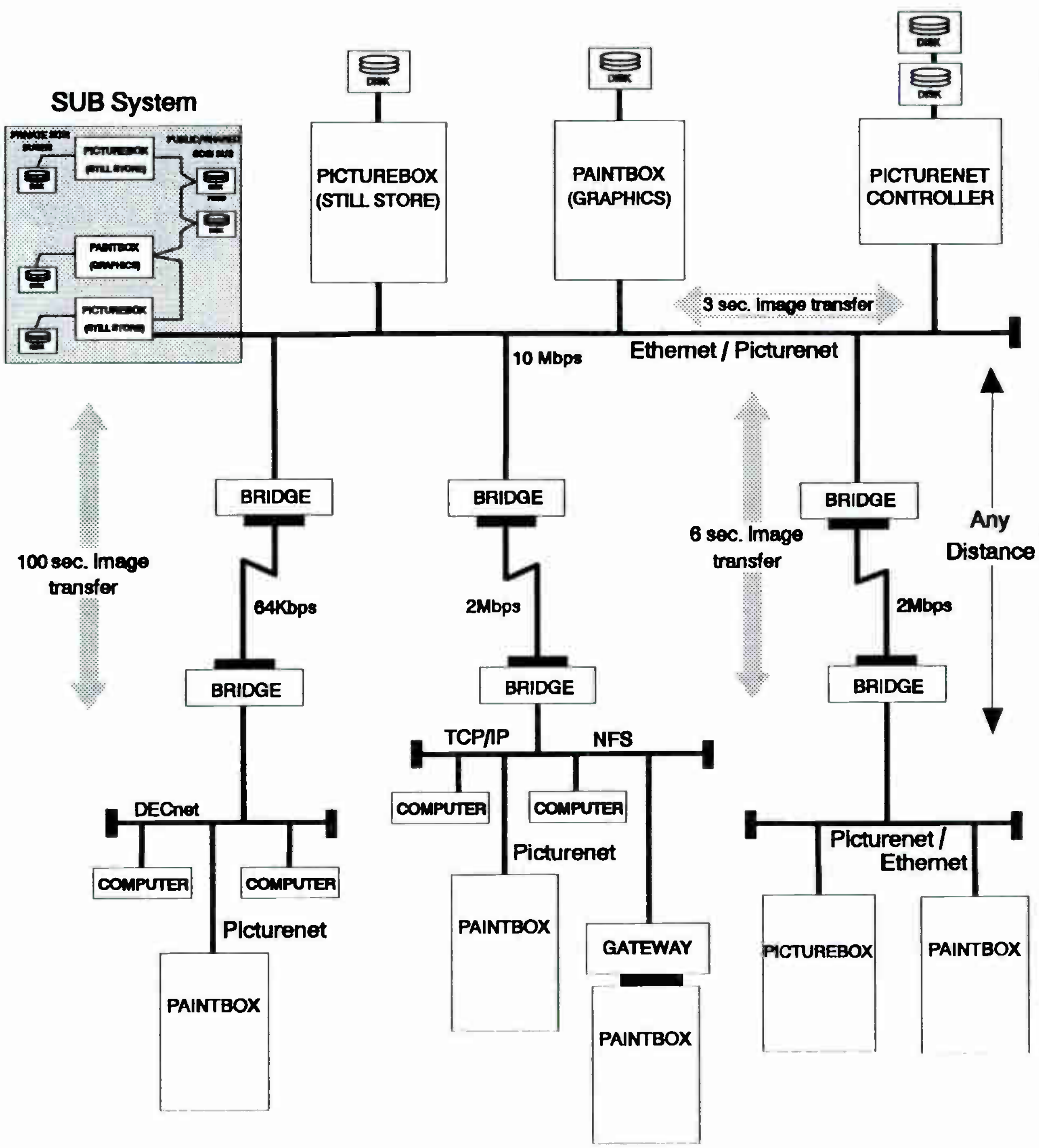

Fig. 3 A NETWORK USING PICTURENET, SUB, GATEWAY & OTHER COMPUTER LINKS

files held on a computer or 3D system, the design of the Gateway is modular and tends to be built on an individual basis.

Further Expansion

It is possible to convert to thick wire ethernet and add more stations and more distance for the network. As extra is added so there becomes the need to increase the total size of the central store at the Picturenet Controller. The disk storage can be supplemented but, at some point, the capacity of the computers in the individual machines is no longer sufficient to handle the load. For the current design the break point is assumed to be around 10,000 pictures- a number quoted several times already. The addition of a dedicated computer to operate the database then becomes necessary, so allowing picture capacities to rise to meet any practical requirement.

Additional storage may well be in the form of removable disks. For automation optical juke boxes offer a means for vast capacities being available on a semi on-line basis. The central computer can keep records of all the images including those off-line. The system could also store compressed sized images for browsing all pictures, including those off-line.

It is possible to connect networks together. Normally most of the traffic will be contained within the individual networks but, by allowing traffic to flow into others, the access capability of each is greatly enhanced.

CONCLUSIONS

Networking on any scale can be economically achieved. At the outset the main task was to make an efficient system to connect still stores and Paintboxes to make the important connection between the major elements of graphics creation, storage and presentation. It proved to be that with this solution it was not easily practical to directly include a very wide range of other graphics equipment. Rather than changing the design which already worked well, it was better to adopt different solutions to fit the various circumstances and requirements that are in the TV graphics environment. Checking against the quality measure at the beginning of this paper, the group of solutions arrived at does very well. The point to note is that there is no single ideal but a group of solutions that can work together for an effective and efficient total networking solution.

A STILL-ANIMATION FILE SYSTEM EMPLOYING A VIDEO SOLID RECORDER

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ABSTRACT

For ongoing election coverage, NHK has developed what's called a "Still-Animation File System", capable of instantaneous generation and recording of various analytical graphics based on collected data from all over the country, randomly selected and transmitted.

This system features rapid coverage and tabulation, with graphics that are displayed at terminals and continuously updated with each new count and revision, and images that are generated and transmitted instantaneously via a simplified procedure.

INTRODUCTION

Overall Flow of NHK's Quick Election Reporting System

Data is accumulated through a network of on-line election sites all over the country and amassed at the host computer at NHK's Tokyo headquarters. Based upon this data, images are created: candidate profiles, vote tallies and analytical graphics divided by various criteria - political parties and sex for easy visual comprehension. These images and graphics are then broadcast. Many of the analytical graphs are animated to present a more dynamic, attractive display.

System Features

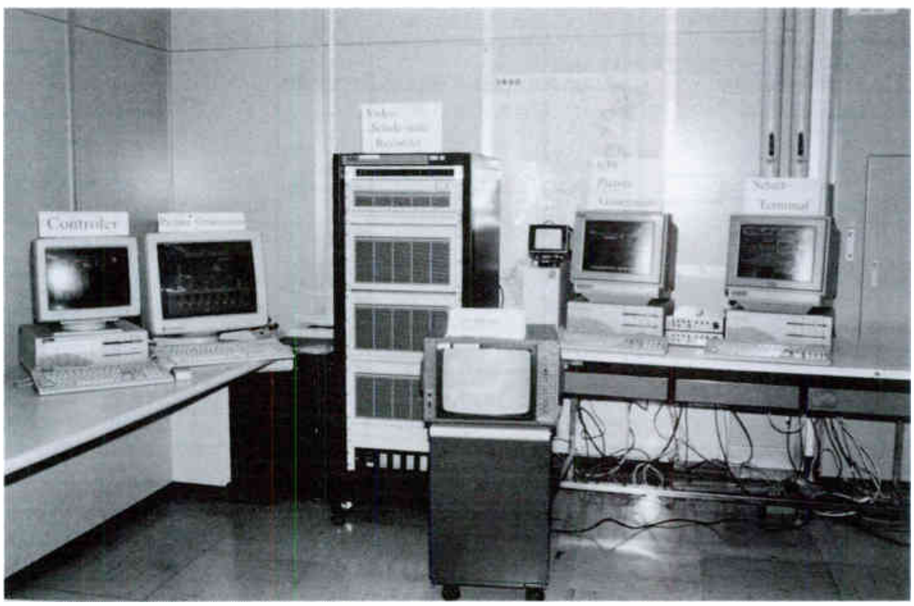

Fig.1 The Entire System

(1) A single system is able to send out three separate channels of media

Independently produced election programs are aired by NHK's nationwide surface broadcasts, local broadcasts, and satellite broadcasts. Thus still and animated images can be transmitted simultaneously through this system.

(2) Analytical graphs detail all changes and updates as they occur

Terminal monitors offer the viewers a number of reduced images, so that as various data are updated, the overall situation is easily followed.

(3) User friendly process for image transmission

One simply touches the surface of the desired picture or graphic on the monitor display itself and that picture is instantaneously brought up for transmission. In this way, even those without technical training can easily key up images.

SYSTEM CONFIGURATION

Fig.1 is a photograph of the entire system. Fig.2 shows the system configuration. This system employs a video solid recorder, which uses a semiconductor memory as the recording medium, for writing and reading various analytical graphics.

(1) The Flow of the System

Tabulated election data is brought in from the host computer on-line, upon each recount and new tally (at approximately 10 second intervals). Based on the data received, graphs of still or approximately 3-5 second animated pictures are generated and then stored in the memory of the video solid recorder. All picture graphics stored in the memory are monitored and can be retrieved when selected.

(2) Component Sections

This system consists of graphics generating workstations, a picture recording-playback section in the memory in which still and animated pictures are stored, picture selection terminals for selecting graphics, a multi-picture generating section which displays up to 16 multi-pictures per

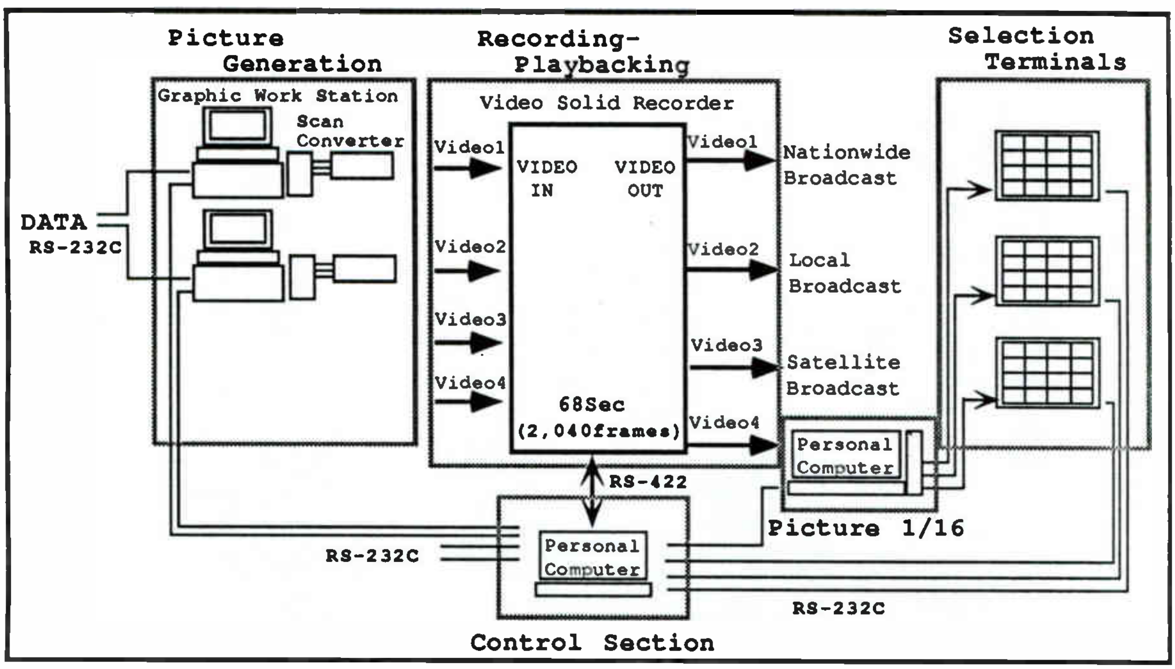

Fig.2 System Configuration

screen on the monitor, and a control section connected to all components of the system.

(A) Graphics generator

In this section still as well as animated graphics are generated instantaneously. Using an election reporting application as an example, about 40 kinds of still pictures and about 15 kinds of animated pictures are generated. The graphics generator creates all kinds of graphics; in data updating, the pictures which are selected at the picture selection terminal are given priority of generation. When animated graphics are requested, the graphics for all the frames are immediately generated and recorded in the picture recording-playback section.

At the graphic picture generating unit, two graphics workstations (Silicon Graphics IRIS4D Series) are applied, and up to 4 workstations can be connected. Even with only one graphics workstation this system is operable; additional workstations are also used as "standby"s in case of failure of any one workstation. What graphic is to be generated by what graphic picture generating unit is determined with a signal from the control section, which actuates the graphic picture generating program corresponding to that graph.

(B) Picture recording-playback section

In this section, graphs of still and animated pictures are stored in the memory. A video solid recorder using a semiconductor memory as the recording medium is used (VSR-10).[1] This recorder has a capacity of recording 68 seconds (equivalent to 2,040 frames of still pictures), which can be extended up to 136 seconds. It has four video input/output ports each, which permit different video signals to be written or read simultaneously through them.

Fig.3 details the memory area and animation management. The memory holding all graphs is secured in the case of still pictures; but in the case of animated graphics, the memory for each individual motion is managed as one block area, and by connecting these blocks, animation occurs with intermediate pauses between each individual movement. If no further blocks are available, animation picture blocks which are not in use at the time can be diverted so as to accommodate a few dozens more animated pictures.

Normally the unit is provided with a controller of its own, but for this system, a unique interface to control the unit from the control section directly has been developed.

(C) Multi-picture generating section

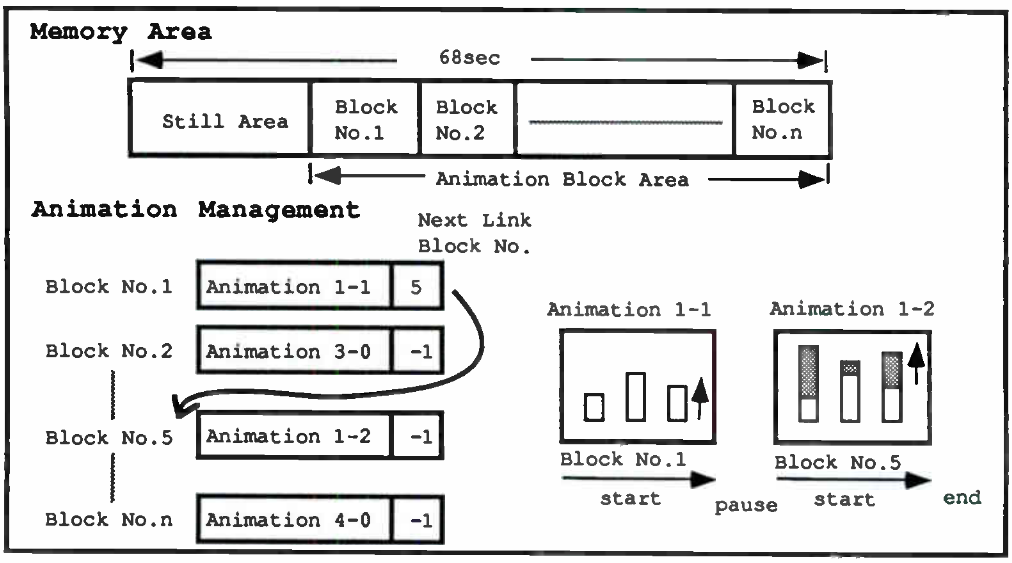

Fig.3 Memory Area and Animation Management

In this section, pictures to be displayed in a form divided into 16 sections on the picture selection terminal monitor are generated. The terminal monitor uses a 14-inch CRT, and it has been judged from the test results that with this size, 16 is the highest viable number of divisions. Other graphs which can not be displayed simultaneously (more than 16) are viewed on separate pages at the terminal. The picture to be displayed is taken out of the fourth video output port which is not in use for broadcast on the video solid recorder. Corresponding to the page which is being selected by the picture selection terminal, it is managed that what graphs should be read out and reduced for display.

(D) Picture selection terminal section

In this section, graphs stored in the memory of the video solid recorder are selected. In correspondence to each video output channel, one picture selection terminal is provided. On the terminal monitor, multi-pictures reduced to 1/16 size are displayed.

Fig.4 shows a monitor screen.

When there are several dozen graphs, they can be viewed on separate pages. There is an editing function that allows for registering in into whichever specific position on the page is wanted. In the case of animated picture graphs, the last still picture (after movement has ceased) is displayed, so that the final graphic can be viewed at a glance. Graph selection is done by simply touching the surface of that particular graph. Moreover, since all graphs are monitored, it is possible, as in our election program, to take in the overall situation.

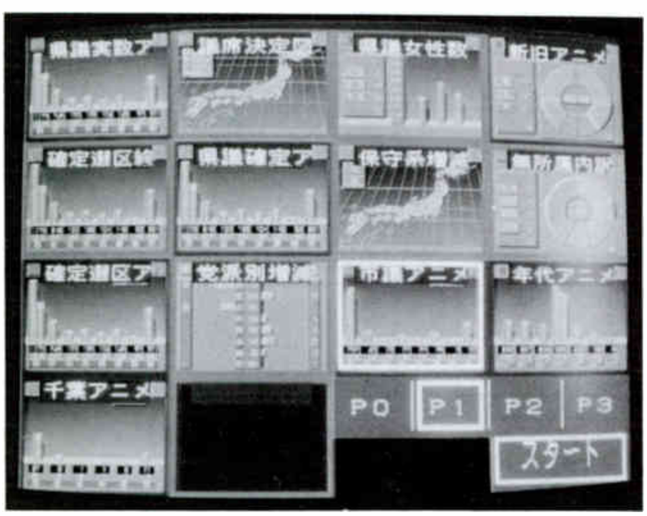

Fig.4 Picture Selection Terminal Screen

(E) Control section

This section is the central section of the new system. It manages read selections made at the picture selection terminal, write management of the graphic picture generating section, manages the locations of the video solid recorder, etc. For graph write and read commands, 1 byte data are used.

Since requests are taken from eight directions - such as picture selection terminals, graphic picture generating units, etc., it must be treated within a one-field period. Even with one-byte data, a few different commands and up to about 240 kinds of graphs can be selected.

The control section is connected to the picture selection terminal section and the graph generation section through RS-232C. RS-232C is applied as a general interface, and provisions are made so that modified picture selection terminals or modified graph generating sections in the future may be used.

Address (field location code) control of the video solid recorder is required for what picture fields are to be read or written within each field. With no addresses renewed, still pictures result, and with addresses renewed in succession, animated pictures result. For all input/output ports, this process is controlled for each field. In

TRASMISSION LINE	RS-422
ASYNCHRONOUS	
BIT LENGTH	8BIT
STOP BIT	1BIT
PARITY	EVENPARITY
BAUD RATE	76.8Kbps

Fig.5 Communication Standard

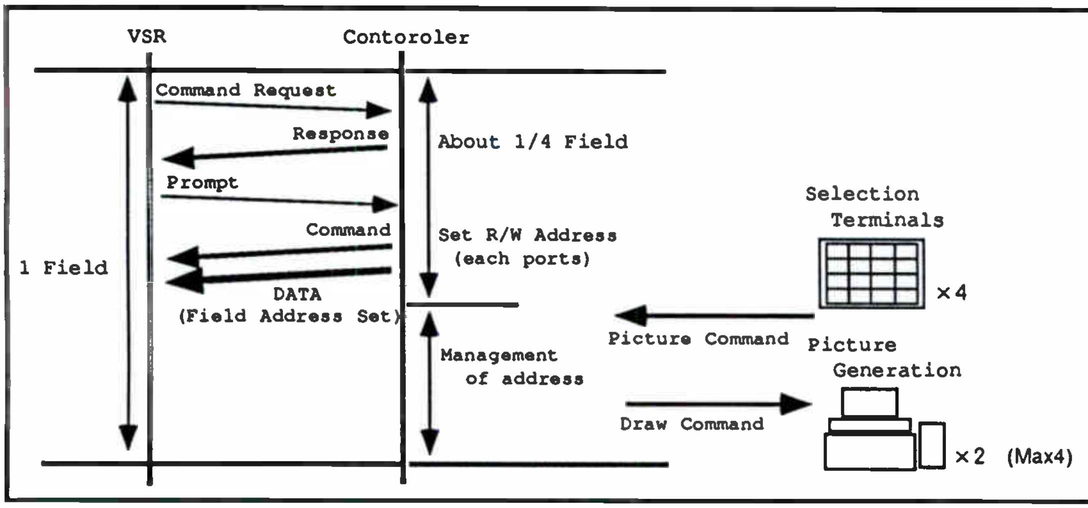

Fig.6 Communication Procedure

practice, to prevent the flickering inherent to NTSC, address control is done in the color frame mode in which one cycle comprises four fields.

The control section is connected to the video solid recorder through RS-422. The interfacing conditions are as shown in Fig.5. The baud rate as high as 76.8k bps is used for communication so that write and read control can be completed within the one-field period. Fig.6 shows the procedure for communication between the control section and the picture recording-playback section.

(3) System Back-Up

This system is designed to be operable with a single graphic picture generating unit alone. Therefore, if any one of the plural graphic picture generating units fails, any other remaining unit can complete the task. In the event that the video solid recorder fails, in the video system, the graphic picture generating unit sends the picture directly, while in the control system, the graphic picture generating unit is connected directly with the picture selecting terminal so that graph selection can be conducted.

AFTERWORD

This system was used by NHK in election coverage during the simultaneous local elections held in April of 1991 throughout Japan (at a scale of about 3,000 seats for 46 prefectures) and proved to be very effective in the analysis of a constantly varying situation.

This system, which is able to produce graphics analyzed in various respects of updating data instantaneously, is very effective for not only election broadcasts based on data, but also programs based on polls, etc.

Existing problems yet to be Dealt with

(1) To increase the number of reduced multi-pictures per screen on the picture selecting terminal monitor. A concrete example is to employ a large size monitor of high resolution.

(2) To improve the function of the graph generating section for faster generation of higher quality graphs.

(3) To make use of the functions of M/K amplifiers built into the video solid recorder, such as image dissolving and wiping effects, etc. for more effective transmission of pictures.

REFERENCES

1. NEC VSR-10 Instruction Manual

REDUCING STATION OPERATING COSTS

Wednesday, April 15, 1992

Moderator:
Dennis Ciapura, Noble Broadcasting, San Diego, California

***HOW TO BARGAIN WITH THE POWER COMPANY AND OTHER METHODS TO REDUCE POWER COSTS**
Patrick J. O'Hare
Cost Analysis, Inc.
Newport Beach, California

***HOW TO GET THE MOST OUT OF TELEPHONE AND DATA SERVICES**
Steve Pilling
Telecom Consultants
San Diego, California

***HOW TO OBTAIN THE GREATEST NUMBER OF TUBE LIFE HOURS**
John Sullivan
Econco
Woodland Beach, California

***DEMAND SIDE ENERGY MANAGEMENT**
John Jensen
Kinetech
Burlington, Massachusetts

*Paper not available at the time of publication.

FAA/FCC WORKSHOP

Thursday, April 16, 1992

Moderator:
John F.X. Browne, John F.X. Browne & Associates, Inc., Bloomfield Hills, Michigan

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THE AIRSPACE ANALYSIS MOOEL (AAM)
David F. Morse and George K. Sakai
Federal Aviation Administration
Washington, District of Columbia
Walter D. Phipps
Ohio University
Athens, Ohio

Panelists:

Richard Smith
Federal Communications Commission
Washington, District of Columbia

Edward W. Hummers, Jr.,
Fletcher, Herald and Hildrith
Washington, District of Columbia

THE AIRSPACE ANALYSIS MODEL (AAM)

David F. Morse and George K. Sakai
Federal Aviation Administration
Washington, District of Columbia
Walter D. Phipps
Ohio University
Athens, Ohio

ABSTRACT

The Federal Aviation Administration (FAA) uses the computer-based Airspace Analysis Mathematical Model (AAM) to determine the electromagnetic compatibility (EMC) between FM broadcast stations (88-108 MHz) and the aeronautical radio services (108-137 MHz). The predictive capability of the AAM is based on a characterization of the existing base of equipment and signals. This characterization is discussed and examples of the AAM database are presented.

INTRODUCTION

Many previous techniques used to determine the electromagnetic compatibility (EMC) between FM broadcast stations (88-108 MHz) and the aeronautical radio services (108-137 MHz) have by necessity used a simplified approach to the many calculations involved in analyzing the compatibility problem. The ready availability of computers has provided the impetus to develop the computer-based Airspace Analysis Mathematical Model (AAM) with the capability of providing a full analysis of any given compatibility question.

The AAM differs significantly from earlier methods of analyzing compatibility in that a complete three-dimensional analysis is performed. This analysis takes into consideration the vertical radiation patterns of the FM broadcast signals as well as the vertical structure of the localizer service volume. The output of this model consists of computer-generated plots which indicate areas within the three-dimensional service volume where interference is predicted.

The existing model forms a foundation intended for future additions. While the initial implementation of this model is intended to evaluate the compatibility between FM broadcast emitters and the Instrument Landing System (ILS) localizer and VOR, methodology is currently being developed and further data acquired which will allow compatibility analyses between the FM broadcast band and VHF/UHF air-ground communications systems. Plans are also underway to expand the capabilities of the model to include the effects of VHF and UHF television signals on various aeronautical services. Consequently, the current model is considered a "living" vehicle. Maximum flexibility has been designed into its composition to accommodate any future advances.

DESCRIPTION OF MODEL

The basis of the model is a set of databases acquired from various sources involved in the investigation of the compatibility between FM broadcast and aeronautical radio services. As these investigations progress, the databases will grow and additional information will become available.

Transmitting Antenna Considerations

The first set of data included in this model characterizes the vertical radiation patterns of various FM broadcast antennas. These data were provided by manufacturers and cover 61 different antenna types encompassing a broad range of configurations. The basic parameter is the number of bays which range from 1 bay to 14. Each of these basic types can be further specified to have either full-wavelength spacing or half-wavelength spacing, and each of these categories can be specified to have 0 degree, 0.5 degree, or 1.0 degree beam tilt. Additional types can be added as required. The default pattern used in the AAM is based on a 6-bay antenna with full-wavelength spacing and no beam tilt. Figure 1 shows a representative vertical pattern for this antenna.

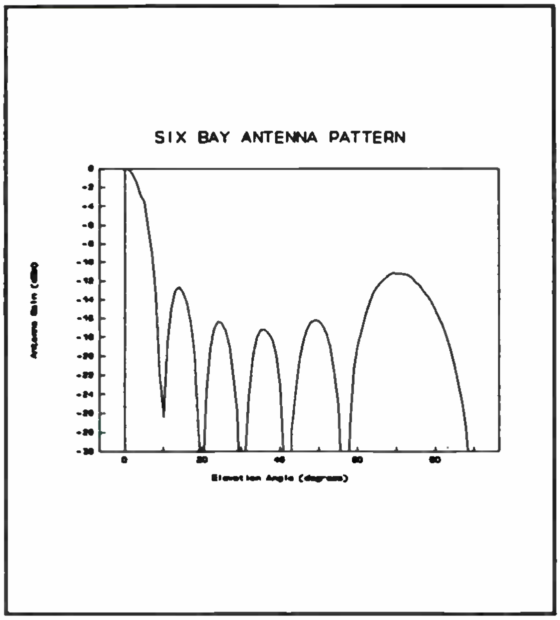

Figure 1. Default Vertical Gain Pattern Used for Transmitting Antenna.

Unless specific information is available, the horizontal radiation patterns for each of the antenna types above are assumed to be omnidirectional. For cases when a station is not omnidirectional, the AAM allows entry of the relative horizontal gain every 5.0 degrees in azimuth. This horizontal pattern is stored for future use. It can be rotated as needed to match the actual installation.

Receiving Antenna Considerations

The second set of data characterizes the frequency response of aircraft navigation antennas in the 88-118 MHz band. These measured data were taken by Transport Canada, the FAA, and various avionics manufacturers using similar methods.

An appropriate representation of the aircraft navigation antenna is crucial to the accuracy of any compatibility model. Minimum field strengths specified for the ILS localizer and VOR transmissions are external to this antenna. Since the receiver measurements used to establish interference thresholds are based on signal levels at the receiver terminals for both the desired and undesired signals, the gains and losses in the receiving antenna and cabling must be accounted for in order to relate calculated signal levels to measured interference criteria. The AAM uses a factor L_R which is the net antenna gain minus cable losses. The AAM generic case is shown in figure 2. The equation for this curve can be expressed as:

$$L_R = 2717.9+(-84.607*F)+(.8571*F^2)-(.0028470*F^3)$$

where,

L_R = Antenna gain (in dB)
F = Frequency (in MHz)

Navaid Considerations

The third set of data used by the AAM characterizes the gain characteristics of various ILS localizer and VOR arrays. The gain (in dBi) and horizontal radiation patterns are stored for 19 different localizer array types, and other types can be added as needed. At this time, VOR horizontal radiation patterns are defined as omnidirectional. Vertical radiation patterns for a typical localizer array and VOR are also available in the AAM. Figures 3 and 4 demonstrate the typical vertical pattern used for the localizer and VOR, respectively.

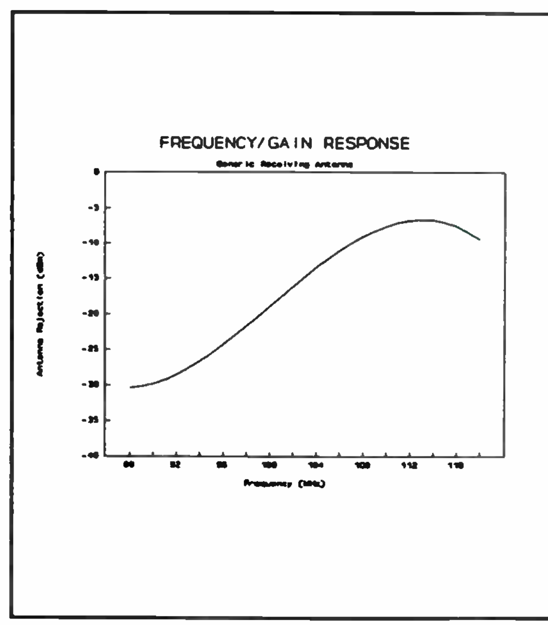

Figure 2. Frequency/Gain Response Of Generic Receiving Antenna.

Receiver Considerations

The fourth set of data characterizes receiver performance. The results of a joint Canadian/United States ILS/VOR receiver test program are incorporated into the model. These results are contained in Report 929, Annex 2, of the International Radio Consultative

Committee (CCIR). Additional testing has been accomplished since then, and these results are also included in this database.

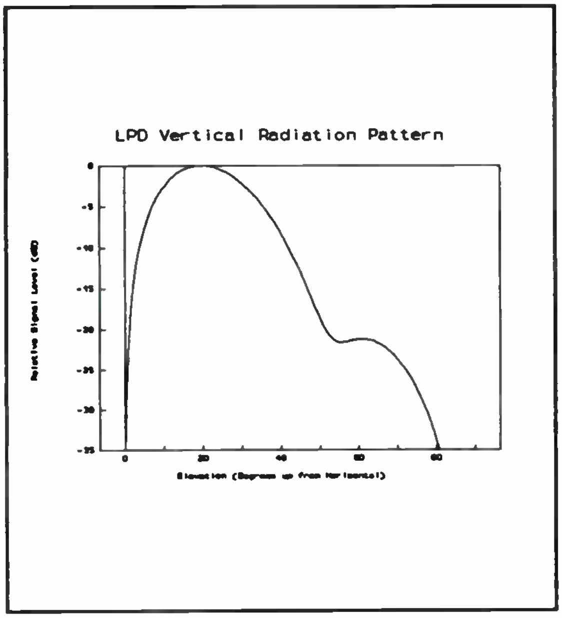

Figure 3. Typical Vertical Gain Pattern for Localizer Array.

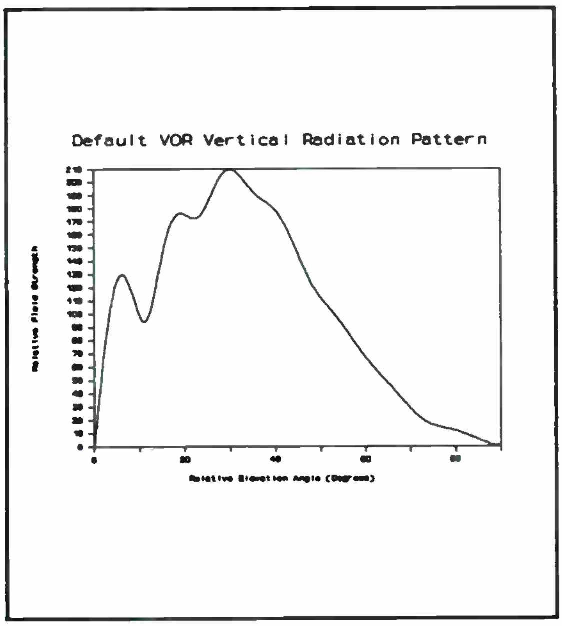

Figure 4. Typical Vertical Gain Pattern for VOR Array.

<u>A2/B2 Interference Thresholds.</u> The AAM incorporates thresholds for A2/B2 (overload/desensitization) and B1 (intermodulation) interference that are based on the measured results of the receiver tests. In general, the AAM uses an average of the three most sensitive receivers. Curve-fitting programs are used to generate analytical expressions for the data, and the equations for the curves are used in the AAM to represent the performance of a generic receiver.

A simplified equipment setup for an A2/B2 test is shown in figure 5. The equipment setup for B1 tests discussed below differs in that multiple signal generators are required to simulate the FM broadcast stations.

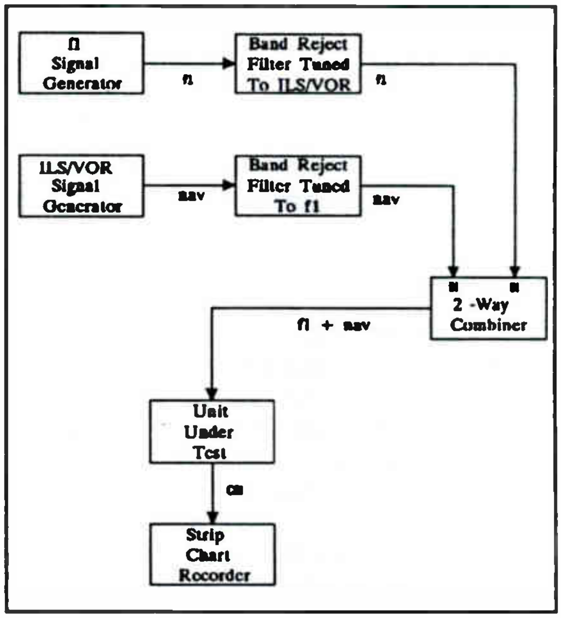

Figure 5. Typical Equipment Setup for A2/B2 Measurements.

For each bench test of an ILS localizer receiver, the amplitude of the simulated broadcast signal is set to an insignificantly low level, and then gradually increased until the strip chart recorder indicates a 10% error in the aircraft crosspointer (CDI) while the desired signal remains constant. The level of the broadcast signal at this point is the threshold of interference for the receiver being tested. These tests are repeated at various frequencies and various desired signal levels until it is certain that the receiver performance has been fully characterized. The receiver models tested are estimated to represent 95% of the total installed fleet.

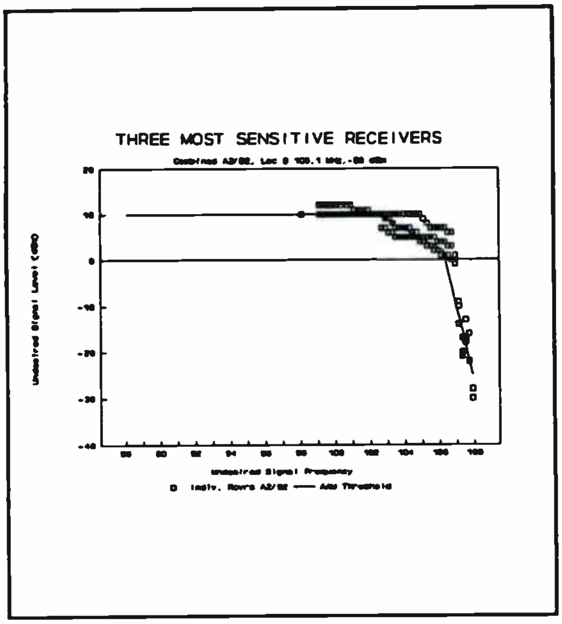

Figure 6. Three Most Sensitive Receivers for A2/B2 Measurements with Regression Lines.

Figure 6 shows the results of the three most sensitive receivers at each broadcast frequency along with the regression lines drawn through these points. These regression lines are the interference threshold used in the AAM. If the signal level from a broadcast station (as measured at the localizer receiver terminal) is above the regression line, then A2/B2 interference will occur. The equations for the regression lines in figure 6 may be expressed as follows:

For $F \leq 102.335$ MHz

$$TH_{BASE} = +10.0$$

$102.335 \text{ MHz} < F < 106.3$ MHz

$$TH_{BASE} = (-2.011 * F) + 215.782$$

For $F \geq 106.3$ MHz

$$TH_{BASE} = (-15.952 * F) + 1696.31$$

where,

TH_{BASE} = A2/B2 interference threshold (in dBm)
F = frequency in MHz

The data shown in figure 6 were collected using a localizer signal level of -86 dBm (as measured at the receiver terminal) and a localizer frequency of 108.1 MHz. Other tests were conducted using different localizer frequencies and levels to determine the relationship between the interference threshold and these variables. An analysis of all measurements to date indicates that the change in the A2/B2 threshold relative to the change in localizer signal level from -86 dBm is .86 dB/dB i.e. a change in localizer level from -86 dBm to -76 dBm (10 dB) will increase the A2/B2 interference threshold by 8.6 dB.

The measurements supporting this correction have been made for broadcast frequencies at or above 107.1 MHz and thus the AAM only uses it for frequencies of 107.1 MHz and above.

An examination of the A2/B2 data shows that the interference threshold increases with an increase in localizer frequency also. Measurements have shown that the improvement factor is given by:

$$I = [(1.85 * F) - 195.745] * (F_{LOC} - 108.1)$$

where,

I = the improvement factor
F_{LOC} = localizer frequency (in MHz)
F = the undesired signal frequency (in MHz)

Therefore, the modified A2/B2 interference threshold can be calculated for broadcast frequencies above 105.8 MHz by:

$$TH_{NEW} = TH_{BASE} + [((1.85 * F) - 195.745) * (F_{LOC} - 108.1)]$$

where,

TH_{NEW} = the revised interference threshold
TH_{BASE} = the baseline interference threshold

The improvements due to changes in the localizer level or frequency are independent and additive. The result of the first calculation becomes the baseline value for the second calculation.

B1 Interference Thresholds. Figures 7 and 8 show the combined two-frequency and three-frequency intermod data, respectively. The horizontal axis represents the $\log_{10}$ of the product of the frequency differences as follows:

$$X = \log_{10} [(F_1 - F_{LOC}) * (F_2 - F_{LOC}) * (F_3 - F_{LOC})]$$

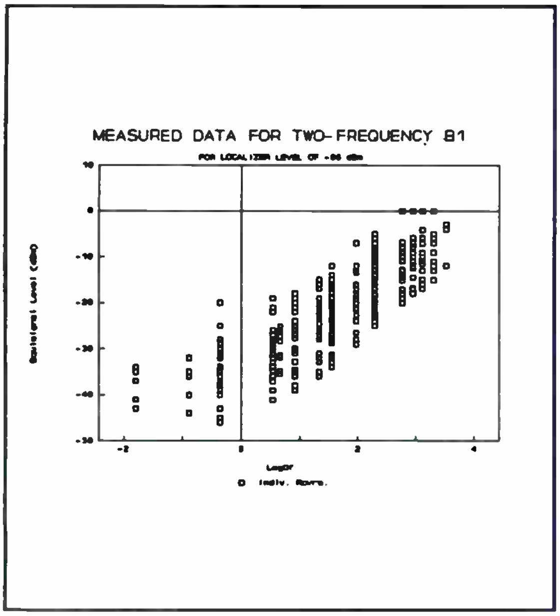

Figure 7. Plotted Results of Two-Frequency B1 Equisignal Measurements.

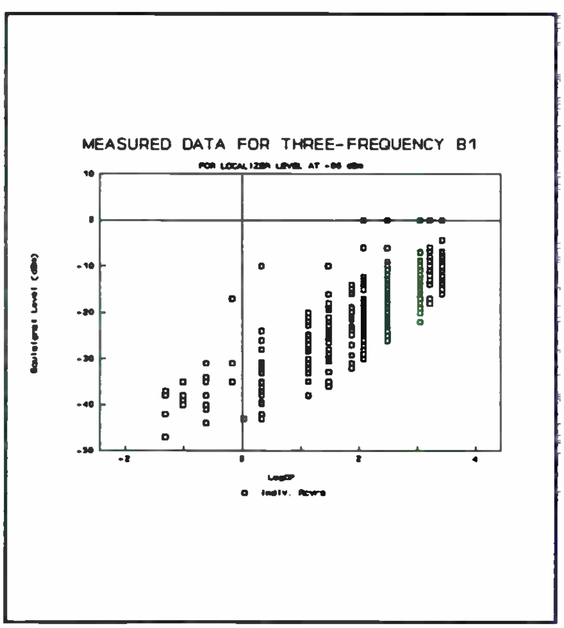

Figure 8. Plotted Results of Three-Frequency B1 Equisignal Measurements.

where,

F_i = Frequency of broadcast station i (in MHz)
F_{LOC} = Frequency of localizer (in MHz)

The test setup used to collect intermod data is similar to that shown in figure 5, except that multiple signal generators are required. The frequency of the localizer and the individual generators must be selected such that:

$$F_{LOC} = F_1 + F_2 - F_3$$

For the two-frequency case, $F_1 = F_2$, so only two generators are required. As in the A2/B2 tests, the signals (N_i) from the generators, are first set to insignificant and equal values. The generators are then increased equally in 1 dB steps until the localizer receiver crosspointer indicates a 10% error. The level from the generators (N_1, N_2, N_3) at this point is referred to as the "equisignal level". The interference threshold, K_3, is exceeded when:

$$N_1 + N_2 + N_3 > K_3$$

Measurements using non-equal signal levels for each station have shown that the above relationship holds for all cases until the highest of the two or three FM broadcast signals reaches the A2/B2 interference threshold. Thus, it is necessary only to sum the signal levels (in dBm) from each broadcast station involved in an intermod, and compare the sum to K_3. Intermodulation interference will occur when the sum of N_i exceeds K_3.

Using the same techniques discussed for the A2/B2 interference thresholds, data for the three most sensitive receivers have been identified at each frequency combination for use in defining the performance of a generic receiver. An analysis of these data points for each unique LogDF value shows that the following equation for the two-frequency equisignal level best fits these points:

$$EQS_{86} = -40.259 + (2.9728 * X) + (1.6895 * X^2)$$

where,

EQS_{86} = the equisignal level for any LogDF value with a -86 dBm localizer level.
X = the LogDF product as described earlier

As previously discussed, the signal levels (in dBm) from each broadcast station involved in the intermod are summed in order to compare them to the interference threshold. Since the EQS_{86} level is present at each signal generator, the total threshold to which the sum of broadcast signals is compared is given by:

$$3 * EQS_{86}$$

Therefore, the two-frequency intermodulation threshold can be expressed as:

$$TH_{86} = -120.777 + (8.9184 * X) + (5.0685 * X^2)$$

By applying a curve-fitting program to the measured three-frequency data, the specific equation for the two-frequency equisignal level is found to be:

$$EQS_{86} = -40.1634 + (2.1977 * X) + (1.5668 * X^2)$$

and the three-frequency interference threshold can be expressed as:

$$TH_{86} = -120.4902 + (6.5931 * X) + (4.7004 * X^2)$$

where the terms are defined as in the two frequency case.

A plot of the data for the three most sensitive receivers for the two-frequency case along with the regression curve is shown in figure 9, while the plot for the three-frequency case is shown in figure 10.

Figures 7, 8, 9, and 10 are based on a localizer signal level of -86 dBm. As in the case for the A2/B2 interference, additional tests were conducted to determine the relationship between the B1 interference threshold and the level of the localizer signal. The results for the two-signal case measured with a localizer signal level of -49 dBm are shown in figure 11, while the results for the three-signal case measured with a localizer signal level of -49 dBm are shown in figure 12.

Using the same curve-fitting program as before, the two-signal equisignal level can be expressed as:

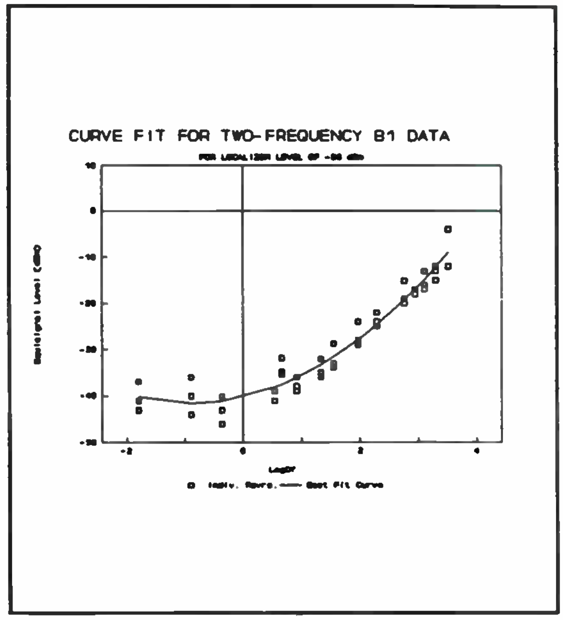

Figure 9. Two-Frequency Results with Regression Curve, -86 dBm Localizer.

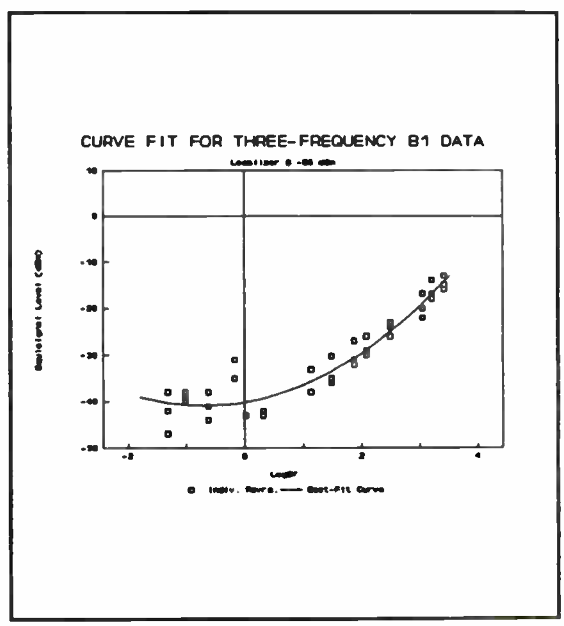

Figure 10. Three-Frequency Results with Regression Curve, -86 dBm Localizer.

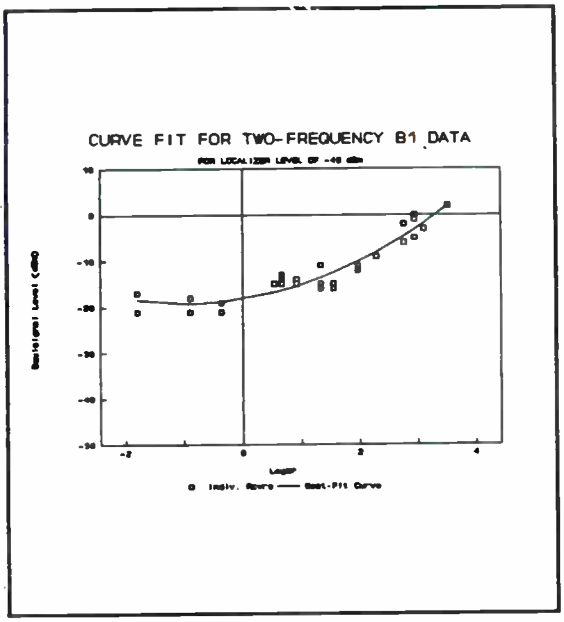

Figure 11. Two-Frequency Results with Regression Curve, -49 dBm Localizer.

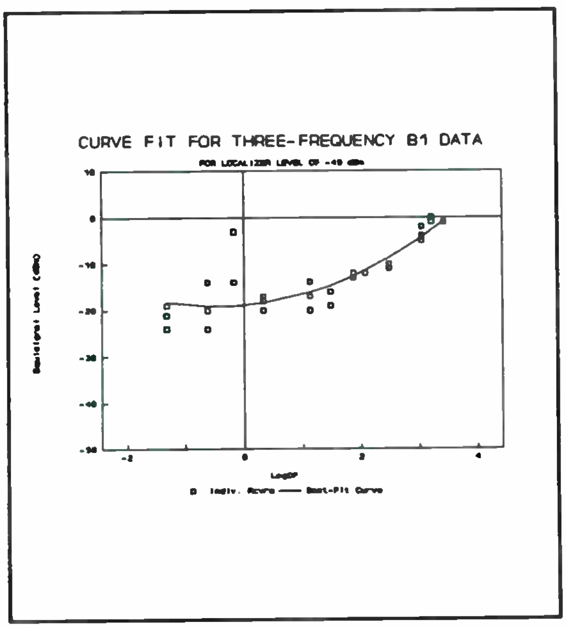

Figure 12. Three-Frequency Results with Regression Curve, -49 dBm Localizer.

$$EQS_{49} = -18.1980 + (2.0070 * X) + (1.0596 * X^2)$$

where,

EQS_{49} = the equisignal level for any LogDF value with a -49 dBm localizer level.
X = the LogDF product as described earlier

The two-signal interference threshold based on a desired signal level of -49 dBm can be stated as:

$$TH_{49} = -54.5940 + (6.0210 * X) + (3.1788 * X^2)$$

For the three-signal case, the equisignal level based on a localizer level of -49 dBm can be expressed as:

$$EQS_{49} = -18.9296 + (1.0941 * X) + (1.2215 * X^2)$$

and the interference threshold based on a desired signal level of -49 dBm can be stated as:

$$TH_{49} = -56.7888 + (3.2823 * X) + (3.6645 * X^2)$$

Given these equations, the B1 interference threshold for any localizer sgnal level can be interpolated by:

$$TH = ((-86 - NAV)/37) * (TH_{49} - TH_{86})$$

where,

TH = the corrected B1 interference threshold
NAV = the localizer signal level stated in dBm as measured at the receiver terminal
TH_{49} = the two-signal or three-signal B1 interference threshold as (appropriate) for a localizer signal level of -49 dBm
TH_{86} = the two-signal or three-signal B1 interference threshold as (appropriate) for a localizer signal level of -86 dBm

The above B1 interference criteria only apply to those cases where the intermodulation frequency falls on the localizer frequency. The case must also be considered when the intermodulation frequency is near, but not on, the localizer channel. For these cases, the AAM currently uses the threshold correction values for off-channel intermodulation products from CARR 1(2), Final Report, Annex 2, paragraph 7.6.5.3 where N_1, N_2, and N_3 as expressed above are corrected by:

$$N_i \text{ (corrected)} = N_i - \text{(correction term)}$$

This has the effect of allowing 3 * (correction term) of additional undesired signal level before the interference threshold is reached. The correction terms are shown in table 1. Recent bench measurements indicate that these correction terms may have to be modified, but additional measurements are required before a final determination can be made.

Table 1. Correction Terms Used for Off-Channel Intermodulation Products.

Frequency Difference Between Localizer and Intermodulation Product	Correction Term for Equisignal Level
0 kHz	0 dB
50 kHz	2 dB
100 kHz	8 dB
150 kHz	16 dB
200 kHz	26 dB

The correction for VOR interference thresholds is automatically made in the above equation due to the higher transmitter power of VOR stations. The same correction for changes in the navaid level is made for off-channel intermod products as well. Finally, the intermod threshold equations apply only if each signal in the intermod is above a minimum "cutoff level". Preliminary results indicate that the worst-case cutoff level is approximately equal to the navaid signal level, but the exact relationship between the cutoff level and frequency or LogDF value has not yet been determined. As of this release of the AAM, the cutoff level is a dynamic value equal to the navaid level at each grid location.

Measurements and Validation

The last set of data incorporated in the model is the result of an airborne measurement program and has been used to validate the equations used in the model. Approximately 300 hours of flight measurements were conducted during the development of this database. These data were used to determine the suitability of the analysis technique by comparing measurements over similar flight paths for repeatability and analyzing the propagation calculations by comparing measured and calculated data.

The characteristics of various individual flight-inspection aircraft are part of the databases utilized by the model. By invoking the characteristics of specific receivers, airframes, and antenna systems from the various databases, the simulation can be made to represent a specific aircraft. This technique must be used to validate the model, since the exact combination of generic criteria used to make general interference predictions may not exist.

SUMMARY

As close as possible, the model emulates airborne measurements. After all necessary input data are supplied regarding the aeronautical facility and the FM broadcast facilities in the area, the AAM generates an grid of imaginary test points arranged at regular intervals throughout the three-dimensional volume of interest. The AAM then conducts a simulated flight through each grid point and computes the desired and undesired signal levels reaching the receiver from each emitter of interest. Grid locations are flagged where any combination of signal levels exceeds the interference thresholds established by the receiver measurements. A plot of the flagged grid locations indicates areas where interference would be expected to occur.

FIBER OPTIC WORKSHOP

Thursday, April 16, 1992

Moderator:
Otto Claus, Claus & Associates, Parkton, Maryland

***FIBER OPTICS AND ITS APPLICATION TO BROADCASTING**
James J. Refi
AT&T Network Cable Systems Services
Norcross, Georgia

*Paper not available at the time of publication.

CONTRACT ENGINEERS WORKSHOP

Thursday, April 16, 1992

Moderator:
John Bisset, Multiphase Consulting, Falls Church, Virginia

Panelists:

Mark Persons
M.W. Persons Associates
Brainerd, Minnesota

Mike Patton
Mike Patton & Associates
Baton Rouge, Louisiana

Grady Moates
Loud and Clean
Boston, Massachusetts

CAMERA WORKSHOP

Thursday, April 16, 1992

Moderator:
Hank Volpe, WBAL-TV/AM & WIYY, Baltimore, Maryland

***HOW TO OBTAIN THE BEST PERFORMANCE FROM YOUR CAMERA**
Fred Himmelfarb
Panasonic
Secaucus, New Jersey

*Paper not available at the time of publication.

SATELLITE UPLINK WORKSHOP

Thursday, April 16, 1992

Norman Weinhouse
Norman Weinhouse Associates
Bell Canyon, California

AUTHOR INDEX

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