1991 PROCEEDINGS

45th Annual Broadcast
Engineering Conference
Proceedings

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April 1991

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TABLE OF CONTENTS

TELEVISION PRODUCTION AND POST PRODUCTION

Monday, April 15, 1991

INTERNATIONAL TECHNICAL UPDATES AND AGENDAS

Monday, April 15, 1991

BROADCAST AUXILIARY AND SATELLITE SYSTEMS

Monday, April 15, 1991

TELEVISION AUTOMATION

Monday, April 15, 1991

ADVANCES IN FM SYSTEM DESIGN

Monday, April 15, 1991

BROADCAST/AERONAUTICAL COMPATIBILITY
Tuesday, April 16, 1991

DIGITAL AUDIO BROADCASTING—SYSTEM CONCEPTS
Tuesday, April 16, 1991

UHF TRANSMISSION
Tuesday, April 16, 1991

DIGITAL VIDEO AND TRANSMISSION SYSTEMS
Wednesday, April 17, 1991

FM SYSTEMS ENGINEERING AND IMPROVEMENT
Wednesday, April 17, 1991

COMPUTER APPLICATIONS FOR BROADCAST ENGINEERS
Wednesday, April 17, 1991

DIGITAL AUDIO INTEGRATION

Sunday, April 14, 1991

Moderator:
Donald Lockett, NPR, Washington, District of Columbia

SIGNIFICANT TECHNICAL CONTRIBUTIONS TO RADIO BROADCAST TECHNOLOGY*
Larry J. Cervon
Broadcast Electronics, Inc.
Quincy, Illinois

A TUTORIAL ON RECORDABLE COMPACT DISC
Laura J. Tyson
Denon America, Inc.
Parsippany, New Jersey

LOW COST DIGITAL SAMPLE-RATE CONVERTERS
Sangil Park
Motorola Inc.
Austin, Texas

SPECTRUM EFFICIENT DIGITAL AUDIO TECHNOLOGY (SEDAT)™
William Koos, Larry Koos, and Kent Malinowski
Scientific-Atlanta, Inc.
Melbourne, Florida

AN INTEGRATED DIGITAL SYSTEM FOR BROADCAST AUDIO
David J. Evers
Broadcast Electronics, Inc.
Quincy, Illinois

AN ALL DIGITAL CD QUALITY STUDIO-TRANSMITTER LINK FOR THE 950 MHZ BAND
Howard Friedenberg and Jamal Hamdani
Moseley Associates, Inc.
Santa Barbara, California

PERFORMANCE TRADEOFFS FOR SATELLITE AND TERRESTRIAL DIGITAL ENCODED AUDIO CIRCUITS*
William W. Rollins
Intraplex, Inc.
Littleton, Massachusetts

*Paper not available at the time of publication.

A TUTORIAL ON RECORDABLE COMPACT DISC

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ABSTRACT

As the compact disc medium establishes a stronger position in the broadcast industry, broadcasters may need to explore new recording formats compatible to existing CD playback hardware. This paper serves as a tutorial on the new CD-WO (Write Once) format, and discusses its comparisons to CD-Audio. The paper will include an overview of the physical and optical properties of the recordable disc. The CD Cart Recorder will be studied as an example of this technology, and examples of how the CD recorder can be useful to broadcasters will be presented.

THE WRITE ONCE RECORDABLE DISC

From the practical standpoint, a recordable disc offers the same benefits as conventional discs, including ease of handling and storage convenience (vs. Open Reel Tape). The audio quality of a recorded or partially recorded disc is compatible with CD Audio specifications.

For the description of a recordable disc, the Taiyo Yuden Co. "That's" Compact Disc Recordable CD-R63/CD-R74 is used as an example. Figure 1 shows a cross section of the "That's" Recordable disc. The Recordable CD can be described according to 3 stages: the Blank disc, the Partially Recorded disc, and the Recorded disc.

Blank Disc

The Blank disc contains a pregroove, a spiral track starting from the center, which has been modulated with a slight wobble. This wobble allows the CD Recorder to maintain accurate constant linear velocity (CLV) and timing during recording. The wobble occurs at a frequency of 22.05 Khz. The timing information imbedded in the pregroove is referred to as Absolute Timing In Pregroove (ATIP). This serves as a preliminary time code which allows the blank disc to be recorded, and the partially recorded disc to be played back in the CD Cart Recorder.

Partially Recorded disc

Data on the Partially Recorded disc is organized into the following sections:

Laser Power Calibration Area - The first section of the partially recorded disc is dedicated to calibration of the laser power for each recording. To achieve consistency of laser intensity for each recording, the intensity of the laser of the first recording is stored in the Power Calibration Area (PCA). During subsequent recordings, the power of the writing laser will be matched to that of the first.

Program Memory Area - When several tracks are recorded on a disc at different times, but prior to the Table of Contents (TOC) being added, the start times for each of those tracks is stored in the Program Memory Area. Once the disc is fully recorded, the final track location information will be stored in the TOC.

Lead In Area - The Lead In area accommodates the TOC when the disc is fully recorded, and no other recordings will be added.

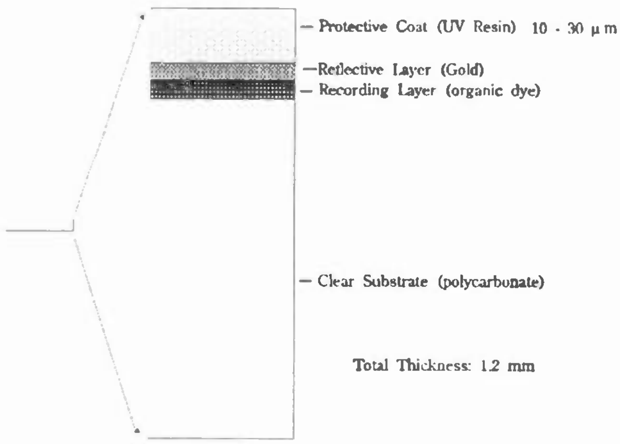


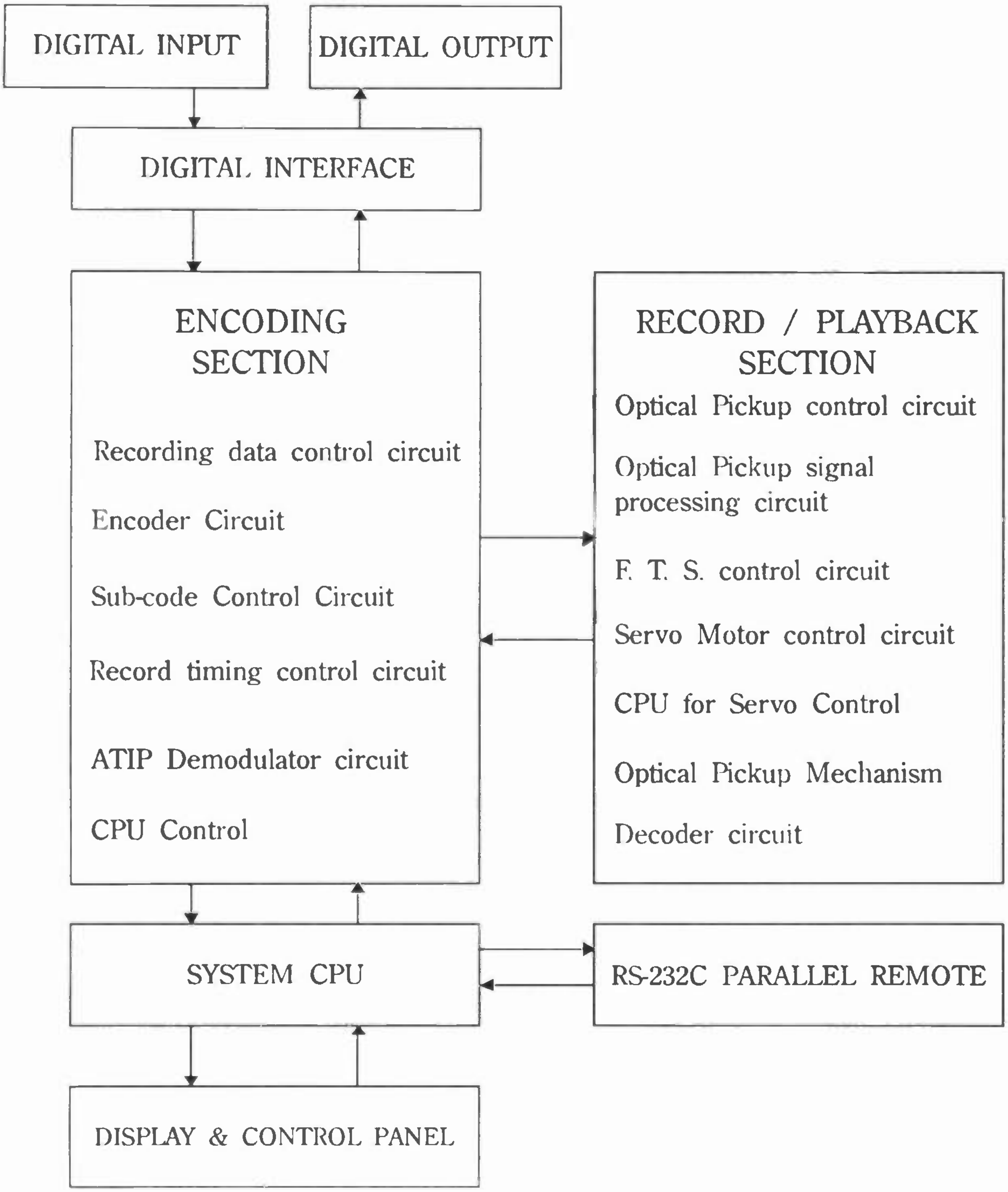
Figure 1
Cross Section of
"That's" CD-R63/CD-R74
Recordable Compact Disc

	CD Audio Disc	CD WO Disc
READ OUT METHOD	via reflection	same
TRACK SHAPE	One, Spiral	same
OUTER DIAMETER	120 mm	same
ECCENTRICITY	+/- 0.2 mm	same
DEFLECTION (WARP)	+/- 0.4 mm	same
CENTER HOLE DIAMETER	15 mm	same
TOTAL THICKNESS	1.2 mm	
REFLECTION	>70%	same
WAVE-FRONT DISTORTION	NA	less than 0.05 λ
TRANSPARENT SUBSTRATE:		
REFRACTIVE INDEX	1.55	same
MAX BIREFRINGENCE	100 nm	same
RECORDING AREA:		
STARTING DIAMETER OF PROGRAM AREA	50 mm	44.7 mm
MAX DIAMETER OF PROGRAM AREA	116 mm	118 mm*
MAX START DIAMETER OF LEAD IN	46 mm	same
TRACK PITCH	1.6μm	same
DIRECTION OF ROTATION	C.Clockwise	same
SCANNING VELOCITY	1.2 -1.4 m/s	same
MAX VELOCITY VARIATION	+/- 0.01 m/s	same
SHELF LIFE:	N/A	10 years

*118 mm for Unrecorded disc, 116 mm for Recorded disc

Note: Exact allowable tolerances for measurements are not indicated above. For specific tolerances, consult the Orange Book Specifications.

Figure 2



BLOCK DIAGRAM CD RECORDER

Figure 3

Program Area - The Program area contains the actual audio or data which has already been recorded.

Recordable User Area and Leadout Area - The Recordable User Area represents the available remaining space on the disc which can still be recorded. The Leadout Area is determined only after the TOC is finalized, and serves as a means for the CD player to determine where the recorded area ends. The Leadout Area occurs following the last recorded track.

Recorded disc

Once the disc is fully recorded and the TOC has been added, it fulfills the same requirements as a "Red Book" disc, and can be played back in any CD player. Figure 2 shows a comparison chart between CD Audio and CD Write Once discs.

DATA ORGANIZATION

According to Red Book Specifications, data is arranged on the disc under the Eight to Fourteen Modulation (EFM) format. In this method, the digital audio information, containing parity information, is grouped into 8 bit symbol. Each of these 8 bit symbol is replaced with a 14 bit symbol, according to a fixed table. Then an additional 3 merging bits is added to each 14 bit symbol. The EFM assures that no pattern of less than 3 or more than 11 "0"s will occur between each "1". Thus, a pattern of EFM and Merging bits such as 00000000000100010 is allowed. A pattern of 00000000000001001 (too many "0"s in a row) or 00010100010010001 (not enough "0"s in a row) is not allowed.

The "1" in the EFM translates to a physical change from land to pit or pit to land on the disc, and to optically "read" a pit corresponding to "101" would be nearly impossible with currently available technology. Therefore, EFM is required in order to physically represent the data on the disc so that it can be read using laser technology presently available.

ERROR CORRECTION

Error correction on the Write Once Recordable Disc is achieved via the same method as regular playback discs. The recorded data on the disc has been "scrambled" according to the Cross Interleave Reed-Solomon Code (CIRC). The CIRC encoding rearranges and adds parity bits to the data prior to the EFM stage, according to a specific pattern. Upon playback, the EFM signal is decoded, and the data is reorganized by the CIRC decoding process. Parity bits are stripped away from audio bits. All CD players perform CIRC decoding, but the method or algorithm by which this is done may differ from one CD player to another.

OVERVIEW OF THE CD CART RECORDER

The Denon CD Cart Recorder model DN-7700R will be used as an example of CD Write Once hardware. Figure 3 shows a block diagram of the DN-7700R CD Cart Recorder. A few of the key sections are described as follows:

AES/EBU Decoder - Converts serial digital information to parallel.

Recording Data Control - Responsible for the arrangement of data. The timing of parallel data can be adjusted prior to transfer to the Encoder.

EFM Encoder - Data is encoded to fulfill the Orange Book CD format.

Optical Pickup Control - Transmits the encoded data in a serial format, so that pits recorded on the disc conform with the Red Book CD Format. Controls laser intensity during recording.

Subcode Control - Adds the PQ subcodes (Table of Contents, Timing and location information for each Track and Index) to the subcode area which was generated by the system CPU.

Record/Playback Timing - Adjusts the timing of the data to be recorded consistent to the timing of previously recorded data, providing continuity of playback timing.

Optical Pickup Control - Controls Focus, Tracking, Slide Motor position, and Constant Linear Velocity (CLV) of Spindle Turntable.

Servo Motor Control - Adjusts the timing of the major Servo Control (Focus, Tracking, etc.) signals.

The remaining playback circuits (EFM decoding, CIRC decoding, audio processing, etc.) are very similar to conventional CD players.

The optical pickup of the DN-7700R carries the same optical requirements as the pickup of a playback only CD player. The wavelength is specified to fall between 770 and 830 nm.

USING THE CD CART RECORDER

First Recording

The CD Cart Recorder uses the same cartridge format as the CD Cart Player. A new, unrecorded disc should be removed from its jewel box, and placed in the cartridge (Model ACD-5B) for handling and recording. The cartridge protects the blank disc from fingerprints, dirt, and damage. The cartridge may be labeled or coded according to to the program material to be recorded. The disc itself can be written on, but only with a soft tip marker (and not with a ball point pen).

Audio is recorded onto the disc in real time. On a new blank disc, least 5 minutes of audio must be recorded continuously. If only a few minutes or seconds of audio are recorded, during playback a CD player may not be able to locate the tracks due to their extremely small physical size. Once the first 5 minutes have been recorded, any duration recording may be done thereafter.

During the recording process, the emitted laser beam heats up sections on the blank disc, and the heated sections create "pits" in the recording layer. The laser output is modulated according to the data stream to be recorded. The reflectivity of the disc is decreased where the pits are located on the disc.

During playback, the laser follows the track of pit and land areas, and the intensity of the signal varies according to the reflectivity. The intensity of the reflected laser beam is measured by a quadrant of photo diodes, and the resulting current recreates the EFM signal.

The Partially Recorded Disc

Once the first 5 or more minutes of audio is recorded on the disc, the result is a Partially Recorded disc. The Table of Contents (TOC), has not yet been recorded. At this point, the Partially Recorded disc cannot be played back in a conventional CD Player, with the exception of the Denon DN-970FA Production CD Cart Player. The DN-970FA contains special software which instructs the player to skip the "READ TOC" sequence if the TOC cannot be located, and to locate and playback audio data. Even if the TOC has not been written, the DN-970FA can still cue up to selected tracks, and play any recorded audio on the disc. The CD Cart Recorder can also be used to play back the Partially Recorded disc.

The Fully Recorded Disc

A maximum of 74 minutes of audio may be recorded on a blank disc. This information can be divided into up to 99 different Tracks, and 99 Indexes within each track. Once the disc has been recorded up to its desired capacity, the TOC must be added. The CD Cart Recorder has a WRITE TOC function built in. Manual creation of the TOC is also possible during the recording process. Once the TOC is finalized, the disc may be played in any conventional CD player. However, no additional recordings may be added to the disc. (Additional recordings would require new TOC information, and at this stage, the TOC may not be modified.) Therefore, it is most efficient to use up as much disc space (time) as possible prior to writing the TOC.

USES FOR A CD CART RECORDER

Spots, PSA's, Station ID's

The CD Cart Recorder makes it possible for a radio station to truly be 100 % CD. All of the stations spots, promotional announcements, station ID's, etc., can be placed on compact disc. For example, radio station WXXX has as an advertiser Bob's Car Dealership. All of Bob's commercials are gradually placed on one disc, as they are created. At any time before the disc is full (and the TOC is added), it may be played back in the DN-970FA CD Cart Player. Later, more spots can be added and played back. Using Bar Coding on the cartridge, the traffic director of the station can control which spot is played at what time.

For example, perhaps during the month of July, Bob's Cars wants to run their special 4th of July promotional spot. The 4th of July spot is located as Track 7 on the disc. Using the Auto Track Select Feature on the DN-970FA, the traffic director could code the cartridge containing the Bob's disc as "Play TNO 7 ONLY", using the appropriate bar code. When the disc is loaded into the DN-970FA, Track 7 would automatically cue up, with no effort from the jock. With the correct bar coding, the DN-970FA would prohibit manual selection of another Track. In August, Bob's decides to run their Summer Clearance spot, which is recorded on the disc as Track 3. Now the traffic director changes the code on the cartridge to "Play TNO 3 ONLY", and that spot would automatically cue up when loaded into the DN-970FA. The station could be 100% confident the Christmas Special spot would not accidentally run during the President's Birthdays week.

Archival

For broadcast stations having large CD libraries, the CD Cart Recorder allows more efficient storage of the music library. For example, a Classic Rock radio station may have 20 or more different Beatles CD's. On each of those CD's, perhaps only a few tracks will ever be played on the air. The selected tracks from the 20 or more CD's could be recorded onto one blank disc, thus providing the station with a "Master" Beatles disc. Now, the Master disc replaces the 20 or more in the music library. The size and storage of the CD library could be scaled down considerably. Selections from other artists could be recorded onto a disc according to style or format.

Stations which run several different formats throughout the day could have a smaller number of discs dedicated to Jazz, Classical, Oldies, etc., rather than having several thousand discs in the music library. "Master" discs for each format could be compiled in a logical manner. Several hundred albums per format could be reduced to perhaps 100 discs (over 700 minutes of music).

Mastering

Using a DN-7700R, a recording studio can provide a client with a "take home" copy of his recording on compact disc, entirely compatible with the client's car or home CD player. This allows immediate monitoring of the recording in alternate environments than the recording studio control room, allowing the artist to better evaluate the tonal quality, mix, and performance.

CONCLUSION

This paper has discussed a CD Cart Recording system. A CD Cart Recorder has many practical applications for broadcast studios. With the onset of Recordable CD, the broadcast station only needs one format of audio playback in the studio. This paper has shown how the CD Cart Recorder can be used for production of spots, PSA's, station ID's and other "custom" materials. Thus it allows the station to use the CD format exclusively for playback of materials in the air studio. The cartridge format is consistent with the CD Cart Player currently in use for playback of audio discs in radio stations.

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DENON **DN-7700R** 日本コロムビア株式会社

CD CART RECORDER DN-7700R

LOW COST DIGITAL SAMPLE-RATE CONVERTERS

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ABSTRACT

This paper describes a method for generation of an interpolated signal for use in "low cost" sample rate conversion applications on digital-signal-processors (DSP's). The particular example used is a conversion form 44.1 kHz (CD) to 48 kHz (DAT) and is implemented on the Motorola DSP56001, a 24-bit DSP. The speed of the algorithm is dictated by the number of instruction cycles needed to implement it, which is variable. A comparison is made of the number of instruction cycles required for implementation to the resulting signal to total harmonic distortion + noise ratio (STHDNR). The assembly code for the routine and the filtering method to generate interpolated samples are included.

1.0 Introduction

The concept of a sampling rate (sampling frequency) is always important in applications which use digital data as a signal. In the past few years, there has been increasing attention given to the problem of matching the sampling rates between two media which are trying to communicate with each other [1-5]. Even when the two machines are using the same sampling frequency, the digital systems are not necessarily synchronous, and must be synchronized to ensure a proper data transfer. However, many transmission systems such as, satellite links or digital networks, may not be synchronized by their nature, since their sampling frequencies might drift due to fading channel characteristics.

There are many industry standard sampling rates including: telephone quality speech at 8 kHz, wide-band telecommunication uses 16 kHz, video systems use 24 kHz rate, the CD-I format uses 37.8 kHz, while compact disc players and DAT players use 44.1 and 48 kHz (which is also universally accepted for most digital audio applications), respectively. It is essential, therefore, to have a converter which can change and synchronize two different sampling rates without losing any of the useful information in the original digital signal. In this paper a simple and effective digital technique to convert 16-bit digital data between two sample rates is discussed. In particular the memory and DSP instruction cycle requirements are analyzed and tabulated to obtain an optimum "low cost" sample-rate converter in trade of STHDNR performance degradations. The software and hardware design issues for the real-time implementation are discussed in sections 4 and 5.

2.0 Conventional sample-rate converters

2.1 "Direct" Technique

There are three techniques conventionally used for sample-rate converters. The first technique is the "direct" method, in which the sampled signal is simply output at the different sample rate. Figure 1 shows the block diagram of such method. Only single memory is shared for input and output. The input samples are received by the in-

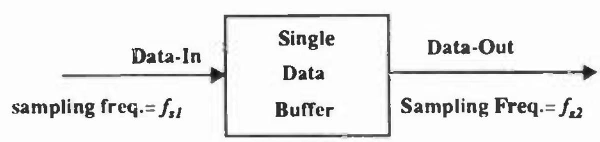

Figure 1. Sample-rate conversion via the "direct" method

terrupts created at the sampling rate of f_{s1} and the internal timer generates interrupts to send out the value in the data buffer at the output sampling rate of f_{s2}. Figure 2 illustrates the problem which results when using this method. When two sample rates are very close each other and the signal which will be converted has relatively low frequency content, the jitter created by the phase difference between the two different clocks sould be small. However, when the frequency content of the input signal is relatively high, the output signal will contain discontinuies as shown in Figure 2. Clearly this method can only used in low performance applications.

2.2 D/A and A/D (resampling technique)

Conversion through the analog domain using a digital-to-analog (D/A) converter and an analog-to-digital (A/D) converter is intuitively simple. Figure 3 shows the block

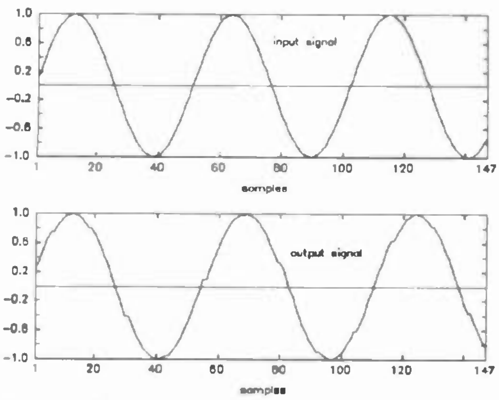

Figure 2. Input and output signals by the "direct"

diagram of this method. The input digital signal is convert-

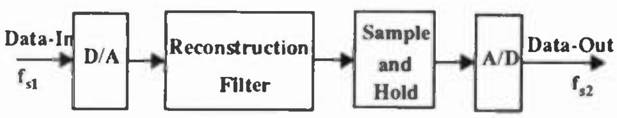

Figure 3. Sample-rate conversion via D/A and A/D method

ed to an analog signal and then resampled at the desired sampling rate. Normal D/A converters create *"image"* frequency components (the D/A equivalent of the A/D frequency modulation process) above the input sampling rate at integer multiples of the sample rate in the analog domain. Because of these images, a reconstruction filter combined with an anti-aliasing filter, is needed between the D/A and A/D converters and is designed with a bandwidth compatible to the lower sampling rate. The primary drawback of this approach is that this analog reconstruction filter is extremely expensive and complex for 16-bit performance. Also, the harmonic distortion and background noise of the D/A and A/D converters are additive and tend to degrade the overall performance of this conversion method.

2.3 M/N method

The third conventional technique has been used to digitally convert sampling rates in a simple fixed ratio M/N [1]. There are many papers which have been written using this approach with different trade-offs [2,3]. Figure 4 illustrates the block diagram of the M/N method. The reconstruction

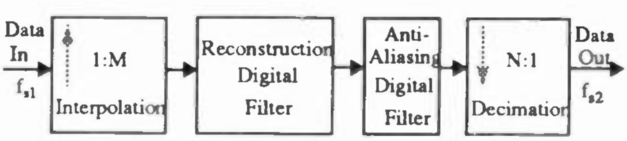

Figure 4. Sample-rate conversion using the M/N method

and anti-aliasing filters are operating in the digital domain unlike the analog approach described in section 2.2. This approach is practical when the ratio of two sampling rates can be represented by two rational numbers which are computed by the least-common-multiple of the two sampling rates. The ratio of 44.1kHz and 48 kHz, for example, can be expressed as 147:160. Thus, the sampling rate of the input signal is first interpolated up by a factor of $M=160$ using digital filters and decimated back down by a factor of $N=147$ to obtain the final predefined sampling rate. This approach can only be utilized in practice when the integers M and N are manageable numbers typically each less than 10. For most digital audio applications, the sampling rates are already so high there is no room to interpolate more than 10 times, so this approach cannot be utilized effectively at 44.1 kHz or 48 kHz.

3.0 Fractionally Interpolated Data

A real-time method to generate a fractionally sampled signal for the sampling conversion from CD rate to DAT rate was introduced by the author [5]. This paper extends that analysis and implementation to the conversion between two arbitrary sampling rates. In this section an interpolation method to create a value between two sampled data points, which can be viewed as fractionally sampled data, is discussed. In order to derive solutions for the *discrete-time domain* interpolation, a mathematical derivation in the *analog domain* is explained followed by *discrete-time domain* equations.

Let $s(t)$ and $s(t-\tau)$ be an analog input signal and its interpolated version, respectively. The Fourier transform (*FT*) of $s(t)$ is defined as [6]

$$S(\omega) = \int_{-\infty}^{\infty} s(t)\, e^{-j\omega t} dt \tag{1}$$

where ω is the radian frequency assumed to satisfy $0 \le |\omega| \le \pi$. The *FT* of $s(t-\tau)$ can be obtained by

$$FT[s(t-\tau)] \triangleq \int_{-\infty}^{\infty} s(t-\tau)\, e^{-j\omega t} dt = e^{-j\omega\tau} S(\omega)\,. \tag{2}$$

Since multiplication in the *frequency domain* is viewed as a convolution operation in the *time domain*, the convolution theorem shows that [6]

$$s(t-\tau) = IFT[e^{-j\omega\tau}] \otimes s(t) \tag{3}$$

where IFT and $\otimes$ denote the inverse Fourier transform and convolution operator, respectively. Using the IFT definition, $IFT[e^{-j\omega\tau}]$ in (3) can be rewritten as

$$IFT[e^{-j\omega\tau}] \triangleq \frac{1}{2\pi} \int_{-\infty}^{\infty} e^{-j\omega\tau} e^{j\omega t} d\omega = sinc(t-\tau) \tag{4}$$

where

$$sinc(x) \triangleq \frac{\sin \pi x}{\pi x}\,. \tag{5}$$

Thus, the interpolated signal in the *analog-domain* can be found using the convolution operation with two obtainable signals as

$$s(t-\tau) = \int_{-\infty}^{\infty} sinc(\upsilon-\tau)\, s(t-\upsilon)\, d\upsilon . \qquad (6)$$

Derivation from (1) through (6) can be applied to discrete-time domain signals. When signal $s(t)$ is sampled at the sampling frequency f_s, the discrete signal $s(nT)$ can be obtained and defined as

$$s(t)\big|_{t=nT} \triangleq s(nT) \qquad (7)$$

where the sampling period $T=1/f_s$. Using discrete Fourier transform (DFT) representation, (4) can be rewritten as

$$e^{-i\omega\tau} \triangleq DFT[sinc(nT-\tau)] = \sum_{n=-\infty}^{\infty} sinc(nT-\tau)\, e^{-i\omega nT} \qquad (8)$$

For a simple mathematical derivation purpose, let $T=1$. When τ is a non-integer (fractional) number, the discrete version of $s(t-\tau)$ in (6) can be obtained by

$$s(n-\tau) = \sum_{m=-\infty}^{\infty} sinc(m-\tau)\, s(n-m) . \qquad (9)$$

Figure 5 shows the discrete (sampled) signal $s(n)$ and a fractionally interpolated value $s(n-\tau)$ at time index n.

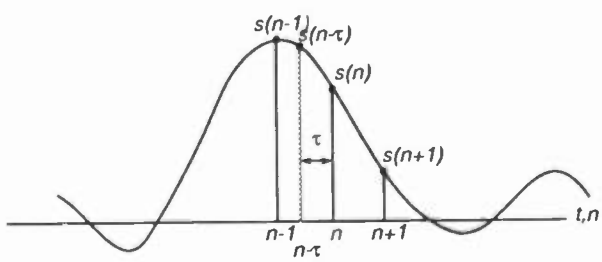

Figure 5. Discrete signal and its delayed value

Using frequency domain interpretation, the delayed signal $s(n-\tau)$ in (17) can be viewed as passing the input signal $s(n)$ through a time-invariant filter whose transfer function is

$$H_d(\omega) = DFT[sinc(m-\tau)] . \qquad (10)$$

Taking DFT of (9) with the definition in (10), we have

$$DFT[s(n-\tau)] = H_d(\omega)\, S(\omega) . \qquad (11)$$

Thus, $s(n-\tau)$ can be exactly realized by the filter which has an infinite number of FIR filter coefficients whose values are plotted in Figure 6. Note that the peak of the *sinc* function is at $m=\tau$. However, since the sinc function is discrete in time and τ is a fractional number, the discrete values of the sinc function are skewed but centered by $m=\tau$.

Clearly, it is not practical to estimate an infinite number of coefficients. However, since the function $sinc(m-\tau)$ in (9) approaches zero as $|m|$ increases as shown in Figure 6 and the maximum delay value τ is up to one sample delay, the series summation in (9) can be truncated at a predetermined finite number with predictable performance degradation. Thus, (9) can be written to

$$\hat{s}(n-\tau) = \sum_{m=-M}^{M} sinc(m-\tau)\, s(n-m) \qquad (12)$$

where $\hat{s}(n)$ is an approximation through the truncation and M is a design parameter which can be chosen by optimization of the performance criteria [7].

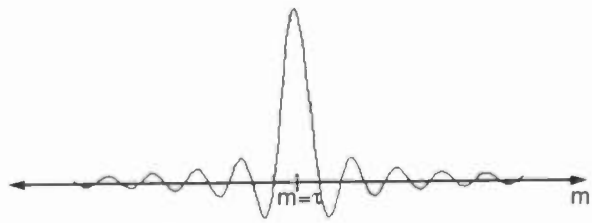

Figure 6. The sinc function

3.1 Error Minimization

The truncation in (12) can create the leakage phenomena which can be smoothed by a window function [6]. In this implementation, the Blackman-Harris window has been used, and it optimizes for maximum side-lobe attenuation. The Blackman-Harris window values are plotted in Figure 7. Using the window function $w(n)$, equation (12) can be rewritten as

$$\hat{s}(n-\tau) = \sum_{m=-M}^{M} w(m)\, sinc(m-\tau)\, s(n-m) \qquad (13)$$

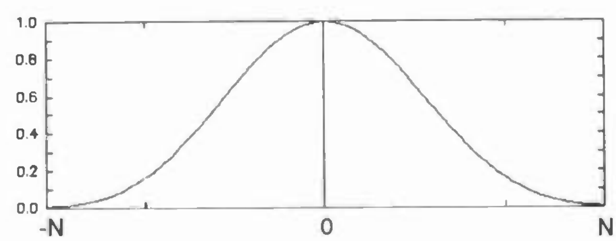

Figure 7. The Blackman-Harris window function

where the Blackman-Harris window function is defined as [8]

$$w(n) = 0.358 + 0.488\cos\left[\frac{\pi n}{N}\right] + 0.141\cos\left[\frac{2\pi n}{N}\right] + 0.011\cos\left[\frac{3\pi n}{N}\right] \qquad (14)$$

for $n = -M, \ldots, -1, 0, 1, \ldots, M$. On defining new filter coefficients which are windowed original coefficients shown in (9), we have

$$h(n) = w(n)\, sinc(n-\tau) , \qquad (15)$$

And (9) can be rewritten as

$$s(n-\tau) = \sum_{m=-\infty}^{\infty} h(m)\, s(n-m) . \qquad (16)$$

Since $h(m)$ has a finite number of elements due to the definition of the window function in (14), the summation in

(16) can be simplified to

$$\hat{s}(n-\tau) = \sum_{m=-M}^{M} h(m)\,s(n-m). \qquad (17)$$

Thus, the discrete signal $s(n)$ is related to its windowed and interpolated version $\hat{s}(n-\tau)$ through a filter whose coefficients are $h(n)$, the values of which are dependent on τ.

3.2 Time-varying delay function estimation

The above concepts for the constant delay function τ can be extended to the case where the delay τ is a function of the time index n. In other words, the value of τ varies sample-by-sample (time-varying). This is achieved via a bank of time-invariant filters. When the individual set of filter coefficients is formed as a vector, the filter bank then can be formulated as a matrix whose rows correspond to the time varying interpolation value $\tau(n)$. If the time-varying delay function is $\tau(n)$ where n is the index for the new sample rate, then the nth filter has the two-dimensional transfer function $H_d(\omega, n)$ as

$$H_d(\omega, n) \triangleq e^{-j\omega\tau(n)} \qquad (18)$$

Thus, the individual impulse response function vector can be found by

$$h_d(n, m) = IFT\left[e^{-j\omega\tau(n)}\right] = sinc\left[m-\tau(n)\right]. \qquad (19)$$

Redefining n to be an output time index and k to be an input time index, from (16) the output of the nth filter at time k is given by

$$s_d(k, n) = \sum_{i=-\infty}^{\infty} h_d(n, i)\,s(k-i)$$

$$= \sum_{i=-\infty}^{\infty} sinc\left[i-\tau(n)\right]s(k-i) = s\left[k-\tau(n)\right]. \qquad (20)$$

The timing relationship between k and n will be discussed in the next section. Since (20) can not be possible due to the infinite number of coefficients, a similar approach described in Section 3.1 can be applied. From (13)-(17) and (20), the converted signal $y(n)$ which is the delayed version of input sample values can be expressed by the approximation (truncation) with the window function $w(m)$ in (14) as

$$y(n) \triangleq \hat{s}_d(k, n) = \hat{s}\left[k-\tau(n)\right]$$

$$= \sum_{m=-M}^{M} h(n, m)\,s(k-n) \qquad (21\text{-a})$$

where

$$h(n, m) = w(m)\,sinc\left[m-\tau(n)\right] \quad for\ n = 0, 1, \ldots, N\text{-}1 \qquad (21\text{-b})$$

From (21), the time-varying filter coefficients $h(n,m)$ can be interpreted as a weighting function whose maximum value occurs at $m = \tau(n)$. Figure 8 shows the matrix formation of the bank of coefficient vectors where $H(n,m)$ is a N by $2M+1$ matrix and N denotes the total number of different $\tau(n)$ values. Therefore, $y(n)$ in (21) can be viewed as being the output of the time-varying filter which is generating a sample-rate converted (new sample rate) signal.

3.3 Aliasing problems in sampling-rate converters

It is important to note that when the input sample rate is larger than the output sample rate, a large portion of the the higher frequency components will be aliased back to lower frequency range at the new sampling rate. This phenomena is due to the decimation (sampling rate reduction). Thus, a band-limiting lowpass filtering must be employed with the conversion process [9,10].

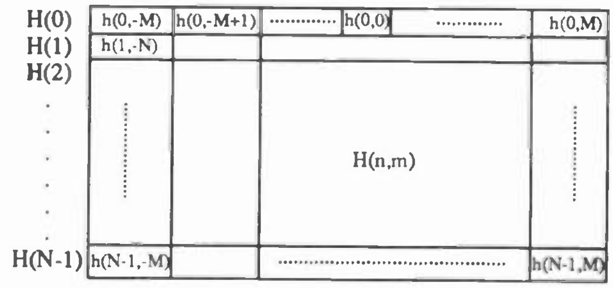

Figure 8. Time-varying coefficient matrix

From (8), it is clear that $h(n)$ are the samples of the continuous-domain function $sinc(t-\tau)$, with the maximum at $t-\tau = 0$, as shown in Figure 6. In other words, given coefficients $h(n)$, the delay τ is the value of t at which the maximum of $h(t)$ (where $h(t)$ is the *continuous-time domain* impulse response function), can be reconstructed from the *discrete-time domain* impulse response function [11]. Since the interpolated signal $h(t)$ has to be band-limited by the new Nyquist sampling rate, $f_{s2}/2$, the interpolation (*sinc*) function has a band-limiting factor β as [12]

$$h(t) = \sum_{m=-\infty}^{\infty} h(m)\,\frac{\sin\pi\beta(t-m)}{\pi\beta(t-m)} \qquad (22)$$

where β is the highest allowed frequency in the new sampling rate, normally less than equal to $f_{s2}/2$.

Applying this result to (15) with discrete-time domain interpretation, we have

$$h(n) = w(n)\,\frac{\sin\pi\gamma(n-\tau)}{\pi\gamma(n-\tau)} = w(n)\,sinc\left[\gamma(n-\tau)\right] \qquad (23)$$

where $\gamma = f_{s2}/f_{s1}$. However, when $f_{s2} > f_{s1}$, $\gamma = 1$ has to be applied to (23), since there is no need for band-limiting.

4.0 Real-time operation considerations

Consider when the algorithm in (21) is applied in a real-time processor. When the output must be generated at the output sample rate f_{s2}, the value of τ and current data pointer for the input buffer must be determined. Figure 9 shows an example of the input and output timing relationship when the output sampling rate is greater than the input sampling rate. When $y(0)$ is ready to output, $s(0)$ can be copied to the output buffer because the timing difference between input and output is zero. However, when $s(1)$ is ready to output, the value τ_1 has to be estimated, and then the processor has to compute the $2M+1$ coefficient FIR filter in (21) whose coefficients correspond to the value of τ_1.

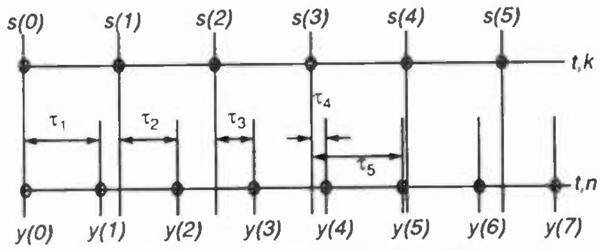

Figure 9. Input and output timing relationship

Also, note that the same input data starting from $s(3)$ are used to generate $y(4)$ and $y(5)$ values with different time delay parameters τ_4 and τ_5 as shown in Figure 9. Thus, the estimation of the time variant delay function and the starting point of the input data vector are very important for this real-time implementation.

Since the current delay value τ_n for the output sample index n is always referenced to the current index k of input data sample, the τ_n can be defined as

$$\tau_n = nT_{s_2} - kT_{s_1} \tag{24}$$

where $Ts_1=1/fs_1$ and $Ts_2=1/fs_2$. Thus, τ_n is always a non-negative number with the following range

$$0 \le \tau_n < T_{s_1} = \frac{1}{f_{s_1}}. \tag{25}$$

When two arbitrary sampling rates are delivered to the converter the discrete values of τ_n may be infinite. However, in practice the two sampling rates are normally known, so that the number of values for τ_n may be finite. Assuming there are N number of fractional values which is linearly spread between an input sample period Ts_1 as shown in Figure 10, the τ_n can be computed as

$$\tau_{i+1} - \tau_i = \tau_0 = \frac{1}{N \cdot f_{s_1}} \quad \text{for } i = 1, 2, \ldots, N-1 \quad . \tag{26}$$

If $2M+1$ coefficients are used for the FIR filter in (21), the memory required for the interpolation coefficients is $(2M+1) \cdot N$ words. Figure 11 illustrates the memory location for the coefficients.

4.1 Estimation of τ by external circuitry

Let's consider how τ can be estimated. Figure 12 shows an external circuit which was built from three common Programmable Logic Devices (PLDs). Basically, this circuit is a bus-addressable, 24-bit synchronous counter. The counter is reset by the input data interrupt which is generated by the SSI Receive port of the DSP56001. The routine

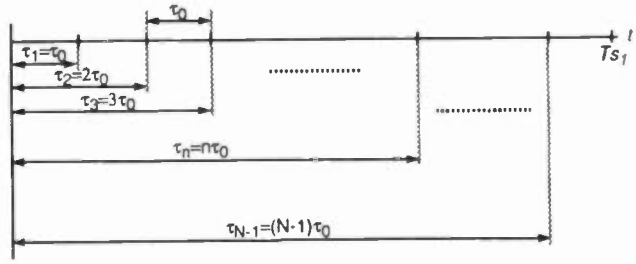

Figure 10. Linearly distributed time delay function

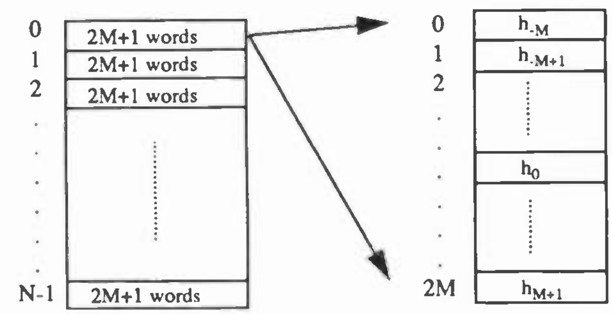

Figure 11. Memory location of the bank of coefficients

which services this interrupt places the new input sample in a circular input buffer and updates the buffer's pointer in anticipation of the next input sample. Because better temporal resolution achieves lower phase estimation jitter, the counter runs from a 27 MHz clock, the same clock which drives the DSP56001.

The SSI Transmit port of the DSP56001 generates the output interrupt. When this interrupt occurs, the DSP56001 reads the contents of the counter through its memory-mapped address and translates this count to the value of the current τ. In turn, the corresponding set of FIR filter coefficients is located, the filter is applied to the contents of the input buffer and the new output sample is delivered to the SSI Transmit hardware—all in real time.

4.2 Filtering Process

To generate new signal with converted sampling rate, an FIR filtering stage is required at the output sampling rate using the current $2M+1$ input data samples. From equation (21) the output $y(n)$ can be obtained using an FIR filter with tapped-delay-line (TDL) structure as shown in Figure 13, where z^{-1} and z blocks make one past sample (one sample delay) and one future sample, respectively. Since the structure shown in Figure 13 which is a the implementation of (21) also uses future values, from $s(k+1)$ to $s(k+M)$ at k, the filtering process is *non-causal*. In other words, it can not be implemented in real-time, because the future values are not available. Thus, a *causal* TDL structure can be implemented as shown in Figure 14. The only difference is that the output is delayed by N samples compared to the non-causal implementation shown in Figure

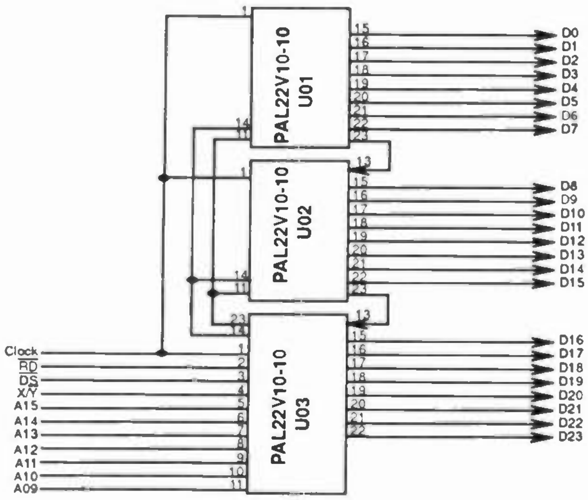

Figure 12. External circuitry for phase estimation

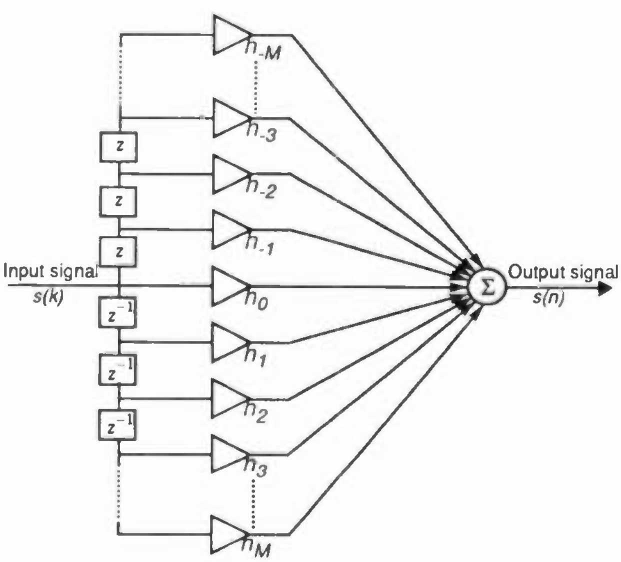

Figure 13. Non-causal implementation of the TDL structure

13. Also, note that since the filter coefficients are made using the symmetric window design rule, the linear-phase through out the frequency spectrum is guaranteed. Figure 18 shows the entire block diagram explained in this section.

5.0 An Experimental Example

In consumer audio applications, there are two popularly used sampling rates: 44.1 kHz for compact disc (CD) players and 48 kHz which is universally accepted for most other digital audio applications including digital audio tape (DAT) recorders. This section discusses, as an example, the sample rate conversion of a 16-bit digital output from CD players to DAT recorders. The ratio of two sampling rates can be represented by two integers which are computed by the least common multiple of the two sampling rates. The

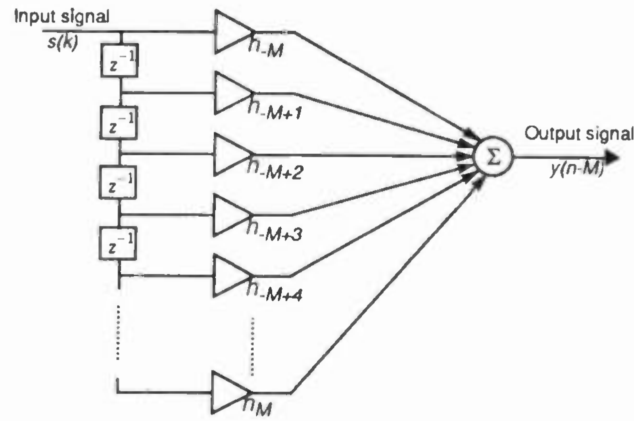

Figure 14. Causal implementation of the FIR filter

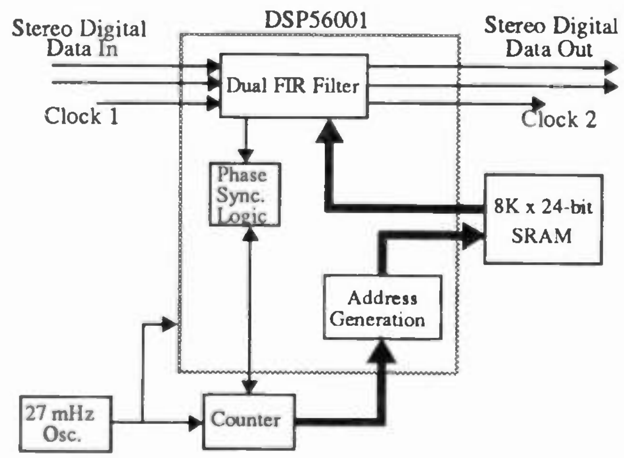

Figure 15. Hardware connection diagram

ratio of 44.1 and 48 kHz, for example, can be expressed as 147:160. In other words, at every 160 output data transfers, the input and output will be synchronized again as shown in Figure 9. Thus, the value of N can be 160 which is defined in (21) and for this implementation let M be 31 (*i.e.*, 63-tap FIR filter).

Figure 16 shows a block diagram of the hardware setup for the real-time implementation. In order to prevent the linear interpolation between samples from introducing errors greater than the input quantization error, it is necessary that the number of coefficients and the number of bits per coefficient be sufficiently large. These conditions can be satisfied only when the exact sample-rate change is known. However, the DSP56001 offers filter coefficients having 24-bits of resolution to assure 144 dB of dynamic range along with 16-bit input resolution. Also, during the FIR filter computation in (21), the 56-bit accumulator has been utilized to guard against unnecessary truncation or rounding— a common deficiency with popular 16-bit DSP processors.

As shown in Figure 16, the proposed method has been implemented using a DSP56000ADS application development system and a converter board which has two DSP56ADC16's (Motorola's 16-bit Sigma-Delta A/D converters) [13-15] and two 16-bit D/A converters. Software has been developed and down loaded on a SUN3/160 workstation, utilizing 10K words of memory to store the coefficients described in section 3.0. The input sample rate of 44.1 kHz is driven by the A/D converters and the output sampling rate of 48 kHz can be created by an external 48 kHz oscillator or generated by the DSP56001.

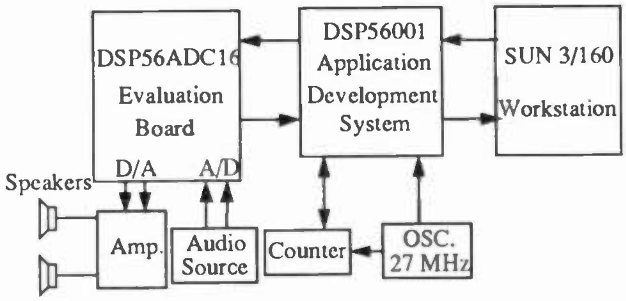

Figure 16. Block diagram for real-time implementation

Now, consider the software implementation of the technique. With a 27 mHz clock, the DSP56001 executes 13.5 million instructions per second (MIPS) [16]. In other words, the execution time for a 64-point FIR filter is only 4.8 μs. The input delay-line buffer is updated by the input sample rate indexed by k, while the output samples are computed in terms of output sample rate indexed by n. The input sample stream and the output sample stream are aligned relative to each other every 160 output samples. Thus, 160 blocks of coefficients are necessary to perform proper sample rate conversion. Since each block consists of 63 coefficient points as shown in Figures 8 and 11, a total 10080 words are required to store all coefficients. However, since the time-delay function of the *sinc* function in (21) has an odd symmetry with reverse coefficient indexing as

$$sinc(m-\tau(n)) = -sinc(-m+\tau(n)) \tag{27}$$

and the Blackman-Harris window in (14) is an even symmetric function, the total length of memory required for coefficient storage can be reduced by half, fitting into 5k (5120 actual) words of memory.

The optimized DSP56001 assembly code for the FIR filter in (21) can be written as

```
clr   a        x:(r1)+,x0   y:(r4)+,y0   ;clear a
rep   #62                                ;repeat next 62 times
mac   x0,y0,a  x:(r1)+,x0   y:(r4)+,y0   ;a=a+c(m)*x(n-m)
macr  x0,y0,a                            ;a=a+c(63)*x(n-62)
```

where **r1** is the register pointing to the input buffer which is modulo-addressed to accommodate only 63 current data points, while register **r4** is pointing to the current coefficient location. The Modifier Register **m1** is set to be 62 (input buffer size-1). This stage requires only 64 instruction cycles per output sample period, which yields a processing requirement of only 3.072 MIPS at 48 kHz of output sample rate. To accommodate stereo signals for sample-rate conversion at least 6.144 MIPS of computations are required with some extra set-up and reset instructions.

5.1 Computer Simulation Results

The performance of the proposed sample-rate conversion technique is tested using a single sinewave created with 16-bits of resolution. To estimate the STHDNR the following steps are used. First, find 2048 FFT (fast Fourier transform) with the 4-term Blackman-Harris window and normalize each bin by the peak value so that the input signal frequency always has 0 dB power strength. Due to the window effect the signal power is smeared to side bands so that 47 bins (23 lower frequency bins, 23 higher frequency bins and the middle (signal) bin itself) has to be removed from the FFT output bins prior to the total noise power calculation. Since the input signal is always real, the first 1024 bins, which represent from DC to the Nyquist frequency, can be used for evaluation. The next step is to sum the individual values of the rest of the bins and multiplied by 1024/(1024-47) which compensates the power of the removed bins.

Figure 17-a shows the spectrum of the input signal with 44.1 kHz sample rate when the signal is a sinewave at 3 kHz and the STHDNR is measured as 94.56 dB. The signal is then processed with the proposed sample-rate conversion routine with several sets of coefficients for performance comparison. The spectrum of the signal converted to 48 kHz is illustrated in Figures 17-b through 17-g which correspond to the number of coefficients, $2M+1$ = 7, 15, 31, 63, 127 and 255, respectively. Similar comparisons were performed with different signal frequencies and the resulting estimated STHDNR of the output spectra are tabulated in Table 1. As Shown in Figures 17-b through 17-g, the spectral peaks which may not be the harmonics of the input signal are decreased as the number of coefficients is increased. Also note that the input STHD-

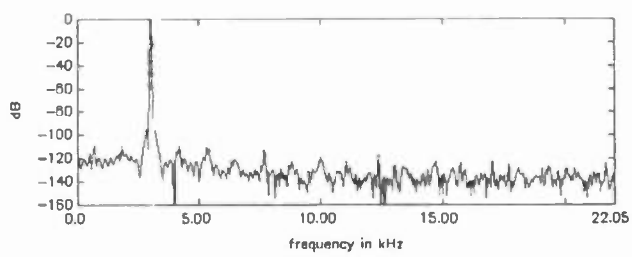

Figure 17-a. Spectrum of input data at 44.1 kHz

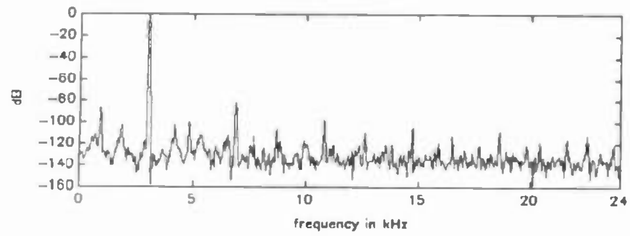

Figure 17-b. Spectrum of output signal when 2M+1=7

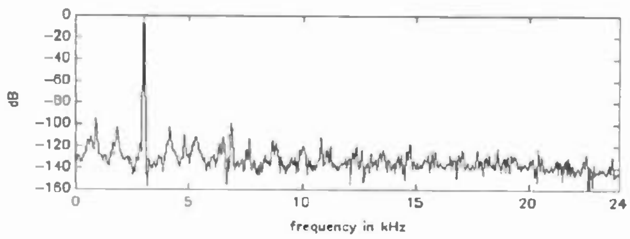

Figure 17-c. Spectrum of output signal when 2M+1=15

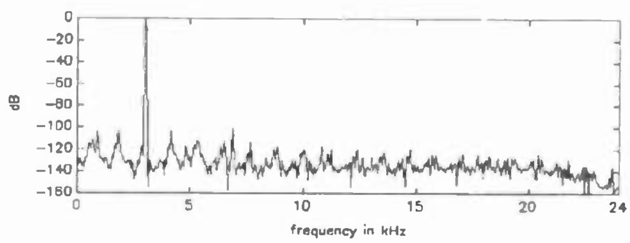

Figure 17-d. Spectrum of output signal when 2M+1=31

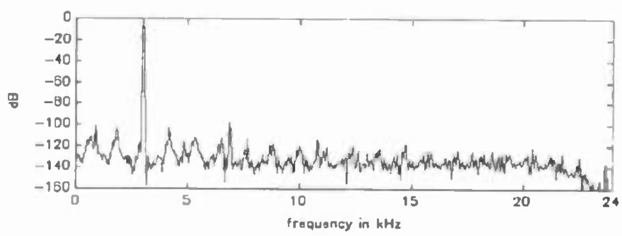

Figure 17-e. Spectrum of output signal when 2M+1=63

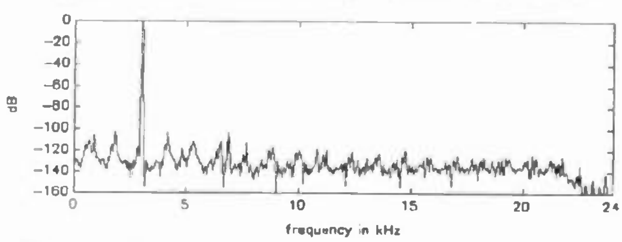

Figure 17-f. Spectrum of output signal when 2M+1=127

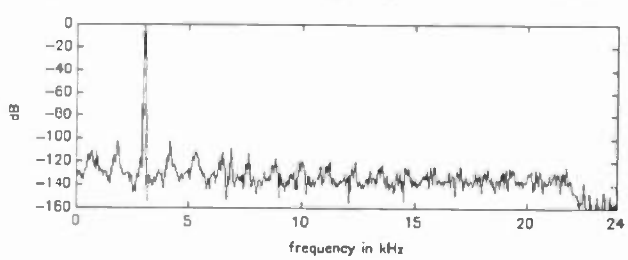

Figure 17-g. Spectrum of output signal when 2M+1=255

		Frequency			
	# of taps	1 kHz	3 kHz	10 kHz	20 kHz
Output STHDNR	255	94.69 dB	92.38 dB	88.81 dB	85.42 dB
	127	93.11 dB	91.29 dB	87.58 dB	84.23 dB
	63	89.21 dB	89.46 dB	86.46 dB	81.48 dB
	31	85.34 dB	86.72 dB	81.74 dB	72.55 dB
	15	78.89 dB	74.57 dB	75.58 dB	69.34 dB
	7	76.49 dB	70.58 dB	64.01 dB	61.08 dB
Input STHDNR		**95.10 dB**	**94.56 dB**	**91.56 dB**	**89.88 dB**

Table 1 STHDNR performance of the converters

NR were different even though the same 16-bit sinusoidal signal was generated. Thus, the difference between input and output STHDNRs gives a guideline for optimization. The memory requirements and number of instruction cycles used for different implementation are tabulated in Table 2. The number inside the parenthesis shows the capacity of the DSP56001 (27 MHz) to run the routine in real-time.

# of taps	# of words (24 bit)	# of inst. cycles/sec
255	20480	12.29 MIPS (95 %)
127	10240	1.14 MIPS (47 %)
63	5120	3.07 MIPS (24 %)
31	2560	1.54 MIPS (12 %)
15	1280	0.77 MIPS (6 %)
7	640	0.38 MIPS (3 %)

Table 2 Hardware requirements for the converters

For performance evaluation, two more sinusoidal signals which are the lowest (20 Hz) and the highest (20 kHz) signals in most of digital audio applications, are applied when the number of coefficients are 63. Examining the low frequency results in Figure 18, shows that, as expected, better than 16-bit performance is obtained. Usually the worst case performance is expected in the high frequency range due to the very limited resolution which can be utilized to represent the wave form. However, even when a 20 kHz signal is applied to the converter, there is a negligible increase in harmonic distortion as illustrated in Figure 19.

6.0 Concluding Remarks

A digital sample-rate conversion technique has been described which is based on a table lookup method. In the case where the two sample frequencies are a rational number, the table entries can be exact since all τ values are known. In the most general case a table having a finite size can be used and the filters with a τ close to the actual τ can be used. I has been shown that t he resulting performance is not only a function of the table size, but also the number of coefficients used in the FIR filtering. Depending on the

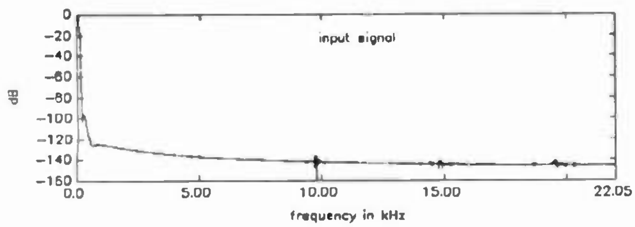

Figure 18-a. Freq = 20 Hz at 44.1 kHz sps

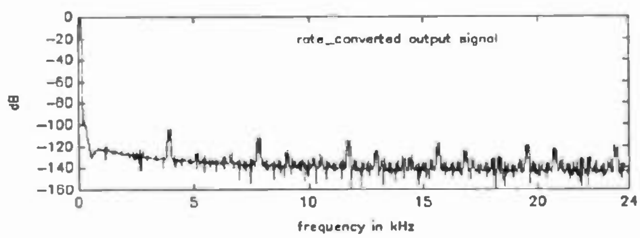

Figure 18-b. Freq = 20 Hz at 48 kHz sps

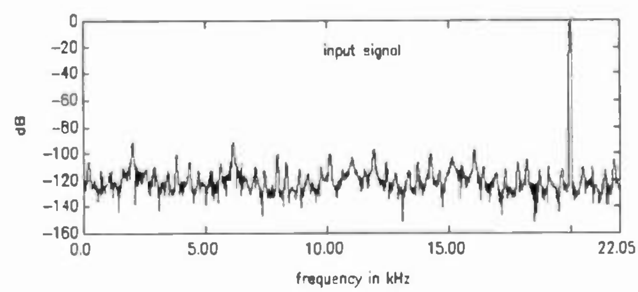

Figure 19-a. Freq = 20 kHz at 44.1 kHz sps

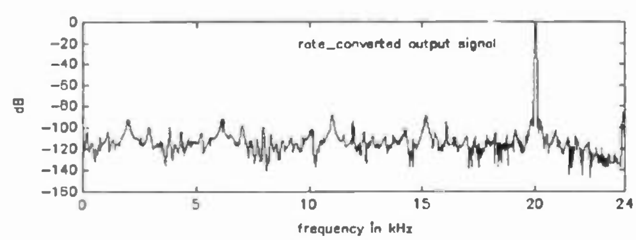

Figure 19-b. Freq = 20 kHz at 48 kHz sps

application, where more or less STHDNR is preferred, it can be optimized by choosing right system parameters using Tables 1 and 2. Also the feasibility of implementing the proposed method using the DSP56001 based system has been presented. By taking advantage of 24-bit coefficient precision, the system is capable of converting a digital signal with 44.1 kHz sample rate to a DAT compatible sample rate with 90 dB of signal-to-(noise+distortion). Although the DSP56001 can handle a stereo 127-coefficient FIR filter computation, the memory requirement may be much larger than the applications which can use only 15 coefficients to achieve about 70 dB of STHDNR. The approach described in the paper has the following design advantages: 1) it uses only digital filtering techniques; 2) it has linear-phase response for no phase distortion; 3) it executes in real-time with two-channels using only a single DSP56001; and 4) it obtains signal-quality compatible with 16-bit digital audio. The entire DSP56001 assembly code is included in Appendix A. Although an uniform input sample period was assumed for analysis and implementation, it is conceptually straightforward to extend this technique to non-uniformly sampled signals with unknown input sample rates.

Acknowledgment

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Appendix A

```
          page 132,60,3,3
          opt cc
start     equ    $0040     ;the scene of the crime
InBuffer  equ    $0100     ;starting Addr for Input Buffer
InBuffLen equ    $0100     ;size of input buffer
Table     equ    $0000     ;ptr to table of coefficients
TableLen  equ    64*160    ;length of coefficient table
K_PTR     equ    $00800    ;mpy constant for ptr computation
CLR_CNTRequ      $FFF0     ;R/W address to clear the counter
RD_CNTR equ      $FFF1     ;Read address for delay count
M_IPR     equ    $FFFF     ;Interrupt Priority Register
M_BCR     equ    $FFFE     ;Bus Control Register
M_PCC     equ    $FFE1     ;Port C Control Register
M_CRA     equ    $FFEC     ;SSI Control Register A
M_CRB     equ    $FFED     ;SSI Control Register AB
M_SSIDataequ     $FFEF     ;SSI Data Register(s)
;*************************
;* SSI Receive Vectors *
;*************************
          org    P:$0000C
          movep  X:M_SSIData,X:(r0)+  ;get the SSI data word
          movep  a,Y:CLR_CNTR         ;clear the delay counter
                                      ;the circuit ignores data
          movep  X:M_SSIData,X:(r0)+  ;SSI Rx w/ Exept. Status
          movep  a,Y:CLR_CNTR
;*************************
;* SSI Transmit Vectors *
;*************************
          org    P:$0010
          move   #0,r7                ;flag that a new sample is needed
          movep  b1,X:M_SSIData       ;ship the LAST filtered sample out
          move   #0,r7                ;SSI Tx w/ Exception Status
          movep  b1,X:M_SSIData
;*************************************************
;* Initialize DSP56001 and define pointers/buffers *
;*************************************************
          org    P:start
srconv    clr    a                    ;you'll need this soon...
          movep  a,X:M_BCR            ;BCR = 0, no wait states
          movec  a,sp                 ;init stack pointer
          movec  a,sr                 ;clear loop flag
          move   a,X:M_PCC            ;zero PCC to cycle it, reset SSI
          movep  #$4104,X:M_CRA       ;16-bit words,2 words/frame,
                                      ;SSI Clk=osc/4/(4+1)=osc/20
          movep  #$F1B0,X:M_CRB       ;Tx/Rx enabled, both Ints. enabled
                                      ;normal,cont.clk,async,Rx,FS(bit)
                                      ;Tx,FS(word),MSBout 1st,int clk
          movep  #$01FF,X:M_PCC       ;enable all SSI & SCI functions
;*************************
;* Initialize all pointers *
;*************************
          move   #InBuffer,r0         ;r0 points to input data
          move   #InBuffLen-1,m0      ;InBuff Is a circular buffer
          move   m0,m1                ;filter ptr needs same modulus
          bset   #1,r7                ;flag set = nothing to do...
          movep  #$3000,X:M_IPR       ;set SSI Interrupt Priority Level
          andi   #$FC,MR              ;unmask all interrupt levels
loop      jset   #1,r7,*              ;now, loop waiting for data...
;*************************************
;* Interpolate Next SSI_Tx Sample *
;*************************************
SSI_TX
          movep  Y:RD_CNTR,x0         ;x0 = delay in ticks from ext. cntr
          move   #K_PTR,y0            ;y0 = mpy constant to adjust ptr.
          mpyr   x0,y0,a r0,r1        ;compute ptr to filter coeff.s
                                      ;r0 may change...r1 = input ptr
          move   a1,r4                ;r4 will point into coeff. table
          bset   #1,r7                ;set flag for new sample needed
;*********************
;* APPLY FILTER *
;*********************
          clr    b      X:(r1)+,X0 Y:(r4)+,Y0;get data, coeff, Init B
          rep    m0
          mac    x0,y0,b  X:(r1)+,X0 Y:(r4)+,Y0    ;apply filter
          macr   x0,y0,b                    ;leave next output in "B"
          jmp    loop                       ;continue
          END    srconv
```

SPECTRUM EFFICIENT DIGITAL AUDIO TECHNOLOGY (SEDAT)™

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INTRODUCTION

In the early 1980s, Scientific-Atlanta engineers invented and patented the world's first satellite *digital audio transmission system* (DATS). The company developed DATS in concert with the major U.S. radio networks, ABC, CBS, NBC and RKO (now Westwood Radio Networks). Because each network adopted DATS, this technology has remained the de facto standard for satellite-delivered digital audio. Thousands of Scientific-Atlanta DATS earth stations provide the nation's radio network feeds.

In the last few years, in response to market conditions, Scientific-Atlanta initiated a development program to improve on its PCM-based DATS standard. Market research had indicated that radio broadcasters were interested in three areas of technology improvements:

- Spectrum efficiency
- Audio quality (20 KHz vs. 15 KHz)
- Network management capabilities

As a result, Scientific-Atlanta set out to deliver a new audio compression technology and product line. This was intended to transmit(transparently and efficiently) compact disk (CD) quality audio over typical satellite links.

Initial studies indicated that this technology should be capable of the dynamic range of CD players over a wide range of bandwidth and bit rate combinations. In other words, broadcasters wanted to achieve transmission of one source at 7.5-10 KHz bandwidth using 1 to 3 bits-per-sample, and another source at 15-22 KHz using averages of 2 to 4 bits-per-sample. To reach this kind of efficiency with little or no degradation to the perceived audio quality, Scientific-Atlanta research determined that the technology had to exploit the properties of the Human Auditory System (HAS). Further, the HAS frequency noise masking property would need to be fully exploited. In addition, it was recognized that the technology should be easily implementable on standard third generation Digital Signal Processors (DSPs) becoming widely available to the market at decreasing prices .

Theoretical and empirical results from early investigations indicated that a form of Adaptive Transform Coding (ATC)[1] was optimum, not only in terms of exploiting the noise masking property of the HAS, but also in terms of coding efficiency, i.e., using a minimum number of bits per sample.

Scientific-Atlanta engineers achieved breakthroughs in coding technology. These breakthroughs (proprietary enhancements to basic coding techniques) improved ATC's capability to exploit noise masking properties of the HAS. Additional algorithm development resulted in extremely efficient implementations that were easily capable of running on any third generation DSP platform.

These resulting algorithms, along with the technology to implement, encode, transmit, decode and manufacture supporting systems, is a Scientific-Atlanta technology family called SEDAT™, for Spectrum Efficient Digital Audio Technology.

In the fall of 1990, the ABC and CBS Radio Networks jointly announced decisions to implement SEDAT. Each network will upgrade its existing Scientific-Atlanta DATS system with new SEDAT encoders at the networks' headends and decoders at each affiliate's downlink. Conversion of affiliate decoders has been scheduled for the second quarter of 1990. This paper outlines one part of SEDAT technology, the SEDAT algorithm.

SPECTRUM EFFICIENT DIGITAL AUDIO TECHNOLOGY SEDAT™

Properties of the Human Auditory System (HAS)

Researchers in psychoacoustics discovered many years ago that the HAS processed signals very much like a bank of filters or a spectrum analyzer.[2] Anatomically, this was due to the interaction of the inner hair cells and the basilar membrane of the cochlea, which produce a bank of resonators. The overall effect of the basilar membrane, the individual hair cells and the individual fibers of the auditory nerve results in a resolving power which can be said to be equivalent to a bank of bandpass filters which have become known as critical bands.

Although the critical bands have continuously overlapping center frequencies, certain frequencies have become common in analysis. These can be compared to one-third-octave filters with 23% bandwidth. It is common to see 24 critical bands listed in various tests across the audible spectrum.

Closely related to the critical bands is the phenomenon known as noise masking. Fletcher's original procedure for defining the critical bands involved mixing wideband noise with a pure tone and asking listeners to determine when the tone was just audible.[3] The bandwidth of the noise required to mask the tone could be used to define the critical band.

It was also discovered that, when a tone was only 20 dB or so higher than the surrounding noise, the HAS would mask the noise, meaning that the noise became inaudible. The degree of masking is also a function of its frequency displacement from the tone. As the noise is moved further from the tone the amount of masking decreases.

The lesson in this for audio compression is that, if the quantization noise caused by the compression can be constrained to small frequency bands, then the signal to quantization noise within the band only has to be about 20 dB down for the HAS to ignore it.[4]

Adaptive Transform Coding (ATC)

ATC has been applied in speech and image coding since the mid 70s.[1,5] Obviously, the transform theory was widely available before this time, but first generation DSPs began to make it practical to implement inexpensively.

A basic principle of transforms is the decomposition of a block of time domain signals into a set of independent components called *basis vectors.* If the goal is to represent the most signal power with the least number of basis vectors, then an optimum transform in terms of minimizing the mean square reconstruction error is given by the Karhunen-Loeve Transform (KLT).

The KLT expands the input signal into a set of basis vectors which are ordered by the energy of the vector present in the signal. For compression, we could therefore use only the lower n components (n being dependent on the amount of compression) for transmission and reconstruction. Unfortunately, the KLT is computationally prohibitive because the transform is dependent on the input signal. Fortunately, the Discrete Cosine Transform (DCT) is a very good approximation to the KLT for the class of signals in which we are interested and can be computed very efficiently.

Although we cannot guarantee ordering of basis vectors by power in the signal (the ordering is always fixed), the DCT does exhibit the energy compression characteristic of the KLT. In other words, adjacent vectors tend to have nearly equal values and the overall spectrum tends to fall off smoothly toward the higher frequency components.

The adaptability of the ATC comes from adapting many quantizers for the quantization of the transform coefficients. The quantizers are adapted with respect to the number of bits used and the step size or scaling. By the use of many quantizers, we can constrain the quantization errors to small frequency bands, thereby optimally exploiting the noise masking property of the HAS. It is this combination of near optimal efficiency due to the DCT and optimal exploitation of the noise masking property which makes ATC so desirable for audio compression.

The SEDAT Algorithm

A block diagram for the Scientific-Atlanta SEDAT encoder and decoder is shown in *Figure 1*. Audio samples are input to a Time Domain Aliasing Cancellation (TDAC) analysis function, which takes in blocks of 1024 time samples and applies to it a time domain window.[6] The composite block is transformed by a 1024-point modified DCT, out of which only 512 samples are unique. The analysis window is then moved by 512 time samples relative to the previous block – and the operation is repeated. Thus, for every 512 input samples, 512 transform coefficients are produced and encoded. (The synthesis portion of the TDAC technique is performed in the decoder.)

TDAC allows one to use a longer overlapping analysis window and guarantees complete elimination of the time domain aliasing produced by the overlapping blocks, without increasing the number of coefficients to be compressed. Longer windows decrease the coefficient leakage into adjacent coefficients, thereby providing higher frequency resolution.

Encoding consists of quantizing the TDAC coefficients using a bank of quantizers across the frequency domain. The TDAC coefficients are divided into frequency bands or sub-bands such that all the coefficients in the sub-band will be quantized with the same quantizer. The TDAC analysis provides very high frequency resolution across the audible spectrum, allowing these sub-bands to be closely matched in bandwidth to the critical bands whose bandwidths increase with frequency.

This gives SEDAT a decided edge in both bit efficiency and perception over traditional sub-band approaches, which typically use a small number of uniformly spaced sub-bands. Variable bandwidth sub-bands allow improved bit utilization without perceptual degradation.

Proprietary bit allocation and scaling algorithms are run against the TDAC coefficients to pick the number of bits and scaling for each quantizer. The actual transmitted information block thus contains quantized coefficients and side information necessary for the decoder to dequantize them.

In an error-free environment, we could simply send this block to the decoder for dequantization and reconstruction. Unfortunately, in any real world communication link we are faced with the prospect of bit errors which corrupt the transmitted data. SEDAT uses several levels of error correction and concealment to allow its use at bit error rates as low as 10^{-4} with imperceptible degradation. The side information and the Most Significant Bits (MSBs) of the low frequency coefficients are protected with a Reed-Solomon code.[7]

The SEDAT decoder must perform two functions before it can get on with the basic job of dequantization and reconstruction. First, it must acquire and maintain block synchronization, i.e., it must know where the beginning of the block is. In SEDAT, synchronization is achieved using information in the error correcting codes, and thus requires no additional overhead bits. Once block sync is achieved the decoder begins normal block processing.

Block processing commences by checking the coded information for errors. Four results are possible:

1) no errors have occurred;
2) correctable errors have occurred;
3) detectable but non-correctable errors have occurred;
4) non-detectable errors have occurred.

In the first two cases, the potential errors are corrected and the processing continues. In the third case, the bit allocation and scaling of the previous correctable block are substituted as an error-concealment strategy and processing continues. In the fourth case, non-detectable errors disguise themselves as either no errors or correctable errors and the algorithm proceeds accordingly. This last case has an extremely low probability of occurrence (one of the strengths of the Reed-Solomon code).

After error correction (and possible concealment), the decoder reconstructs the frequency domain coefficients from the compressed coefficients and the side information. The resulting coefficients are transformed back into time domain samples, and the synthesis half of the TDAC technique is performed by overlapping and adding the results of successive blocks.

The robust ability of the Reed-Solomon decoder to detect non-correctable errors is the basis for the block synchronization and audio muting functions. Simply put, if the decoder detects a sufficiently large number of non-correctable blocks in a given length of time, it determines that the link quality has degraded below an acceptable level and mutes the audio output.

If the error correction performance is poor enough (typically occurring during extended intervals with bit error rates of 10^{-2} or worse), the decoder assumes that it has lost block synchronization and begins re-acquisition.

What's next?

The SEDAT algorithm has applications in many other areas. For example, digital audio compression is possible in terrestrial backhaul, telephony, digital music storage, broadcast automation, digital radio broadcast, simples RF transmission, cellular telephone and a host of others.

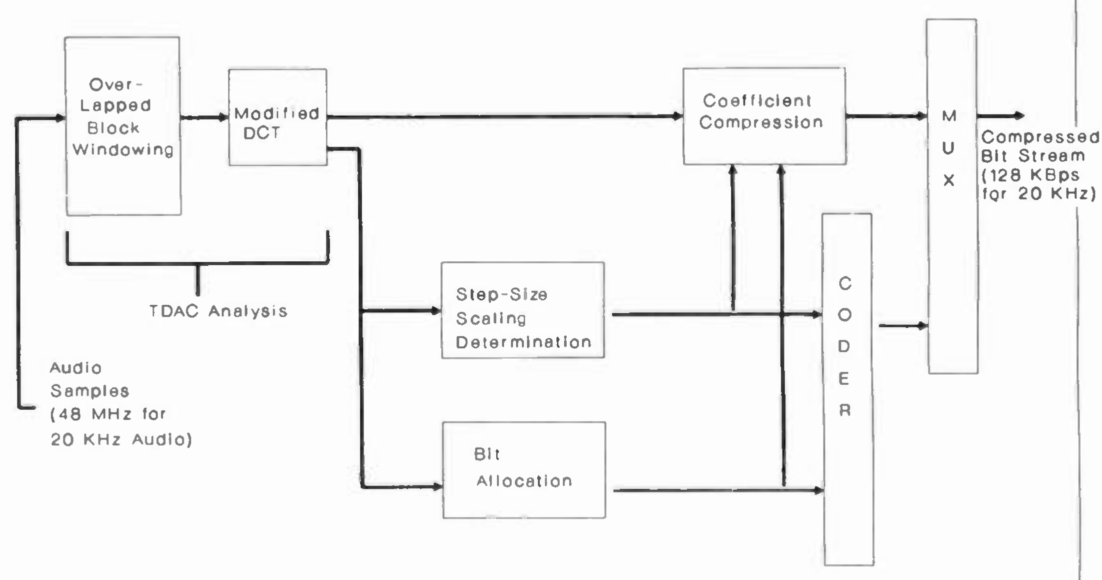

Figure 1 - SEDAT Encoder

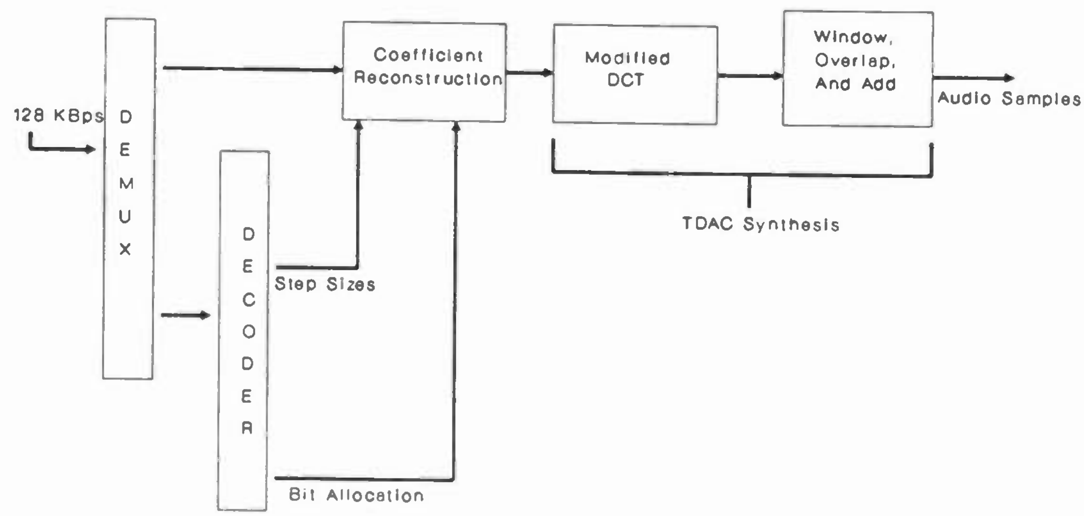

Figure 2 - SEDAT Decoder

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AN INTEGRATED DIGITAL SYSTEM FOR BROADCAST AUDIO

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A variety of digital audio "solutions" for the broadcaster have been proposed in the past few years; but none have attained widespread acceptance by the industry. Perhaps the primary reason is that broadcast audio deals not just with the basic requirements of recording and playback of program material; but with a number of control and management issues as well. A true solution must address such concerns as scheduling, format enforcement, ease of access, security, and integration with other programming sources; in addition to the fundamental ability to record and play high quality audio. And all of these issues must fit within an acceptable conceptual "model" that provides a comfortable and familiar environment for the broadcast user.

This paper outlines a solution to the digital audio needs of the broadcast industry which addresses all of these issues in an integrated fashion; providing high quality, efficient audio storage and retrieval within a framework that fits into the broadcaster's world instead of forcing him to learn to live in a new one.

Digital Audio in Broadcasting

The ability to capture and replay audio in digital form has been available to the professional recording and broadcast industry for some time now. The basic concepts of Analog to Digital/Digital to Analog conversion, with data storage on various media in between, are well documented and applied to dozens of existing products; ranging from PCM (Pulse Code Modulation) data stored in pseudo-video form on video recorders to huge and expensive multitrack systems in DASH (Digital Audio - Stationary Head) format used in recording studios.

In recent years, this expensive technology has made its way to the consumer market in the form of Compact Disks as a playback-only medium and DAT (Digital Audio Tape) for full record-play functionality. These have "backed into" the broadcast marketplace even though few products are specifically aimed for it. And most recently, we have seen the introduction of Mass Storage Digital Audio devices using hard disk media to store a complete inventory of commercials and other short audio segments.

In attempting to analyze why this clearly superior technology hasn't been more widely accepted in the broadcast industry, the following three areas of concern seem to stand out:

Access

> A product must offer a means of locating and readying for play the exact selection the operator is looking for; in a quick, simple and efficient manner.

Programmability

> In addition to being able to randomly select something at a moments notice; an operator can benefit from having those elements of his show that are known in advance (such as his spot schedule) pre-programmed for him. Programmability also allows the product to run for some length in an unattended mode.

Control

> The product must allow for outside control over the functions of Access and

Programmability. Simply having these capabilities provided internally is not enough; the product must be able to operate in conjunction with other elements of a system.

With these criteria in mind, let's examine the current state of the art in digital audio equipment to see why none have literally taken the market by storm.

Control and Source Equipment

There are two types of equipment used in the studio of a radio station today: Source Equipment and Control Equipment. Source Equipment covers the range of products that are a source of audio, such as Cart Machines, CD Players, and DAT Recorders. Control Equipment encompasses those products used to control audio sources, including Consoles, Automation Programmers, and Live Assist Systems.

In most instances, there is a clear distinction between Source and Control Equipment; each filling a part of the overall puzzle. However, digital technology tends to blur this line; simply because the power to do more things is available. Consider the fact that most CD players have built-in programming capability to select the order in which tracks can be played. For the home user, this provides a convenience; however, in professional use this capability is seldom used. It is preferable instead to have individual remote control capability in each *source* that can be centralized at an appropriate point of *control*.

The product area where this division is most indistinct, however, appears to be in Mass Storage Digital Audio; where a number of short selections (typically commercials) are held in digital form on a large storage medium such as hard disk. Because of the unique nature of such a device, it is difficult to distinguish whether it is a piece of source equipment, control equipment or both. Obviously, it is a source of audio; so it

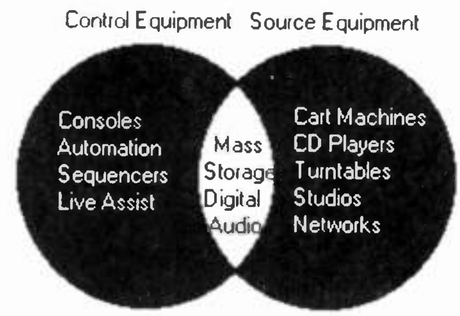

Figure 1 Mass Storage Digital Audio devices have attributes of both Source and Control Equipment.

must qualify as source equipment. However, most systems now available have significant levels of programmability. In addition, most of these systems are also geared towards use in satellite programming situations where they not only provide control over the *internal* material played, but also for *external* sources such as the satellite audio and cart machines.

The question with such devices is not whether to provide control and programming capability; these systems need it to provide for the access and programmability criteria defined earlier. The real problem is determining *where to stop*! Once a certain level of capability is reached, additional program and control functionality is redundant and, in some cases, detrimental to the performance of the product in the station. More capability usually means higher cost. Sometimes it also means sacrificing some measure of performance or making the product more complex and harder to use. In all products of this type, there is a fine line beyond which any additional capability is simply a burden.

In the development of Broadcast Electronics' contribution to the Mass Storage Digital Audio market, the **AudioVAULT**, great concern was placed on the appropriate division of source and control responsibility. As the balance of this paper will explain, the answer involved not just a single product; but rather the shared capabilities of two separate but interrelated technologies.

AudioVAULT

As with all present entries in the Mass Storage Digital Audio area, the Broadcast Electronics AudioVAULT is a record & playback device using hard disk storage as a random access medium for short duration audio element such as commercials, jingles, ID's and liners. The basic design criteria for the AudioVAULT encompassed several key requirements which were felt necessary to provide a suitable environment for radio station use:

1) The unit would be required to allow recording and playback of audio material simultaneously.

2) The unit would allow multiple, asynchronous access from two to four stereo output channels from the stored "inventory" of recorded material. One copy of an audio selection could be played to four separate output pairs independently.

3) Access to the stored inventory would be virtually instantaneous; involving no cumbersome searching and cueing procedures.

4) The unit would be equally at home in a controlled environment, such as an automation system where all selections are programmed in advance and sent to the device as needed; or in an interactive "Live Assist" situation where the operator accessed selections one at a time.

5) The unit would incorporate a graphical user interface to provide necessary feedback for system operation; such as dynamic meters during recording and "progress displays" during playback.

To meet all of the above requirements involved the selection of adequate hardware and software platforms. The hardware consists of a computer section, which is basically a 286 or 386 PC compatible subsystem of CPU, Hard Disk Drive, and VGA Color Graphics; and an audio section, which contains the actual A/D, D/A and storage "engine" running under the direction of the computer. The audio section is a subsystem manufactured by WaveFrame, Inc.; a widely recognized leader in the field of digital audio. This component provides a working equivalent of an 8-in, 8-out digital multi-track deck; however, in this instance, it is used more like 4 independent 2-track decks accessing the same storage inventory.

For the software platform, the choice was made to use Microsoft Windows 3.0; for a variety of reasons. First, it offers a stable graphical development environment with device-independent control of input and output. Second, as a cooperative multi-tasking environment, it offers the potential to co-exist with other Windows-based programs. And finally, Windows offers a consistent user interface; so customers familiar with the rapidly increasing number of Windows programs have less of a "learning curve" to begin to use the program.

Outside of the hardware and software platform issues, a choice also had to be made about the "model"; or the way the AudioVAULT would be made to simulate "real world" products. In the radio broadcasting environment, there is only one model to build upon: the Cart Machine. While there are many physical differences between the way the AudioVAULT works as compared to a cart machine; there are many conceptual similarities. We chose to exploit these resemblances in order to make the operation of the AudioVAULT more intuitive to station personnel already comfortable with the use of traditional equipment.

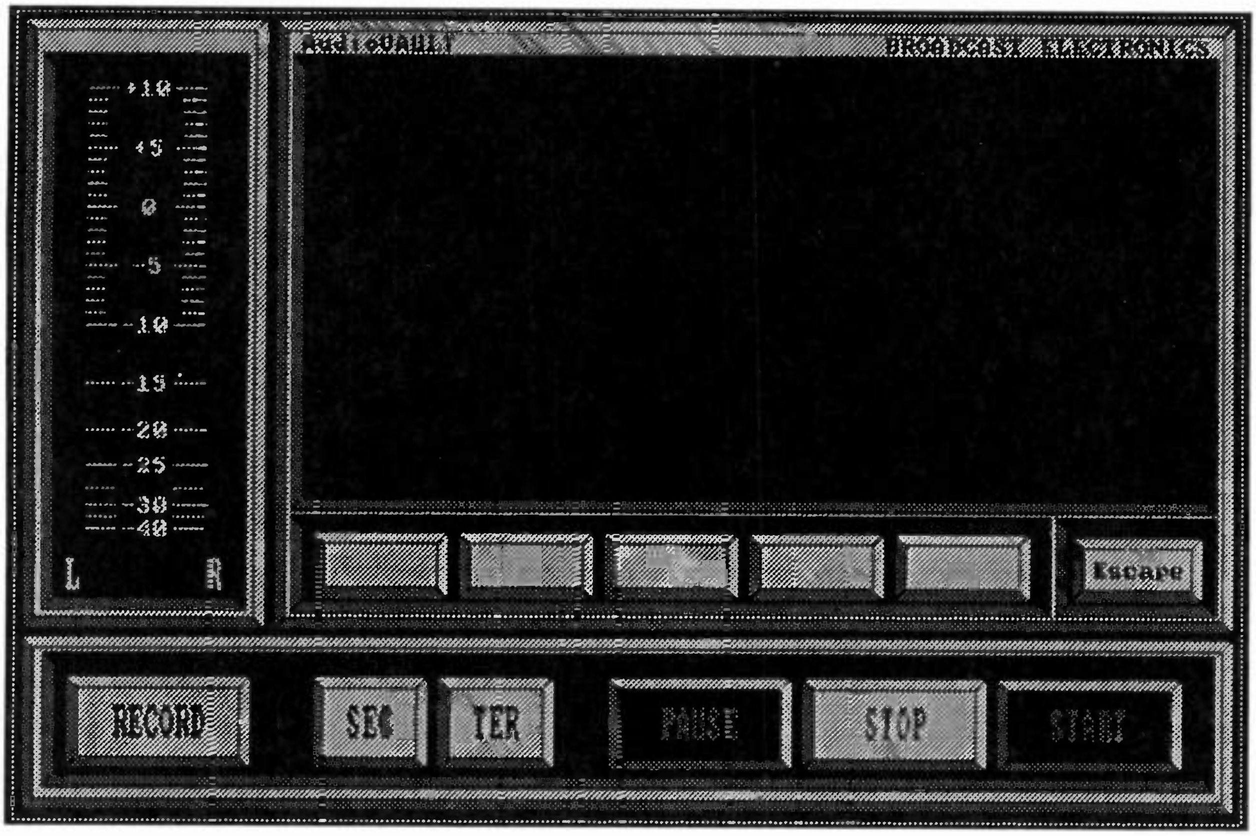

Figure 2 A Screen Display from the Broadcast Electronics AudioVAULT

The "Cart Model"

With any type of Mass Storage Digital Audio device, the processes of recording and playing back audio material are significantly different than those for cart machines and other analog recorders. For one thing, the media is invisible - there is no physical handling, erasing, recording, and storage like there is with an analog cart. Audio simply "disappears" into the inside of the system; where it magically reappears when you request it for playback.

However, the operator must feel comfortable with the processes involved in getting audio in and out of the system. For this reason, Broadcast Electronics decided to make the operations and terminology mimic those that the operator was already familiar with. The software platform provided this capability in the form of "object orientation"; in which software representations of real-world "objects" are made to behave as if they were truly physical entities manipulated by the operator.

For example, a basic object in the AudioVAULT is the "Deck"; a software representation of a Cart Machine. The Deck has the attributes you would expect for a Cart Machine: A START button, a RECORD SET button, meters, a STOP button, RUN and READY lights, and a place to put a "Cart" object to be played. The operator manipulates these familiar controls in exactly the same manner as he would a Cart Machine; and he gets the results he would expect, even though the internal operations involve foreign concepts like hard disks and Digital/Analog conversion.

Likewise, the audio selections are represented by objects in the AudioVAULT software. The most fundamental object is a "Cut": a single mono or stereo audio segment. The operator records a Cut in a familiar manner: he presses the RECORD SET button, adjusts the record level on the meters, and presses the START button. The Cut is concluded by pressing STOP. A Cut has additional attributes such as a Cut Title, a Length, and information about the time and date when it was recorded; serving the same functions as a paper label.

Most Mass Storage Digital Audio devices provide the capability of recording the equivalent of a Cut on the AudioVAULT. However, because of the sheer capacity of the storage media involved, a device may end up with literally thousands of individual audio selections. Selecting the desired Cut from this inventory becomes a major chore for the operator.

Analog carts handle this problem by allowing multiple Cuts on a single cart; and then rotating the Cuts automatically when the cart is played. The AudioVAULT provides exactly the same capability by allowing one or more Cuts to be assigned to a "Cart" object. Selecting a given Cart will play its assigned Cuts in rotation; although any Cut may be directly accessed if desired. The effect is to subdivide the overall inventory of selections into a more manageable group of Carts; much in the same way subdirectories divide up file storage on a PC's hard disk.

Stacks and PlayLists

Because of the power available in the AudioVAULT, it was possible to implement higher-level objects made up of Carts to extend the level of programmability to the operator. Two mechanisms are provided to allow for multiple Cart objects to be manipulated: "Stacks" and "PlayLists". A Stack is exactly what its name implies: a Stack of Carts. Much in the same way as an operator might stack up analog carts for easy access to selections he knows he may need quick access to (such as jingles or PSA's), the AudioVAULT allows setting up Stacks made up of individual Carts. A unique benefit of these Stacks, however, is that a single Cart may exist in more than one Stack; something an analog cart could not do without duplicates. A Stack is an inherently random access mechanism; allowing any Cart to be quickly addressed and played. In effect, Stacks provide a way to subdivide the inventory of Carts the way that Carts subdivide the inventory of Cuts.

A PlayList, on the other hand, is more of a sequential access mechanism. This is an object that specifies the particular order that certain Carts are to be played; and has the ability to enforce that order. A PlayList may be created by the operator "on the fly", or it may be prepared for him "off-line" in the form of a simple ASCII text file which is later loaded into the AudioVAULT.

The most obvious use of the PlayList is to define the scheduled commercials for airplay. This is typically information that is known well in advance; and is usually of an important enough nature to try to ensure that it is played in the order scheduled. By having all his commercials pre-scheduled for him in a PlayList, the operator can concentrate on the balance of his show.

PlayLists also form the fundamental basis of the AudioVAULT's programmability. The operator need not be present to execute the elements of the PlayList. When so scheduled, the PlayList can execute automatically for any desired period; allowing for a short break for the operator or even providing the ability to program the system for full 24 hour automatic operation. This, incidentally, raises one of the most likely uses of the AudioVAULT: Satellite Programming.

Radio From Space

Satellite programming has been around since about 1982; and has become a major force in broadcasting today. It's advantages are obvious: it allows any station, even in the smallest markets, to sound like a major market station with high-caliber talent and programming. For the price of a dish in the back yard and a monthly fee, 100% of a stations programming needs can be met.

From the very beginning, automation technology has played a large role in the development of satellite radio. After all, it made little sense to replace an announcer with the satellite; and still have a person feeding commercials into cart machines for the spot breaks. However, most automation systems were overkill for the relatively simple requirements of satellite formats; most were too expensive; and few had the ability to handle the automatic changing of local liner's and ID's when the satellite announcer changed; meaning someone had to attend to the system several times a day.

Virtually all Mass Storage Digital Audio devices have been directed in some fashion towards satellite programming. Since all systems have some level of programmability, they allow for the pre-scheduling of the local spot breaks. Since all of these systems offer "instant access" to the inventory of material, they are well suited to playing the ID's and Liners which come at the whim of the satellite announcer. And since the storage capacities of these units allow for holding all Liners and ID's for all announcers; they provide an easy method of switching between them at an announcer change on the satellite format.

Ironically, it is exactly this capability that forms the paradox of Source Equipment vs. Control Equipment orientation for the product. At what point does the addition of features designed for providing control capability such as the pre-scheduling of spot breaks or announcer changes begin to detract from the overall utility of the device for its intended purpose: as a source of audio?

Broadcast Electronics struggled with this question and arrived at a simple and elegant answer; which can be stated succinctly as follows:

Rule #1 of Source/Control Technology

> **"A piece of Source Equipment ceases to be a piece of Source Equipment and becomes a piece of Control Equipment when it is required to provide machine control to other sources outside itself."**

The key word in this statement is machine control. This infers that some type of physical control; such as a contact closure or control data stream, is being provided by the unit to control a source other than itself. Whenever this occurs, the system's complexity is necessarily expanded beyond it's original intent, by a margin that may never be truly defined. What kinds of external source control is allowed? What happens when a new type of external source comes along that has radically different machine control requirements? Where does it all end?

You will notice that sources which do not require machine control are not prohibited. These would be (by definition) limited to audio feeds, such as studios, networks, and (surprise) satellite feeds. Since these sources merely involve the switching of audio signals on and off, they add no complexity to the system, and are guaranteed to remain simple forever.

In many Mass Storage Digital Audio systems, the switching of audio outside the system is accomplished by an external audio switcher. This can be considered a violation of Rule #1 of Source/Control Technology because the control of the audio switcher requires machine control. Broadcast Electronics addresses this requirement in a much simpler way. The AudioVAULT has the capability to support up to 8 audio inputs (in 4 stereo pairs); each fully addressable within the AudioVAULT as a digitized audio signal capable of being routed to any of the 4 stereo output pairs (using the digital loop-through or source monitoring function). In effect, all audio switching is accomplished by fully digitizing the input signal and routing it in the digital domain! The command to perform this switching function is entered in the PlayList as CH2&3 (indicating Channels 2 and 3).

To accomplish the playing of Liners and ID's, a special Stack known as a Pre-Load Stack is used. This takes a Stack of up to 8 Carts and pre-loads them into the system's cueing mechanisms; so that each may be started directly upon receipt of a contact closure from the satellite decoder. A given contact closure is directed to a given Cart; and normal rotation of Cuts within the Cart insures a different Liner or ID each time it is called upon. When the time comes for a new announcer to appear on the satellite, a simple Stack Change command is entered in the PlayList to abandon the current Pre-Load Stack and replace it with a new one; holding the Liners and ID's for the upcoming announcer.

Therefore, all requirements for most satellite formats (and perhaps many other types of formats as well) can be accomplished using only the AudioVAULT without violating Rule #1. But what happens when the specific demands of the station require that multiple sources of audio; some needing machine control, must be addressed? In this instance, Broadcast Electronics decided that the most appropriate response is to relegate the AudioVAULT back to its original intended purpose: as an audio source product; and to provide for the control requirements with a product specifically designed for this function.

The CORE 2000

The Program Automation business has been around for 25 years or more; evolving from relatively simple electro-mechanical systems to complex dedicated microprocessor control configurations. In the mid-80's, it became apparent that it was time for a new generation of program control products, for a variety of reasons:

1) To take advantage of the developing PC industry to provide a powerful control center with attractive features (disk storage, large memory capacity, and color graphics displays) at a reasonable price.

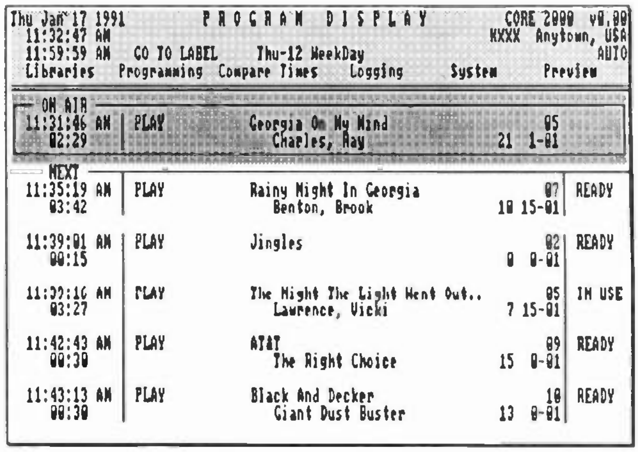

Figure 3 The Core 2000 Program Display Screen

2) To address the rapidly evolving source equipment situation; including exotic sources such as CD jukeboxes and RDAT's.

3) To provide a new level of control software that allowed not only for sophisticated automation control; but laid the groundwork for unprecedented Live Assist capability using sophisticated sources.

The CORE 2000 was developed to address these requirements. Consisting of a new RS-232 controlled audio switcher/mixer and a standard PC as the "brains"; the system was designed with the following basic requirements in mind:

- Addressability for up to 36 source inputs; each with flexible machine control and full audio control including VCA level adjustment under computer/database control. This was important for sources such as CD jukeboxes which could require level adjustment on an event-by-event basis.

- Independent "Smart" Source Interfaces for each input; with the capability for sophisticated parallel I/O (for such sources as Sony CDK-006 CD Changers) and Serial I/O (RS-232, RS-422, and infrared-remote-controlled sources).

- Backward compatibility with all present automation source equipment (Reel to Reels, Single Play Cart Machines, and Random Access Sources).

- Forward compatibility with known and unknown sources; using full microprocessor control capability on each source interface.

- Standard 286 or 386 PC compatibility, allowing for tremendous power at an unprecedented price.

- Extensive database-supported programming capability; allowing operators to work with the system in their own terms (Song Titles and Artist, Commercial Names, and exact event durations).

In short, the CORE 2000 becomes a Control Equipment partner to the AudioVAULT when the particular format requirements would exceed the control capabilities of the AudioVAULT.

Traditional automation systems included a wide variety of source equipment; such as Reel to Reels for music; Random Access Cart sources form commercials, and Single-Play Carts for such elements as Jingles, ID's, News, and Weather. Using new source equipment such as CD changers like the Sony CDK-006, and Mass Storage Digital Audio sources like the AudioVAULT; a Core 2000 system could provide significantly more capability in significantly less rack space. And given the fact the such a system is configured with complementary Source and Control equipment technology; the appropriate division of task responsibility is maintained.

In what cases, then, would a Mass Storage Digital Audio device like the AudioVAULT alone be appropriate? In following the general guideline of Rule #1, one use could be for a satellite-only format in which the only external source inputs were those for the satellite itself, and perhaps with a separate network news feed. Another appropriate configuration might be in a Live Assist situation in which the announcer intends to remain in full control; and the only cases of semi-automation are for spot breaks.

Conversely, an inappropriate situation for a lone Mass Storage source would be a part-time satellite operation in which the off-satellite programming was originated from sources requiring machine control; such as contact closures to one or more cart machines. In such a case, a clearly more flexible configuration would add a system controller such as the CORE 2000 to the system; to more effectively manage the multiple requirements of the various sources.

Conclusion

It is tempting to look at the capabilities of the PC's that are so common in todays broadcast product market; and to assume that with just the addition of "a little software" a given product could be made to do more than it was originally intended. Indeed, many products today demonstrate this philosophy. However, sometimes it is better to step back for a bit and examine whether or not the addition of some feature or capability might actually detract from the overall value of the product; because it is suddenly more complex and harder to use.

The trend today sometimes appears to be heading towards a "one product does all" approach. Sadly, this concept ultimately portends products which may do everything; but are too difficult to understand and use. It may be wise at this point to realize that while "one might fit all", perhaps "two might fit better".

AN ALL DIGITAL CD QUALITY STUDIO-TRANSMITTER LINK FOR THE 950 MHZ BAND

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Abstract - A new digital modem has been developed which converts CD-quality audio to a spectral-efficient digital signal that is ready for transmission over standard analog FM STLs. Excessive bandwidth requirements have in the past made transmission of digitized high-quality program material incompatible with 950 MHz Studio-Transmitter Link radio usage. Performance tests have shown the resulting digital FM signal to be compatible with existing analog services. This paper describes the considerations that drove the development of this modem as well as performance and compatibility tests.

INTRODUCTION

Digital Audio Tapes (DATs), Compact Discs (CDs), digital carts and other digital storage media have made significant inroads toward the culmination of the "all digital" radio station. The benefits of this metamorphosis are generally appreciated and accepted. The digital media is inherently more robust and avoids the noise and distortion accumulation that accompanies analog stages in the audio chain. This trend toward digital component replacement of their analog counterparts has initially targeted the weak spots in the processing chain. One stage in that chain that could benefit from digital quality is the Studio-Transmitter Link (STL). Resting at the end of the processing chain, it sets many of the crucial chain characteristics. But digital improvements in STL technology have been limited by the current investment in 950 MHz analog FM STLs. Also, the large bandwidth required by digital audio obviates the use of this band when using conventional STL technology.

Recent advances in digital signal processing have solved this dilemma and fueled the development of a 16-bit digital stereo STL that operates through existing 950 MHz analog STL equipment. Figure 1 illustrates the typical configuration using the digital modem over an STL radio link. The digital encoder converts audio program and auxiliary data channels into a shaped digital baseband signal. From this, the STL transmitter can generate a spectrally compact digital FM signal that is compatible with existing analog FM services. At the receiver, an inverse operation is performed to recover the original program information with perfect digital accuracy. The digital link delivers audio with exceptional transparency that has not been attainable with strictly analog links. Direct PCM inputs and outputs facilitate a 100% digital processing chain.

This report describes the benefits, technology considerations and performance requirements that lead to the development of the Moseley DSP-6000 digital STL system.

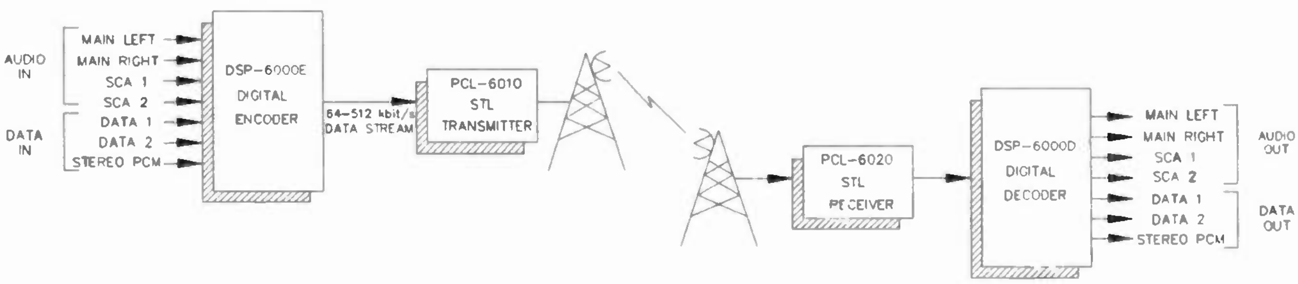

Figure 1
Digital STL in Retrofit Configuration.

BENEFITS OF DIGITAL TRANSMISSION

Degradation-free multiple hops

A standard analog STL system adds a certain amount of noise to the audio program (though these contributions may be acceptable in a well designed system). For multiple STL repeater hops, these noise and distortion products add. The signal-to-noise ratio (SNR) can never be better than the first hop and will continue to worsen with each hop. In a digital system, so long as there is sufficient SNR to regenerate the data accurately, the last hop will produce the same data as the first without degradation of the SNR distortion or frequency response. The 20th hop will be as clean as the first.

Constant audio SNR during fades

Audio program SNR is approximately proportional to the received carrier power in analog radio transmission. In digital transmission, so long as the data is received cleanly, SNR remains constant at its maximum level. Fades have no effect on SNR.

Higher system gain

Most analog signals are considered unacceptable when their Carrier-to-Noise (CNR) reaches 30 dB. Most digital systems will continue to regenerate data correctly at a CNR ratio of 15 dB. Their audio SNR will remain constant until that threshold is reached.

No crosstalk

Because left, right and auxiliary audio program channels are multiplexed digitally rather than in an analog fashion, there is no crosstalk between any of these channels. Thus, concerns for STL effects on composite stereo generators or SCA subcarriers are eliminated.

No background chatter

During those periods of the day when your signal has faded and your neighbor's hasn't, adjacent or co-channel interference may be a problem. The digital system will operate cleanly and quietly under these conditions in contrast to the analog system which may generate audible background chatter or birdies.

No phase distortion

The extreme phase linearity associated with the use of linear PCM encoding techniques and phase-linear FIR filters results in negligible phase distortion. Thus, peak overshoot is minimized allowing for safe use of maximum carrier deviation. Also, zero differential channel phase eliminates the phase synchronization problems normally associated with stereo or dual mono transmission.

SOURCE AND CHANNEL CODING

Linear 16-bit PCM for broadcasting (15 kHz audio) requires 512 kbit/s/channel at 32 kHz sample rate. A typical STL configuration might require two 15 kHz channels for stereo and one 7.5 kHz SCA channel yielding an aggregate data rate of 1280 kbit/s. It is this high data rate that has made previous attempts at digital STL usage in the 500 kHz (and 300 kHz) bandwidths of the 950 MHz band impractical. The required spectral efficiency suggests that coding techniques of greater complexity than conventional FSK with binary signaling are necessary to squeeze the data into standard broadcast channels. Such coding may be partitioned between source and channel encoding techniques to reduce the burden of complexity on either. Currently, state-of-the-art source coding hardware offers bit rate reduction by a factor of four. Channel coding formats yielding spectral efficiency of 1 bit/s/Hz will allow 320 kbp/s transmission in 500 kHz bandwidths.

Source coding

For high quality audio, there are three kinds of source coders— transform coders, sub-band coders and hybrids.[1,2,3] Choice of source coder is governed by requirements for quality, bit rate, delay, bit error protection, post processing capabilities and hardware implementation complexity. The relative importance of each of these parameters is very application-specific. For 'live' transmission purposes, the coding delay is a critical parameter. The maximum tolerable delay for these applications is on the order of 10 milliseconds. Longer delays present problems for off-air monitoring such as when using IFB or cueing channels.

Transform coders involve the conversion of a block of consecutive samples into the frequency domain. This strategy enables a reduction in the redundancy of the audio signals. Transform coders rely heavily on the computationally intensive FFT algorithm for frequency analysis of the signal. A certain amount of error protection overhead must also be included to enhance performance in poor environments due to the unforgiving nature of the coding. Though very effective at bit rate reduction, transform coders have long coding delays, typically greater than 50 ms, making them impractical for our application. Sub-band coders divide the broadband signal into a number of sub-band signals with a suitable filter band. This method avoids time-frequency-time conversion and therefore, reduces the degree of signal processing complexity. Depending on the implementation complexity, coding delays are between 1 ms and 20 ms.

Currently, the DSP-6000 digital modem is based on an implementation of sub-band ADPCM with linear prediction and backward adaptive quantization. Like the afore mentioned coding schemes, sub-band ADPCM exploits the considerable natural redundancies of voice and music to achieve the substantial 4-to-1 reduction in bit rate (for further details see [4]). This system satisfied the critical design considerations for our needs. Complexity is moderate. Coding delay is 3.8 ms which is quite reasonable for this application. Transparency was exceptional. Also, an important by-product of sub-band coding is excellent bit error immunity and gentle error handling which offers great benefit in broadcast radio transmission.

Channnel Coding

Efficient spectrum utilization is a primary consideration for the choice of modulation format. The format must co-exist favorably in the primarily analog frequency-modulated 950 MHz band. Thus, the out-of-band spectral energy must be well suppressed to avoid interference of the digital carrier with existing analog services. Linear modulation formats such as QPSK or QAM will regenerate sideband energy when confronted with non-linear Class-C amplification found in conventional FM STLs thereby limiting their usefulness. In contrast, continuous-phase frequency shift keying (CPFSK), a broad class of digital frequency modulation, is particularly well suited to FM transmission with non-linear amplification due to its constant-envelope property.

Spectral efficient schemes, such as tamed FM, partial response FM and GMSK, belong to a class of CPFSK known as correlated or partial response CPFSK. This class may be thought of as binary FM that has been very heavily band-limited prior to modulation. The result is a controlled amount of intersymbol interference (ISI) in pulse shapes that last a few bit periods. The pulse value at a given time becomes correlated to the value of previous pulses producing a composite multilevel FM signal.

The simplest form of partial response FM is known as duobinary FM. This scheme gives rise to intersymbol interference from only one previous pulse to yield a baseband signal that has three levels at the sampling instants. Duobinary signaling provides a 2:1 bandwidth compression relative to bi-level or binary signaling. The Nyquist criteria places the maximum symbol packing rate for bi-level binary signaling at 2 symbols/s/Hz but practical filter realization sets this rate closer to 1 symbol/s/Hz. With duobinary signaling, the ideal Nyquist rate of 2 symbols/s/Hz can once more be attained with practical filters that are tolerant of production, temperature and time variations. The price for this increased signaling efficiency is a small loss in robustness for adding the third signalling level. (For comparison, to produce the same spectral efficiency as duobinary, a corresponding zero memory or full-response multi-level signal would require 4 signalling levels thereby further reducing system robustness.) Using duobinary baseband signaling with the FM modulator's peak deviation set for one quarter of the data rate (h=0.5), an RF spectrum is created with a spectral efficiency of 1 bit/s/Hz. It is should be emphasized that spectral efficiency, as generally defined, does not refer to the 3 dB bandwidth that is considered by Carson's rule but to the "infinite attenuation" bandwidth which is more practically taken to be the 20 dB to 50 dB attenuation points depending on the interference requirements of the application.

Along with high spectral efficiency, duobinary signaling has been shown to have good error performance[5] and speed tolerance. Speed tolerance is defined as the amount of increase in signaling speed that will just cause overlap between adjacent levels (eye pattern closure). For ordinary duobinary, this speed tolerance is 43%. This translates to robustness in withstanding circuit and channel perturbations such as filter variations and unintentional intersymbol interference due to co-channel and adjacent channel interference and transmission channel distortions. This robust quality has been proven in several long haul communication links over the years and proven to be very reliable.

Another benefit of duobinary signaling is that error detection may be readily obtained without introducing redundancies into the data stream. This is accomplished by monitoring the received data for violations in its correlation properties. This error signal aids in determining the quality of the overall transmission path.

Error detection ability, constant envelope, high data rate packing, perturbation tolerance and efficient spectral shape have made duobinary signaling an appropriate technique for use in this modem to compliment existing STL technology.

DSP-6000 IMPLEMENTATION

Source Coding

The source coder portion of the modem is shown in Figure 2. The modem accepts up to four audio program inputs (main left and right and two SCAs) in 15 kHz or 7.5 kHz bandwidths. All audio inputs and outputs are active balanced XLR type. The audio program is converted to 16-bit linear PCM data by a 64 times oversampling dual A/D converter with digital linear-phase anti-alias filtering capable of 95 dB dynamic range. Alternately, the encoder accepts direct 16 bit linear PCM data from a digital source for contiguous digital transmission of the main program channels. On the decoder side, the PCM signal is ouput as either AES/EBU formatted or unformatted serial data. Analog outputs are also available by use of a 4 times oversampling dual D/A convertor. Two low speed asynchronous data channels (up to 9600 baud) allow for supervisory functions, low data rate services, and other possible applications. All timing and control functions are handled by field programmable gate-array (FPGA) logic (not shown in figure) which greatly reduces parts count to enhance reliability and allows for future upgrades.

Channel Encoding

The duobinary channel encoder is illustrated in Figure 3. Note the simplicity of the encoder. The encoder accepts data from the digital audio encoder or from an external data source. The external data input allows the modem to be configured as a repeater. It also allows transmission of any external data source from 64 kbits/s to 512 kbits/s thereby facilitating the transmission of low data rate services such as basic rate ISDN, Musicam source coders, etc.

A scrambler is the first operation that the input data encounters. The FCC requires that all digital radios randomize their carriers to prevent spectral lines, caused by particular data formats, from interfering with adjacent channels. The next block performs precoding of the data which is necessary to prevent error propagation in the receiver due to the correlative properties of each received bit. The data is Nyquist shaped by a raised cosine low-pass filter with 100% roll-off which is well known to minimize ISI distortion. The shaped data drives the frequency modulated oscillator of the STL transmitter.

Baseband Recovery

At the STL receiver, discriminator detection is used for baseband demodulation since it is best aligned with present hardware. Also, it avoids the carrier synchronization problem associated with coherent detection. Though coherent recovery does a good job against ISI and provides better static BER performance by 2 to 4 dB, it requires complex hardware and has been found to be problematic in multipath environments such as those that characterize terrestrial communications[6]. Other attractive features of discriminator detection are that it offers immunity to center frequency drift[7] and can handle arbitrary values of modulation index without realignment.

Channel Decoding

The tri-level baseband is recovered at the discriminator output. This signal enters the channel decoder, as displayed in Figure 4. Here it is first noise averaged by raised cosine shaped low-pass filtering. Ideally, optimal data detection requires a matched filter system to maximize the ratio of output signal power to output noise power. The filtering would be partitioned equally such that the transmitter and receive filter characteristics are identical. But since spectral efficiency is of greater importance than optimal data detection, most of the filtering is performed during the data shaping process in the encoder. Receiver filtering is set to approximately 1.5 times the cutoff of the transmit filtering to band-limit channel noise while avoiding additional eye pattern

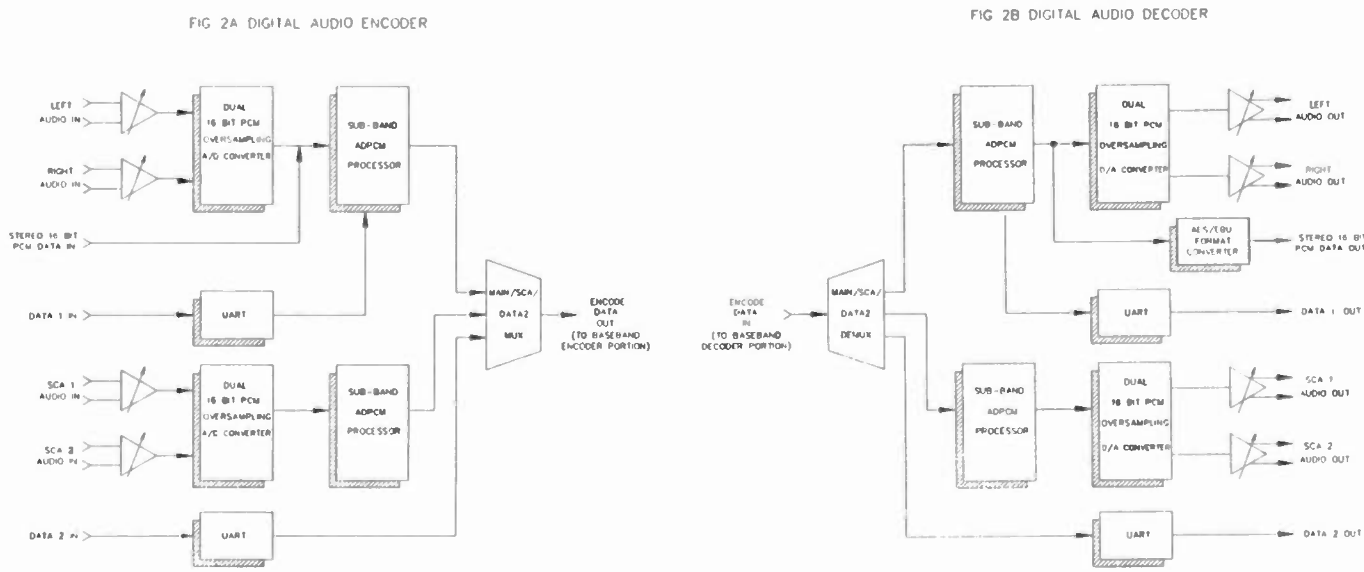

Figure 2
Digital Source Coder Portion of Modem showing a) encoder and b) decoder.

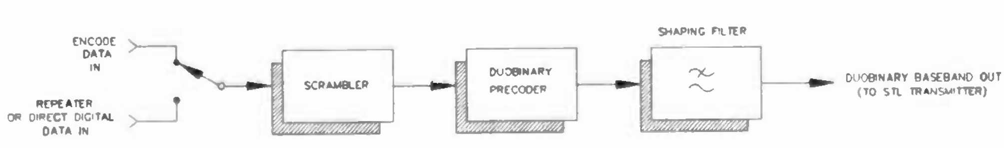

Figure 3
Duobinary Channel Encoder Portion of Modem.

distortion. The loss in C/N performance is only about 1 dB for this filter partitioning versus optimal partitioning[8].

Clock Recovery

From the recovered tri-level baseband signal at the discriminator output, both bit clock and data recovery are accomplished. The bit clock synchronizes the rest of the system to the recovered data. The duobinary signal is similar to NRZ data in that it contains no significant spectral energy at the clock frequency. A non-linear circuit is used to generate the necessary clock information. The resulting signal has been corrupted by pattern related jitter due to ISI and channel noise jitter. This signal is pre-filtered to 6 kHz and synchronized by a narrowband 2nd order phase-lock loop. The prefilter greatly improves the input SNR to the PLL allowing continuous clock synchronization in the face of deep channel fading behavior to error rates less than 1E-1. An extra pole was placed in the loop to reduce the effect of high frequency or "infinite variance" jitter which may result in potential cycle slips. The locked loop band width of 206 Hz and damping factor of 0.8 provide a lock time of 63 milliseconds. These loop dynamics attenuate the jitter component for a tracking error less than 0.5 degrees rms after 10 repeater hops.

Data Recovery

Bit-by-bit detection is performed on the filtered baseband signal by means of a digital sample-and-hold. This method was judged the most effective for data recovery yielding adequate BER performance with the least amount of circuitry complexity. Following data sampling, decode logic reconstructs and descrambles the data for output to the digital audio processor. Bit-error and BER threshold detection are also provided by the decoder for signal quality indication as described above.

PERFORMANCE TESTS

Spectral Occupancy

Figure 5 shows the RF spectrum produced by the digital STL transmitter for 256 kbits/s and peak deviation of 64 kHz (h=128/256=0.5) measured in a 3 kHz bandwidth. Two FCC compliance masks are overlayed on this spectrum. The hatched overlay mask represent the emission boundries for FM aural STL transmission under Part 74.535 in 500 kHz spacings. The dashed line overlay represents the emission boundries for digital microwave transmission (Docket No. 19311, FCC Rules Part 21.106) for 500 kHz channels. In this docket, a measurement bandwidth of 4 kHz is specified. The correction for the 1 kHz difference in bandwidth is to add 1.24 dB to the spectrum shown in the figure. In any case, the spectrum falls well within either emission mask.

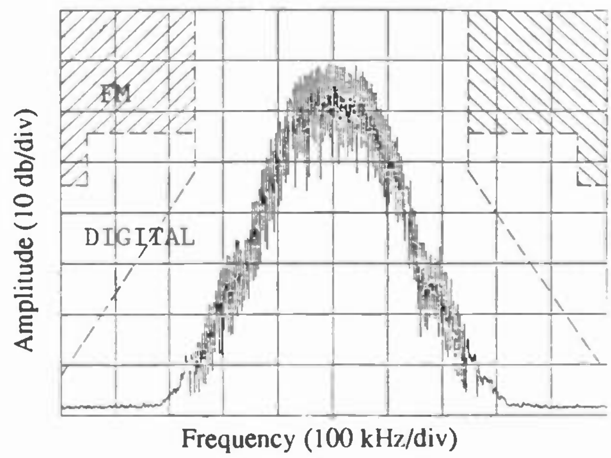

Figure 5
Digital STL Emission with FCC Emission Masks.

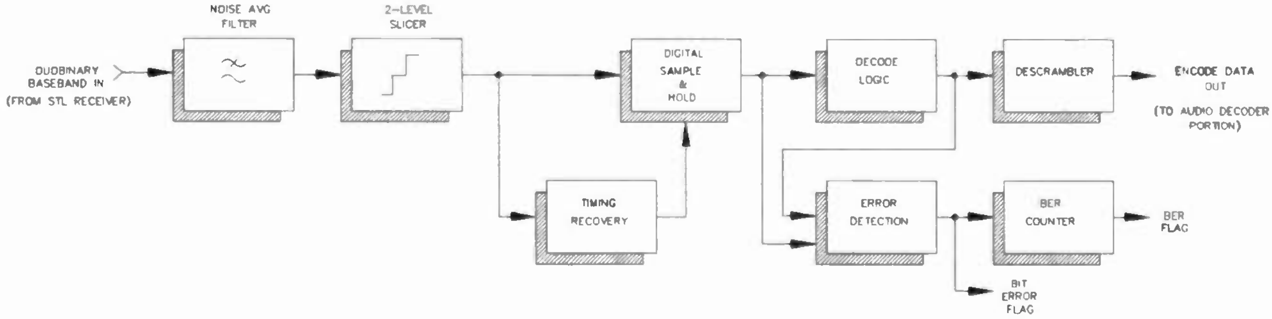

Figure 4
Duobinary Channel Decoder Portion of Modem.

Transparency

Back-to-back performance measurements were taken on the digital STL. Most of these tested the limits of our measurement equipment. Frequency response was +/-0.1 dB from 10 Hz to 15 kHz. Static SNR was greater than 90 dB. Stereo seperation was greater than 80 dB over the entire 15 kHz bandwidth. Differential phase and amplitude deviation between left and right channels were negligible. Group delay was 1.89 ms +/-0.001 ms over the bandwidth. Several critical listening tests were performed using A-B comparison to a CD source. Experienced impartial listeners certified the high sonic quality of the overall system.

Threshold Performance

Sensitivity is an important measure of overall coding efficiency of this system. A baseline configuration with two main channels was selected for this test. Data rate was 256 kbits/s and h=0.5. Since no perceivable degradation in audio quality occurs for BER < 1E-4, this level was chosen as the system threshold. For a receiver IF bandwidth of 500 kHz, this threshold was reached for a measured RF input level of 5 microvolts or -93 dBm. For comparison, a composite STL would produce a de-emphasized SNR of 32 dB at this received signal level.

Compatibility With Existing Analog STL Radios

a. Digital STL with adjacent digital STL interferer.
In gauging the effects that digital STLs have on adjacent digital STLs, the digital receiver BER threshold was examined in the presence of an adjacent fully modulated digital carrier. The encoder was configured for two main channels (256 kbit/s data rate). The reference STL carrier was set for a modulation index of 0.5 at 950 MHz. The interfering digital STL was configured similarly. Its carrier was placed at 500 kHz and 300 kHz offsets from the reference carrier. The adjacent signal was increased until the 1E-4 BER threshold was reached. The difference between the two carrier levels (in dB) at this point is the Interferer-to-Desired-Carrier (I/C_d) ratio. The results are displayed in Figure 6.

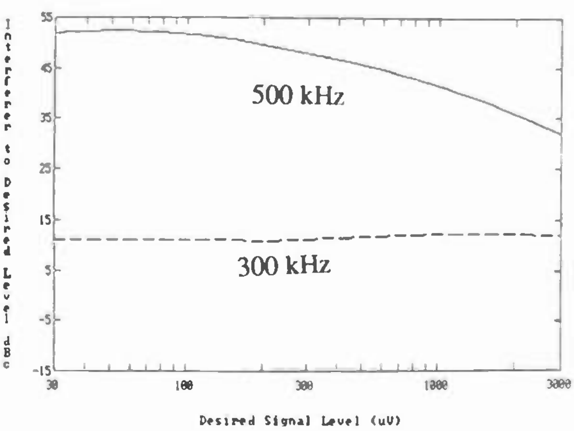

Figure 6
I/C_d for a Digital STL with Digital STL Interferer.

The results show that the adjacent signal power needs to be at least 30 dB higher for all input levels (500 kHz spacing) to degrade the BER threshold noticably. As expected, I/C_d is lower for the narrower 300 kHz spacing yet BER degradation still requires an adjacent carrier level 11 dB above the reference signal level. This was judged to be more than adequate margin for most applications.

b. Digital STL with adjacent composite STL interferer.
In this case a composite FM STL was used as the interfering STL to the digital reference STL. The digital STL was configured as previously described. The interfering composite carrier was modulated by 2 kHz tone (i.e., L+R and L-R channels) at 50 kHz peak (100%) deviation. The results are displayed in Figure 7.

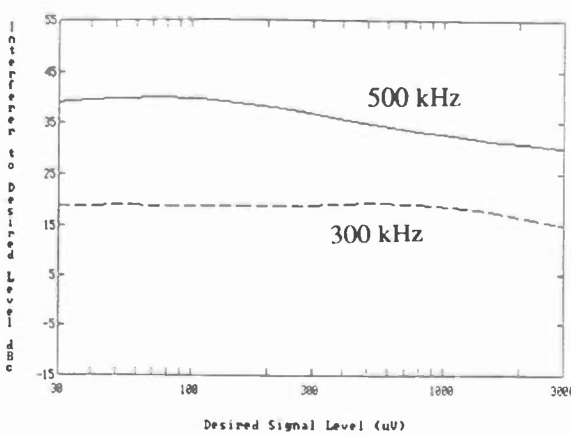

Figure 7
I/C_d for a Digital STL with Composite STL Interferer.

The results show that I/C_d remains relatively constant for both channel spacings. In 500 kHz channels, an adjacent composite FM signal must be at least 30 dB larger than the digital STL's level. In 300 kHz spacing, BER degradation requires an adjacent composite signal 15 dB above digital carrier level. These results are considerably better than for an analog STL with a similar analog interferer. Thus, the digital carrier appears very capable in face of adjacent analog STL interference.

c. Analog STL with adjacent digital STL interferer.
This test is essentially reversed from the preceding test. The test parameters were identical to those previously described. Monitoring the composite SNR, the adjacent interfering signal was increased until the composite SNR dropped by 3 dB. The results are displayed in Figure 8.

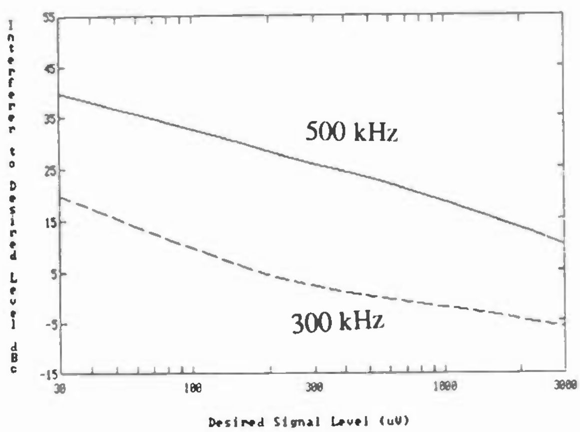

Figure 8
I/C_d for a Composite STL with Digital STL Interferer.

The I/C_d dependence on the composite carrier level results from SNR degradation with lower carrier levels (i.e., composite SNR drops from 76 dB at 3.2 mV to 52 dB at 32 uV). In essence, it requires more interference noise from an adjacent signal to swamp the channel noise.

At a typical received power of 1mV for 500 kHz spacing I/C_d is +19 dBc which is approximately comparable for a composite FM STL with a similar composite FM interferer. By extension, this suggests that the digital STL is compatible with existing analog STLs in 500 kHz spacings.

For 300 kHz spacing, I/C_d is lowered to -2 dBc at 1 mV received level. This might suggest that an adjacent 300 kHz digital channel must always be lower in power than the primary analog FM carrier. This is not an unreasonable assumption, however. Under normal conditions, a given STL channel will exhibit an RF signal level about 10 to 20 dB stronger than the adjacent channels on either side. This is mainly due to the effects of alternating the antenna polarization from one channel to the next and assumes the use of similar equipment (i.e.,transmitter power, antenna gain, etc.) and similar path lengths. This certainly seems to be the case for the majority of users in large metropolitan areas and is a result of their frequency coordinating committees. However, during the course of time the common path will take fades, and because of the space diversity between users, the effect will not be uniform. The desired signal will take 18 dB fades 1% of the time while the adjacent interfering signals will remain unaffected. It would therefore stand to reason that an adjacent digital carrier, as configured, could co-exist next to a composite FM carrier in 300 kHz spacing without much problem if proper planning of channels and signal levels has been done.

CONCLUSION

Digital transmission of CD quality audio over a 950 MHz studio-transmitter link is now a reality. The benefits of robust digital transmission and the reliability of radio STLs are married in one system. The digital modem as described and tested solves several inadequacies that presently exist in analog FM STL transmission. The digital quality may be obtained even using an existing analog STL. The system has proven to be very robust and flexible. Spectrally speaking, it co-exists admirably with analog FM transmission.

We would like to acknowledge Dr. Douglas Hogg and Dan Barnett for their guidance and assistance during the course of the work and writing of this paper.

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AM SYSTEMS ENGINEERING AND IMPROVEMENT

Sunday, April 14, 1991

Moderator:
Milford Smith, Greater Media, Inc., East Brunswick, New Jersey

FCC AM REGULATIONS UPDATE*
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MODERN METHODS IN MEDIUMWAVE DIRECTIONAL ANTENNA FEEDER SYSTEM DESIGN
Ronald D. Rackley
du Treil, Lundin & Rackley, Inc.
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USING ISOLATION TRANSFORMERS TO LEASE AM TOWER SPACE
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Kintronic Laboratories, Inc.
Bluff City, Tennessee

IMPLEMENTATION OF ANTISKYWAVE ANTENNA TECHNOLOGY BY EXTREME TOP LOADING OF SHORT ANTENNAS IN A DIRECTIONAL ARRAY
Timothy C. Cutforth, P.E.
Vir James Consulting Radio Engineers
Denver, Colorado

PREVENTIVE MAINTENANCE FOR AM RADIO TOWERS
Robert E. Sundius, Jr.
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Owen Ulmer
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NOISE FREE RADIO FOR AM BROADCASTING
George W. Yazell, P.E.
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Lakeland, Florida

*Paper not available at the time of publication.

MODERN METHODS IN MEDIUMWAVE DIRECTIONAL ANTENNA FEEDER SYSTEM DESIGN

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INTRODUCTION

Advances in computer modeling of antenna arrays and feeder systems allow study of medium-wave directional antenna characteristics not possible little over a decade ago. Exact theoretical solutions for the two separate, but related, areas of impedance bandwidth and pattern bandwidth are possible, allowing optimization of antenna performance as it impacts the delivery of high quality over-the-air audio within the entire coverage area of a station.

With the advancement of computational capability have come new approaches to design of power dividing, phasing, and matching networks. This paper is intended to provide an overview of the current technology and show, by an example, the differences in performance between systems designed using modern methods and the best of traditional methods.

Modern design tools and circuit concepts will be covered in survey fashion. Because of the broad scope of this paper, it will not be possible to examine every concept mentioned in great detail. A list of suggested books for further reading about several key concepts may be found at the end.

PRIOR METHODS OF ANTENNA MODELING

The prediction of antenna electrical characteristics involves very complicated mathematical analysis. As a simplifying assumption, it has been traditional to consider the current in a linear antenna element to be distributed sinusoidally along its length. Among other things, this assumption has a very obvious problem for towers exactly one-half wavelength in height: the base current would have to be zero for any radiated power level! It stands to reason that impedance predictions for towers with height other than one-half wavelength can significantly miss the mark also. This is indeed the case.

For towers in directional arrays, the sinusoidal current distribution assumption becomes even more problematic. Even if it is assumed that a nondirectional antenna tower driven at its base has a sinusoidal current distribution characteristic, the same tower, undriven but bathed in incident field from another source, could exhibit a very different current distribution. In the receiving mode, the current distribution is a function of the terminating impedance at the base.

The fact that each tower in an array operates both to radiate energy itself and to receive energy from the other towers means that the individual tower current distributions will not be identical. This means that, not only can the sinusoidal current distribution assumption not be relied upon for reliable base impedance predictions, but one cannot even make the convenient assumption that drive current ratios and phases are the same as field ratios and phases.

Using the traditional methods involving sinusoidal current distribution assumptions, precise prediction of antenna system characteristics cannot be done prior to construction because, by and large, the precise determination of operating currents and impedances must be left to trial-and-error. Feeder system designs cannot be optimized for specific base operating impedances and drive parameters predicted with a high degree of confidence, but must be capable of meeting a wide range of conditions.

MODERN ANTENNA MODELING- THE MOMENT METHOD

Moment method antenna modeling has proven to be a very useful tool in overcoming the limitations of traditional antenna theory. Moment method modeling, very basically speaking, divides each radiator into a large number of individual segments for which corresponding current values can be calculated, freeing the designer from the necessity of making any assumptions about current distribution. The Numerical Electromagnetics Code (NEC) and its descendent, MININEC, are the two most commonly available programs for moment method analysis.

It is now possible to design and adjust antenna systems using moment method software with little, if any, experimentation, if the conditions at the site approach the ideal in terms of flat terrain and an absence of nearby reradiating objects. Even where conditions are not ideal, moment method modeling is a very useful tool in relating current drives to field parameters and reducing the amount of trial-and- error work necessary. It also makes possible very reliable base impedance predictions for use in optimizing array bandwidth predictions.

COMPARISON OF PRIOR AND MODERN ANTENNA MODELING METHODS

Figure 1 shows predicted antenna current using both the moment method and sinusoidal assumptions for a three- tower array with each element having a height of one-half wavelength. Significant differences between the tower currents predicted by the two methods are obvious.

The moment method current distributions make it appear that the towers differ in electrical height, even though their physical heights are identical. This is very commonly found in such arrays. There is a five-to-one difference in base current between the tower shown on the left and the one shown on the right, even though their field ratios are identical in the pattern design. The drive current phases differ over a range of approximately 40 degrees, even though, by sinusoidal assumptions, there should be no variation.

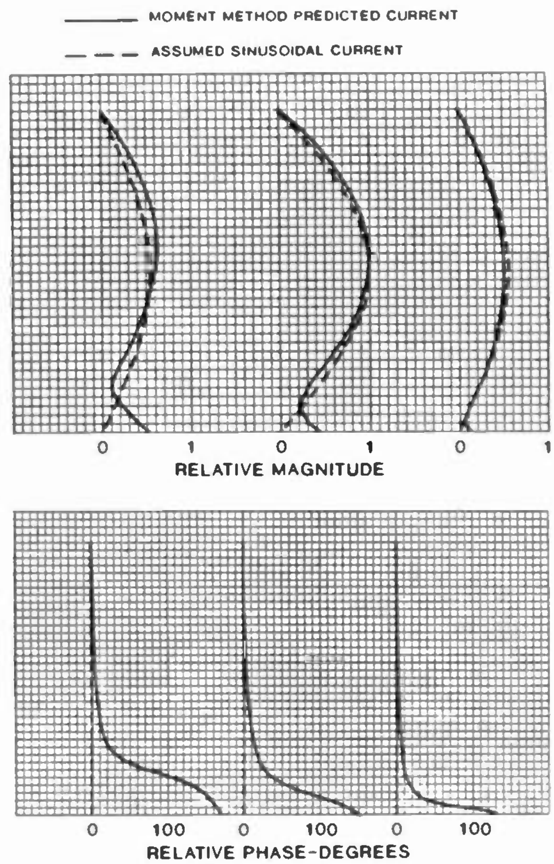


Figure 1. Comparison of the Three Tower DA Current Distributions for Moment Method and Sinusoidal Assumptions

Measured current distributions, operating base impedances and base drive requirements have been found to agree well with the moment method predictions for directional antenna systems. The importance of having a method for predicting directional antenna element currents and impedances with more sophistication than possible with the sinusoidal current assumption is obvious from Figure 1. This is particularly true if it is desired to optimize the feeder system design for realistic base drive values.

NODAL ANALYSIS OF NETWORKS

The technique of nodal analysis has been well known in the field of electrical engineering for a long time. The complexity of the system of networks necessary to feed a directional antenna array, however, severely limited its use for this field in the pre-computer age.

In nodal analysis, network branch currents are defined in terms of voltages at the points where branches interconnect, or nodes, and the branch admittances. Figure 2 is a general network example with two independent nodes, A and B. Figure 3 shows the standard form for the resulting node equations for a network with N independent nodes. The linear equations in this form lend themselves to simultaneous solution using the method of determinants, which must be accomplished with matrix algebra.

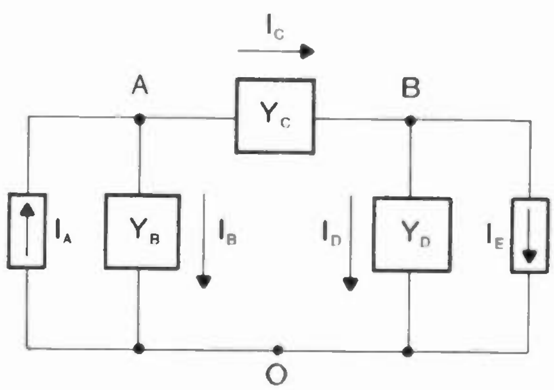


Figure 2. Network Example with Two Independent Nodes, A and B

Nodal analysis is of great use in predicting the bandwidth performance of directional antenna phasing and coupling equipment, since admittance values can be given for each component and the tower bases can be modeled as nodes with self and mutual admittance values determined using moment method analysis. An exact solution for carrier and sideband currents and impedances can be found for every branch in a system, bypassing the "chicken and egg" problem of simpler analysis techniques which must assume a set of base current parameters to determine operating impedances which, when presented to the system of networks, yield a different set of base current parameters and render the starting assumptions invalid.

There is a catch, though. In order to perform the necessary matrix algebra, an N by N matrix (where N is the number of independent nodes) must be inverted. Since even a simple two-tower phasing and coupling system can easily have 20 nodes, the nodal analysis technique is not practical for directional antenna analysis without the aid of a computer. The author cannot recall manually working with a matrix larger than about five-by-five, even in engineering school.

$$\begin{aligned}
Y_{AA}V_A + Y_{AB}V_B + Y_{AC}V_C + \cdots + Y_{AN}V_N &= I_A \\
Y_{BA}V_A + Y_{BB}V_B + Y_{BC}V_C + \cdots + Y_{BN}V_N &= I_B \\
Y_{CA}V_A + Y_{CB}V_B + Y_{CC}V_C + \cdots + Y_{CN}V_N &= I_C \\
\cdots\cdots\cdots\cdots\cdots\cdots \\
Y_{NA}V_A + Y_{NB}V_B + Y_{NC}V_C + \cdots + Y_{NN}V_N &= I_N
\end{aligned}$$

Figure 3. Node Equations for N Independent Nodes

Fortunately, we today have computing equipment and software to take over the burdensome work of matrix algebra and free the design engineer for tasks better suited to the human mind, such as exercising judgement in the selection of networks and components to optimize not only impedance and pattern bandwidth but also adjustability with phasing system controls.

TRADITIONAL CIRCUITS FOR POWER DIVIDING, PHASING, AND MATCHING EQUIPMENT

Before discussing new techniques in phasing and coupling system design, a review of traditional approaches is in order. Figure 4 shows the two types of power divider used for the majority of phasing systems designed prior to the late 1970s. The "tank" or "jeep coil" type of power divider goes back to the very earliest days of radio and the "parallel" or "Ohm's law" design became popular during the 1950s.

Both circuits of Figure 4 function primarily as power dividers, with separate networks necessary for phase adjustments. Both can introduce high system Q, thus possibly restricting bandwidth. The "tank" circuit circulates all of the power fed into the system through a parallel tuned antiresonant circuit and the "parallel" circuit can result in relatively high circulating current due to the low resistance presented when several tower feeds are paralleled. Of course, the high Q of such circuits could serve to counteract bandwidth problems inherent in an array design. This would require careful system modeling to be effective. Such modeling was not practical in the era when most such systems were built.

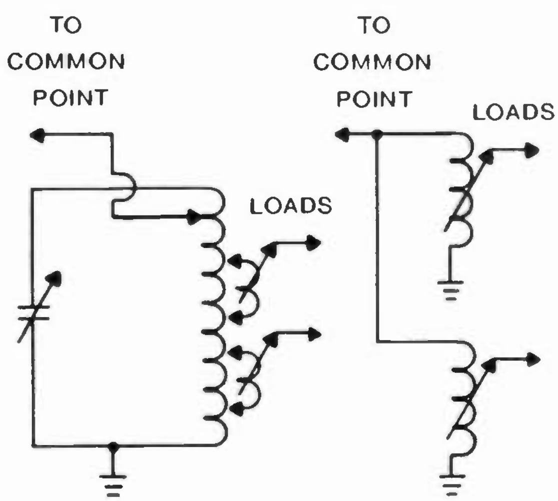


Figure 4. Traditional Power Divider Types

For phase shifting and antenna matching, 90 degree T networks were common building blocks. They were believed to be the optimum circuit for matching two differing impedances, although this author can find no reason for using them other than the mathematical simplicity of their design and their ability to be readjusted over a fairly wide range of phase shifts.

MODERN POWER DIVIDER CONCEPTS

Figure 5 illustrates the general principle of operation for all power divider circuits. If the common feed for all power dividing circuits is considered to be a voltage buss, the power delivered to each tower is determined by the conductance value presented to the buss by that tower's power dividing circuit. The voltage for the desired buss impedance can be determined and then the circuits necessary to present the required conductances, when terminated in the transmission lines, can be designed.

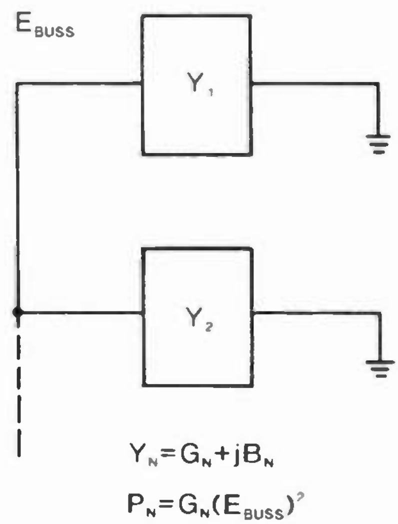


Figure 5. General Power Divider Principle

It is usually desirable to design for a buss impedance of 50 ohms where 50 ohm transmission lines are used, unless another factor suggests otherwise. Such an alternative situation would arise where one tower in a system needs much higher power than any of the others and could be fed directly off of the buss without adjustment capability and satisfy the requirements for optimum overall phase shift. For instance, a 25-ohm buss would feed half of the power delivered to it directly to a 50-ohm transmission line.

MODERN POWER DIVIDER CIRCUITS

Any network which can adjust the conductance presented across the buss for a tower feed can be used as a power divider circuit. There is no need to have the same type of power divider network for every tower in an array. Indeed, it may be desirable to have different types intermixed in a system from the standpoints of adjustability and bandwidth.

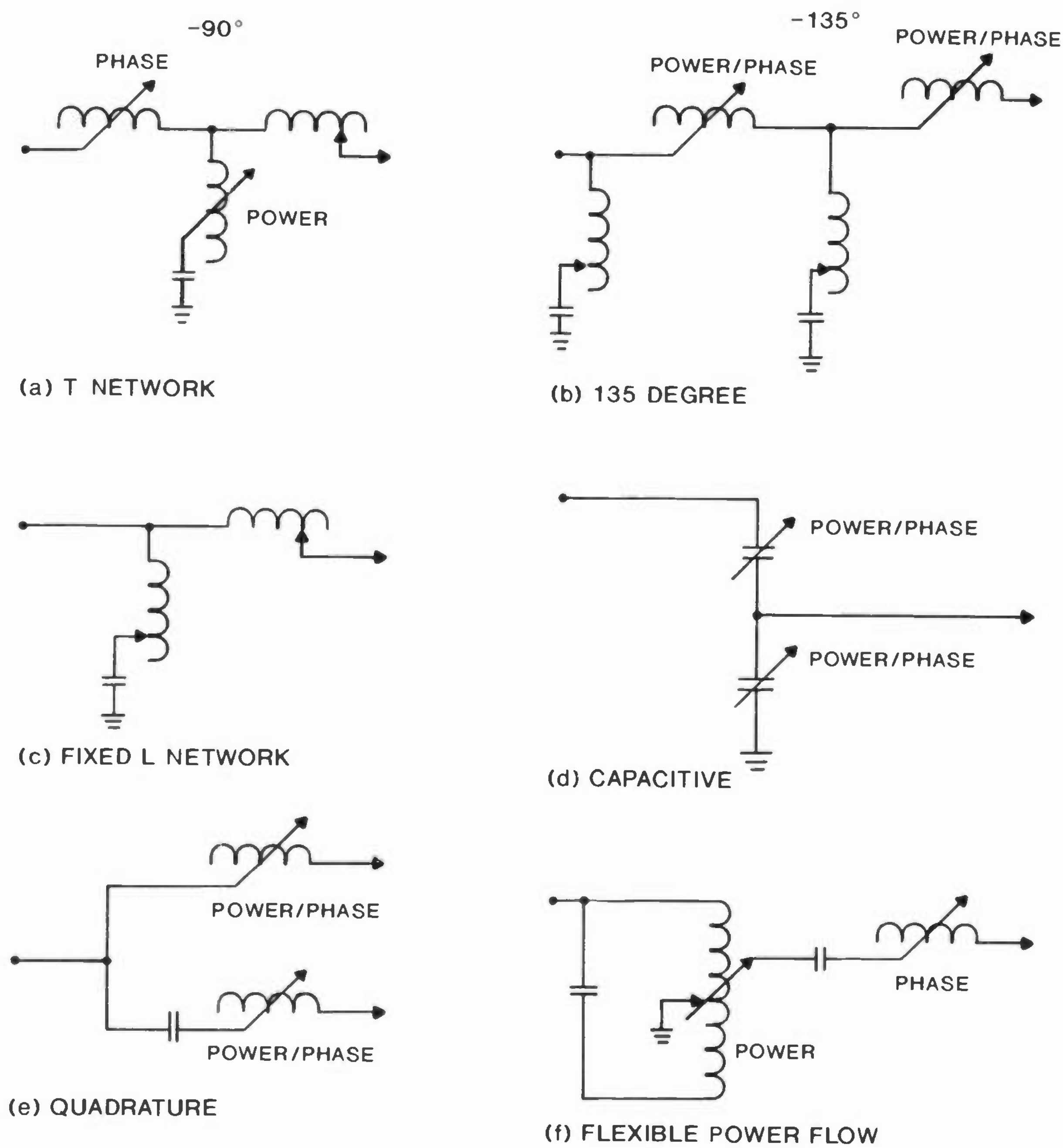
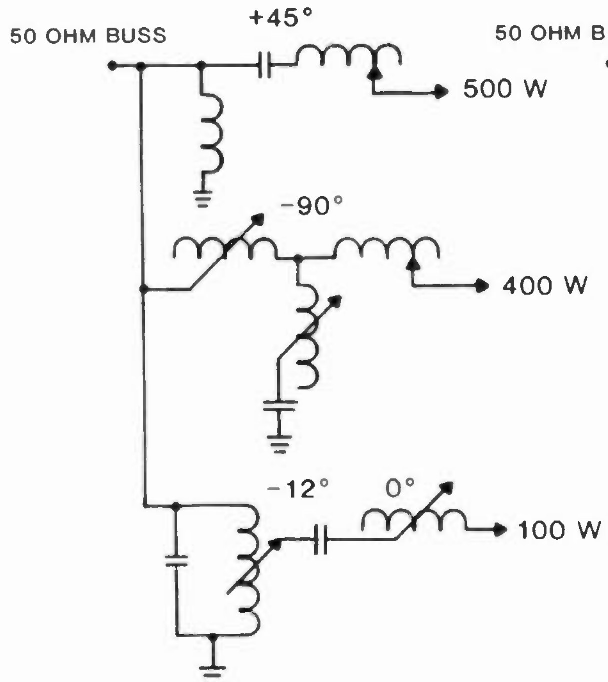
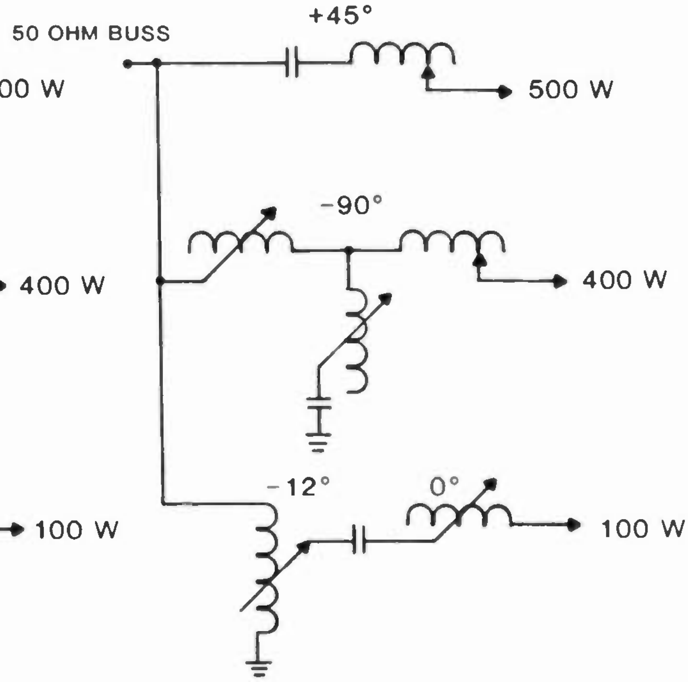


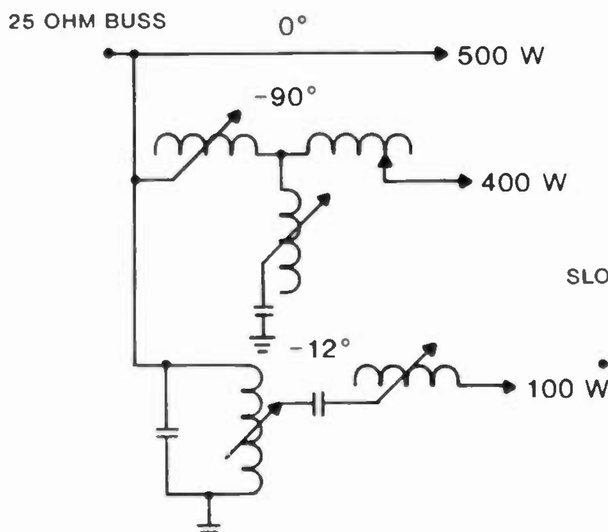
Figure 6. Modern Power Divider Circuits



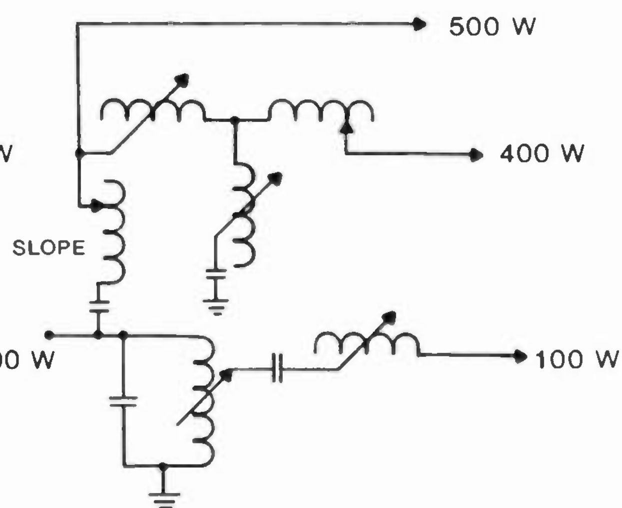
(a) MIXED USE OF POWER DIVIDER TYPES



(b) SIMPLIFICATION BY ELIMINATION OF PARALLEL COMPONENTS



(c) DIRECT FEED TO HIGEST POWER TOWER



(d) SPLIT BUSS WITH PATTERN BANDWITH IMPROVEMENT

Figure 7. Modern Power Divider Examples

Figure 6 shows several power divider circuits which have become popular in recent years. Each one shown is capable of serving for control of both power and phase, making separate phase adjustment networks unnecessary. If properly applied, the circuits of Figure 6 can generally lead to lower power dividing and phasing network Q than attainable with either power divider of Figure 4.

Of the networks shown on Figure 6, circuit A, an adjustable T network, is very popular for towers with power ranging from 20 percent to 50 percent of the total system power. This circuit provides a non-reactive load to the buss at design center branch values.

Circuit B, which consists of cascaded 45 degree L and 90 degree T networks, makes it possible to adjust both phase and power division over a wide range while maintaining an essentially non-reactive load for the buss, with the slight disadvantage of not having independent control of power and phase. It is useful for towers with as low as 10 percent, and sometimes less, of the total power.

Circuit C, an L network, is a good choice for the highest power tower of an array, where some transformation is necessary in order to allow for a 50 ohm buss. Front panel adjustment is often not required in such a case.

Circuit D, a capacitive power divider, is economical in high power service, where the vacuum variable capacitor cost is justifiable. It is not possible to have a non-reactive buss with this circuit unless a parallel inductance is added.

Circuit E, the quadrature power divider, is a very simple circuit to use where two towers are related in power by a factor of three-to-one, or less. The two outputs, for design center adjustment, differ in phase by 90 degrees.

Circuit F gives very good control of base current and phase for a tower which may operate with either positive or negative power flow because its operating resistance is near zero. In order to realize this advantage, it is necessary that the total phase shift of the transmission line and networks between the buss and the tower base be a multiple of one-half wavelength.

Most of the circuits of Figure 6 do not have separate controls for power and phase. This is not a great disadvantage, because the circuits that do only offer totally independent control when connected to load impedances that remain constant. This is not the case for any power divider which is feeding elements in an array, due to the effects of mutual coupling between the elements. The controls for circuits B, D and E are very predictable. Generally, turning both controls in the same direction will predominately change either phase or power while turning them in opposite directions will predominately change the other parameter.

The low Q circuits of Figure 6 are popular modern alternatives to the traditional power dividers of Figure 4. For some directional arrays with highly volatile power division, however, the traditional power dividers, with their higher Q, may be desirable if easy adjustability is valued. Proper system modeling could be used to minimize the high Q effects or actually use them to improve overall system bandwidth.

MODERN POWER DIVIDER DESIGN

Figure 7 shows how the basic power divider circuits of Figure 6 can be applied in phasing system design. Circuit A offers good control and a 50 ohm buss, but can be simplified to circuit B if the proper value is chosen for the lowest-power tower's power divider coil so that the capacitor necessary to antiresonate it is of the same reactance magnitude as the top tower's fixed L network shunt coil. This would be possible in a case where the lowest-power tower would not change power flow direction. In the process, the power divider Q is lowered by the elimination of a parallel antiresonant circuit across the buss.

Circuit C shows how, if the phase shift requirements allow it, the top, high-power tower feed can be connected directly to the buss, eliminating the three components of the L network. Circuit D is identical to circuit C, except that the buss has been divided with a series L-C slope network. In the case shown, the high-power towers need to have the phase shift of their feeds tailored to track the lower-power tower in order to preserve pattern bandwidth. This is the purpose of the L-C slope network shown.

As can be seen from circuit D of Figure 7, high Q circuits can be inserted at appropriate locations in phasing equipment to effectuate broadbanding. It must be understood, however, that such processes require modeling of total system performance, such as with nodal analysis, in order to be effective. In many cases, it will be necessary to improve pattern bandwidth with high Q circuits added after the common buss, with an additional network to improve impedance bandwidth included in the common point matching circuit.

MODERN CONCEPTS OF PHASING AND MATCHING

Absent the determination of a good reason to insert high Q circuitry for bandwidth improvement, it is generally a good idea to minimize circuit Q for the feeder system. Figure 8 shows a conventional T network, which is the basic building block for antenna matching and phase shifting functions. Although it is popularly assumed that optimum bandwidth performance results with the phase shift of a T network adjusted to 90 degrees, the family of curves shown on Figure 9 indicate otherwise. There is clearly an optimum T network phase shift for each transformation ratio. These values are generally lower than 90 degrees.

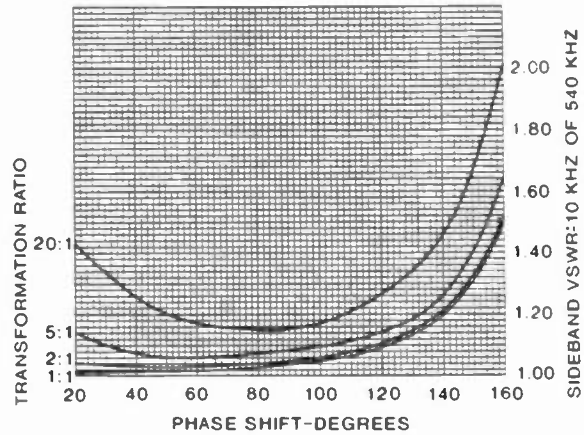


Figure 9. T Network Sideband VSWR Versus Phase Shift for Various Transformation Ratios

The high VSWR bandwidth for a twenty to one transformation ratio T network suggests that there might be a better way to achieve such a match. Figure 10 shows how two networks can be cascaded to achieve a gradual step-up of resistance that requires that only one additional shunt branch be added alongside the normal tee network configuration. Such a circuit may be designed from either L and tee or L and L cascaded networks. Figure 11 shows the bandwidth performance of such a circuit designed with two cascaded networks. For the cost of an additional network branch, there is approximately a three to one improvement in sideband VSWR.

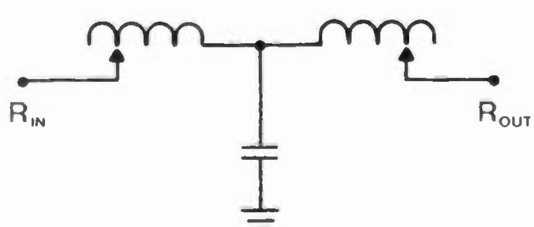


Figure 8. T Network: Basic Circuit for Impedance Matching

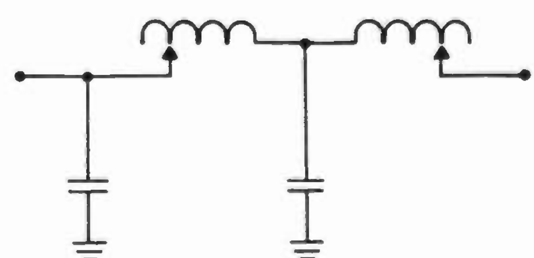


Figure 10. Cascaded T and L Networks for Optimizing Phase Shift and Transformation Ratio

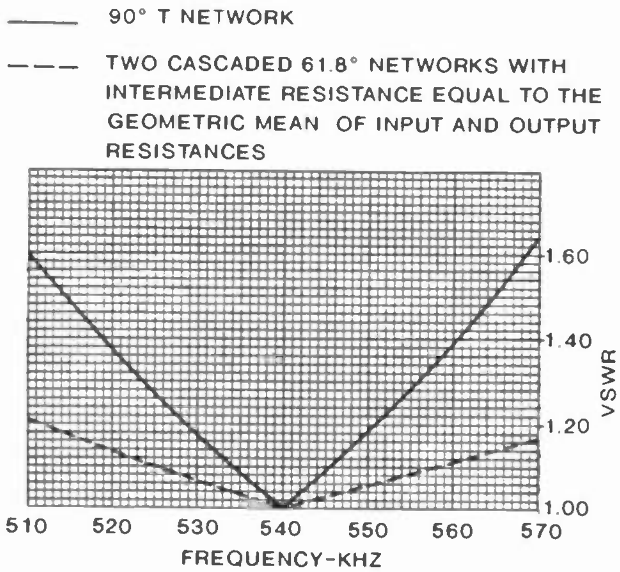


Figure 11. VSWR Introduced by Network for Transformation Ratio of 20:1

COMPARISON EXAMPLE

Figure 12 shows a conventional phasor design for a two-tower array which was constructed during the early 1970s. The array consists of two 59 degree towers which are spaced 70 electrical degrees and driven to produce a near-unity field ratio with a phase difference between the tower fields of 162 degrees. The design uses a conventional shunt power divider and tee networks with phase shifts near 90 degrees at all circuit locations.

Figure 13 shows a design using modern concepts. A quadrature type power divider is utilized and cascaded networks are employed for the tower matching units. Impedance bandwidth correction circuitry is employed at the common point.

It was found, through nodal modeling, that proper selection of total system phase shift beyond the common buss had a large impact on pattern bandwidth. The pattern bandwidth improvement resulting from system phase shift optimization exceeded that from reduction of matching circuit Q.

It has long been recognized that transmitter sideband performance with narrowband loads can be optimized by careful selection of phase shift between the final amplifier stage and the load. The same principles may be used to optimize the individual tower feeds as presented to the power divider in a directional antenna, so as to improve the stability of the antenna parameters as frequency is varied. Such pattern broadbanding is only possible with whole-system modeling, as with nodal analysis.

Figure 14 shows predicted carrier and sideband patterns for one kilowatt carrier power and predicted common point impedance values for the traditional feeder system design of Figure 12. The same information for the modern broadband design is presented on Figure 15.

As can be seen from Figures 14 and 15, both the pattern and impedance bandwidth may be improved dramatically using modern design approaches. The movement of null azimuth with frequency seen for the traditional design produces very high envelope distortion within a wide range of azimuth near the nulls. Because of sideband imbalance, phase modulation to distort stereo reception occurs in all areas, most particularly in the null and minor lobe regions. Since both sidebands are significantly attenuated, several dB of de-emphasis occurs in both lobes.

For the modern design, the "null talk" is almost completely eliminated. Additionally, sideband balance is greatly improved at all azimuths and de-emphasis effects are greatly reduced.

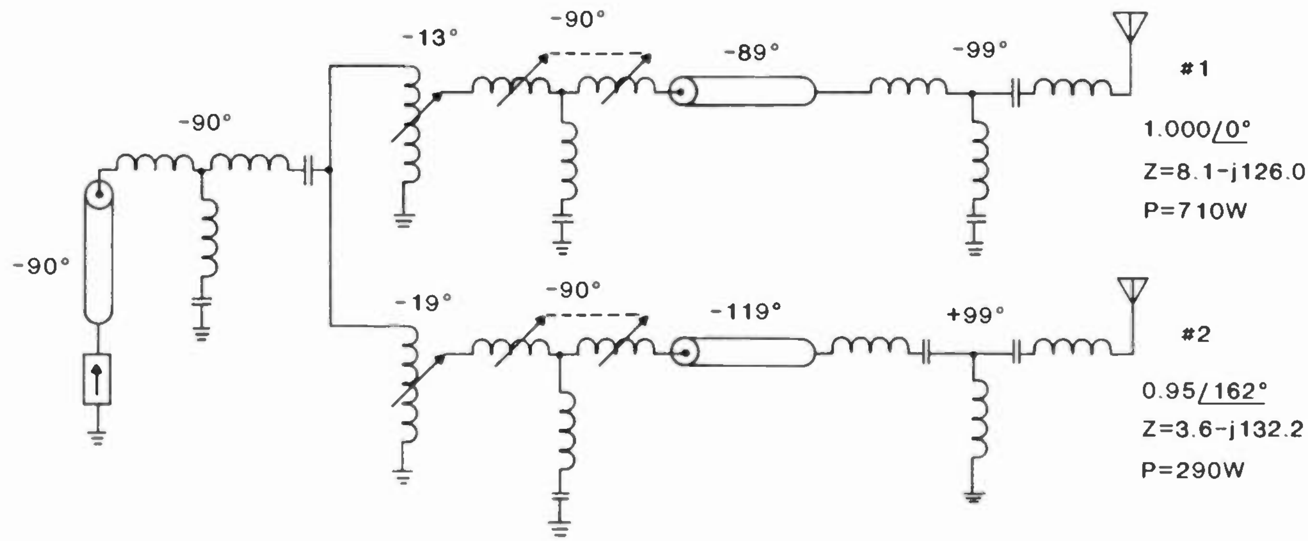


Figure 12. Traditional Phasing System Design

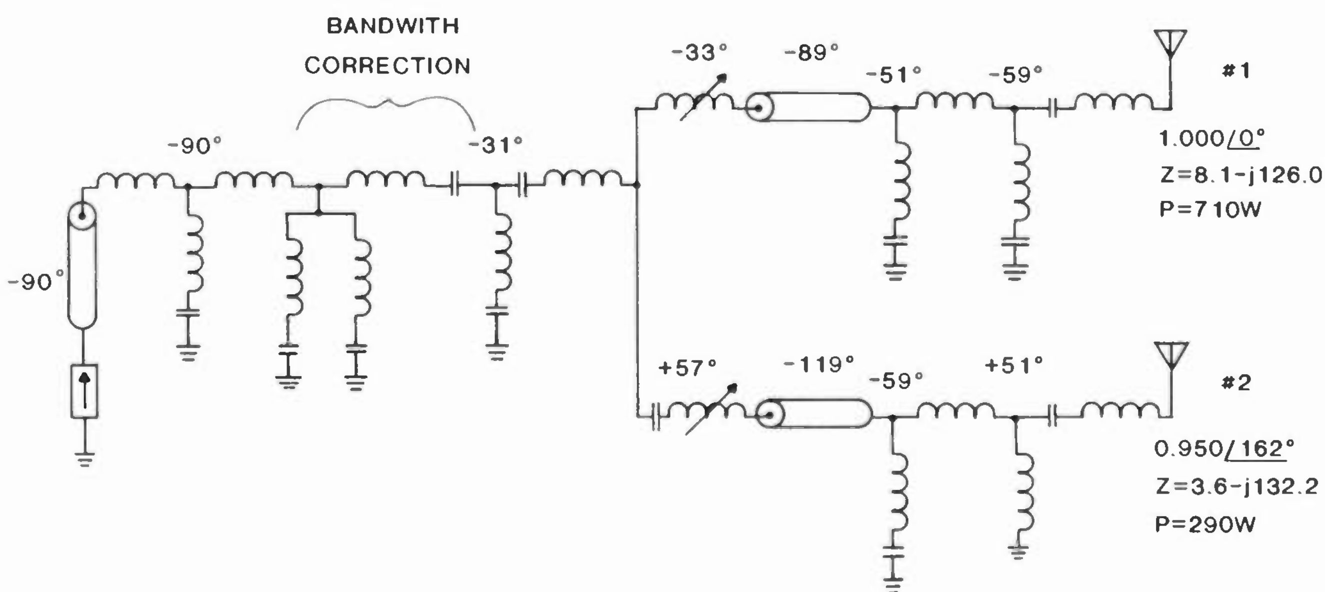
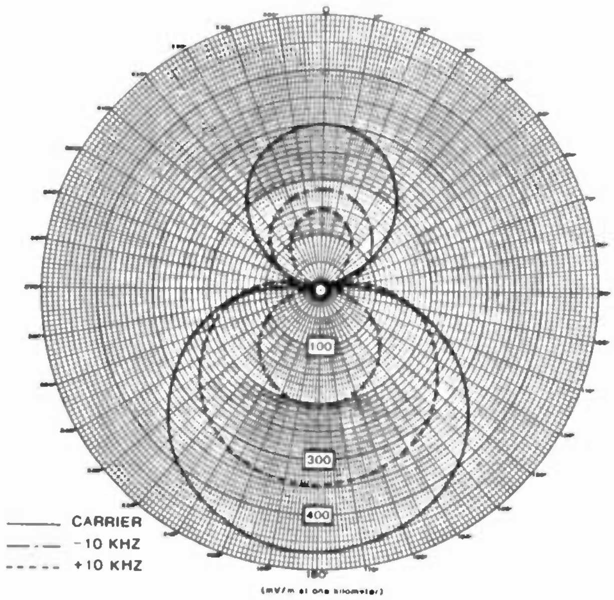
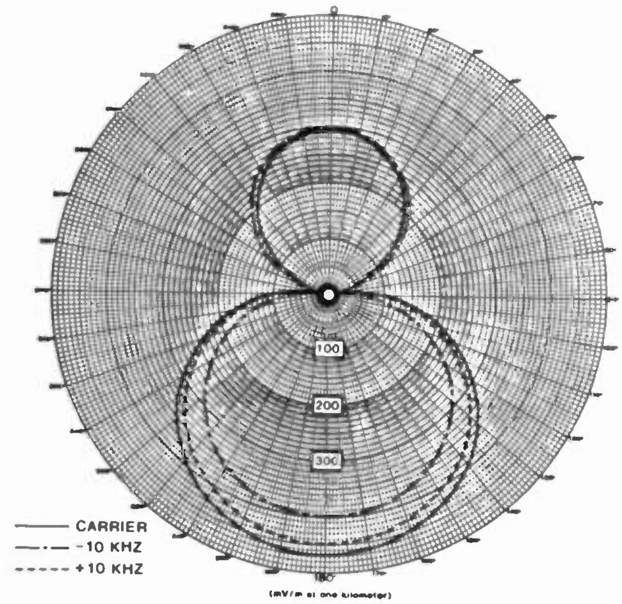


Figure 13. Modern Broadband Phasing System Design



FREQUENCY	COMMON POINT IMPEDANCE	VSWR
-10 KHZ	149+j83	4.00
CARRIER	50+j0	1.00
+10 KHZ	32+j42	2.97

Figure 14. Carrier and Sideband Patterns for Traditional Phasing System



FREQUENCY	COMMON POINT IMPEDANCE	VSWR
-10 KHZ	50-j16	1.38
CARRIER	50+j0	1.00
+10 KHZ	44+j12	1.34

Figure 15. Carrier and Sideband Patterns for Modern Broadband Phasing System

CONCLUSION

As has been shown, modern computational capability has led to vast improvements in directional antenna feeder system design capabilities. Studies aimed at improving bandwidth performance and adjustability have suggested modern circuits which are often simpler than those which composed the standard technology of the past.

When planning a new facility, or improvements or refurbishment of an existing antenna system, careful attention to the phasing and coupling system design can yield great rewards. The additional engineering costs should be viewed in the proper perspective. Much is often spent on audio processing and stereo generating equipment, where the possibilities of making great improvements in null-region coverage and overall over-the-air sound by antenna system improvement are left unexplored.

SUGGESTIONS FOR FURTHER READING

Many articles dealing with specific areas of antenna feeder system design have been published in the trade press within the last 15 years. Although it is impossible to enumerate them all here, three of the most active authors are Grant Bingeman, Dane Jubera, and Jerry Westberg.

The following textbooks are listed because they treat the basic principles discussed herein and at least some of them should be available at any library with an engineering section:

Moment Method Modeling

Stutzman, W. and Thiele, G., Antenna Theory and Design, John Wiley & Sons, 1981

Balanis, C., Antenna Theory, Analysis and Design, Harper & Row, 1982

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Johnson, R. and Jasik, H., Antenna Engineering Handbook, Chapter 43, McGraw-Hill, 1984

USING ISOLATION TRANSFORMERS TO LEASE AM TOWER SPACE

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Abstract- With the emergence of today's mobile communications technology-cellular telephone, paging systems, trunking systems, etc.--the need for vertical height of the associated antennas above average terrain has presented a possible new source of income for the owners of AM broadcast stations. Also as a result of a growing opposition to the erection of new tower structures, the installation of the mobile communications antenna(s) on an existing AM tower is a low cost and fast solution in getting the mobile site up and operating. A comparison of several types of isolation transformers appropriate for this application will be discussed.

INTRODUCTION

In this paper, the standard AM broadcast tower is a series-fed, base insulated, guyed or self-supporting tower. The point at the base of the tower above the base insulator, to which RF power is applied, presents a lumped impedance load to the RF source. This impedance is referred to as the self impedance for an omni-directional antenna and as the drive point impedance for an element in a directional antenna. In order to add any ancillary antenna to the AM tower structure requires that a corresponding transmission line be routed across the base insulator and up the tower to the antenna. This antenna and transmission line must be added to the tower in such a way that minimum change in the tower impedance and radiating characteristics results. This is in fact the purpose of an isolation transformer. From basic network theory if an impedance of magnitude greater than 10 times the tower base impedance magnitude is applied in parallel with the tower base impedance, the resulting impedance will be approximately equal to the original tower base impedance. In addition to effectively presenting a high impedance across the base of the tower, an isolation transformer must also present minimum insertion loss to the RF source driving the ancillary antenna or to the received RF signal coming from the antenna, whichever may be the case.

Three types of isolation transformers will be compared in this paper--the quarter-wave stub, the isocoupler and the multi-coax isolation inductor. This comparison will assess the relative merits of these three approaches with regard to the impedance presented across the tower base, insertion loss, peak voltage rating, throughput power rating, installation procedure and cost. Also the applicable FCC rules will be discussed.

The Quarter-Wave Stub

The quarter-wave stub is the oldest technique that has been used for isolating transmission lines across an AM tower base insulator. Figures 1A and 1B illustrate two approaches of quarter-wave stub installation for tower electrical heights greater than or equal to 90 degrees or less than 90 degrees, respectively[1]. For tower heights in excess of quarter-wave, the transmission line is insulated from the tower up to the quarter-wave point at which the outer conductor is

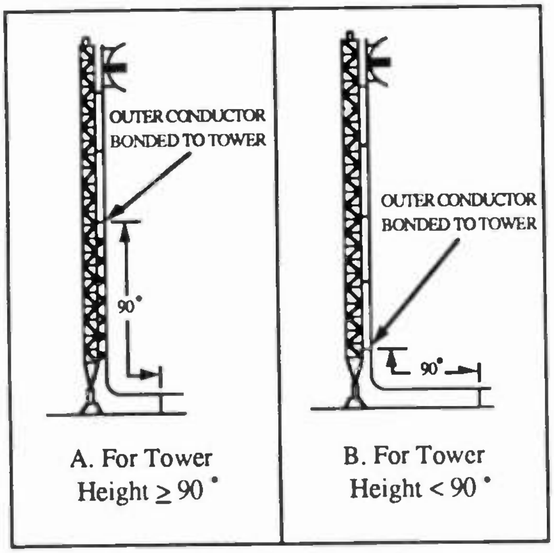

Fig.1 Typical Quarter-Wave Stub Installations.

bonded to the antenna ground just below the base insulator. The combination of the outer conductor of the coaxial cable in parallel with the tower leg electrically equates to a quarter wave section of transmission line. By bonding the outer conductor of the coaxial cable to the tower at the end of the quarter-wave section, a shorted quarter-wave transmission line is formed, which in accordance with classical transmission line theory,[2] presents an open circuit to the RF source, i.e., the AM transmitter.

It is too often the case that quarter-wave stubs are improperly installed on the tower resulting in poor impedance isolation and reduced tower radiating efficiency. If the coaxial cable is not bonded to the tower at the quarter wave point, a variable vacuum capacitor or variable inductor can be installed between the coaxial cable outer conductor and the tower base to effectively increase or decrease, respectively, the electrical length of the transmission line stub. This adjustment can be made to minimize the change in the original tower base impedance.

For tower heights less than a quarter-wave, the quarter-wave section of transmission line is insulated above ground as opposed to being insulated off the tower leg as illustrated in Figure 1B. The outer conductor is bonded to the tower just above the base insulator and to the antenna ground at the quarter-wave point toward the transmitter. This approach is rarely used due to the separation distance required between the tower and the transmitter.

The major drawback of the quarter wave stub isolation approach is the influence of the insulated quarter-wave coaxial cable outer conductor on the shape of the tower radiated field. The RF current flowing in the tower leg, which supports the insulated quarter-wave stub, produces an electromagnetic field. The presence of the outer conductor of the quarter wave stub in this field results in an induced current flow on the outer conductor as illustrated in Figure 2 below. The resulting interaction of the radiated field from the tower with the re-radiated field from the quarter-wave stub outer conductor will tend to directionalize the overall radiated field of the tower.

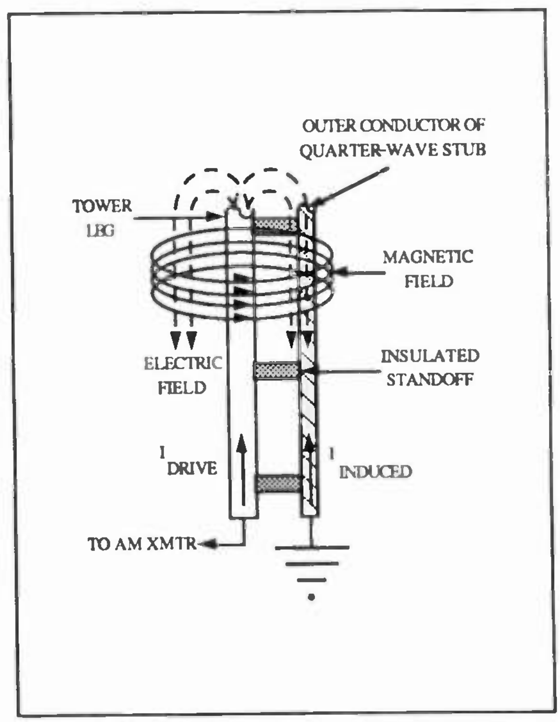

Fig. 2 Induced Current in the Quarter-Wave Stub Outer Conductor.

This is a particular concern if the quarter-wave stub is installed on a tower in a directional array. One way to minimize the effects of this induced field is to route the quarter wave stub to the inside of the tower structure. Also an alternate approach to the typical installation is to insert a 90 degree length of transmission line between the transmit or receive equipment and the base of the tower.[1] The outer conductor would be bonded to the tower just above the base insulator and grounded at the transmit/receive end, which would normally be the case anyway.

The relative merits of the quarter wave stub approach are summarized in Table 1 below.

Table 1. Advantages and Disadvantages of the Quarter Wave Stub

Advantages	Disadvantages
*Simple, Does Not Require Separate Isolation Device	*Reradiation From Insulated Outer Conductor Distorts Pattern Shape
*Not Bandwidth Limited	*Not As Conducive To Multiple Line Isolation As Other Approaches.
*Not Limited In Utilization By Peak RF Voltage Across Base Insulator	*Critical Install Procedure To Achieve Good Impedance Isolation
*No Additional Insertion Loss	*Tower Bond Point May Be At or Near Voltage Maximum, Hence May be Subject To Arcing

Isocouplers

The second approach for isolation of ancillary antennas on an AM broadcast tower is the isocoupler. An isocoupler may be described in electronic terminology as a double-tuned, inductively-coupled isolation transformer. These devices were originally used in the early 60's to isolate FM antennas on an AM tower. The same design concept has been expanded to higher frequencies to include remote pickups (450-455 MHZ), trunking systems (800 MHZ) and studio-to-transmitter links (948-955 MHZ) as well as other applications in the 24-1000 MHZ region.

Figure 3 below shows the internal components of a typical isocoupler.

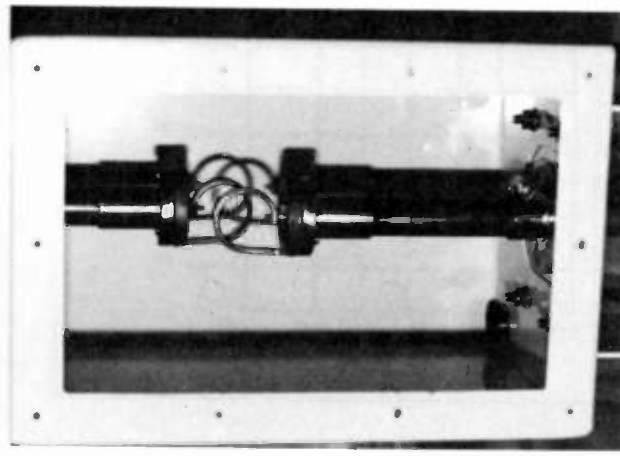

Fig.3 Internal Components of a Typical Isocoupler.

The isocoupler input and output transmission line sections are terminated in a series inductor and capacitor that form a low Q series resonant circuit at the pass frequency of the isocoupler soas to yield a 50 ohm termination in the desired passband. The terminating copper loop constitutes the inductor whereas the brass stud threaded into the base of the loop and extending into a teflon dielectric in an expanded end of the center conductor assembly equates to a variable capacitor, which is adjusted to yield the desired 50 ohm termination at the pass frequency. This capacitance value is fine-tuned in accordance with the mutual inductance that exists between the two coupled loops to yield the desired passband characteristics with minimum insertion loss, which is typically 0.2-0.8 db depending on the throughput frequency and the power rating. The throughput power rating of this isocoupler design is limited by the voltage breakdown of the variable capacitor assembly, which also impacts the operational passband of the isocoupler. Power rating is traded off for bandwidth in the design of the variable capacitor to yield the best possible performance.

The typical passband of an isocoupler utilizing the electrical design as shown in Figure 6 is illustrated by the oscilloscope trace shown in Figure 4 below.

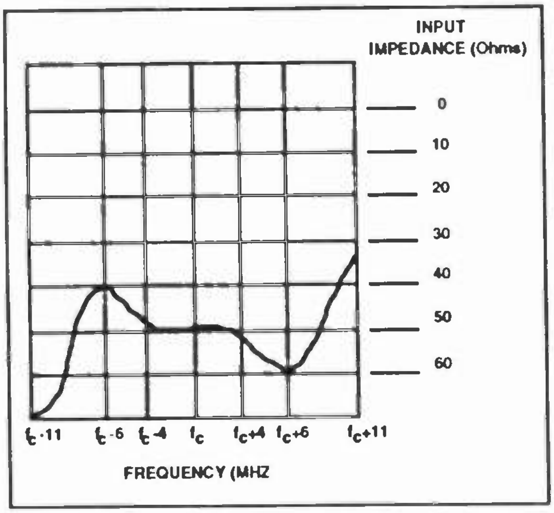

Fig.4 Typical Frequency Passband of Isocoupler.

This trace was achieved by coupling a sweep generator to the isocoupler input with the isocoupler output terminated in a 50 ohm load. The output of the sweep generator was fed into the X-input of an oscilloscope set up to operate in an X-Y format. The Y input was originally calibrated with a 50 ohm load prior to inserting the isocoupler in the test fixture. This passband contour illustrates the response of the input LC loop assembly on the left half of the graph and of the output LC loop assembly on the right half. As is evident in the sweep trace, the bandwidth over which 50 ohms input impedance is maintained is 8 MHZ.

Another consideration in determining the suitability of an isocoupler for installation on an AM tower is it's peak voltage rating in relation to the peak RF voltage that appears across the tower base with maximum AM transmitter peak modulation. The isocoupler voltage rating is determined by the minimum spacing between the conducting elements that are effectively across the tower base. This voltage rating ranges between 6 KV and 35 KV depending on the passband, power rating and manufacturer. A minimum safety factor in isocoupler voltage rating of twice the peak base voltage should be specified to accommodate lightning related voltage transients.

The impedance presented by an isocoupler across the base of an AM tower is typically represented as a lumped capacitance ranging in value from 6 to 200 picofarads, depending on the passband, power rating and manufacturer. In the AM band this will constitute a capacitive reactance between -j675 and -j40,000, which typically should result in minimal change in the tower base impedance. This fact also makes the isocoupler a more acceptable isolation approach for the installation of multiple transmission lines across an AM tower base insulator.

An excellent example of a multi-isocoupler installation is located at the directional antenna site of WROK Radio in Rockford, Illinois. The WROK array is a three-tower inline operating at 5 kilowatts daytime on 1440 kHz. The center tower of this array is 225 degrees electrical height or 440 feet. A total of twelve isocouplers are installed across the base of this tower as illustrated in the block diagram (see Figure 5 below) provided by Mr. Marvin Beasley of Sinclair Radio Laboratories.[3]

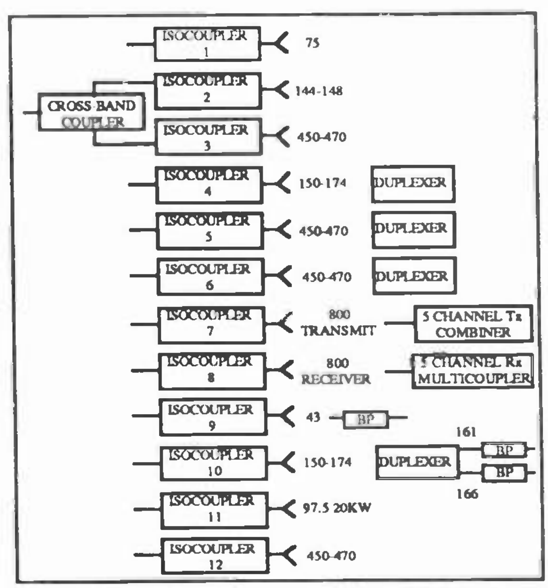

Fig.5 Block Diagram of Multiple Isocoupler Installation at WROK Radio.

As shown in Figure 5, these isocouplers are used for a variety of applications, including FM transmission, land mobile communications, remote pickup units and an 800 mHz trunking system, which has recently been increased to its maximum ten channel capacity due to increased user demand. According to Mr. Dennis Carter of Rock River Service Company, which is under contract to maintain this site, the operational reliability has been excellent since the original installation in 1986. Photographs of the WROK installation are shown in Figures 6A and 6B.

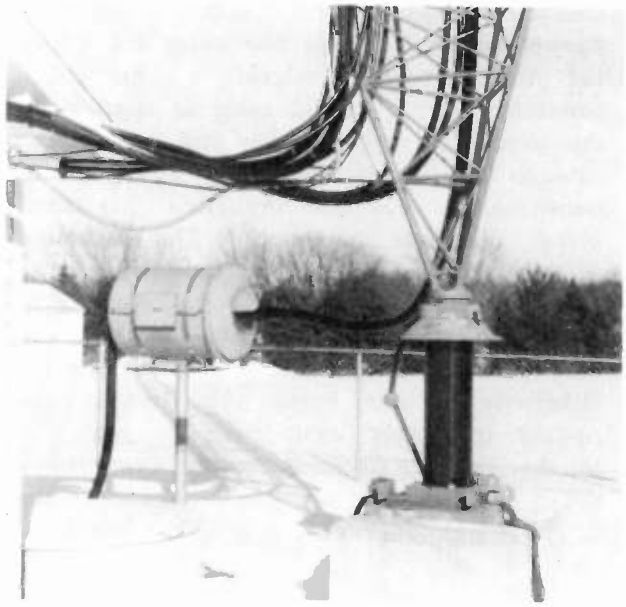

Fig.6B WROK FM Isocoupler and Ancillary Antenna Transmission Lines Routed Across Tower Base Insulator.

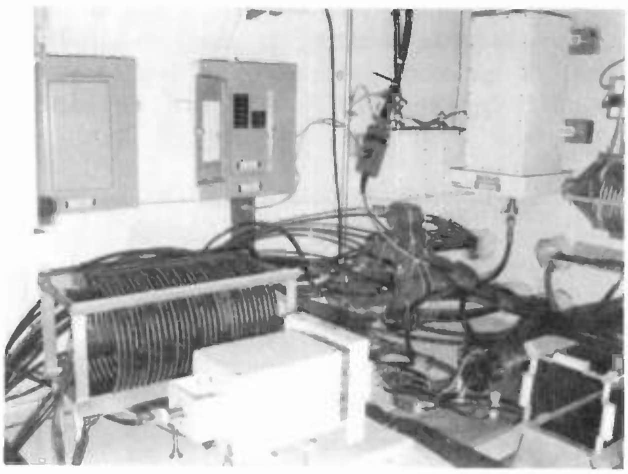

Fig.6A Multiple Isocoupler Installation at WROK Radio.

Regarding the proper installation of an isocoupler the outer conductor of the upper transmission line section of the isocoupler is bonded to the tower, and the outer conductor of the lower section is bonded to the tower ground system. The outer conductor of the transmission line between the isocoupler and the antenna on the tower should be bonded to the tower above the base insulator, at the antenna end and at intervals not to exceed 50 feet in between. The transmission line on the tower should appear electrically as an integral part of the tower structure soas not to effect the omni-directional shape and magnitude of the radiated field.

The pros and cons of the isocoupler isolation transformer are summarized in Table 2 below.

Table 2. Advantages and Disadvantages of Isocouplers

Advantages	Disadvantages
*High Impedance	*Susceptible To Lightning Damage
*Low Insertion Loss	*Insufficient Voltage Rating For Some Applications
*Acceptable Bandwidth For Most Applications	*Limited In RF Throughput Power Rating
*Peak Voltage Rating ≥ 15KV	

Multi-Coax Isolation Inductor

The third isolation transformer approach to be considered is the isolation inductor. This method of isolation has been historically associated with the use of sampling loops installed on the towers in an AM directional array. An isolation coil is simply an inductor wound with the same or different size of coaxial cable as that being fed across the tower base insulator. The outer conductor of the coaxial cable is attached to the tower RF feed on one end and to the antenna ground on the other end soas to constitute a lumped inductive reactance across the base insulator. The inductive reactance should be sufficiently large compared to the base impedance magnitude to result in a less than 2 percent change in the base resistance or common point resistance. Otherwise a new Form 302 showing the revised impedance value must be submitted to the Federal Communications Commission (FCC) in accordance with section 73.45 of the FCC rules.[4]

In a typical trunking system or cellular telephone installation a minimum of two transmission lines are required for the associated transmit and receive antennas. The bandwidth and throughput power ratings of isocouplers may be insufficient for this application, and the installation of multiple quarter wave stubs may be impractical as well as unadvisable in maintaining the AM station's coverage. The solution for this situation is a multi-coax isolation inductor as illustrated in Figure 7.

The intent of the multi-coax isolation inductor is to merge several isolation inductors together so that electrically they equate to one isolation inductor as opposed to several isolation inductors in parallel, which would reduce the effective impedance across the tower base to an unusable level. This is accomplished by forming a large copper tube with a slot in its side into an inductor. Individual coaxial cables are placed through the slot inside the copper inductor, and the outer conductors of each line section are electrically connected to the copper inductor winding at each end. The ends of each line section are terminated in a standard type N or UHF connector. The outer

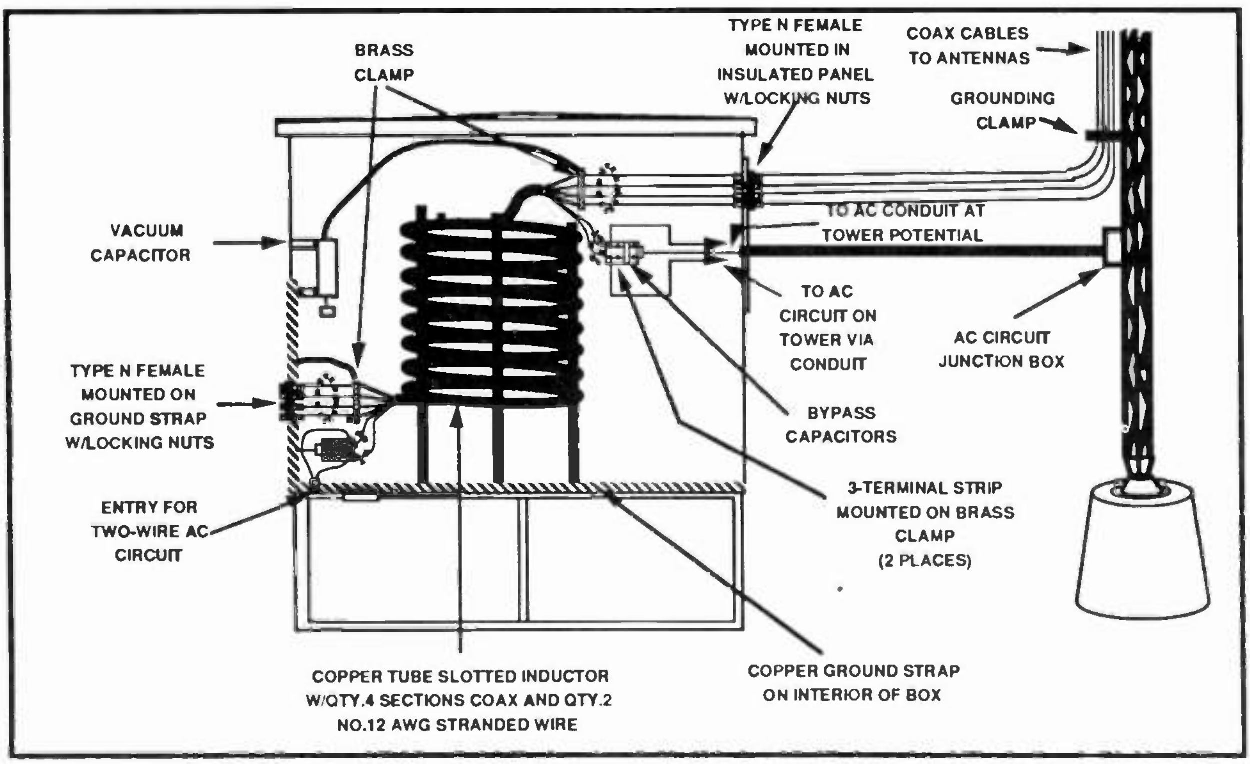

Fig.7 Multi-Coax Isolation Inductor Configured for Cellular Telephone Installation on AM Broadcast Tower.

conductors of all the lines going onto the antennas and at a maximum interval of 50 feet. The outer conductors of all the line sections, going to the transmitter building should be connected to the antenna ground.

Another feature of this isolation scheme is that other AC insulated wiring can be laid in the slotted copper inductor and terminated in bypass capacitors at both ends in a similar configuration to a standard lighting choke. This wiring could be used for tower lighting, antenna heaters, antenna rotors, preamplifiers or other applications.

In the event that the tower electrical height is in excess of 120 degrees resulting in a high base impedance, a variable vacuum capacitor could be connected in parallel with the slotted copper inductor to yield an anti-resonant tank circuit. In this case the reactance of the variable vacuum capacitor would be set equal to that of the copper inductor assembly at the AM carrier frequency. The resonant Q of the tank circuit should be low soas to maintain a high impedance at the AM carrier as well as at the $\pm$10 kHz sidebands. This will also help minimize the reactive volt-amperes that are produced in the tank circuit as a result of the AM transmitter modulation. It is also important to note that the peak voltage rating of the anti-resonated isolation inductor is limited to that of the variable vacuum capacitor.

The only disadvantage of this isolation transformation approach is the insertion loss added by the coaxial cable in the isolation inductor. Typically 1-5/8" foam coaxial cable is utilized to feed cellular antennas soas to minimize the attenuation in their operational frequency range. The slotted copper winding would be prohibitively large to allow for two or more sections of 1-5/8" line; hence smaller coaxial cables with higher attenuation must be used. The loss will typically range between 1.5 and 2 db, including the connector losses at each end. With additional preamplification, this loss can be offset if necessary.

An example of a multi-coax isolation inductor installation is shown in Figure 8.

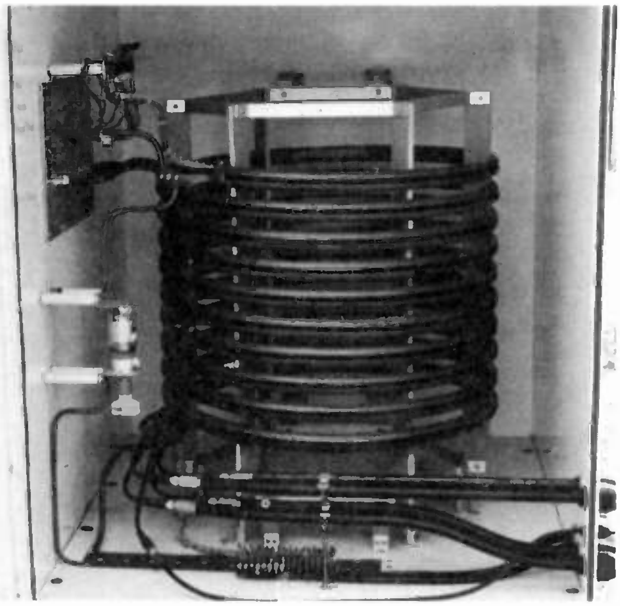

Fig.8 Multi-Coax Isolation Inductor Installation At WBET Radio in Brockton, Massachusetts.

This unit is installed at the base of one of the self-supporting towers in the two-tower directional array of WBET Radio in Brockton, Massachusetts. This slotted isolation inductor has the following cables and wire inserted in it:

A. Qty.5 Pieces of 3/8" Foam Transmission Line For:
   a. Sampling Loop
   b. 455.8 MHz Antenna
   c. 450.7 MHz Antenna
   d. 173.225 MHz Antenna
   e. 945 MHz STL Dish

B. Qty.3 No.10 AWG Wires For Tower Lighting

C. Qty.8 No.14 AWG Wires For An Antenna Rotor

D. Qty.3 No.12 AWG Wires For FM Antenna Heater

This example testifies to the versatile use of the multi-coax isolation inductor. A summary of the advantages and disadvantages of the multi-coax isolation inductor are listed in Table 3 below.

Table 3. Advantages and Disadvantages of Multi-Coax Isolation Inductor.

Advantages	Disadvantages
*Multi Transmission Line Isolation	*Higher Insertion Loss Than Isocoupler or Quarter-Wave Stub
*Accessory AC Circuit Isolation	
*Very High Effective Impedance Across Tower Base	
*Not Bandwidth Limited	
*Comparable In Cost To Multiple Isocoupler Installation	

Summary

For a comparison of the relative merits of the three isolation transformer approaches that have been discussed the quarter-wave stub, the isocoupler and the multi-coax isolation inductor--please refer to the bar chart in Figure 9.

If the throughput power rating and or tower base peak voltage exceed the ratings of available isocouplers, a properly installed quarter-wave stub is the best solution. For the majority of single or double transmission line applications, the isocoupler is the most cost effective solution. For applications involving more than two transmission lines and/or wide bandwidth and high throughput power requirements that exceed isocoupler ratings, the multi-coax isolation inductor is the best solution.

In closing, it is also important to remember the following FCC policy regarding towers in a directional antenna--any change above the base insulator requires a partial proof of performance.[4] This added cost factor makes non-directional towers the most desirable choice.

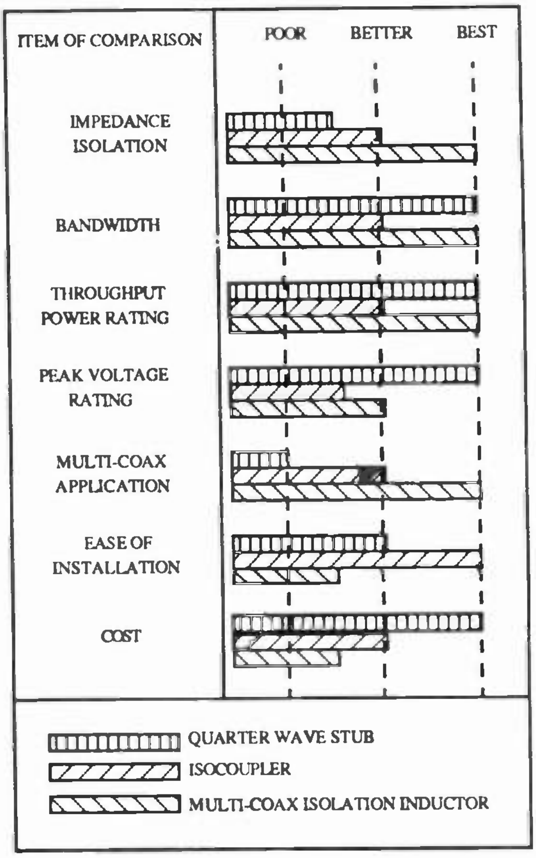

Fig.9 Comparison of the Relative Merits of the Three Isolation Transformer Approaches.

Acknowledgements

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IMPLEMENTATION OF ANTISKYWAVE ANTENNA TECHNOLOGY BY EXTREME TOP LOADING OF SHORT ANTENNAS IN A DIRECTIONAL ARRAY

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ABSTRACT

Top loading of short vertical antennas is usually limited to groundwave field enhancement. A successful method of top loading to achieve antiskywave properties is detailed. The allocation criteria of the KNWZ Thousand Palms, California nighttime directional array includes protection of a station in the direction of the major population center in the service area making antiskywave design essential. The high radiation efficiency achieved in the KNWZ nighttime array is also attributable to the antiskywave characteristics in the antenna design. A similar nondirectional toploading implementation at KIAM Nenana, Alaska achieved dramatic improvement in fringe reception both daytime and nighttime.

ANTI SKYWAVE ANTENNA

The well known 5/8 wavelength tower is in effect a gain vertical colinear array because of the current distribution. The gain and the change in vertical section is largely due to the phasing of the currents along the tower. In the far field a 5/8 wavelength tower and its ground reflection behave as a 3 element broadside array. The additional gain clearly comes from the suppression of the otherwise wasted high angle radiation. A similar broadside vertical array can be created with a rather wide range of "element" spacings (or tower height) while retaining significant gain and desireable improvement in the vertical section characteristics. This can be achieved by shifting the normal current distribution with top loading on towers shorter than 5/8 wavelength. As the tower becomes considerably shorter than 5/8 wavelength the method of top loading becomes more important. In the case of KNWZ Thousand Palms, California the deepest required protection was to XEAZ Tijuana, Mexico at 52 degree vertical angle in the direction of Palm Springs. Top loading from 93 degrees of physical height to 216 degrees electrical height provided the needed high angle skywave suppression while increasing gain for the KNWZ array.

MEASURED PERFORMANCE

Measured gain beyond theoretical for the ND and DA-2 arrays indicated gain of 1.5, 1.2 and 3 db was actually achieved at KNWZ. A similar toploading installation was done at KIAM. The KIAM ND tower in Nenana, Alaska achieved 1 db measured gain as constructed when toploaded from 68 degrees physical height to 208 degrees electrical height. KIAM has regular listeners and FM translators as much as 200 miles distant in areas with signal strengths of as little as 25 microvolts per meter. Before installation of the toploading the signal strength at the Tanana translator was S7 on the receiver and varied downward to S5. Since the toploading installation the receiver indication is S8 with no visible fading day or night. Other fringe locations indicate a dramatic improvement in reception with some locations reporting full time listenable signal where there was little or no discernable signal before. This is clearly greater improvement than can be attributed to the measured 1 dB signal strength increase as measured in the nondirectional proof of performance. We attribute the much larger fringe reception increase to changes in the high angle skywave interference to the groundwave. There was also some increase in the actual transmitter modulation ability resulting from the excellent antenna bandwidth.

DEFINING THE ANTENNA

The tower characteristic and the amount of top loading is defined most clearly by the location of the current null on the tower. Measurement of the current is facilitated by the construction of a shielded torroidal loop of coaxial cable which can encompass the tower section. This type of loop is much more tightly coupled to the actual tower current than a conventional sample loop or field meter antenna loop and will easily define the current null location with accuracy of better than one degree. After setting the current null at the exact location desired the loop is moved up and down the tower at convenient increments to measure the actual current distribution. Relative current distributions measured were not sinusoidal but appeared to have the approximately the same current area under the curve that should result from a theoretical sinusoidal current distribution.

The large degree of top loading was accomplished using a relatively small capacity to ground. The KNWZ array used 5m of the top guy wires. The KIAM tower used a 20 ft diameter capacitance hat. The resulting capacitance is insulated from the top of the tower in each case but attached via a coaxial transmission line to lumped reactances at the bottom of the tower for ease of adjustment. An adjustable reactance at the tower base allows for easy adjustment of the current distribution over a wide nearly infinitely variable range of current null locations.

METHOD OF ADJUSTMENT

The most practical method of adjustment is to place the current sample loop at the desired tower elevation and then to adjust the loading reactance for a minimum sample current. Sample current nulls as much as 30 db below the current maximum have been observed. The severely top loaded antenna is relatively broadband with driving point resistances considerably higher than the standard tower of the same physical height and have a rather low base reactance resulting in a low Q driving impedance. The KNWZ daytime and nighttime directional arrays both exhibited good bandwidth without specific effort going into field adjustment of the bandwidths of the common point networks or antenna tuning units. The KIAM antenna characteristic provided 10 kHz sideband VSWR's of 1.7 and 2.0 at the tower and was rduced to 1.04 and 1.50 using a standard three component "T" network from parts on hand. The resulting measured KIAM modulation sideband symmetry was down 2 db at 622 kHz relative to 638 kHz with 8 kHz tone modulation and the plate of the Collins 21E transmitter tuned just slightly on the high efficiency side of the dip as is customary. Overall, these antennas have been successful in all respects.

PREVENTIVE MAINTENANCE FOR AM RADIO TOWERS

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INTRODUCTION

The very nature of an AM tower as a radiating entity presents unique circumstances in maintaining and conducting visual inspections of the tower. Such inspections should be conducted on an annual basis to ensure compliance with current FAA regulations and to prevent minor maintenance needs from developing into serious and costly repairs.

ANNUAL VISUAL INSPECTION

A comprehensive annual tower maintenance inspection should review all facets of the tower from the ground system and the tower structural integrity to the safety systems.

The Tower Foundation

The tower base should be visually inspected for structural integrity. It should be checked for signs of cracking, erosion, and undue settling or movement. The foundation should be checked for adequate drainage away from the base of the tower.

Guy Wire Encasements and Anchors

An inspection should be conducted for signs of rusting to the anchor heads, as well as checking that appropriate protection of the anchor heads exists to prevent any rust from developing. The cotter pins, lock nuts, and any other securing components should also be examined for signs of rust and displacement. Guy wire encasements should be examined for any signs of cracking, erosion, and settling. As with the foundations, it is important for the extended life of these encasements, that there is adequate drainage away from the encasements. If there is no concrete around the anchor shaft, a check should be made to determine if acidic soil is eroding the anchor shaft.

Grounding System

As there exists extensive grounding systems for an AM tower, each should be individually evaluated. It is important to ensure that optimal grounding exists, for this will impact the performance of the signal. The radial grounding system should be examined for complete and appropriate connections and continuous soil coverage. The type of soil covering the grounding wires can also affect the electrical disbursements. Acidic soil can also erode the grounding after a period of time.

Base Insulators and Guy Wire Insulators

The base insulators and the guy wire insulators should be checked for any cracking or damage. The insulators should be examined for any grease or other conductive films or materials. The ball gaps on the base insulator should be check for proper spacing. Guy insulators should be inspected using high power binoculars. A visual inspection during a thunder storm can also be revealing if arcing is noted. Special attention should be given to non-metallic guy materials such as fiberglass rods. Contaminated rods can arc and burn through causing failure of the guy and possibly the tower. The lighting choke or transformer should checked. the rings on the transformer should be properly spaced to prevent potential damage.

Structural Members

The entire tower should be inspected to ensure that none of the nuts and bolts are missing and that each is tightly and appropriately in place. All welds should be checked for cracking and failures. Each structural member should be examined to see if it is bowed, bent or loose, which would affect the integrity of the tower.

All members should be examined for potential arcing hazards. Arcing can be produced by two loose, crossing members or two members that are not adequately grounded. This condition can result in structural integrity damage to the member. Welds at the tower flanges made for continuity should be inspected. If cracks are observed, alternate methods can be employed to assure continuity.

Corrosion Protection

The galvanizing or the paint coverage on a tower is critical to ensure extended life of the tower. On a painted tower, continuous maintenance of the paint should be conducted to prohibit extensive corrosion of the tower from gaining foothold. As with so many things, maintenance is far less costly than the solution of a badly rusted tower: complete sand blasting and repainting of an entire tower. Any evidence of rust should be thoroughly cleaned and coated with a good rust inhibiter. In addition to corrosion prevention, tower paint also serves as an obstruction warning system, as required by the FAA. While the paint visually may appear in good condition, it may not meet the FAA color spectrum regulations due to fading. Damage to the galvanizing should be cleaned thoroughly and repaired with cold galvanizing compound.

Lighting System

In both the strobe light and red light systems the FAA sets out specific operating parameters for obstruction warning systems. Either light system must be examined to determine that these criteria are being met. In the red light system the bulbs and lenses must be changed and checked on a regular basis. Also, the conduit and connections should be inspected to ensure that there is adequate protection from the elements.

The strobe light system should be examined for proper interval flashing. Also, the intensity of the lights should meet the specified candelas, as set by the FAA.

The lighting and paint regulations of the tower for obstruction warning have become more vigorously enforced by the FAA than in the past due to increasing potential liability problems. FAA specification describes both painting and lighting requirements. With the use of medium intensity strobe lights, paint is not required on towers under 500 feet. Consideration of future maintenance could affect a decision on whether to change the lighting system.

Ladder and Safety Devices

Federal requirements by OSHA are becoming more extensive each year for those individuals working on or inspecting towers. These regulations are not only rapidly changing, but can be different for each state. The existing safety ladders and devices should be examined for their structural integrity, and an evaluation should be conducted to determine if the tower meets the most currently established safety regulations.

Plumb and Tension

It is important to determine that the tower is plumb and that the guy wires are correctly tensioned, so that the designed live and dead loads caused by wind, snow, ice, etc. can be met.

Other items that can affect the structural integrity, plumb, and tension of the tower can include additional equipment, (antenna/dishes) installed on the tower for which the tower was not originally designed. A complete inventory of the additional equipment on the tower should be compared to the original design to determine that the amount of antennas, their height, the size of their transmission lines and mounting brackets does not exceed the original tower design loads. If the current antenna loads exceed the original design loads a structural analysis should be done to determine if the tower is overloaded.

Summary

A visual inspection, reviewing at minimum the discussed categories, should be done on an annual basis. This will provide the station a current and annual log of their tower's condition and service as an annual maintenance opportunity for the tower. It will also provide forewarning of potential major repairs or changes that may need to be done. These then can be undertaken in a manner commensurate with need and budget parameters versus on an emergency basis.

The station should expect a written summary report of their tower's inspection accompanied with photographic documentation of any noted and significant problems. The report also should outline maintenance and remedial steps that need to be taken for the tower's proper, optimal and extended operation. Along with this, appropriate consideration should be given to unmet or newly enforced safety regulations to ensure that the safety needs of those working on or operating around the tower are being met.

NOISE FREE RADIO FOR AM BROADCASTING

George W. Yazell, P.E.
Noise Free Radio
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Abstract: Presenting a practical way to **add** frequency modulation to AM broadcast transmitters, providing all the advantages of FM in a broadcast band channel, only 10kHz wide, while retaining compatibility with all existing AM radios. Description of early laboratory experiments and a report of tests authorized by FCC and conducted by WQYK(AM), Seffner, FL. Discussion of a plan to expedite implementation of NFR.

EVOLUTION OF NOISE FREE RADIO

I retired in 1987, after 48 years in Broadcasting and communications. The social security check arrived monthly. There were few deadlines to meet. I had time to cerebrate the stories in the trade press about the decline in ratings, revenue and resale value of AM stations.

I was intrigued by discussions of no-sky-wave antennas, NRSC filters, AM stereo, audio pre-emphasis, etc. But, I was not convinced that any of these could solve the dilemma. There must be a better solution.

About three years ago, I got the wild idea of substituting narrow band frequency modulation for amplitude modulation, on the medium-wave (AM) broadcast band. The experiment worked, but left a lot to be desired. While there was some improvement over AM when the signal was received on an NBFM radio, the audio quality was inferior to wideband FM. The NBFM signal could be received on a manually tuned radio (slope detection), but not on a synthesized tuner.

Next came the idea of **combining** NBFM with AM. That would be compatible with all existing AM radios. When received on a specially designed **Noise Free Radio (NFR)**, it would provide two audio services, which could be used as simulcast or for separate programming.

Testing this theory showed improvement, but there were still problems. I used 5 kHz deviation for the FM component and 100% amplitude modulation. There was cross-talk between AM and FM, and while the FM quality was excellent, the "capture effect" did not appear to be as effective as I had expected.

Reviewing frequency modulation theory convinced me that to achieve the quality of an FM broadcast, my receiver would require delivering a wideband signal to the FM demodulator.

My next "transmitter" had the AM limited to 90% modulation, and the FM deviation limited to 3 kHz. For an NFR receiver we modified an AM tuner by retuning the IF transformers from 455 kHz to 428 kHz. The 428 kHz IF signal with 3kHz deviation was processed through a series of frequency multiplier stages where the IF was multiplied 25 times. This produced a new IF signal at 10.7 MHz with 75 kHz deviation. That signal was then injected into the IF amplifier of a standard FM broadcast receiver for demodulation.

This combination sounded so good that I started giving public demonstrations to broadcaster groups, ham radio clubs, and finally at the SBE National Convention in Kansas City in October, 1989.

On November 16, 1989, the FCC held an open hearing, with all seven commissioners present, to discuss the many suggestions being offered to improve the viability of the AM band. I submitted my comments outlining the NFR experiments I had been conducting, and suggesting the FCC look favorably on requests by AM licensees for permission to conduct "on the air" experiments with NFR, or other modifications of Standard Broadcast Band transmitters and receivers; modifications that could result in a more effective broadcast service. Authorization for such experiments should require a detailed report of results achieved in the experiments.

On December 10, 1989, WQYK(AM), Seffner, Florida, submitted a request for permission to conduct NFR tests over a period of 6 months.

March 22, 1990, the FCC issued a Temporary Experimental Authorization for Narrow Band FM modulation of the AM Broadcast Carrier of WQYK(AM). The authorization further states, "Initial analysis indicates that your proposal offers the prospect of significant technical advancement and benefit to the AM broadcast service such that the public interest would be served by allowing you to test the proposed system under rigorous, real world, broadcast conditions."

Frank Berry, Director of Technical Operations at WQYK(AM) and I collaborated on design of the NFR exciter used in the tests. The major technical improvement we made was in using an FM exciter on 101.0 MHz with 100 kHz deviation. The output of this exciter was processed in two "divide by ten" frequency divider chips, resulting in a signal on 1010 kHz with 1 kHz deviation. Limiting the deviation to only 1 kHz effectively eliminated the FM to AM crosstalk problem.

This reduced FM modulation level required modification of the receiver also. We used a normal 455 kHz IF which was multiplied 75 times, resulting in a new signal of 34.125 MHz with 75 kHz deviation. That signal was heterodyned with a crystal oscillator on 23.425 MHz, producing a new IF of 10.700 MHz with 75 kHz deviation, which was then demodulated in a standard FM broadcast receiver.

Those tests have been completed and a report on our findings have been submitted to the FCC.

BASIC FM THEORY

In FM the instantaneous frequency of the RF output wave differs from the unmodulated carrier frequency by an amount proportional to the instantaneous value of the modulating wave. For example, consider a 100 MHz carrier modulated by a 1000 Hz audio tone with a peak amplitude of 1 volt, that results in a peak deviation of 10 kHz. If the amplitude of the audio input is increased to 2 volts, the frequency deviation will become 20 kHz.

An important term to understand in FM is "MODULATION INDEX". With a sinusoidal modulating tone, the modulation index is equal to the frequency deviation divided by the modulating frequency. In FM broadcasting, the FCC has established that a frequency deviation of 75 kHz constitutes 100% modulation

For example, when a monophonic FM transmitter is modulated 100% by a 50 Hz tone, the modulation index is 1500 (M = 75,000 / 50 = 1,500).

When the carrier is modulated 100% by a 15kHz tone, the modulation index is 5. (M = 75,000/15,000 = 5).

The significant point here is that the modulation index is inversely proportional to the frequency of the modulating tone.

When an RF carrier is frequency modulated, sideband frequencies are generated, theoretically an infinite number of sidebands. These sidebands consist of pairs of sidebands spaced above and below the carrier by multiples of the modulating tone. Example: The carrier is modulated by a single 5 kHz sinusoidal tone. Sidebands are created at 5, 10, 15, 20, 25, 30 kHz, etc., etc. above and below the carrier.

When the modulation index is low (<0.5) the second and higher order sidebands are so small that the output consists essentially of the carrier and the pair of first order sidebands. When the modulation index is increased by raising the level of modulation, the higher order sidebands become more prominent.

Bessel Functions

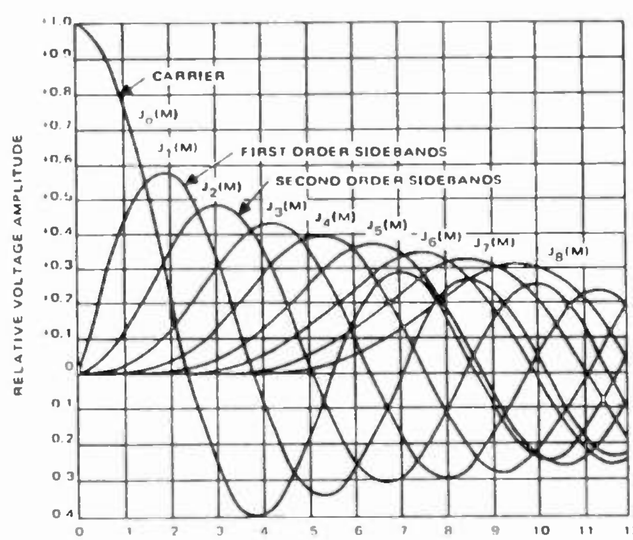

Fig. 1 MODULATION INDEX

Relationship of carrier and sideband amplitudes to modulation index.

The amplitude of the carrier and the sidebands can be expressed mathematically by the use of Bessel functions. Fig. 1 shows how the carrier and first eight pairs of sideband components vary with modulation index. (This is simply a plot of Bessel functions, and applies to single tone modulation only.) The first order sidebands (a pair, one above and one below the carrier frequency) are displaced from the carrier by an amount equal to the modulating frequency (5 kHz). The second is twice the modulating frequency away from the carrier, and so on.

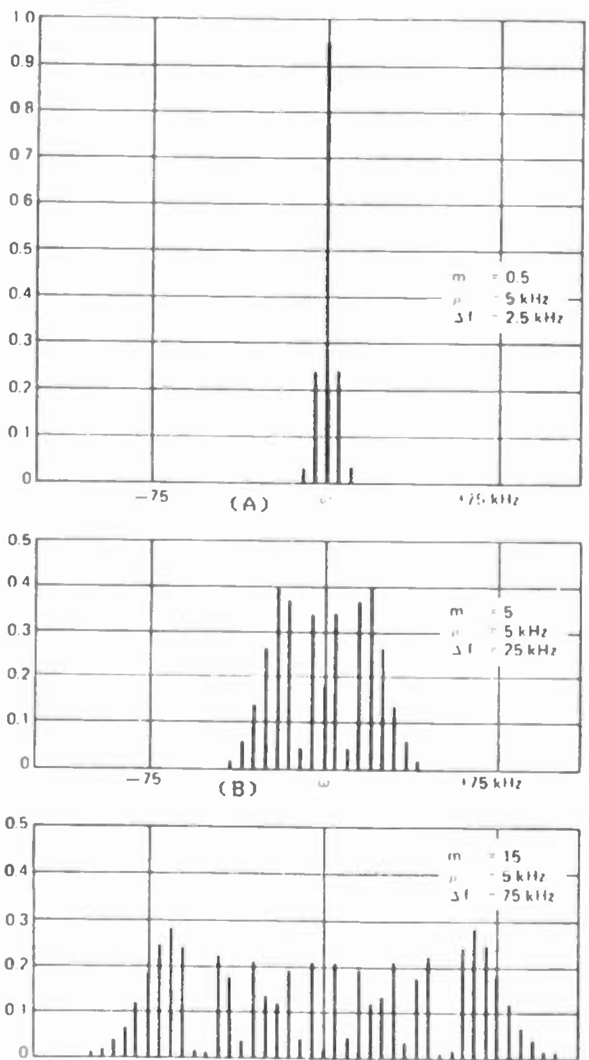

Fig. 2

RF spectrum with modulation indexes of 0.5, 5.0, and 20.0.

The three graphic charts in Fig. 2 illustrate the frequency components (and spectrum occupancy) for modulation indexes of 0.5, 5, and 15. Notice that the carrier strength varies with the modulation index. At a modulation index of 2.405, the carrier disappears entirely. It then becomes "negative" at a higher index, meaning that its phase is reversed compared to the phase without modulation.

In frequency modulation, the energy that goes into the sidebands is taken from the carrier, the total power remaining the same regardless of the modulation index.

Since there is no change in total amplitude with modulation, an FM signal can be amplified without distortion by an ordinary class-C amplifier. The modulation can take place in a very low level stage, and the signal can then be amplified by either frequency multipliers or straight-through amplifiers.

If the modulated signal is passed through one or more frequency multipliers, the modulation index is multiplied by the same factor as the carrier frequency. For example, if modulation is applied at 10 MHz and the final output is on 100 MHz, the total frequency multiplication is 10 times, so if the frequency deviation is 7.5 kHz at 10 MHz it will be 75 kHz at 100 MHz.

If the frequency, modulation index and spectrum occupancy can be multiplied, then surely they can be divided.

It is logical then that if we were to have a signal on 100 MHz modulated to a deviation of 75 kHz, we could pass that signal through two "divide by ten" frequency dividers and produce a new signal on 1 MHz with a deviation of .75 kHz.

FREQUENCY MODULATION VS AMPLITUDE MODULATION

Amplitude modulation (AM) has been replaced by frequency modulation in nearly all communications systems, except AM broadcasting, some aviation channels and the video transmission in television broadcasting.

FM broadcast technology is a superior medium that delivers the entire audible range of sounds, excellent fidelity and almost complete freedom from natural and man-made noise. The most important characteristic of FM broadcast receivers is the ability of the receiver demodulator to "capture" the strongest signal on a channel and ignore other signals. Most FM broadcast receivers have a "capture ratio" of 2.0 or better.

AM broadcast receivers detect and convert to audible sound any signal that varies in amplitude and passes through the selective circuits of the tuner. In a typical AM receiver the received signal is reproduced relatively free of noise and interference only if it is at least 20 times as strong as the interference.

The obvious question is, "Why don't we convert the AM broadcast band to FM?"

The obvious answers are:

(1) AM broadcast channels are only 10 khz wide and broadcast quality FM requires a channel 200 kHz wide.

(2) FM transmissions would not be compatible with existing AM radios.

There may be other objections to an AM to FM conversion, but if we can solve these two major problems, the others can also be overcome.

First, consider the compatibility problem. My suggestion is that we add narrow band frequency modulation to the AM carrier. We know this is practical since Motorola C-QUAM stereo utilizes two modulation modes, AM and PM (Phase Modulation), to transmit the main and stereo information channels. Existing monophonic AM radios still perform as usual.

Next, consider "channel width". FM broadcast signals are 200 kHz wide. 100% modulation is defined as carrier deviation of 75 kHz. "Audio bandwidth" is from 30 Hz to about 100 kHz.

The desirable characteristics of FM are accomplished in the receiver demodulator. In order to achieve these characteristics, the intermediate frequency amplifier (IF) of the FM receiver delivers a signal on 10.7 MHz, with 75kHz deviation to the demodulator

In NFR (Noise Free Radio) we can frequency modulate the carrier of the AM transmitter. This requires replacement of the transmitter's crystal oscillator with a frequency modulated oscillator. Then we can transmit both FM and AM through the same transmitter, either simulcast or separate programming.

In order to meet the severe spectrum occupancy limitations of the AM broadcast band, frequency deviation of the carrier will have to be limited, perhaps to as little as 1 kHz. Then we expand that deviation to 75 kHz by frequency multiplication in the IF amplifier of an NFR receiver.

Early FM transmitters used exciters like the GE Phasitron, or the Gates Serrasoid modulator. They used a low frequency crystal controlled oscillator between 100 and 125 kHz. This oscillator was phase modulated with audio (50 Hz to 15 kHz) to a frequency deviation of only 85 Hz! For example, a crystal oscillator on 115.856 kHz with a deviation
of 85 Hz was multiplied 864 times to result in a signal on 100.1 MHz (FM channel 261) with a frequency deviation of 75 kHz.

We have shown that it is possible to multiply a frequency modulated RF signal, thereby increasing the frequency, the modulation index, and the channel width. The opposite is also true. By frequency division it is possible to lower the frequency and reduce the modulation index and spectrum occupancy.

Stated simply, it is possible to compress a 200 kHz wide FM signal by frequency division to fit into the narrow 10 kHz wide channel of the AM broadcast band, transmit that signal, then expand that signal in a receiver to its original parameters.

NFR ADVANTAGES

NFR can provide most of the advantages of wide-band FM. Full audio fidelity, very low noise, reduction of interference, etc. We can expect night-time coverage almost as good as day-time because sky wave interference is eliminated except when the sky wave signal is almost as strong, or stronger than you signal.

It is even possible to have some advantages over an FM station. The most powerful FM stations have coverage of less than 100 miles. In areas of good ground conductivity I have know 250 watt class IV stations to cover more than that.

One of the problems suffered by FM stations in major cities is "multi-path" or "picket-fence" effect, produced by reflection of the very short wavelength FM signals from buildings, hills, and other structures. The long wavelength signals of the AM band have never been bothered with this phenomenon.

Adding an NFR channel to an AM station provides two program channels. Adding NFR to the AM part of an AM/

FM combo provides three program channels.

LAUNCHING NFR!

I have no intention of applying for a patent on any of the ideas I have contributed. The fact that these ideas have been presented in person and in various publications offered for sale renders such ideas public domain. Please feel free to use any of this information for your profit or pleasure.

I believe the techniques and procedures described herein are common practice, and your application of them in private or commercial use would not be an infringement of patent rights or copyrights of others. However, since I have not verified that belief in every event, I cannot accept responsibility for any liability you may incur by the use or application of such circuits, techniques or procedures.

Having said that, let's turn now to establishing NFR as a working communications medium.

I have prepared a list of the things I feel need to be done to get NFR going. Here they are in chronological order.

1. Organize a committee of capable broadcast engineers working together to prepare technical specifications and operating parameters to be submitted to the FCC as a basis for establishing rules and regulations applying to NFR.

2. Obtain the services of a competent attorney to prepare a petition for rule making, authorizing any or all licensees of AM Broadcast Sations to establish and operate an NFR service in accordance with the rules and regulations established by the FCC. No further application, license or fees would be required.

3. Concurrent with the above, launch an all-out publicity campaign to "sell" NFR to radio station owners, transmitter and receiver manufacturers, and the general public.

4. Present seminars and demonstrations at broadcasters meetings and conventions.

5. Establish an NFR "hotline" where interested people can phone in and hear a three minute recorded message, reporting the latest developments in the progress of NFR. Tape updated at least once a week.

6. In general, solicit cooperation and assistance from anyone and everyone who is capable and willing to contribute to making NFR work.

ACKNOWLEDGEMENTS

The author wishes to express his gratitude for the cooperation and assistance of the many who have helped in the development of NFR. Special recognition must go to Frank L. Barry, Ph. D, Director of Engineering Operations at WQYK (AM&FM); Paul Rebmann, Chief Engineer WEZY/WLKF; Dr. Lance Miller, Consultant; Jim Wood, Inovonics, Inc.; John E. Leonard, Jr., J.N.S. Electronics, Inc. and the hundreds who have phoned and written with suggestions, questions and encouragement. I thank you!

TELEVISION ENGINEERING—SIGNAL DISTRIBUTION AND TRANSMISSION

Sunday, April 14, 1991

Moderator:
John Tollefson, WJLA-TV, Washington, District of Columbia

COMPLYING WITH THE OCTOBER 1, 1991 STL DEADLINE
Craig M. Skarpiak
Andrew Corporation
Orland Park, Illinois

LOOKING 100 MHZ AHEAD
Erik Fabricius-Olsen
PESA Industries, Inc.
Huntsville, Alabama

MULTISTAGE DISTRIBUTION SWITCHING SYSTEMS, CLOS AND BEYOND
Marc S. Walker
BTS, Broadcast Television Systems
Salt Lake City, Utah

A UNIQUE ADAPTATION OF THE TRADITIONAL TELEVISION VECTOR DISPLAY
Mark Everett
Videotek, Inc.
Pottstown, Pennsylvania

THE DEVELOPMENT OF COMMERCIAL ECHO CANCELLERS FOR TELEVISION
Stephen Herman
Philips Laboratories
North American Philips Corporation
Briarcliff Manor, New York

REMOTE MONITORS FOR BROADCAST TRANSMITTERS
Roy K. Chrisop
Harris Allied Broadcast Equipment
Quincy, Illinois

AVERAGE POWER RATINGS OF COAXIAL TRANSMISSION LINES
Anthony N. Schmitz
Dielectric Communications
Raymond, Maine

STRIPLINE TECHNOLOGY—FUNDAMENTALS AND APPLICATIONS IN HIGH-POWER TRANSMITTERS
P.C. Turner
LARCAN Communications Equipment, Inc.
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Steven J. Crowley
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COMPLYING WITH THE OCTOBER 1, 1991 STL DEADLINE

Craig M. Skarpiak
Andrew Corporation
Orland Park, Illinois

Abstract-
Owners of Television Auxiliary Broadcast stations see October 1, 1991 as an important date in their future. The Federal Communications Commission (FCC) has mandated that all microwave antenna systems used for studio transmitter links (STL) be in conformance with the standards as stated in the Code of Federal Regulations (CFR) 47, Part 74.641 on that date. This mandate and its impact on new and existing antenna systems has caused uncertainty and confusion for STL owners.

What are the standards and how do they alter antenna selection?

What is the difference between a Category A and a Category B antenna?

Why does the FCC deem necessary the use of only a Category A antenna in certain geographical areas?

Does my antenna system conform with the October 1, 1991 mandate?

In this paper, we analyze the FCC antenna standards for STL microwave systems in an attempt to eliminate the confusion.

ANTENNA STANDARDS

Table 1 below shows the FCC antenna standards for Television Auxiliary Broadcast (Part 74) STL microwave links. Each operating frequency band has its own standards. The specified electrical characteristics include:

Category A and Category B Classifications

Maximum half-power (3 dB) beamwidth requirements

Minimum gain requirements

Minimum radiation suppression requirements

Each of these parameters affects the size and type of antenna that can be used.

Categories A and B

The category of an antenna defines the level of performance required of an antenna. The level of performance indicates the degree to which it minimizes potential interference into an adjacent system by reducing signals transmitted in unwanted directions.

Category B is the minimum antenna performance level acceptable to the FCC. Category A is a higher performance level for lowering potential interference into adjacent systems. The distinctions between Categories A and B are important from two aspects.

Frequency (MHz)	Category	Maximum beam-width to 3 dB points (included angle in degrees)	Minimum antenna gain (dBi)	Minimum radiation suppression to angle in degrees from centerline of main beam in decibles						
				5° to 10°	10° to 15°	15° to 20°	20° to 30°	30° to 100°	100° to 140°	140° to 180°
1,990 to 2,110	A	5.0	n/a	12	18	22	25	29	33	39
	B	8.0	n/a	5	18	20	20	25	28	36
6,875 to 7,125	A	1.5	n/a	26	29	32	34	38	41	49
	B	2.0	n/a	21	25	29	32	35	39	45
12,700 to 13,250	A	1.0	n/a	23	28	35	39	41	42	50
	B	2.0	n/a	20	25	28	30	32	37	47
17,700 to 19,700	A	n/a	38.0	25	29	33	36	42	55	55
	B	n/a	38.0	20	24	28	32	35	36	36
31,000 to 31,300 [2]	NA	[1] 4.0	38.0							

[1] The minimum front-to-back ratio shall be 38

[2] Mobile, except aeronautical mobile, stations need not comply with these standards.

NOTE Stations must employ an antenna that meets the performance standards for category A, except that in areas not subject to frequency congestion antennas meeting standards for category B may be employed. Note, however, that the Commission may require that use of a high performance antenna where interference problems can be resolved by the use of such antennas.

Copied from CFR47, 1989

ANTENNA STANDARDS

Table 1
FCC CFR47, Part 74.641

First, the FCC allows the use of Category B antennas in regions where frequency congestion is low. However, the FCC may later require that the Category B antenna be replaced with a Category A antenna, if necessary, to resolve an interference problem. Second, in order to maximize the use of the frequency spectrum, the FCC requires the use of Category A antennas in geographical regions having high frequency congestion, such as is the case in most urban areas.

Categories A and B are defined by the FCC using the antenna characteristics of half-power beamwidth, gain and radiation suppression.

Half-Power Beamwidth

The half-power beamwidth of an antenna is the angle included between the two points of the antenna's main beam which are 3 dB below the peak signal level. It is graphically illustrated in Figure 1. Beamwidth is primarily a function of the antenna diameter and operating frequency. As the diameter or frequency increases, the beamwidth decreases.

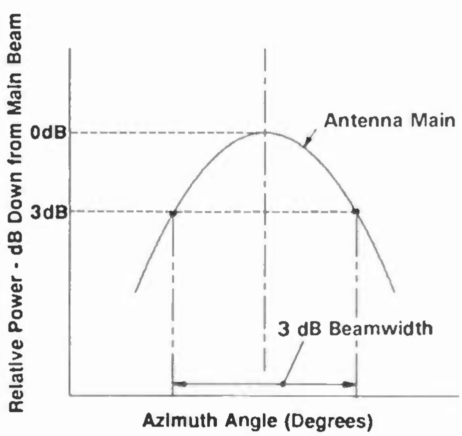

Figure 1
Antenna Half-Power (3 dB) Beamwidth

The FCC specifies maximum half-power beamwidth for both Category A and Category B antennas. For example, the maximum beamwidth for a Category A antenna operating in the 6.875 to 7.125 GHz, Part 74 band is 1.5°, whereas it is 2.0° for a Category B antenna. Refer to Table 1.

Half-power beamwidth is measured and specified by the antenna manufacturer. However, approximate beamwidth can be calculated, based on antenna size and frequency, and we have done so for purposes of the following examples.

The calculated beamwidths are plotted as a function of antenna diameter at 1.990 and 6.875 GHz in Figures 2 and 3.

Figure 2 shows that 4.4 feet is the minimum antenna diameter to satisfy the 2 GHz STL Category B requirement and a minimum 7.1-foot diameter antenna is required to meet Category A. Using commercially available antenna sizes, this means a 6-foot antenna is required for Category B and an 8-foot antenna is required for Category A.

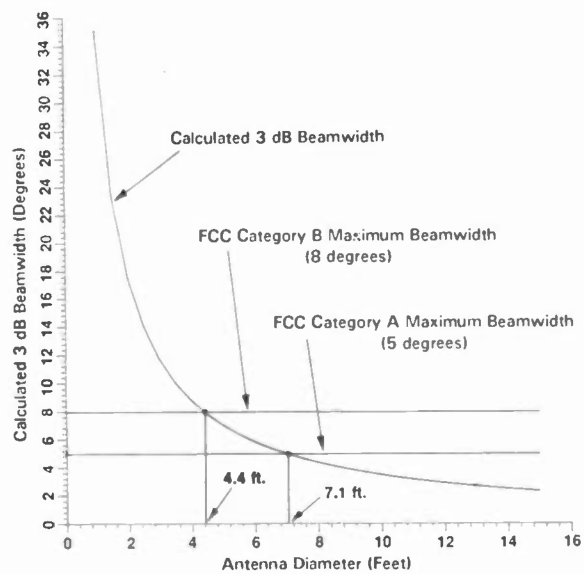

Figure 2
Beamwidth as a Function of Antenna Diameter
1.99 GHz

For the 6.5 GHz STL band, Figure 3 shows that Category B antennas must be at least 5.1 feet in diameter and Category A antennas at least 6.8 feet. Again, using commercially available antenna sizes, this means a 6-foot antenna is required for Category B and an 8-foot antenna is required for Category A.

Antenna Gain

Depending on both the application and the frequency band, the FCC sometimes specifies a minimum antenna gain. The FCC uses the minimum gain specification to control the effective isotropic radiated power (EIRP) from the antenna system.

Antenna gain is not the same as the gain associated with an active device such as a transistor. It is, instead, a passive gain and describes the ability of the antenna to direct its signal into a defined angular region. Gain is also a function of antenna size and operating frequency.

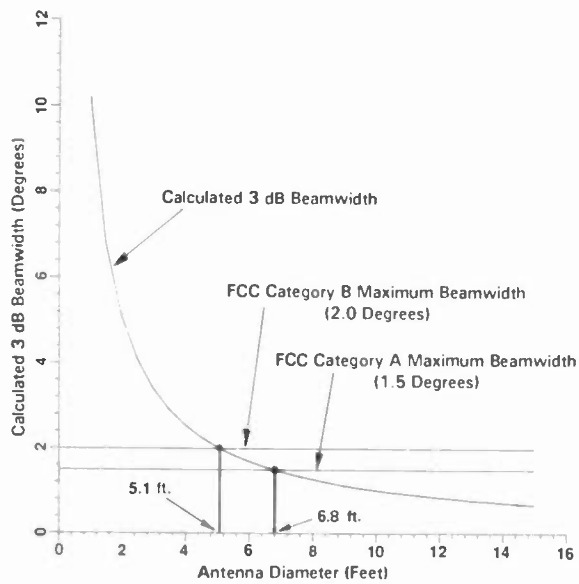

Figure 3
Beamwidth as a Function of Antenna Diameter
6.875 GHz

The FCC normally specifies either the minimum gain _or_ the maximum beamwidth for an antenna. For example, as shown in Table 1, the FCC specifies requirements for maximum beamwidth for 2 and 6.5 GHz bands. No requirements are specified for minimum gain. However, for 18 GHz, they specify only minimum gain. Because gain and half-power beamwidth are both functions of antenna size and operating frequency, they are interrelated. As antenna frequency or diameter increase, the gain increases while the beamwidth decreases. Therefore, if one of the two characteristics is specified, the other is, in effect, also specified.

Figure 4 shows the gain for a 10-foot diameter antenna as a function of frequency at various antenna efficiencies. Terrestrial microwave antennas typically have efficiencies of approximately 55%

Radiation Suppression

The FCC specification for minimum radiation suppression is intended to limit the amount of spurious radiation from the antenna in undesirable angular regions. It is a mask applied to the antenna performance and is specified for angular regions from ±5° through ±180° relative to the antenna boresight direction. The FCC adopted these minimum radiation

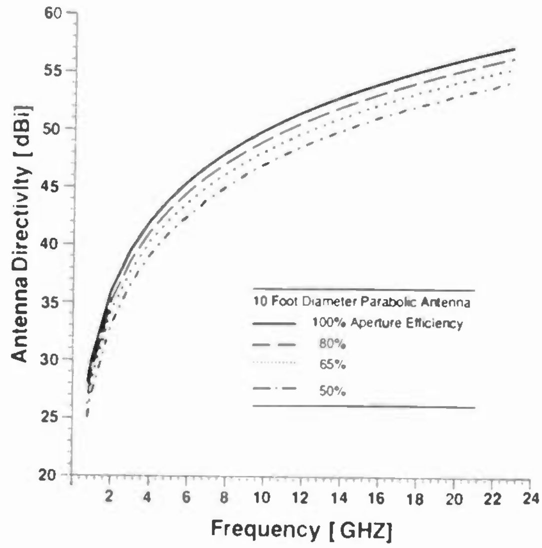

Figure 4
Antenna Gain vs. Frequency
10-Foot Diameter Parabolic Antenna

specifications to minimize potential interference and, thereby maximize the use of the frequency spectrum. Figure 5 shows the Category A and B masks in graphical form. Category A is the more restrictive mask and is, therefore, required for use in areas of high frequency congestion.

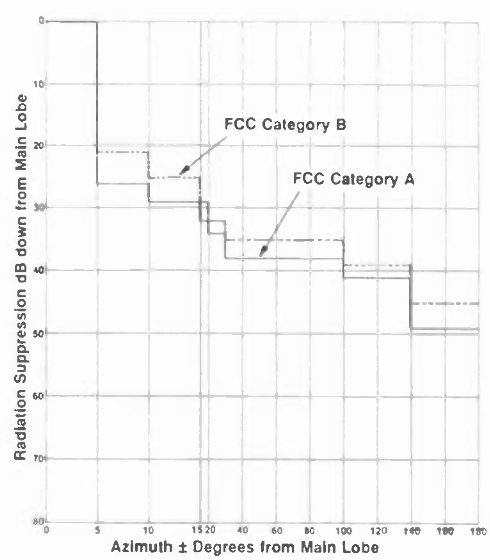

Figure 5
FCC Radiation Suppression Mask
Part 74.641 - 6.5 GHz Band

The FCC radiation suppression specifications are _minimum_ requirements. The burden is on the owner of a proposed system to ensure that the new system does not impair the operation of existing systems. To satisfy this requirement, the new system may need antennas having much better performance than required by the FCC minimum. In today's congested terrestrial microwave environment, this may be the rule rather than the exception.

For coordination of microwave systems, antenna manufacturers supply a radiation suppression document known as the radiation pattern envelope (RPE). The RPE describes the radiation characteristics for an antenna as a function of horizontal angle. Figure 6 shows the RPEs for two different 8-foot antennas operating in the 6.5 GHz STL band along with the FCC mask. While both antennas are better than the FCC Category A minimum, there is a vast difference in the level of performance between the two antennas. The system designer will select the proper antenna based upon calculated adjacent system interference determined from the specifications supplied by the radio and antenna manufacturers.

CONCLUSION

As October 1, 1991 approaches, owners of STL microwave links should review their antenna systems to make sure they conform with the FCC regulations. We hope the above information has made the performance requirements a little more understandable and less confusing. Look to your antenna manufacturer to supply specific antenna information.

As a final statement, when implementing a new or upgrading an existing microwave antenna system, we highly recommend that you specify Category A antennas. Such a system may be initially more expensive, but it offers the best long term protection of your STL communication system.

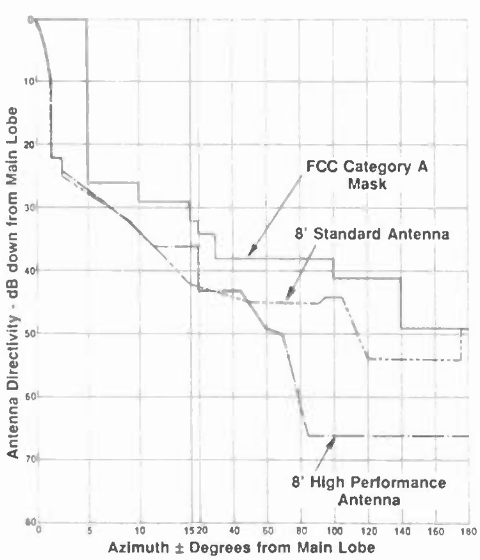

Figure 6
8-foot, 6.5 GHz STL antenna RPEs
Compared to FCC Category A

LOOKING 100 MHZ AHEAD

Erik Fabricius-Olsen
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Huntsville, Alabama

ABSTRACT- This paper will explore the bandwidth needed in future video routing switchers and address the design trade-offs as encountered during design and pilot production.

INTRODUCTION

In recent years advancements in electronic technology have brought many new products that we did not even dream about before. Broadcast TV has been around for over 40 years but in the last decade we have seen the growth and proliferation of more and more advanced graphic devices that provide much higher resolution than has been around before. In TV we are now seeing the introduction of HDTV that requires 30 MHz bandwidth. In high resolution graphics the bandwidth requirements continue to increase as we use larger and larger screens yet we maintain or improve the per inch resolution. Demands for higher bandwidth will continue to increase in areas such as medical electronics with MRI, CAT Scan, and X-Ray equipment, or in government electronics for Radar video or high resolution maps.

As serial digital video grows we will see more and more users starting into this technology. During the time where the industry is changing from analog to digital there will be a need to be able to expand routing switcher capabilities to handle digital signals at 140 to 170 MBits per second for composite video or up to 270 MBits per second for component. If the analog routing switcher could offer the bandwidth capable of switching serial digital video it would become a very attractive investment for those users that want to grow into digital over time, so the need exists today for a switcher even beyond 100MHz.

The following is a summary of the lessons learned at PESA (some were relearned) when we designed our 100MHz 48x96 Video Routing Switcher.

BARRIERS TO HIGH BANDWIDTH

Capacitive loading, series inductance, and resistive losses are the primary limitations to achieving good high bandwidth frequency

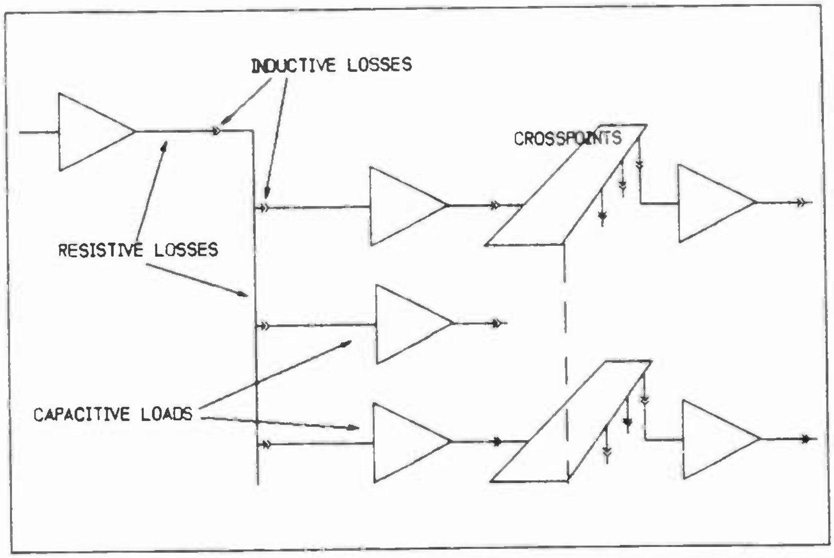

FIGURE 1 ..Video Bus Bandwidth Constraints

response (Figure 1). In video routing switchers it is necessary to have video busses that tie crosspoints together both on inputs as well as outputs. The physical size of the components used is the major driving force behind the length of the video busses. BNC connectors are standard and are now becoming the limiting component to miniaturization. Only so many BNC connectors can be placed in one rack unit. The problems with long video busses are many. Stray capacitance grows with longer busses; resistance and inductance increase with longer busses; long busses cause reflections. This forces the design engineer to make the traces as wide as possible but wide busses increase stray capacitance. When traces are wide the PCB packing density is jeopardized. When stray capacitance is high the amplifier driving the trace must use higher drive currents thus dissipating more power and increasing distortion. When power increases components cannot be packed as tightly. Therefore, we try to limit power consumption only to active paths (Figure 6). This allows tighter packaging and hence improved frequency response. Resistive losses lead to short and wide busses. Inductive losses come from connectors so they must be carefully selected and remain as few as possible. Capacitive loads lead to short busses and small devices. In order for bandwidth variance to be minimized busses should be short and low impedance.

To maintain controlled and consistent impedance along a trace with up to 12 taps on each line is difficult at best. Not only is it necessary to select components that don't load the bus but the circuit configurations (design of amplifiers etc...) must be such that variations are minimized. To maintain constant impedance

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es.

As we continued to improve the performance of the switcher we ran into the common problem of parasitic oscillation because of trace inductance when we changed to a higher performance transistor (Figure 4). It was easily fixed with a series resistor.

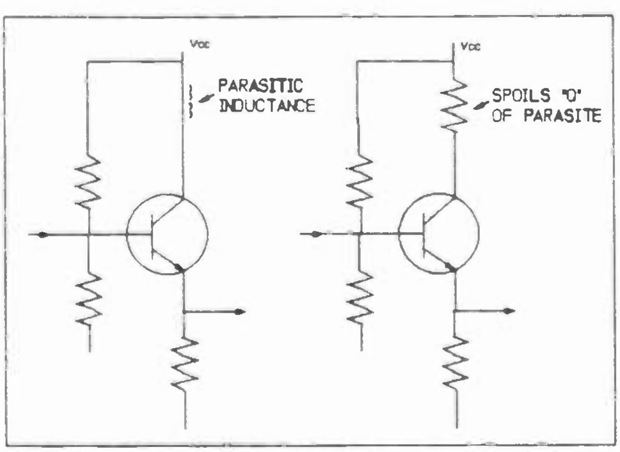

FIGURE 4 ..Elimination of parasitic oscillation

Connectors play an important role in the design of a routing switcher. They can be a source of crosstalk because of close proximity of other video signals even when every other pin is grounded. In many cases the connectors are the limiting factor to packing density so for that reason alone connectors should be avoided where ever possible. Connectors are also a source of signal losses thus reducing bandwidth. However, to keep the product flexible in size and cost it is necessary to cope with connectors in the design. The challenge becomes to minimize the number of connectors in series with the signal path yet still maintain a broadly competitive product.

Developing a 100 MHz switcher is not just a matter of getting the frequency response to look good. Other parameters are just as important if not more so.

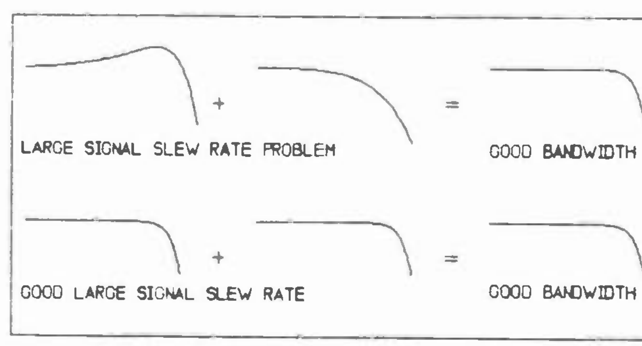

FIGURE 5 ..Large signal slew rate

Slew rate limiting must be carefully watched since the switcher can have excellent frequency response but if some stages have to make up for losses in other stages, it is not capable of this response at full amplitude (Figure 5). Therefore it is best to design each stage in the switcher to have a flat response over the full frequency range at full amplitude.

Crosstalk is a parameter that often conflicts with high bandwidth. Conducted crosstalk can be prevented with good power and ground distribution (Figure 2 and 3). However, the more we miniaturize the design the greater is the opportunity that capacitively coupled crosstalk will develop. To get good crosstalk performance we would like to place video

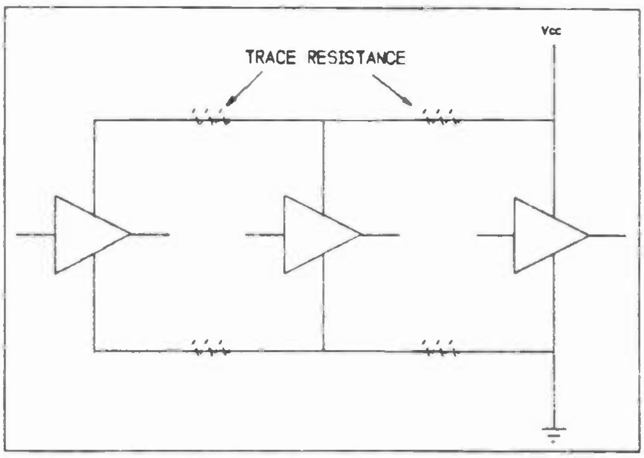

FIGURE 2 ..Power and Gnd losses cause crosstalk

busses far from each other. If that is done we will have long busses with more stray capacitance thus reducing bandwidth. To optimize the trade off between crosstalk and bandwidth requires careful PCB layout with appropriate grounding between signal paths to isolate the signals from each other with minimal capacitive loading on the signal. Capacitive loading reduces slew rate so shielding with ground layers has very limited application.

Consistent delay t
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Little stray capacitance
few active elements in se
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low and consistent.

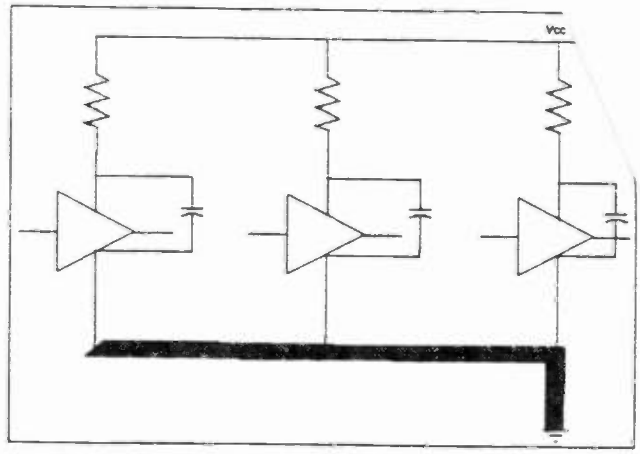

FIGURE 3 ..Improved crosstalk performance

Differential phase and gain must be watched carefully. High bandwidth demands high power drivers that because of thermal reasons should operate with as low a supply voltage as possible. Low power supply voltages, if taken too low, could be the source of distortion and lower dynamic range. High current also causes nonlinearities so current requirements must be minimized.

GROUNDING

Because of the stray capacitances and the loads of the system, large high frequency currents flow through the signal traces. The return paths for these currents are through the grounds. Not even a ground plane is zero impedance so care must be taken in routing the grounds (Figure 2 and 3). If not, serious crosstalk can develop. The same problems exist on the power plane. Of course power and ground planes can be designed into the PCB but because of feedthrough holes and cut-outs in the power and ground planes it is very difficult to maintain high integrity power and ground planes internal to the circuit board without loading the video signal traces.

THERMAL MANAGEMENT

In high frequency switching stray capacitance and high slew rates require high operating currents. Thus careful thermal planning is necessary. First, we try to eliminate all use of power when circuit paths are not used (Figure 6); this is accomplished in the electrical circuit design and component selection. In a 96 output video switcher only 96 crosspoints should be drawing current. Second, we keep the number of active devices in the signal path to a minimum. By carefully organizing the placement of components we avoid hot spots. By using SMT technology throughout the product we have been able to keep the capacitances low because the busses are short, consequently less drive is required and less power is consumed.

Some heat is always generated in the switcher and in high frequency switchers even more so. This heat must be removed as quickly as possible. Cool operating temperatures usually mean higher reliability. We keep temperatures low by careful airflow planning and use of specially designed heat sinks that we build into our PCB shield and support structure. To further enhance the thermal management in the routing matrix we monitor the ambient frame temperature and control the exhaust fan based on the temperature. The result is less temperature variation in the frame and thus more consistent performance of the product. Longer fan life, less noise, and lower filter maintenance are other benefits.

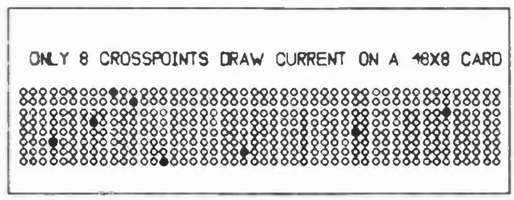

FIGURE 6 ...Low heat dissipation

MODULAR ARCHITECTURE

The demands of the marketplace require that the routing switcher architecture be modular. We have balanced cost, modularity, and technical performance in a 48 input by 96 output video frame. The building blocks are 12 quad Video Input Amps (VIA), 12 crosspoint cards with 48 inputs and 8 outputs (48x8), 72 crosspoint cards with 8 inputs and 8 outputs (8x8), 96 video output cards (VOA), a vertical trigger card, and dual power supplies. No single point of failure will shut the whole switcher down. Cards can be plugged "hot". Expansion slots for option cards are built in for features such as clamping, independent vertical interval switching, and other "special" requirements.

MANUFACTURING

For a product to be competitively priced and have the best specifications current technology permits, particular attention to manufacturing and test issues is needed in the design. Cumbersome hand labor should be avoided like time consuming hand wiring or similar operations.

Use of SMT components requires investments in the order of $ 1,000,000 for a "state of the art" SMT production line; we have selected in stead a contract manufacturer that has world class expertise in electronics manufacturing. The Huntsville area is fortunate to have several of the best contract manufacturers in the world such as SCI, AVEX, Comptronix, and Teledyne Lewisburg. Besides electronic manufacturing these companies offer services in test and environmental stress screening that assures high quality and high reliability at costs that are very attractive.

SUMMARY

Surface Mount Technology is what makes high bandwidth video switchers possible at reasonable costs but the miniaturization has its own set of problems to be solved before a product becomes a reality. The concepts are not new but the smaller architecture sends us back to basics and forces us to rethink our design methodology.

MULTISTAGE DISTRIBUTION SWITCHING SYSTEMS, CLOS AND BEYOND

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ABSTRACT- The average size, and therefore cost, of distribution switchers in television facilities is growing larger each year. In 1953, Charles Clos published a landmark technical paper concerning non-blocking multistage switching techniques, explaining how to reduce the number of crosspoints when compared to a standard switcher. Multistage switching systems have been proposed and used by several companies in the television industry. This paper shows why blocking can occur in television systems, even though Clos type switcher systems are normally considered to be strictly non blocking. It explains the causes of blockage, shows statistical results demonstrating the amount of blockage that may occur, and discusses how these blockage problems can be solved.

INTRODUCTION

The huge number of distribution switchers in use in the world today attests to their value in the television industry. Years ago the average size of distribution switchers was small, in the 20 input by 20 output range. As the years have passed, the average switcher size has increased, and the number of large distribution switchers being built, those larger than 100 inputs by 100 outputs, has grown. The number of crosspoints, and therefore the switcher cost, increase rapidly with switcher size, making it desirable to reduce the number of crosspoints. Multistage switchers are one way reduce the crosspoint count and cost, but blocking can occur if steps are not taken to prevent it. Multistage switchers can do an excellent job when care is taken in their use and blocking is prevented.

SINGLE STAGE SWITCHING

Most distribution switchers use single stage switching, (see Figure 1) where a crosspoint is provided for every possible input output combination. The number of crosspoints in the system is the product of the inputs and outputs. These switchers are "strictly nonblocking," meaning it is always possible to make a connection between any input and any output at any time, regardless of the conditions of the other connections. An interesting characteristic of the single stage switching structure is that multiple crosspoints can be turned on at the same time and mix the selected signals together. Mixing is not usually used in television system switchers, so the crosspoints and control electronics are not usually designed for it.

Single stage crosspoint count and cost increases rapidly with switcher size. To double the number of inputs and outputs of a switcher, the number of crosspoints must be increased to four times the original. For example, a 50 input by 50 output switcher requires 2,500 crosspoints, and a 100 input by 100 output switcher requires 10,000 crosspoints.

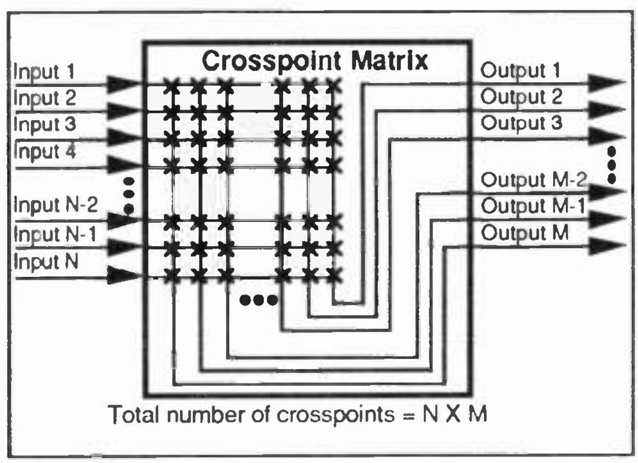


Figure 1. Single Stage Switcher

The design of single stage matrices usually limits the maximum practical size of a switcher. There is a maximum number of outputs that can be driven from an input signal before distribution amplifiers are required. The DAs increase the size and cost of the large switcher system. Inputs are usually limited by what can be switched in one rack frame of matrix boards. Beyond that size, combining switchers are needed, which is a type of multistage switching costing additional crosspoints.

Since DAs and modern switchers usually have equivalent performance, many large switchers are already have three stages, without saving crosspoints. (see Figure 3)

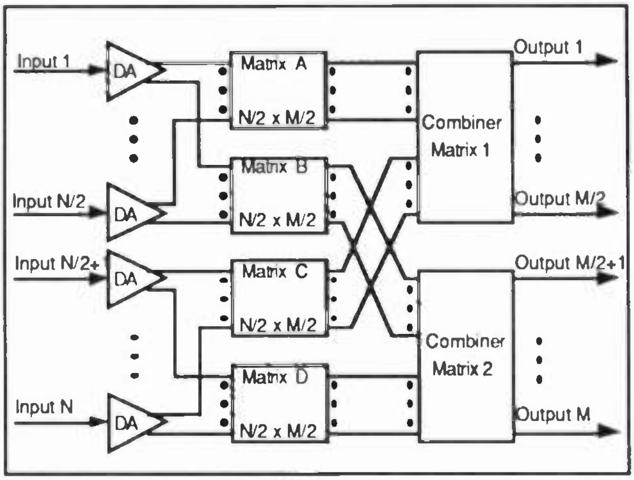


Figure 2. System With Input DAs and Output Combiners

CLOS MULTISTAGE SWITCHING

In 1953, Charles Clos published a technical paper describing a method to reduce the crosspoint count through multistage switching techniques[1]. Since that time, others have proven additional savings are possible. With the Clos technique, it is possible to reduce the crosspoint count of a 100 input switcher by 100 output switcher from the standard 10,000 crosspoints to only 5,700 crosspoints, and larger switchers have even greater savings. A multistage switcher is constructed out of smaller matrices, that are interconnected to form the large system with an an odd number of switching stages. (see Figure 3) This discussion in this paper will be limited to three stage systems, because they are theoretically best for input and output sizes ranging from 36 to over 200, and five stages are not significantly better until over 1000 inputs and outputs. Interconnection cables, and the need for additional output driver cards in video switchers, make the brake even point in cost at a size of about 100 inputs and outputs. The crossover point in audio systems can be lower.

The signal inputs of the switcher system are divided up among a series of input matrices. The outputs of the input matrices are fed to inputs of the middle stage matrices. The middle stage matrix outputs are fed to the inputs of the output matrices. Each input matrix has one and only one connection to each middle matrix, and each middle matrix has one and only one connection to each output matrix. Each output matrix provides a portion of the total system outputs.

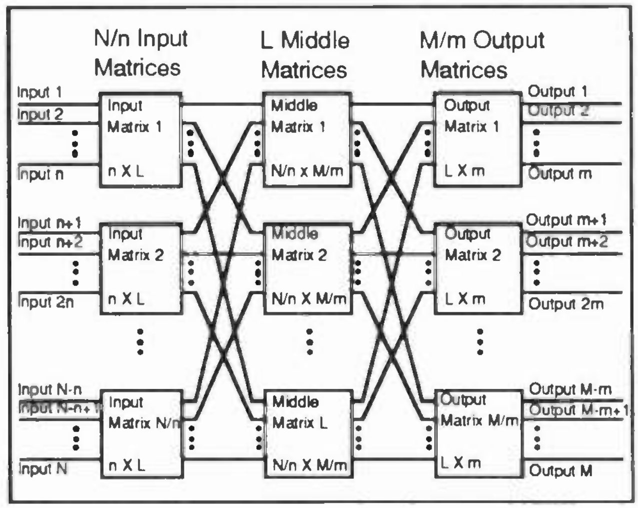


Figure 3. Generalized Three Stage Switcher System

The number of inputs per input matrix, "n," and the number of outputs per out matrix, "m," are usually near the square root of their respective system input and output counts, "N and M,". The number of middle matrices required by Clos is equal to one less than the number of inputs per input matrix (n), plus the number of outputs per output matrix (m). The proof given by Clos is relatively simple. If a path is needed between a given input switcher and a given output switcher, then a maximum of n-1 inputs from the input switcher and a maximum of m-1 outputs of the output switcher will already be in use. Thus a maximum of n+m-2 middle matrices will already be in use. One more path is needed for the new connection, giving a maximum total of n+m-1 middle matrices necessary. These matrix sizes and

System Size	Input Matrices		Middle Matrices		Output Matrices		Board & Crosspoint Counts					
							3 Stage			Single Stage		
	Size	#	Size	Count	Size	#	Mat	Out	Xpts	Mat	Out	Xpts
100x100	10x20	10	10x10	(19) 20	20x10	10	60	50	6000	100	10	10,000
200x200	10x20	20	20x20	(19) 20	20x10	20	160	100	16000	400	20	40,000
300x300	10x20	30	30x30	(19) 20	30x10	30	300	150	30000	900	30	90,000
400x400	20x40	20	20x20	(39) 40	40x20	20	480	200	48000	1600	40	160,000
400x400	10x20	40	40x40	(19) 20	20x10	40	480	200	48000	1600	40	160,000

Table 1. Comparison of 3 stage and single stage switchers

counts are often adjusted to fit standard matrix board sizes, but there must be a sufficient number of middle matrices in order for the system to function.

For the example in Figure 3, with 10 inputs per input matrix and 10 outputs per output matrix, a total of 19 middle matrices is required, with each having 10 inputs and outputs. The total number of crosspoints is 5,700 as predicted by Clos. If we use a standard size matrix board with 10 inputs and 10 outputs, the number of crosspoints increases to 5,900, but 200 of these crosspoints are not used. The total number of circuit boards in this three stage switcher is about the same as a single stage switcher, because additional output boards are needed, but the savings really start to mount up as the switcher system increases beyond 100 inputs by 100 outputs. Table 1 lists the number of crosspoints for various switcher sizes. The circuit board counts are based upon a matrix board size of 10 inputs by 10 outputs, and an output amplifier board with 10 outputs. The middle matrix counts in parenthesis are the Clos numbers, with my recommendations following them. The crosspoint counts are my recommendations. This table shows two ways to build a 400 x 400 switcher. In this case, both resulted in the same circuit board count, but usually one method is better than others.

Finding a signal path through a Clos switcher is a relatively simple process. You determine which input and output matrices need to be connected, and look for a middle matrix with paths open to both of them.. With the conditions Clos described, there will always be at least one middle matrix that satisfies this condition, and therefore a way to make the connection. In fact, others have proven the same capabilities can be achieved with less crosspoints than Clos proposed[2].

Middle matrix usage history for 1 Million takes to a 100 x 100 switcher

Matrix Number or Count	Number of Takes made in this middle matrix	Number of Takes while this matrix was in use	Histogram for the Number of middle matrices in use at one time
1	82405	1000000	0
2	81913	1000000	0
3	80913	1000000	0
4	80026	1000000	0
5	78925	1000000	0
6	77453	1000000	0
7	75920	1000000	0
8	73850	1000000	0
9	71511	1000000	0
10	68558	1000000	0
11	64501	1000000	2
12	58812	999993	4
13	50925	999907	809
14	36874	995164	150933
15	15549	829662	699516
16	1845	169933	147385
17	20	2857	1351
18	0	0	0
19	0	0	0
20	0	0	0

Table 2. Switcher Blocking Tests for Type Point to Point Switching

Table 2 shows the number of middle matrices in use at a time for a run of 1 million random point to point selections on a 100 x 100 switcher. The histogram column shows a minimum of 11 middle stages were in use at one time, with 15 matrices in use 70% of the time, and the maximum of 17 in use for 0.135% of the time. This maximum of 17 middle switchers used is 2 less than the upper limit predicted by Clos. All of this sounds great, we've saved up to 50% or more in crosspoint count and eliminated input distribution amplifiers if they had been needed. This has only cost three passes through the crosspoints and a little complication to the crosspoint control system.

DIFFICULTIES WITH CLOS MULTISTAGE SWITCHING

The work done by Clos and most other researchers has been for telephone systems operating in a Point to Point mode, meaning that each output is connected to only one input at a time, and each input is connected to only one output at a time. The first condition is normal for television distribution switchers, but the second is not.

Television systems usually require all inputs to be capable of feeding any combination of outputs, from none, to one, to several, to many, or even all outputs. This is termed as "Broadcast" mode switching (This has no connection with transmitters or broadcast stations.). At first this sounds easier, because there will usually be fewer different signals to pass through the middle matrix, since many of the inputs may not be in use at a given time. As illustrated in Figure 4, sometimes it is possible to get copies of the input signal at the output matrix, or the

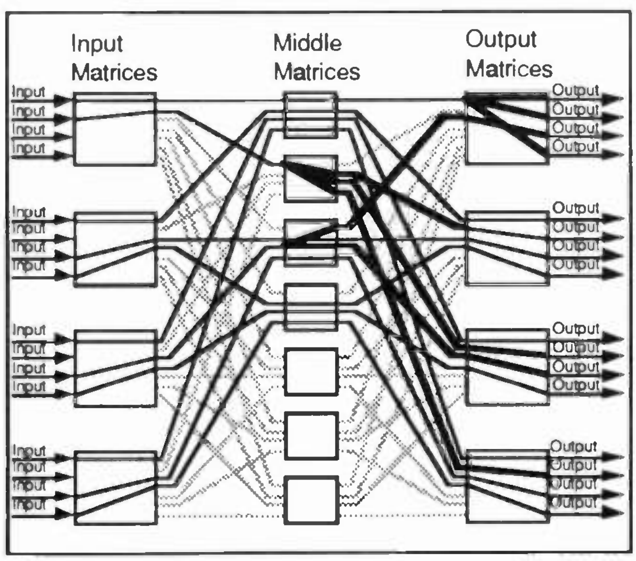


Figure 4. Illustration of Signal Copying in the Middle and Output Matrices

middle matrix, without going back to the input matrix every time the same input is used.

Table 3 shows the middle level usage for a 100 x 100 switcher that was initially set up with each output being connected to its respective input. Then 1 million random takes were made to this switcher by randomly selecting an output number and randomly selecting an input to switch to that output. The maximum number of middle switchers in use at any time was 19, which fits well with the Clos concepts. The 19th middle switcher was used to complete only 6 takes out of the 1 million takes made. Over 600,000 of the takes were made using 15 or fewer matrices at one time, but more middle matricies were needed to complete all of the paths. This appears to be satisfactory at first glance, but television plants do not have random switching.

Middle matrix usage history for 1 Million takes to a 100 x 100 switcher

Matrix Number or Count	Number of Takes made in this middle matrix	Number of Takes while this matrix was in use	Histogram for the Number of middle matrices in use at one time
1	84819	1000000	0
2	83342	1000000	0
3	82492	1000000	0
4	80934	1000000	0
5	79288	1000000	0
6	77840	1000000	0
7	75681	1000000	0
8	73383	1000000	0
9	70064	999992	0
10	66189	999996	0
11	61803	999964	1
12	54917	999560	2
13	46195	997059	3783
14	34112	972791	118584
15	19838	822038	492959
16	7433	417167	331002
17	1461	95475	48982
18	203	12446	4634
19	6	432	53
20	0	0	0

Table 3. Switcher Blocking Tests for Random Input and Output Switching

Input 1 loaded by 4 outputs,
inputs 2-11 loaded by 3 outputs,
inputs 12-21 loaded by 2 outputs,
and all other inputs loaded 0 or 1 outputs.

Middle matrix usage history for 1 Million takes to a 100 x 100 switcher

Matrix Number or Count	Number of Takes made in this middle matrix	Number of Takes while this matrix was in use	Histogram for the Number of middle matrices in use at one time
1	77089	1000000	0
2	74679	1000000	0
3	71980	1000000	0
4	68658	1000000	0
5	66326	1000000	0
6	63335	1000000	0
7	60327	999999	0
8	57235	1000000	0
9	53917	999947	0
10	50291	999733	0
11	45910	998645	0
12	40634	995841	0
13	35261	983746	0
14	31313	964674	0
15	27061	958130	0
16	27272	985890	0
17	27658	988191	9
18	32674	990995	446
19	24620	995402	9364
20	23533	980429	111838
21	29345	975420	565600
22	10882	363761	312743
23	0	0	0

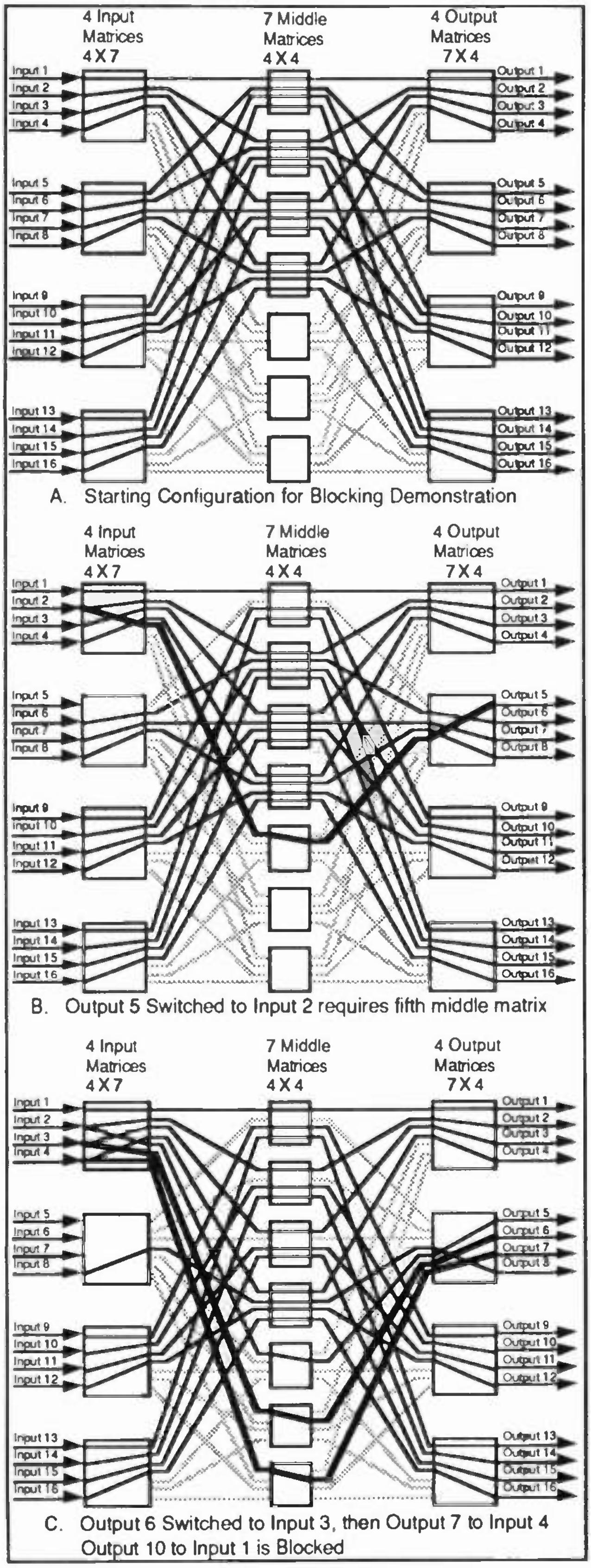
Table 4. Switcher Blocking Tests for Random Switching With Heavily Loaded Inputs

Table 4 shows a test more typical of what happens in a television system, by loading certain inputs heavily, while other inputs were used only once, and other inputs were not used at all. In this case, 31 switcher outputs were always using inputs from the first input matrix., and 21 outputs were always using inputs from the second input matrix, resulting in 52 switcher outputs concentrated on the first 20 inputs. This loading more closely models the usage of inputs in a television facility. To maintain the input loading when making takes, two outputs were randomly selected, and the inputs selected to these outputs were then traded. 1 million random takes were made by doing 500 thousand trades. Table 4 shows that 22 middle matrices were needed to prevent blocking, which is three more than the Clos limit. With standard size matrix blocks of 10 x 10, adding these matrices requires an additional matrix and output card for each input switcher, and an additional matrix board for each output switcher. This brings the cost up to more than a single stage switcher. Why didn't broadcast switching work when it looked easier than point to point switching?

Sometimes it is not possible to reach the desired signal at the middle matrix level, even though it is passing through one or more middle matrices, because the path between the output matrix and the desired middle matrix or matrices is already in use. Then it is necessary to go back to the input matrix to get the signal. As this process continues, all of the outputs of the input matrix may get used up, with more selections pending. This is illustrated by the sequence in Figure 5.

Although this example shows a 16 x 16 switcher for simplicity and clarity, and uses more crosspoints than a single stage matrix, the blocking process is the same as for larger switchers. Figure 5A starts by switching each output to its respective input. This requires only 4 of the 7 available middle matrices.

In Figure 5B, output 5 is switched to input 2, rather than its previous selection of input 5. Note that output 5 is not able to get input 2 within its own output matrix, and is not able to reach the middle matrix already passing input 2, because the path to that middle matrix is already feeding output 6. Thus it is necessary to use the fifth



A. Starting Configuration for Blocking Demonstration

B. Output 5 Switched to Input 2 requires fifth middle matrix

C. Output 6 Switched to Input 3, then Output 7 to Input 4 Output 10 to Input 1 is Blocked

Figure 5 16 x 16 Blocking Demonstration.

middle matrix. In Figure 5C, similar switches are performed for output 6 to input 3 and output 7 to input 4. This has now used up all of the middle matrices. If an attempt is made to select input 1 to certain outputs, such as output 10, a path will not be available and the switcher is therefore blocked!

This blockage always occurs between an input matrix having many heavily used inputs, and the middle matrices. The input matrix is being used as if it had many more inputs than it actually has. The basic procedure to cause blocking is to force the system to use new middle matrices, by careful selection of switching combinations in the proper sequence. It is possible to prevent this blocking by adding more middle matrices and the necessary input and output connections, but the savings in crosspoints is soon lost. I have used these techniques to block a 100 x 100 switcher with 37 middle matrices, having 1100 crosspoints more than a single stage switcher!

Although the worst case condition for blocking 37 middle matrices is possible, it is very unlikely to occur naturally. Blocking can be minimized or even eliminated by careful selection of the switcher input assignments. Spreading out the heavily used inputs, such as black, color bars, program etc, over different input matrices will greatly relieve the problem. To illustrate this, the test shown in Table 5 was done by spreading out the heavily used inputs. Input 1 was used by 4 outputs. Each input

Input 1 loaded by 4 outputs,
inputs 11 21 ... 91 loaded by 3 outputs,
inputs 12 22 ... 92 loaded by 2 outputs,
and all other inputs by loaded 0 or 1 outputs.

Middle matrix usage history for 1 Million takes to a 100 x 100 switcher

Matrix Number or Count	Number of Takes made in this middle matrix	Number of Takes while this matrix was in use	Histogram for the Number of middle matrices in use at one time
1	84078	1000000	0
2	83409	1000000	0
3	82102	1000000	0
4	80838	1000000	0
5	78830	1000000	0
6	77371	1000000	0
7	75116	1000000	0
8	72398	999998	0
9	68322	1000000	0
10	63928	999996	0
11	57373	999767	0
12	48166	998087	0
13	36970	981259	646
14	26450	902608	31702
15	18791	729383	233102
16	29609	763301	512751
17	16249	548956	221799
18	0	0	0
19	0	0	0
20	0	0	0

Table 5. Switcher Blocking Tests for Random Switching with Heavily Loaded Inputs

matrix then had one input used by three outputs, and one input used by two outputs. All other inputs were used by one or no outputs. This is the same input loading as the Table 4 test, but the heavily used inputs are spread out. In this test the maximum number of middle matrices needed was reduced to 17, from the 22 required in the previous test.

Table 6 shows the results when the input loading was increased by one output on each of the heavily used inputs. Although the number of middle matrices required is the same, the histogram peak was moved to a lower level, indicating more middle and output stage signal copying. Unfortunately you can't guarantee that all of the

Input 1 loaded by 5 outputs,
inputs 11 21 ... 91 loaded by 4 outputs,
inputs 12 22 ... 92 loaded by 3 outputs,
and all other inputs by loaded 0 or 1 outputs.
This test is similar to the last table, but the output loading has been increased by one on each of the above named inputs.

Middle matrix usage history for 1 Million takes to a 100 x 100 switcher

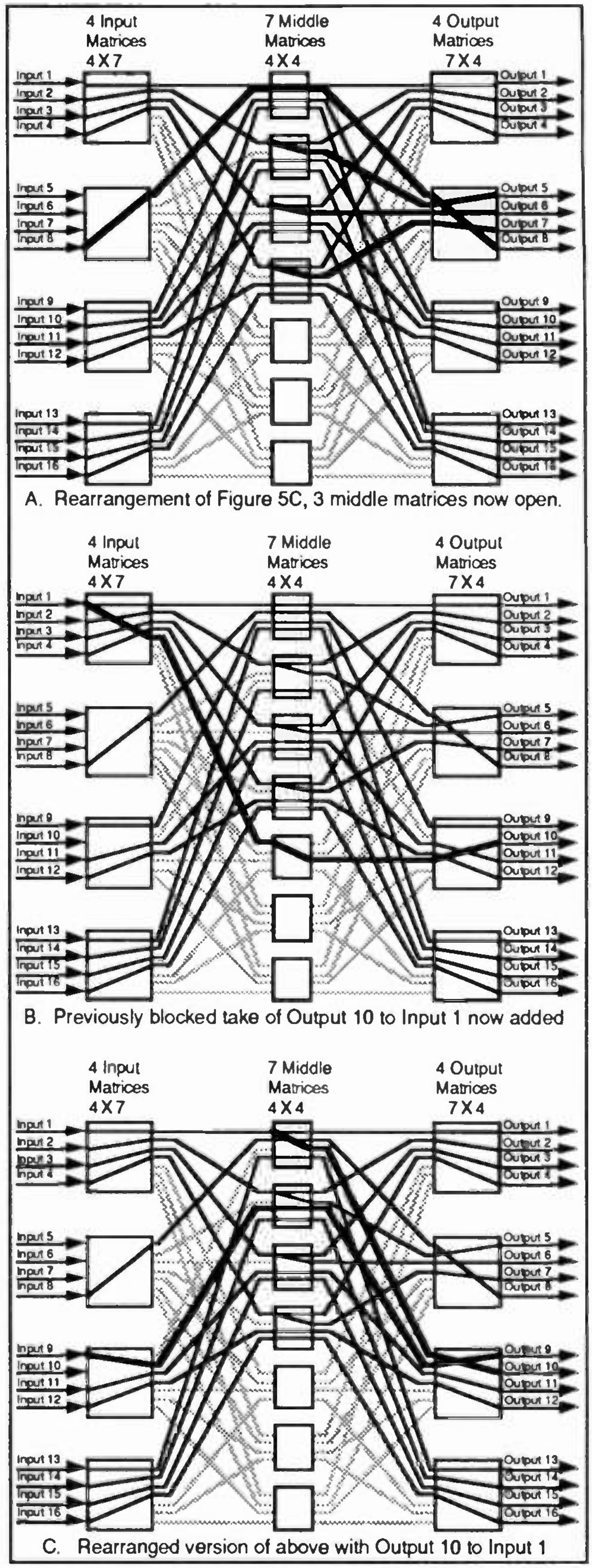
Matrix Number or Count	Number of Takes made in this middle matrix	Number of Takes while this matrix was in use	Histogram for the Number of middle matrices in use at one time
1	89195	1000000	0
2	88583	1000000	0
3	87243	1000000	0
4	84722	1000000	0
5	83249	1000000	0
6	81433	1000000	0
7	79160	999998	0
8	75401	1000000	0
9	70975	1000000	0
10	65130	999972	0
11	56780	998865	0
12	46210	991717	102
13	31466	908940	27844
14	23528	700138	400330
15	36682	938020	554117
16	237	23459	17456
17	6	325	151
18	0	0	0
19	0	0	0
20	0	0	0

Table 6. Switcher Blocking Tests for Random Switching with More Heavy Loading on the Inputs

inputs to a given matrix will not all become heavily used at some time. Thus the problem is not entirely solved yet. A safety net is needed for those special occasions.

REARRANGEABLE SWITCHING

If we examine Figure 5C further, it is now possible to switch output 5 to input 2 by copying the signal in the second middle matrix. This path was opened up when output 6 was switched to input 3. In a similar way, output 6 can now be moved to the third middle matrix. Output 8 can reach input 8 through the first middle matrix with the interconnections originally used for output 5 to input 5. Finally, output 7 can now reach input 4 in the forth middle matrix. This results in the switching pattern



A. Rearrangement of Figure 5C, 3 middle matrices now open.

B. Previously blocked take of Output 10 to Input 1 now added

C. Rearranged version of above with Output 10 to Input 1

Figure 6. Rearranging Demonstration

of Figure 6A, illustrating that even though the switcher had been blocked to new inputs, we can free up 3 of the middle matrices by moving signals to different paths. Adding the connection from output 10 to input 1 will now use the fifth middle switcher, as shown in Figure 6B. If the connection of output 9 to input 9 is moved to the second middle matrix, it is possible to connect output 10 to input 1 in the first matrix, as shown in Figure 6C. Thus it is not just the multiple usage of signals that causes blocking, but also the sequence of switching and using those inputs.

Rearrangeable non-blocking switching networks have been suggested as a way to save more crosspoints in switching systems. These switching systems rely upon the fact that while there may not be a path available, one can be opened up by moving some of the other signals to different paths. This can be done without any loss or disconnection of existing signals, by setting up each new path, and switching to it before removing the old path. The only switch noticeable at the output will be when the output matrix changes from the old to new path for the same signal. (see Figure 7) Thus, any switching glitches or transients caused by rearranging will occur only once per output, not three times. If the switcher provides clean switching with well matched paths, there will not be any noticeable disturbance. This technique can work especially well when switching digital signals, as long as bits are not lost or destroyed beyond recovery by a switching glitch..

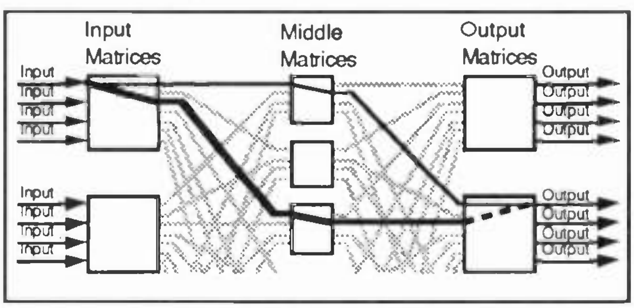


Figure 7. The new path is set up in the input and middle matricies before the output matrix switches

It has been shown that a point to point switcher which is rearrangeably non-blocking can be constructed with n or more middle matrices. This means a 100 X 100 rearrangeably nonblocking point to point switcher will require 10 middle matrices, for a total of 3000 crosspoints. Up to n-1 outputs, 9 for this example, may require rearranging to open the necessary path in this switcher[3].

Since broadcast switching requires more middle matrices, it should be expected that a 100 x 100 rearrangeable broadcast switcher will need more than 10 middle matrices. The number of middle matrices required will depend somewhat upon the amount of rearranging that will be permitted. For a minimum amount of rearranging, my tests show m+n middle matrices are adequate to guarantee a path can be made in a broadcast rearrangeably non-blocking switcher. This means the 100 X 100 broadcast switcher will need 20 middle matrices, for a system total of 6,000 crosspoints. This is less than 5% over the Clos figures, and still 40% less than the standard single stage switcher. Percentage crosspoint savings will be greater in bigger switching systems. The additional middle matrix also reduces the probability that blocking will occur, and therefore the number of times rearranging is necessary. Although the number of middle matrices could be decreased further, it would require more extensive and more frequent rearranging of the signals. I believe n+m is a good compromise for the number of middle matrices.

It is also possible to lock certain outputs of the switcher system to prevent them from being rearranged. This may be desirable with high priority feeds, such as the on air program and its component signals. As the number of locked outputs increases, the complexity of rearranging will increase.

At BTS, we have written software to do rearranging of paths when necessary. We have not been able to block this software in 100 X 100 systems with 20 middle matrices. These algorithms are in use in a number of three stage switchers that BTS has delivered to a variety of customers. Rearranging is seldom if ever necessary, and we have not found any blockages to occur in these systems. For the data patterns described above where 20 or fewer middle matrices are adequate, rearranging will not take place, because sufficient paths are available.

Table 7 shows the results of a test of rearranging using the input loading of Table 3, with 20 middle matrices, resulting in 139 rearrangements necessary out of 1 million take requests. This is about 0.014% of the switching requests. Table 8 shows test results for a more severe output loading. It is interesting that the histogram has moved toward a lower number of middle matrices in use, but the number of rearrangements increased to 2780, or 0.28% of the switching requests.

With proper attention paid to distributing heavily used inputs across all of the input matrices, rearrangements will seldom if ever be required, as shown by Tables 2, 3, 5, and 6. If blocking still occurs, then one or more signal paths may be moved to open a path for the new selection. This will cause a maximum of one switching transient at outputs that have been moved. Other outputs will not be affected. Clean switching characteristics will make the transients small enough to be completely ignored.

Input 1 loaded by 4 outputs,
inputs 2-11 loaded by 3 outputs,
inputs 12-21 loaded by 2 outputs,
and all other inputs loaded 0 or 1 outputs.

Middle matrix usage history for 1 Million takes to a 100 x 100 switcher

Matrix Number or Count	Number of Takes made in this middle matrix	Number of Takes while this matrix was in use	Histogram for the Number of middle matrices in use at one time
1	79210	1000000	0
2	76702	1000000	0
3	70669	1000000	0
4	71944	1000000	0
5	68527	1000000	0
6	65983	1000000	0
7	63128	999993	0
8	60218	999983	0
9	57033	999995	0
10	46402	999892	0
11	49244	999528	0
12	44266	998165	0
13	39193	992642	0
14	34918	982049	0
15	31251	972349	0
16	29299	972113	0
17	28023	967638	108
18	26626	956462	9330
19	26986	966770	175804
20	30378	997633	814758

Rearrange cycles 138

Table 7. Switcher Blocking Tests for a Rearrangeable Switcher with Heavily Loaded Inputs as in Table 4

Inputs 1-20 loaded by 4 outputs,
inputs 21-40 loaded by 1 output,
and all other inputs not loaded.

Middle matrix usage history for 1 Million takes to a 100 x 100 switcher

Matrix Number or Count	Number of Takes made in this middle matrix	Number of Takes while this matrix was in use	Histogram for the Number of middle matrices in use at one time
1	66869	1000000	0
2	65763	1000000	0
3	65043	1000000	0
4	63935	999997	0
5	62714	1000000	0
6	60798	999995	0
7	59626	999996	0
8	56839	999962	0
9	55340	999820	0
10	53079	999726	2
11	50632	999246	7
12	47354	997808	8
13	43955	992680	4
14	40719	984536	9
15	38808	978486	17
16	36645	963620	9
17	35295	941251	1785
18	33164	960847	37439
19	31706	950691	290681
20	31716	860075	670039

Rearrange cycles 2780

Table 8. Switcher Blocking Tests for a Rearrangeable Switcher with More Heavily Loaded Inputs

CONCLUSIONS

Three stage switchers can give excellent performance while saving cost and space. Multistage switchers require that certain tradeoffs be made. First, the signal goes through three matrices on each path through the system, but with high quality matrices this still gives excellent performance. Second, blocking may occur if many inputs on the same input matrix are heavily used. The blocking can be reduced or eliminated by careful assignment of inputs to the matrices. If blocking still occurs, then one or more signal paths may be moved to open a path for the new selection This will cause a maximum of one switching transient at outputs that have been moved. Other outputs will not be affected. Clean switching characteristics will make the transients small enough to be completely ignored.

Multistage switching techniques provide an attractive method of reducing the number of crosspoints, and therefore the costs, of large switching systems. It allows larger switching systems to be built than are practical with single stage switchers, because of the practical size limitations on single stage switching matrices. Input distribution amplifiers are eliminated in large switching systems. System reliability should actually be enhanced because there are fewer crosspoints in the system, and failures in the middle matrices can be bypassed by using other middle matrices. Multistage switching should be considered by anyone looking at large distribution switching systems. It gives the user an attractive alternative to large single stage switchers.

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A UNIQUE ADAPTATION OF THE TRADITIONAL TELEVISION VECTOR DISPLAY

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Abstract - Baseband color video signals are made up of three dimensions of signal characteristics plus time. Traditional vectorscope and waveform monitor views of baseband video signals are limited. Each displays only two dimensions of a video signal. Waveform monitors display total video amplitude and time. Chroma saturation and phase are visible, but quite difficult to measure. Vectorscopes are used to display chroma saturation and phase with no respect to time or luminance. Standard usage requires both waveform and vector displays to accurately judge the total video signal. The 3-D display is based on a vector display with the additional view of the Y (or Luminance) axis. Operator flexibility is allowed to view this improved vector display on all three axes (three dimensionally) or on any two axes (B-Y and Y, for instance). The distinct advantage of this concept is that the operator begins with a familiar display. He then modifies the display to suit his current needs. Tasks like encoder balance and component level sets are made easy with this invention.

BACKGROUND

Composite and Component video systems require both waveform and vector displays to evaluate and adjust video signals. Composite encoders have traditionally had adjustments which modify amplitude, phase and saturation with a single control. We do admit that the interaction is not planned, but it often just happens. Many adjustments require the technician to view the output signal on both waveform and vector displays. He must watch both,at the same time, to view all of the effects of his adjustments. The most common problems are in the white and black balance sets in the two modulators, either R-Y and B-Y or I and Q. The advent of component video has returned the spotlight to this old problem. At least one manufacturer has developed a method to assist the technician in the adjustment of the R-Y and B-Y amplifiers. The operational problem remains due to the radically different form of their display. We need a convenient and clear method to observe luminance characteristics while making changes in the color (or color difference) channels. All of the methods available to the technician today are either not intuitively obvious or require the scrutinizing of two displays at once to assure proper adjustments.

THE BASICS OF THE 3-D DISPLAY

The new Videotek TVM-710 family of combination Waveform Monitor and Vectorscopes have a unique 3-D display. This function allows the user to view a display which appears as a traditional vector display. It allows rotation of the signal on all three axes. The advantage of this 3-D display is that it allows the viewing of any combination of luminance and chroma amplitudes. It holds the relative phase display to sustain user familiarity with the display. The 3-D display is useful in both composite and component systems.

The 3-D display begins as a conventional, on screen, vector display. Operation of the 3-D display is dynamic. The operator maintains visual recognition of his actions as he moves the display from a "normal" view to a less conventional perspective.

The 3-D display concept is based on the traditional composite vector display. In the traditional display, chroma amplitude and phase are displayed as points with vectorial (X and Y) attributes relative to the origin (Screen Center). All luminance attributes are "displayed" on the Z axis and are not apparent to the operator except as a single point at some X and Y position. All synchronization and monochrome portions of any video signal are displayed, on a traditional vector display, as a dot at the origin. The operator has no indication of amplitude of these portions of the signal. Similarly, any portion of a composite video signal with color, will appear with only it's saturation and relative phase displayed. The technician has no indication of the luminance content of the signal.

For the rest of this paper, some terms must be defined. The traditional vector display is often referred to as an X and Y display. We will now refer to the traditional horizontal axis (or X) as the B-Y axis. The vertical (or Y) axis is the R-Y axis. The perpendicular (or Z) axis is the Y axis. The relationships to video are that the R-Y and B-Y are color difference signals and Y is luminance. Color difference signals such as I and Q, or U and V are simply similar expressions of R-Y, B-Y. The main different is the mathematical definition. GBR (or RGB) are primary, not color difference signals. They must be transcoded to luminance and color difference signals for displays such as vector or 3-D. So, for this discussion, we will use R-Y and B-Y as the identification as the color difference signals. We do understand and can provide for the special requirements of other color difference signals.

This new display concept allows the operator to rotate the display on, for example, the B-Y axis to reveal the Y portion of the signal. He still maintains display of the R-Y axis attributes. At some arbitrary rotation point, 45 degrees for instance, the operator will have a display which shows all three axes in an orthogonal or three dimensional view. The 3-D system allows the operator full control of rotation on all three axes. Adjustments of encoders or the set up of component amplifiers in both D.C. offset and signal level are greatly simplified. The 3-D display maintains the traditional vector "boxes" for reference. It indicates the trajectory (or luminance value) of any chroma portion of a signal relative to its position on the familiar two dimensional vector plane. We have begun the process of application for a U.S. patent on the method and apparatus described above.

OPERATIONAL DESCRIPTION

The easiest method to describe the operation of the 3-D mode is to look at a traditional modulated stairstep signal with 5 steps. On a standard waveform monitor display, the signal is five steps of increasing amplitude with subcarrier impressed on each of the steps.

The Vectorscope view is simply a display with a two dots. One concentrated at the origin and the second one at the subcarrier mark. This indicates that all color in this video signal are of a single phase and saturation. (Figure 1)

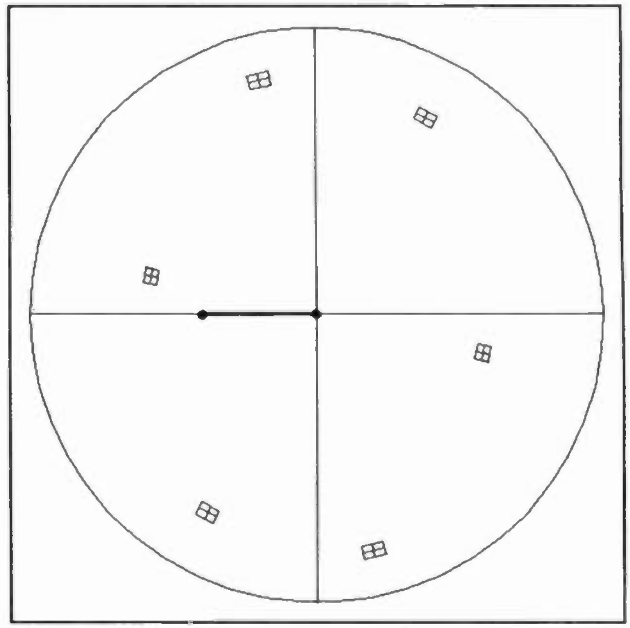

Figure 1 - 3-D Vector view of 5 step modulated stairstep test signal.

With the 3-D mode selected, the operator may choose to rotate the entire signal on the Y axis. The resulting display appears just as a rotation of the phase knob on a conventional vectorscope. The only difference is the electronic graticule rotates with the display. The signal is being rotated on the luminance axis. (Figure 2)

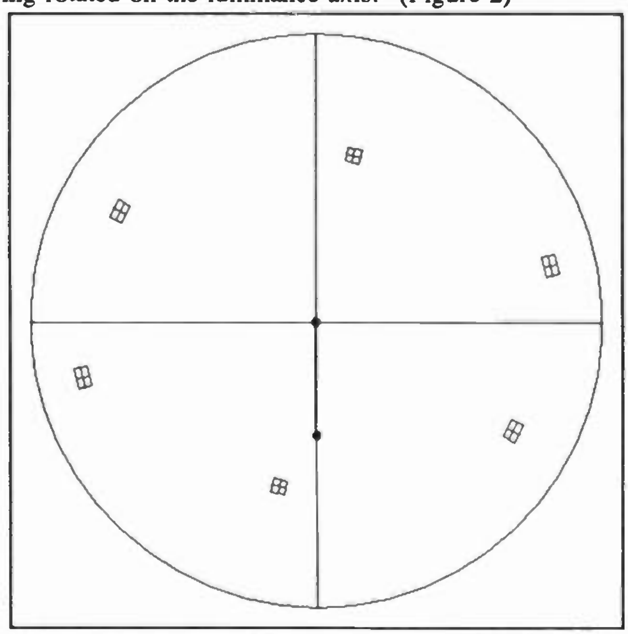

Figure 2 - 3-D Vector view, with Y axis rotated 90 degrees, of 5 step modulated stairstep test signal.

Now, if the operator selects B-Y as the axis of rotation, then the display may be positioned to reveal the Y components while still showing the B-Y and R-Y components. Further rotation on the B-Y axis (at either 90 or 270 degrees rotation) will show no R-Y components as the rotational plane is perpendicular to the to the plane of the display. At either point of no R-Y, the display is a box with one edge coincident with the Y axis. The opposite edge is coincident with the saturation level of burst. Five markers are on that edge coincident with the five steps of the stairstep. Tie lines connect the all of these points together. The last attribute noticed is a small line protruding below the B-Y axis along the Y axis. That line represents the horizontal sync portion of the signal and is a valid portion of the signal. All signal attributes, except time, are embodied in this display. (Figure 3)

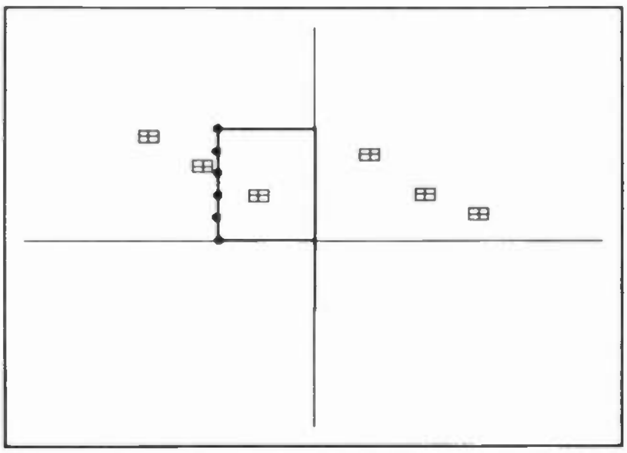

Figure 3 - 3-D view, with R-Y axis rotated 90 degrees, of 5 step modulated stairstep.

A more complex, yet standard, test signal is color bars. For purposes of this paper, there is no operational difference between composite or component color bars. The 3-D display works in NTSC, PAL, 525-60 component and 625-50 component Beta, M-II, GBR, and so on. The method for dealing with various formats is transcoders and demodulators, which are accepted and are not included in the scope of this paper. The 3-D display of color bars begins again as a commonly recognized vector display. The graticule is made electronically (Figure 4). The Y rotational display control again moves the display as the phase knob does in a traditional vectorscope. (Figure5)

Composite phase adjustments are made elsewhere to assure phase matching with the projected graticule. Rotating the display on the B-Y axis will provide the operator with a view of not only the color difference portions of the signal but with the luminance values as well.

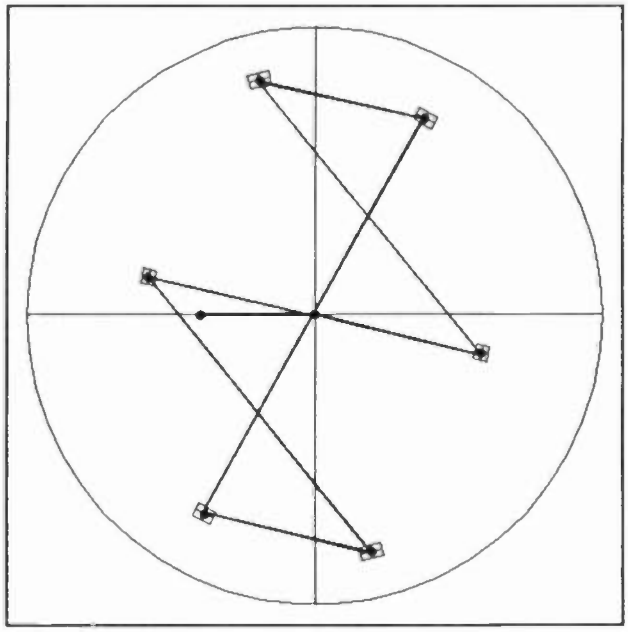

Figure 4 - 3-D Vector view of Color Bar test signal.

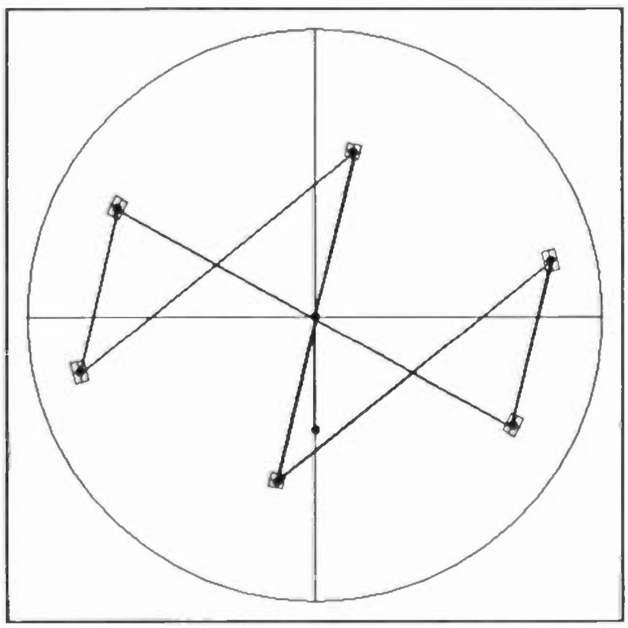

Figure 5 - 3-D Vector view, with Y axis rotated 90 degrees, of Color Bar test signal.

Again, stopping at exactly 90 or 270 degrees will leave only the B-Y and Y components of the display. The display, at this point, appears to be similar to a portion of the "lightning" display. It is different in that it is only one view of a nearly infinite number of views available with then rotating about the Y axis. (Figure 6)

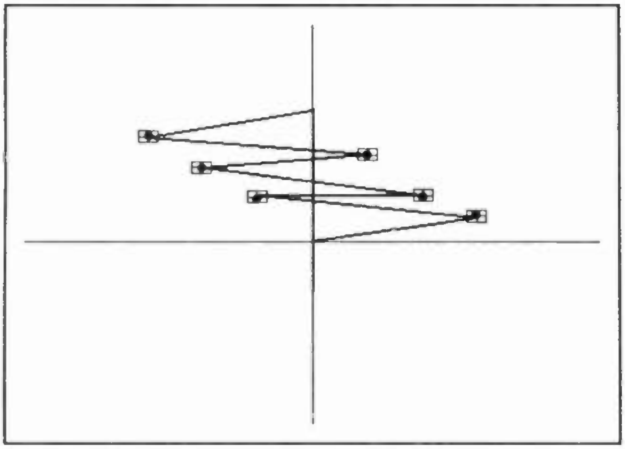

Figure 6 - 3-D view, with B-Y axis rotated 90 degrees, of Color Bar test signal.

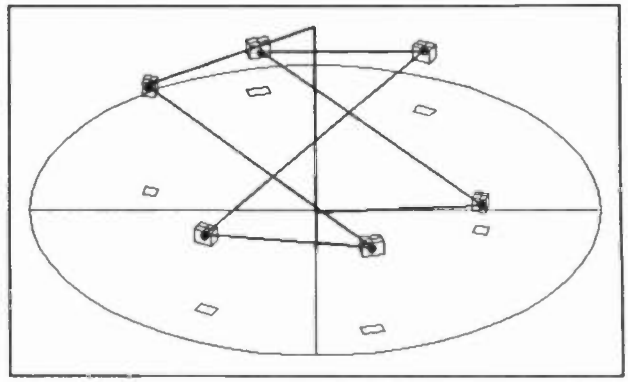

Figure 8 - 3-D view with B-Y axis rotated 40 degrees, of Color Bar test signal and showing all attributes of the signal.

Similarly, beginning from the basic vector view, rotating on the R-Y axis will eventually lead to two locations where the view is similar to the B-Y view of the "lightning" display. These limited, two axes views, are not however the intent of the 3-D display. (Figure 7)

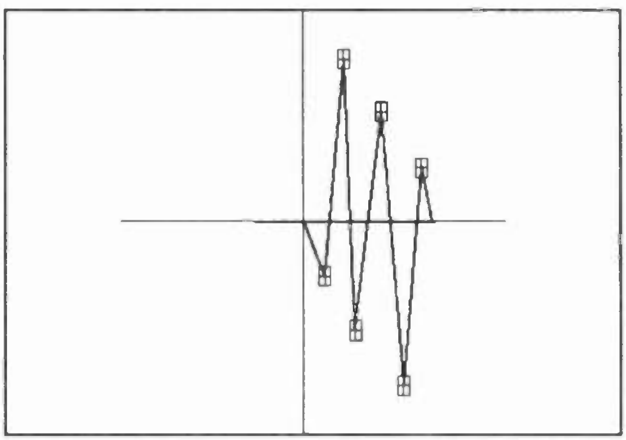

Figure 7 - 3-D view, with R-Y axis rotated 90 degrees, of color bar test signal.

The true value is the simultaneous display of all three axes in an orthogonal view. It allows the technician to view each axis, in it's correct perspective. He can view the results of his work and judge total signal quality on a single display. (Figure 8)

Some have improved their understanding of this process with viewing the display as a cylinder. The standard vector view is just like looking directly at the end of the cylinder. All you see is a round display. As you move you point of view off axis, in any direction, the view becomes elliptical, and the height of the cylinder becomes apparent. While some particular views may be extreme, there are many combinations of views which present understandable and useful information to the user.

FUNCTIONAL DESCRIPTION

The 3-D display system is the result of many engineering discussions. We were looking for an improved method to present complete video information to the user. The 3-D system depends upon the digital line store system used in the TVM-710 family. The line store system actually has the capability of storing three totally different lines of video. The 3-D system depends upon using the line stores. The display is actually a special line store display, but since it is digital, it has none of the traditional low intensity characteristics. Again, since it is a digital line store, the refresh rate is no quicker than once per field. The refresh rate applies only to the information for display, not to the display on screen. The on screen display is refreshed at a much faster rate to maintain a bright, crisp display. Further, the digital samples are taken at a rate of 8 x fsc and using 10 bit resolution. The resulting combined resolution is much greater than most digital display methods. The line store usage further allows the operator to select VITS or VIRS and make meaningful measurements and judgements in the 3-D mode.

Much of the functionality is similar to that of a vector display. The input video is treated, as required, with color difference demodulators, luminance detector and reference lock generator. We end with three digitized components, Y, R-Y and B-Y. The digitized samples are then treated with a trigonometric matrix conversion. The matrix converter is used to calculate, in two dimensions, the 3-D coordinates defined by the sampled video. A major feature of this process is that the calculations include operator actions. We have provided operator control to selectively address the rotational angles to be applied the three axes. Although the process will allow for rotation to be applied to an axis different than R-Y, B-Y or Y, we determined that the resultant might not be helpful to the operator.

Here are the basic calculations used to determine the display. The two dimensional display surface (the CRT face) has display axes X and Y. The input signals B-Y is called $\underline{X}$, R-Y is called $\underline{Y}$, and luminance (Y) is called $\underline{Z}$. The values for X and Y are calculated trigonometrically from the values of $\underline{X}$, $\underline{Y}$ and $\underline{Z}$. We use also the angles by which the projected display is rotated in the $\underline{XY}$, $\underline{XZ}$ and $\underline{YZ}$ planes. The angles for these rotational planes are called $\underline{a}$ for $\underline{XY}$, $\underline{b}$ for $\underline{XZ}$ and $\underline{c}$ for $\underline{YZ}$. The conversion to obtain a two dimensional isometric projection of the rotated three dimensional data can be determined with the following relation:

$$X = \underline{X}(\cos\underline{a}\,\cos\underline{b}) + \underline{Y}(\sin\underline{a}\,\cos\underline{b}) + \underline{Z}(\sin\underline{b})$$

$$\begin{aligned} Y = {} & \underline{X}(-\sin\underline{a}\,\cos\underline{c} - \cos\underline{a}\,\sin\underline{b}\,\sin\underline{c}) \\ & + \underline{Y}(\cos\underline{a}\,\cos\underline{c} - \sin\underline{a}\,\sin\underline{b}\,\sin\underline{c}) \\ & + \underline{Z}(\cos\underline{b}\,\sin\underline{c}) \end{aligned}$$

The next step in the system is to present the calculated values of X and Y to a Digital to Analog Converter for transformation to an analog form. Finally, the analog signals are coupled to amplifiers which provide the current to drive the horizontal and vertical deflection coils of the CRT.

For user convenience, and clarity of the display, we also have included the three dimensional generation of a reference graticule. This is calculated and projected with the video display signal. The traditional vector box targets are given a third dimension to complete the concept and display of luminance value in the projected display.

Videotek expects that this improved vector display will aid in the never ending struggle to produce the best and most accurate video signals possible. We certainly know that the 3-D concept is easily understood. Applying this concept to viewing standard video test signals will enhance the skills and understanding of those in the technical video field.

THE DEVELOPMENT OF COMMERCIAL ECHO CANCELLERS FOR TELEVISION

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Abstract-A system is being developed for cancellation of multipath echoes in television receivers. Towards this goal a prototype aimed specifically at current NTSC broadcasts is already operational and has been field tested with the help of a local broadcasting station. Future plans call for extending the field tests to include cable applications as well. This should be followed by extensions to cancelling echoes in advanced television systems.

The architecture for the NTSC application is based on the use of FIR digital filters for cancelling pre-echoes and nearby post-echoes. An IIR filter is used for post echoes of longer delays. The system has been designed to be flexible so that it can perform cancellation with any ghost cancellation reference that may be adopted by broadcasters. The field test thus far has used the reference signal developed by the BTA in Japan. However, a higher energy reference signal, as well as the hardware needed to broadcast such a signal, is now under development. This will allow more rapid and efficient cancellation of echoes, especially under high-noise and transient conditions. Three algorithms have been developed and tested for calculating the coefficients of the digital echo-cancellation filters. These are 1) a frequency division method, 2) an LMS approach, and 3) a proprietary time-domain method. All three methods undergo further unique optimization procedures that greatly increase the efficacy of echo cancellation over what can be achieved by methods published heretofore.

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. If the various transmission paths are not of the same length then each copy of the signal is distinguished by the time of arrival, relative signal amplitude and phase shift of the RF carrier. The receiver will lock onto the strongest of these signals. All others are called multipath echoes or "ghosts". It has long been recognized that much optimization has gone into NTSC television to make the picture quality as good as possible within the restrictions of the NTSC signal formats. Echo cancellation is one of the few remaining possibilities for achieving significant further improvements in received image quality. Moreover, for HDTV, many of the proposed encoding schemes rely on digital data recovery which can be severely compromised by echoes. This paper will provide a progress report on the development of practical echo cancelling hardware and software aimed at the American scene.

The desirability of cancelling echoes has been recognized for a long time. The basic approaches that are finally being implemented today were already well known more than 10 years ago. The tutorial by Ciciora[1] et all, written in 1979 pointed out that RF-domain echo cancellation is, in general not practical. It indicated the preference of cancelling the echoes in the baseband using either deconvolution or some adaptive algorithms. However, until recently, those methods could not be affordably applied to consumer products because of the costs of the required hardware. However, with the present low cost of signal processing chips the question is no longer affordability but the choice of the best signal processing hardware, the optimum algorithms, and the choice of the required ghost cancellation reference (GCR) signal.

The choice of a national standard GCR was studied at length in Japan by the Japanese Broadcasting Technology Association (BTA). After a lengthy set of experiments, the BTA recommended and Japan adopted a GCR that is the time integral of a sin x/x pulse[2,3] to be transmitted on line 18 of the Vertical Blanking Interval (VBI). Subsequent discussions in the U.S.A. by the Specialist Group on Ghost Cancelling (T3S5) of the Advanced Television System Committee (ATSC) expressed the fear that the BTA choice of a GCR may be suboptimal. Since the energy of this signal is very low, the performance under high noise conditions is poor. Therefore, the processing to remove the noise makes adaptation to changing echo conditions too slow. Since then a variety of high-energy GCR signals have been under study for adaptation by U.S.A. broadcasters. A review of the GCR selection activities and the recent trends in echo cancellation was given by Uyttendaele[4].

Philips Laboratories, Division of North American Philips Inc., together with Philips Consumer Electronics Corporation of Knoxville, TN. have been actively seeking a commercially and practically viable solution to the problem of echo cancellation in NTSC and advanced TV systems. The work has progressed on three fronts: 1)The choice of an optimum high-energy GCR signal, 2) the development of low-cost hardware that can be economically incorporated into home receivers, and 3) the derivation of algorithms that would allow the cancellation of echoes to be performed robustly with minimum hardware costs.

The resulting first generation hardware and software was field tested during the Autumn of 1990 using the BTA GCR, in cooperation with WABC-TV in the New York City area. The results of those field tests were used to improve the design of subsequent generations of echo cancellation equipment. It also led Philips to confirm the conclusions drawn by the NAB during its own field tests of Japanese equipment[5], namely, that viable echo cancellation is possible within the state of the art. However, improvements over current first generation efforts are not only possible but very desirable. Towards this goal, Philips is developing its own proposal for a high-energy GCR to improve the handling of transient echoes and performance in high noise conditions. This new GCR, together with supporting hardware, will be submitted to the NAB for field testing in response to their request for such a proposal[6]. Some of its properties are detailed below. This paper also discusses the two other prongs of Philips' three-prong effort: the development of new hardware and algorithms.

Research into echo cancellation for television began on the mathematical level at Philips Laboratories in 1987. It progressed to computer simulation in 1988. The first laboratory prototypes became operational during the Spring of 1990, and the first field tests were conducted in the Fall of 1990. At this point, work is actively progressing on the design of third-generation chips for an economical implementation of echo cancellation in the NTSC environment. Additional field tests are planned in 1990 in both the terrestrial broadcast and cable environments. Furthermore, the current methods are being extended from NTSC to HDTV applications.

BASIC PRINCIPLES

When several replicas of the transmitted signal are superimposed at the receiver antenna, the strongest signal is usually selected as the main signal. Signals that arrive after the strongest one are called *post echoes*. However, it is possible to receive signals before the strongest one. These are called *pre-echoes*. Pre-echoes can occur for example, via direct pickup, when a weak signal is received over the air before the delayed main signal over cable. The cancellation of pre and post echoes is implemented differently. However, both represent linear distortions which are removed through the use of adaptive linear digital filters.

Cancellation of post echoes

The post-echo generation process may be modeled by the system diagram shown in Fig. 1. The output, X is sum of

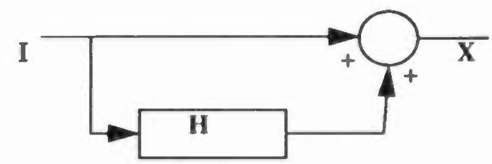

Figure 1. Generation of post echoes

the input I received over the (shortest) direct path and the echo terms modeled in the frequency domain as I•H. Since, X=I(1+H), therefore, the frequency response of the echo channel is H=X/(I-1). Such post echoes can be cancelled by the use of an infinite impulse response (IIR) filter such as the one shown in Fig. 2. The output of this

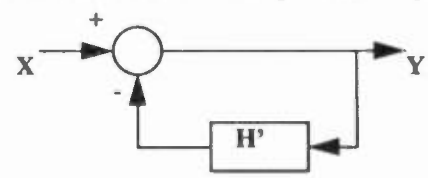

Figure 2. Cancellation of post echoes with IIR filter

filter is Y=X/(1+H'). If H' is set equal to H, the frequency response of the echo channel, then the output of the IIR filter will be

$$Y=X/(1+H)=I(1+H)/(1+H)=I$$

and the cancellation will be perfect.

Cancellation of pre-echoes

Pre-echoes can also be characterized as being produced by a linear filter. However such an equivalent filter, where the response precedes the stimulus, would be non-causal (apparently operating in negative time). The IIR filter that would cancel such pre-echoes is physically non-realizable. Therefore it is necessary to settle for incomplete cancellation of such pre-echoes with a finite impulse response (FIR) digital filter.

Calculation of echo channel characteristic

The use of IIR or FIR filters for cancelling echoes presupposes that the echo-generating channel is completely characterized. Such characterization is done with the aid of the transmitted GCR signal discussed previously in the Introduction. There are a variety of methods for computing the channel characteristic by comparing the received distorted version of the GCR with the ideal transmitted one. Philips begins with the 8-field sequence recommended by the BTA[2]. This takes into account that the received baseband waveform is sampled at 4 times color subcarrier. That means that on successive field the subcarrier information will have shifted by 90° with respect to horizontal phase. Therefore every fourth field will be in phase. So 8 successive fields are stored. The transmitter will transmit the GCR on fields 1,3,6 and 8 and no GCR on fields 2,3,5 and 7. Now we can subtract field 1 from 5, 2 from 6, etc. The result is that all non-changing information such as pedestal levels is subtracted out and 4 copies of the received GCR remain. Depending on the signal-to-noise ratio (SNR)and the choice of the GCR, a number of received GCRs isolated in this fashion must be averaged over a period of time.

Once a received GCR is available with sufficiently high SNR, the echo channel response can be calculated. Towards this goal, we use proprietary improvements of three well known methods: 1) an iterative least mean square (LMS) approach, 2) frequency division as indicated above, and 3) a convolutional time-domain method.

HARDWARE ARCHITECTURE

The block diagram of the basic real-time echo canceller is shown in Fig. 3. The composite video baseband input

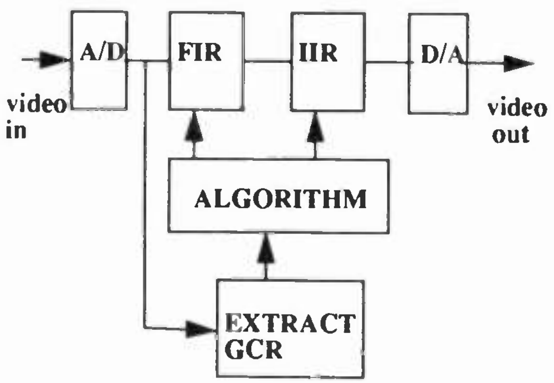

Fig. 3 Echo canceller system diagram

signal is digitized and the echoed GCR signal is extracted. From that an algorithm calculates the coefficients of the digital filters needed to cancel the echoes. On our first generation equipment, these calculations were done with a personal computer (PC). In later equipment it is done with a digital signal processing (DSP) chip. The FIR filter is used to cancel pre-echoes and close post echoes. The IIR filter completes the cancellation process. The filters are implemented with custom designed VLSI chips, optimized for this application. The final D/A converter restores the digital signal to the analog video form.

SYSTEM SPECIFICATIONS

The actual operating specifications of a system are always the result of a set of compromises. The specifications given here are those of our first-generation echo canceller, when operating with the Japanese BTA GCR. We expect considerable performance improvements and some specification changes with the use of a high-energy GCR.

VBI-related issues

The GCR can be in almost any available VBI line. When using Japanese laboratory-based echo generation equipment, we use line 18. During the broadcast field test line 16 is employed. The VBI line prior to the one carrying the GCR must carry a waveform that remains the same from field to field, as required by the BTA. The the GCR is transmitted on alternating fields and 8-field subtraction and averaging is performed, as explained above. Further averaging is done in order to improve the SNR of the received GCR prior to calculating the filter coefficients.

Ghost cancellation performance

The BTA GCR allows a total echo delay range of 45 μs. We chose to apportion this range to yield 3.4μs of pre-echoes and 41.5μs of post echoes. Our algorithms have no threshold for the minimum or maximum ghost levels that can be eliminated. With the BTA reference, we have cancelled ghosts with Desired-to-Undesired D/U ratios as low as 6 db. Since we approach this task from the perspective of a receiver manufacturer, our system architecture is flexible with respect to the number of ghosts that can be cancelled. We expect to aim simpler systems for inexpensive receivers and more complex ones for the top-of-the-line. Our first-generation equipment can cancel 16 simultaneous distinct ghosts. The ghost attenuation is better than 40 db when the received SNR exceeds 40 db. The computation time of the algorithms itself is well under 1 second. The fundamental limitations on adaptation time are the GCR and the SNR. The BTA specifies that with a received SNR of 40 db, 128 fields have to be averaged. This averaging requirement should be reduced substantially when a high-energy GCR is adopted.

Test apparatus

We have built a PC-based test console that will display the following output:

a) Waveform of the average received GCR before cancellation

b) Waveform of the average received GCR after cancellation

c) Graphical representation of the computed filter coefficients

d) Fourier spectrum of the GCR before and after cancellation

e) Graphical representation of the noise accompanying the received GCR

f) Numerical data on received noise level (db), residual ghost level and

g) Perceived Desired to Undesired Ratio (PDUR)

The PDUR is a measure of how objectionable multiple ghosts are in terms of one equivalent ghost. This measure was derived through a set of psychophysical experiments[7]. Of course the PDUR is only defined for the BTA GCR, it would be undefined for most high-energy references.

HIGH-ENERGY SIGNAL DESIGN

Philips has designed, and will propose to NAB and the ATSC, a new type of high-energy signal. It is the next generation improvement over the previously proposed high-energy signals based on pseudo-random sequences such as the M-sequence[8]. It shares many of the advantages of these sequences, while overcoming some of their shortcomings. Its properties are summarized below:

- Can be adjusted to any desired signal length and amplitude.
- The energy level of the received decoded reference is proportional to the signal duration. For example, a signal duration of 35.5 µs would provide a signal-to-noise improvement over the BTA GCR of a factor of 8. However, the duration is not restricted to 35.5 µs.
- The GCR allows the detection of both pre and post ghosts, with no fundamental restriction on the total detection range.
- The GCR can be used as a frame synchronization signal.
- No sensitivity to constant-offset sampling errors due to shifts in the sampling pulse locations.

By contrast, previously-published proposals for a high-energy GCR[8] use cyclic pseudo-random sequences that allow a total ghost-detection range, *pre-ghost plus post-ghost*, of 35.5 *µs*. Furthermore, those signals can be adversely affected by shifts in the sampling pulse locations. They also are restricted in length to 2^N-1 (N=1,2,3,4...), yielding the only practical length of 255 or 35.5 µs.

A sample of one of several proposed waveforms is shown in Fig. 4.

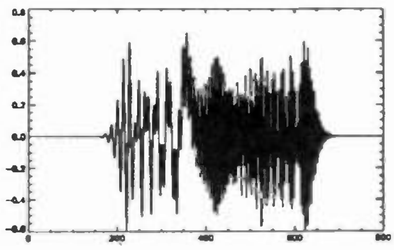

Fig. 4 One of the proposed high-energy GCR waveforms

CONCLUSIONS

Although cancellation of TV echoes has been studied for several decades, it has only become a potentially practical option for commercial home TV receivers with the recent advent of inexpensive and powerful VLSI technology. This new technology has led in Japan to the development of first-generation echo cancellers and to the continuous broadcasts of the associated GCR signal. Our experiments in the U.S.A. have indicated that it is possible to achieve improved performance over the existing Japanese cancellers, even if the BTA GCR is used. This improvement was exhibited in terms of the number and severity of the ghosts that are cancelled and in terms of residual ghosts remaining after cancellation. With the use of an improved, high-energy reference, we expect the speed of response and noise immunity to increase substantially. For example, we should be able to cancel echoes even from some moving reflectors, a task that the long integration time mandated by the BTA GCR does not permit. The Philips nomination for such a high-energy reference, together with the associated hardware and software will be exhibited during a competitive public field test to be sponsored by the NAB starting 10 June 91. The results of those field tests, as well as upcoming tests in the cable environment, will be reported in later papers.

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REMOTE MONITORS FOR BROADCAST TRANSMITTERS

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INTRODUCTION

The design criteria for a remote monitoring system for broadcast transmitters is described. The value of remote monitoring versus remote control, the issue of system security and the various functions required to provide a useful tool to the transmitter operations and field service engineers are also discussed. The desirable features of a graphic user interface suitable for remote monitoring is also presented.

WHY REMOTE MONITOR?

The demand for the capability of remotely monitoring broadcast transmitters has intensified as the availability and use of personal computers (PC) has increased. This, coupled with the availability of transmitters equipped with serial ports gives rise to the possibility of monitoring the operation of a broadcast transmitter from virtually any location accessible via telephone. The equipment required is a suitably configured personal computer and software package, modems at the monitor and transmitter location and a transmitter equipped with a serial port and appropriate software.

Use of a PC based system as an "extended" monitor system may also be desirable. In this instance, the PC would be located up to a few hundred feet from the transmitter, but close enough that a telephone connection would not be required. All required elements of the system would remain the same, except that the modems would not be required.

An obvious extension of these capabilities would be to use the PC as a control element. While such capability is theoretically possible, it is not necessarily desirable. Many stations are routinely equipped with remote control systems that provide capability to monitor and control transmitter operations, usually from the studio location. Adding the capability of control from the PC would lead to possible conflict with the remote control system. In addition, most remote control systems provide much more than transmitter control. Such facility control would raise the requirements on a PC based system to yet another level. The issues associated with system security are also minimized since the user does not have the capability to control the transmitter.
Anxiety about the possibility of inadvertently "crashing" the transmitter are eased with monitor only capability.

SYSTEM SECURITY

The first level of security is established in that the user must have the correct complement of equipment and software. For the typical system this would include a IBM PC/AT or compatible computer equipped with 512K memory, a VGA/EGA video card, color monitor, serial interface card (RS-232C) and Hayes compatible modem. Software requirements include MS-DOS and a copy of the monitor software. While all of this is readily available, many PCs are not so equipped.

Even with suitably equipped system, password protection for the transmitter serial port is a must. The transmitter software must provide an entry field in which to set the system password and prohibit access if the user does not correctly enter the password from the PC. Password modification should only be possible from the front of the transmitter, and should be changed at regular intervals. The password should not be displayed on the PC at any time.

It may be useful to a user to allow access to the monitor system by the manufacturers field service organization. This would permit the manufacturer to aid in trouble shooting when problems arise or to provide a routine monitor service if desired. Obviously, the user must divulge the password to the service engineer in this case. Denial of future access would be accomplished by simply changing the password.

SYSTEM FUNCTIONS AND USER INTERFACE

The ideal PC based monitor system should emulate the monitoring functions of the transmitter as closely as possible. Thus, as the engineer becomes accustom to the display configuration of the transmitter, this familiarity is applicable to the remote monitor system. Since the purpose of the remote monitor is to communicate the operational status of the transmitter in the most expedient manner possible, a user interface that provides graphical representation as opposed to textual format is desirable. The use of graphical interfaces in industrial monitoring has proven itself for many years. It allows more information to be viewed and interpreted in less time than the textual interface.

Another desirable feature is the appropriate use of color. Many of the monitor functions of modern transmitters is via red, green and amber light emitting diodes (LEDs) or other display elements. Thus the availability of color is a key element in determining transmitter status.

Response time is also important. For real time operation the response time for changing screens or updating the data on the screen should be less than one second when using standard telephone lines.

Availability of a print function is also desirable. This may be done automatically on a hourly basis or manually as desired. A permanent record may be used to facilitate trend analysis of operating parameters or to identify patterns behind recurring faults. For the Sentry monitor and Platinum Series VHF TV transmitter, the complete contents of the alarms queue, all analog transmitter metering information and transmitter fault status may be printed.

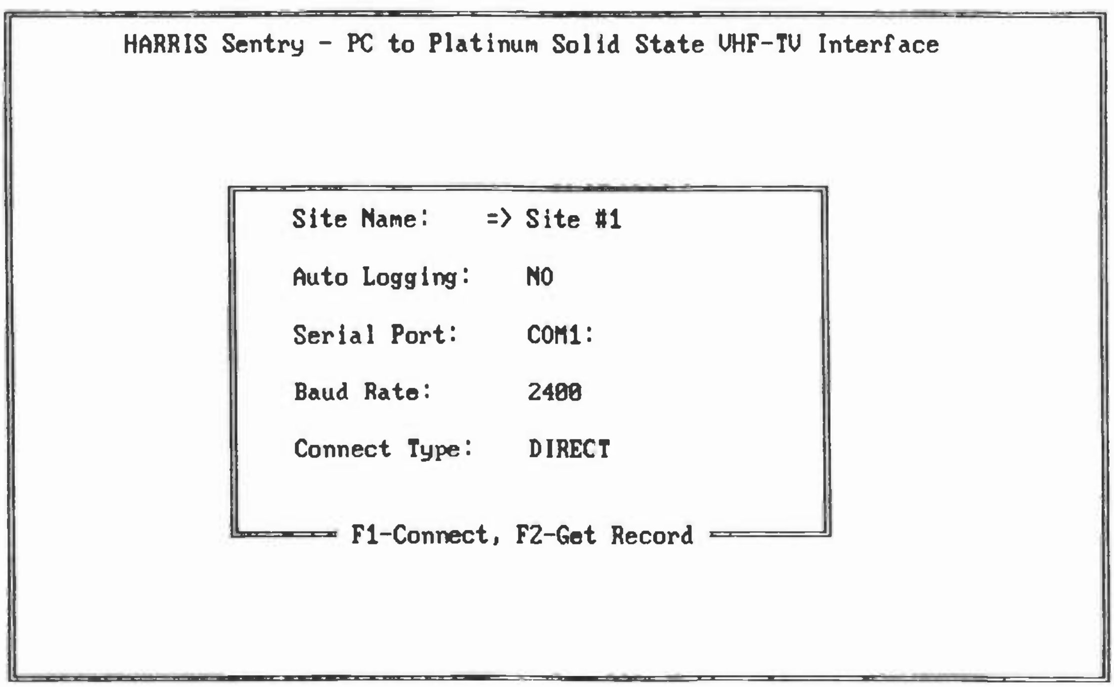
For ease of use and to facilitate monitoring of multiple transmitters a site selection and configuration screen (Figure 1) is required. This screen necessarily permits the operator to specify the site name, the communications port, baud rate, connection type (remote or extended) and the phone number. Once these parameters are specified the operator may give the commands to connect to transmitter. The capability of saving a specific configuration to disk facilitates ease of making repeat connections to the same site.

For modern solid state transmitters, two types of monitor screens are useful. A "Transmitter Display" screen (Figure 2) can show a graphical representation of the entire transmitter, including cabinet, exciter, module and power supply locations. With this screen module, power supply and exciter faults may be displayed as well as exciter switcher status and the active exciter.

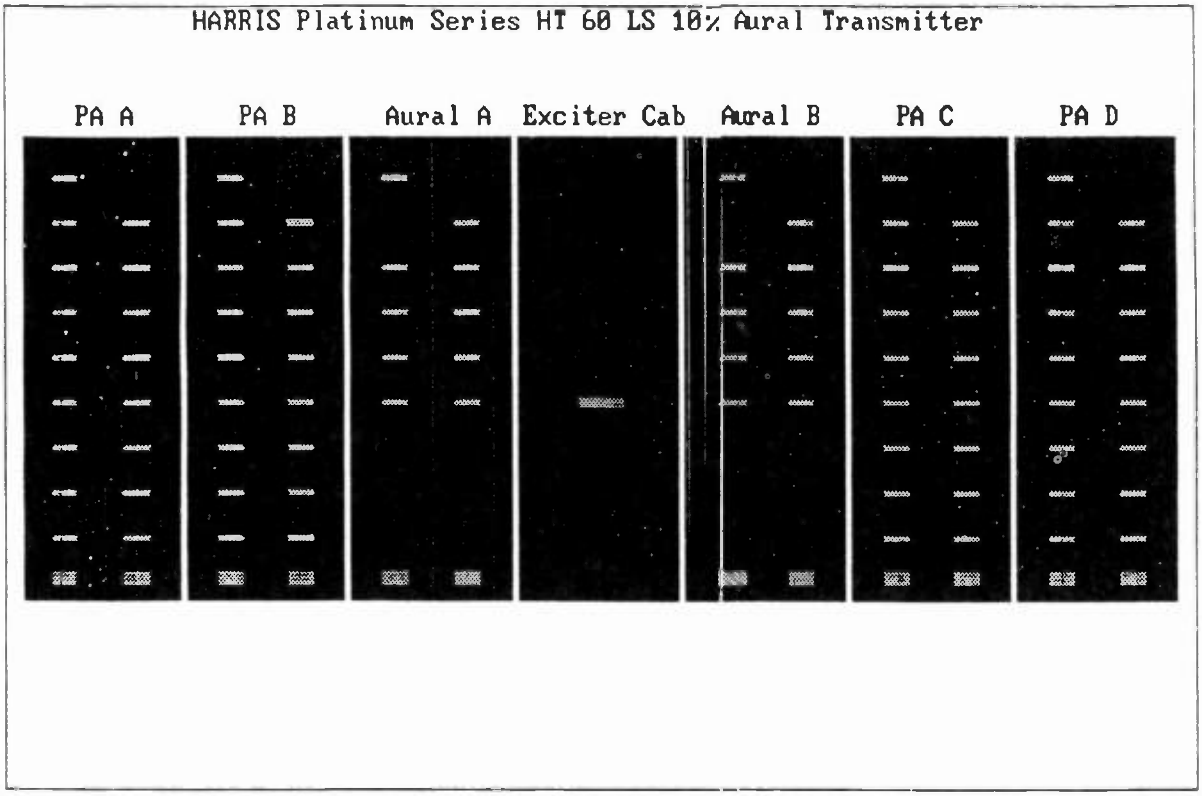
A second screen displaying a graphical representation of the control panel (Figure 3) provides the remainder of the transmitter status information. For the Sentry monitor system and Platinum Series VHF TV transmitter, virtually all operating data desired may displayed. This includes forward and reverse power for both visual and aural, power supply voltages and current, driver power, logic voltages, and AC mains voltages and current. In addition all faults displayed on the control panel are available. When there is a fault the text appears in the status portion of the screen describing the fault with the same text as is used in the transmitter. By using the function keys on the PC, the action of the transmitter soft keys is emulated, making operation of the remote monitor virtually identical to the transmitter monitor. The monitor user has the capability of selecting the default screen and modifying the parameters on this screen without affecting the default screen on the transmitter. These default parameters may be stored in a configuration file on the PC hard disk.

SUMMARY

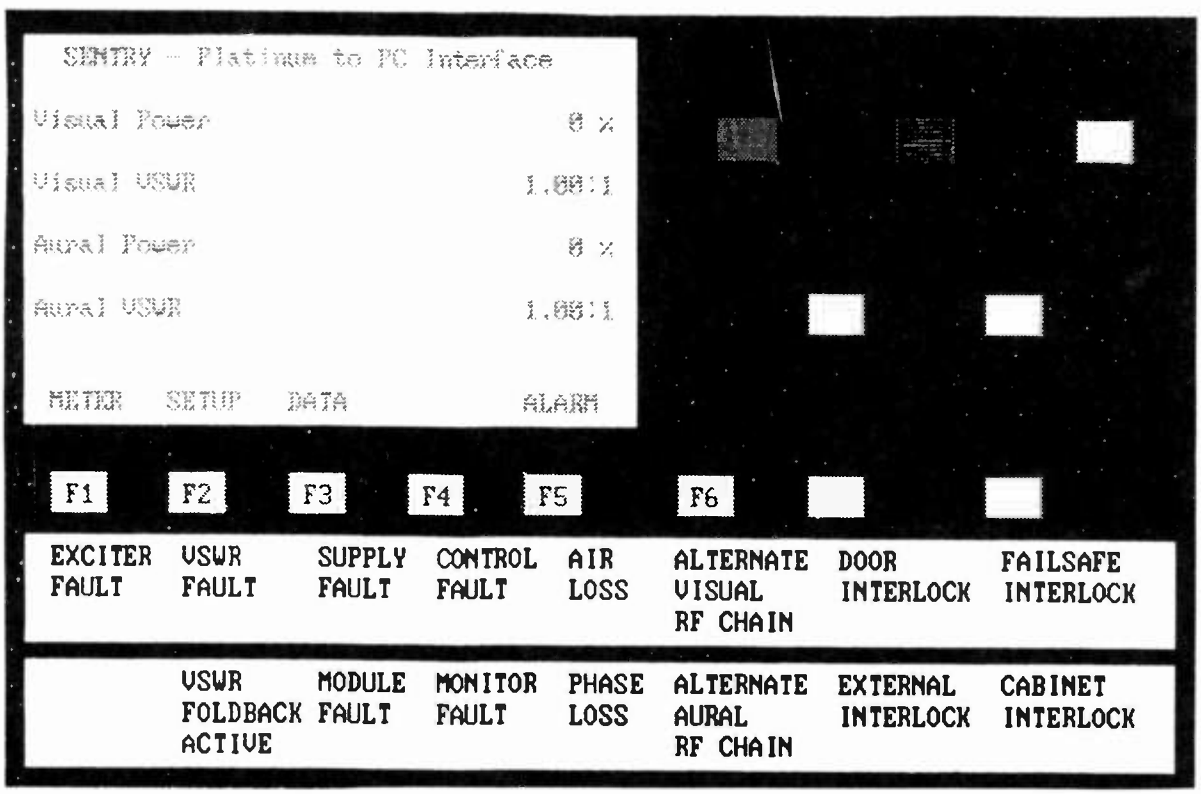
The design criteria and desirable features of a PC based remote transmitter monitoring system have been presented. At present, this concept has been developed for use only with the Platinum Series TV transmitter. However, the concept is applicable to virtually any transmitter, old or new, AM, FM, or TV, provided it is equipped with a serial port. With the continued widespread use of PCs in broadcast stations, manufacturing and the home, systems to monitor other types of transmitters should be forthcoming.



("FIGURE 1")



("FIGURE 2")



("FIGURE 3")

AVERAGE POWER RATINGS OF COAXIAL TRANSMISSION LINES

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ABSTRACT

A computer model of transmission line losses and heat transfer has been developed and verified by careful RF testing. The model considers heat transfer by both radiation and convection along with other factors, such as resistance change with temperature, surface emissivity, and line pressure.

A discussion is presented considering factors, such as types of copper used for inner and outer, effects of painting outer, and general transmission line considerations.

HISTORICAL PERSPECTIVE

In the past, establishing average power ratings for coaxial transmission lines was not an easy task. Most line ratings were determined from limited power testing and the results extrapolated to other line sizes. Ratings varied widely from one manufacturer to another, probably as a result of testing inaccuracies and competitive pressures. In order to solve the heat transfer equations, iterative calculations are required. These calculations took much time for a solution and were subject to error because of the large number of calculations involved. With the widespread use of digital computers in the late sixties, it became possible to handle the calculations in an accurate and time effective manner. RCA at that time developed a computer program to solve the heat transfer equations. The program included the effects of increased losses due to change of resistance of the inner and outer conductor with temperature. Room temperature line losses were determined from measured values of long runs. Dielectric has recently updated the computer program to include calculations for line losses and the effects of VSWR. This program is now the source for Dielectric's line ratings.

DEFINING THE PROBLEM

Under steady state conditions, there will be a balance of incoming and outgoing heat energy. (HEAT IN = HEAT OUT) The HEAT IN is the power loss per unit length, and the HEAT OUT is the heat transfer by radiation and convection per unit length. This energy balance must be done separately for both the inner and outer conductors, but there is an interaction between the two. That is, the heat flow from the inner conductor depends on the temperature of the inner conductor and also on the temperature of the outer conductor. The problem then is to accurately determine the heat in to both the inner and outer conductors and then to set up a thermal balance equation to establish the resulting temperatures. The equations for heat transfer are well known and can be found in any good book on heat transfer such as references 1 and 2 in the bibliography. The equations for transmission line losses can be found in reference 3. I will not bore you with the detail equations in this article, but I would like to discuss some of the factors to which judgement must be applied to implement these equations.

SETTING THE STANDARDS

In order for the average power ratings to have any meaning, we have to define what conditions the ratings are based on. Dielectric, as well as most transmission line manufacturers, base their published average power ratings on the following conditions:

1. Matched line (unity VSWR)
2. Horizontal line
3. Still air (no wind blowing)
4. No solar load
5. Inner conductor temperature of 120° C in a 40° C ambient
6. Zero gauge pressure (unpressurized line)

Changing any of these factors will affect the average power rating. Let us examine these factors one at a time and explore their ramifications.

1. MATCHED LINE

A VSWR on the line will increase the the I^2R losses and produce hot spots on the line at the high current points at half wave spacings. A series of curves (Figure 1) has been developed to correct the average power rating for the effects of VSWR. The curves were established considering conduction from the hot spots to the cooler spots so that the temperature of the hot spots do not exceed the established maximum inner conductor temperature.

Notice that as the frequency decreases, the derating for VSWR increases. This is because the effects of heat conduction become less effective as the distance between the hot spots (high current points) and the cool spots (low current points) increases with decreasing frequency. If there were no heat conduction along the line, the derating at all frequencies would be $((VSWR + 1)/(2 * VSWR))^2$ which would be the value at which the lower end of the curve would become horizontal. This occurs at about 10 MHz. If there were infinite heat conduction along the line, the derating at all frequencies would be $(VSWR+1)^2/2(VSWR^2 + 1)$, which is the approximate value at the high frequency end of the curve. Since we live in a real world where neither zero or infinite heat conduction exists, the curves in Figure 1 are the results.

2. HORIZONTAL LINE

The transmission line power ratings are based on horizontal line mainly as a convenience. It is easier to set up and measure a horizontal line than a vertical line and the results are more uniform. Setting up a thermal model for horizontal coaxial cylinders is also well defined. The difference between horizontal and vertical orientation is small. A 10 ft. section of 4" 50 ohm line was tested in a vertical orientation and compared to a horizontal section while both were carrying full rated power. The inner conductor of the vertical section ran 3° C hotter than the horizontal section. It was also determined that the lower several feet ran 6° C cooler and the upper several feet ran 7° C hotter than the main portion of the verical line. This slight difference was felt to be within acceptable limits especially since the upper portion of a vertical transmission line run always handles less power than the lower portion due to line losses.

3. STILL AIR

Air movement on the outer conductor will greatly increase the convective heat transfer from the outer conductor to the surrounding air. It can easily increase by a factor of ten. Air movement, wind, is almost always present on a tower, but this is not true inside a transmitter building. Hence, average power ratings are based on still air.

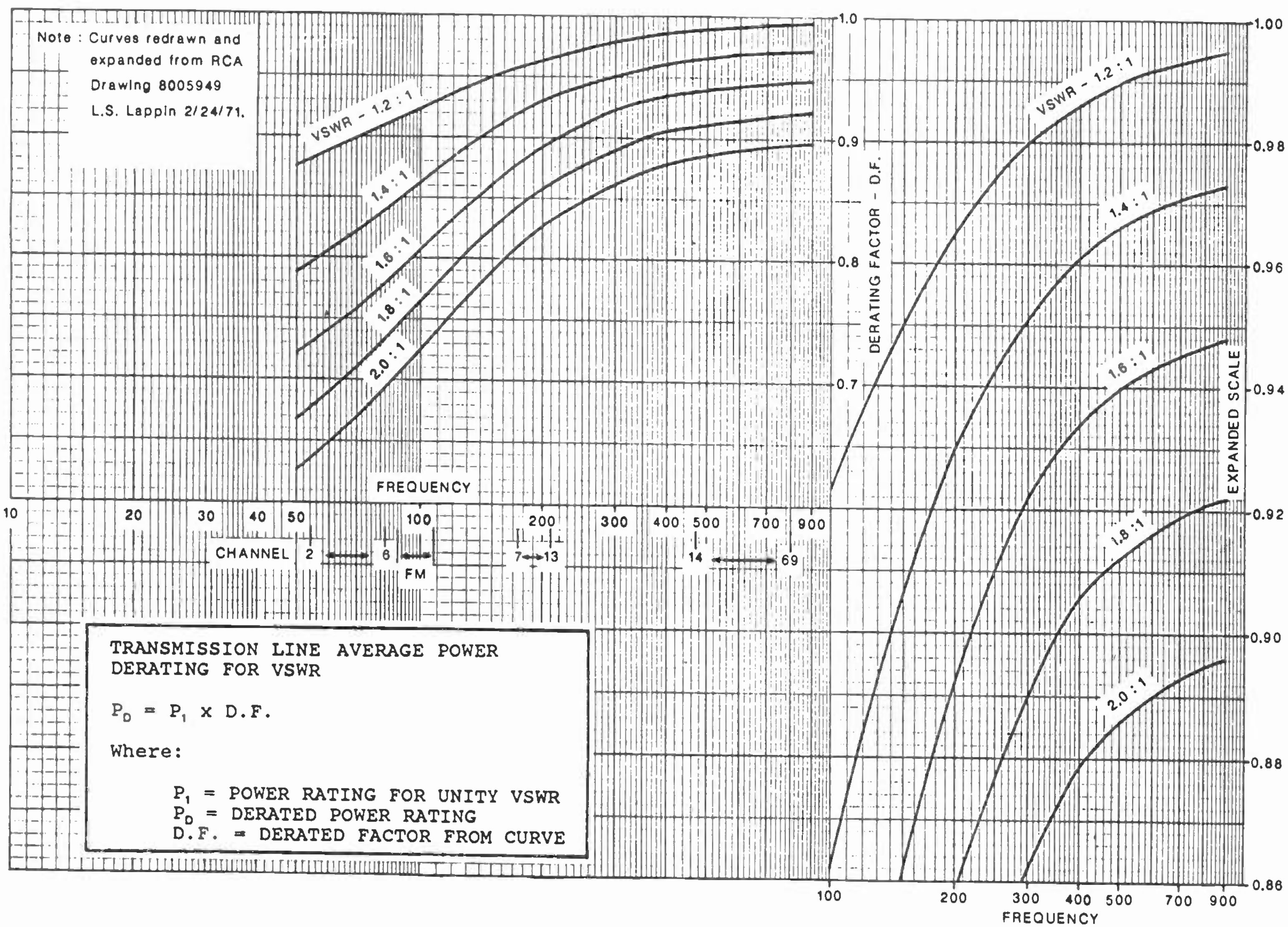

Note : Curves redrawn and
expanded from RCA
Drawing 8005949
L.S. Lappin 2/24/71.
VSWR - 1.2 : 1
1.4 : 1
1.6 : 1
1.8 : 1
2.0 : 1
DERATING FACTOR - D.F.
1.0
0.9
0.8
0.7
FREQUENCY
10
20
30
40
50
100
200
300
400
500
700
900
CHANNEL
2
6
FM
7
13
14
69
EXPANDED SCALE
1.00
0.98
0.96
0.94
0.92
0.90
0.88
0.86
TRANSMISSION LINE AVERAGE POWER
DERATING FOR VSWR
P_D = P_1 x D.F.
Where:
P_1 = POWER RATING FOR UNITY VSWR
P_D = DERATED POWER RATING
D.F. = DERATED FACTOR FROM CURVE

Fig. 1

4. SOLAR LOAD

The effects of solar heating are not included in the thermal calculations to establish power ratings. This is not as bad as one would think at first glance. There are three things which tend to offset this effect:

1. The thermal calculations assume the outer conductor is untarnished copper, therefore, have a low value of emissivity. (In case you have forgotten emissivity is a measure of a surfaces ability to radiate heat energy, and solar absorptivity is a measure of a surfaces ability to absorb solar heat energy.) Fortunately the solar absortivity for shiny copper is also low. This means very little solar energy is absorbed. As the outer conductor becomes oxidized, both the emissivity and the solar absorptivity increases. More solar energy is absorbed but more heat energy is also radiated. One tends to offset the other to some extent.

2. Transmission line outdoors on a tower almost always has some wind. Since line ratings are based on still air, even a very slight breeze will more than offset the increased heating due to the sun.

3. During the hottest part of the day the sun is in a poor position to contribute to heating of vertical transmission line. For instance if the sun were directly overhead, there would be no sun on the vertical surfaces of the line.

5. INNER CONDUCTOR

Forty years of experience have shown that a maximum inner conductor temperature of 120° C is a prudent choice. Copper will undergo very little oxidation at this temperature if kept in a dry environment. At temperatures above this, oxidation tends to occur at a more rapid rate. The higher the temperature, of course, the faster the rate of oxidation. 40° C ambient is another historical number which is adequate for the majority of applications. For ambient temperatures above 40° C, use the following derating factors.

Ambient Temp. C	Derating Factor 50 OHM	75 OHM
40	1.00	1.00
45	.94	.92
50	.86	.85
55	.79	.77
60	.71	.70

6. ZERO GAUGE PRESSURE

This is the normal method of rating the line and indeed unpressurized line is normally used indoors. Outdoor line should always be pressurized to keep it dry. Increasing the pressure in the line increases the density of the gas and thereby increases the convective heat transfer from the inner to the outer conductor. The increase in power vs. pressure for both 50 and 75 ohm line is shown in table 1.

Table 1

Line Pressure PSIG	Rating Factor 50 OHM	75 OHM
0	1.00	1.00
5	1.09	1.08
10	1.16	1.15
15	1.21	1.22
20	1.26	1.28
25	1.31	1.33

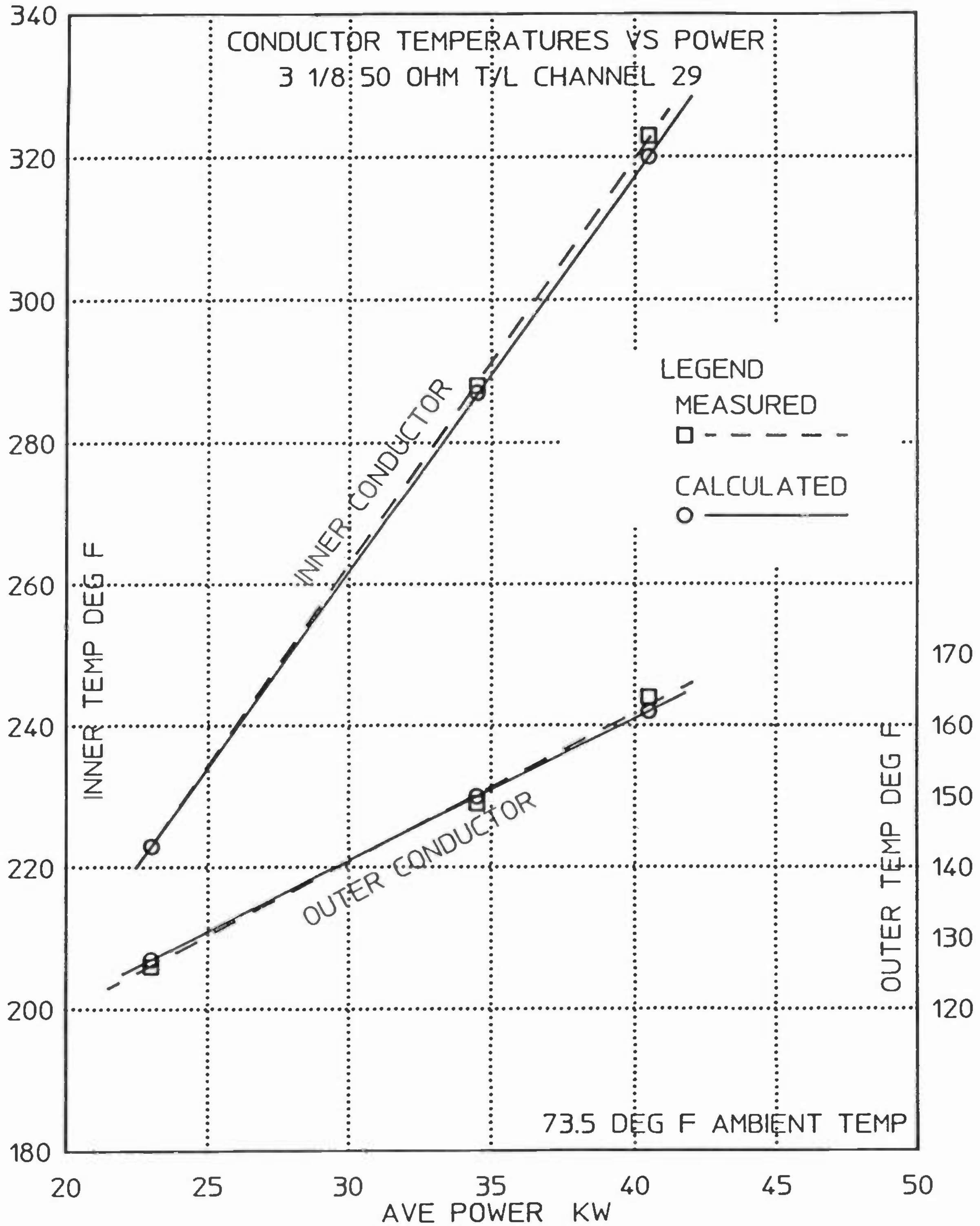
VERIFYING THE RESULTS

Comparing power tests to the calculated values give excellent results. For example, figure 2 shows calculated and measured temperatures for both the inner and outer conductors verses average power for a 3-1/8"-50 ohm line. A recent test at 100 MHz on 10"-50 ohm line with aluminum outers also showed excellent results. The results of these tests along with comparisons of results of the thermal model to past power testing has given us confidence in the accuracy of the thermal model.

As I mentioned earlier in the article, there are several factors in a thermal model which can cause considerable error if not addressed properly. These being thermal emissivity and electrical conductivity. Emissivity for copper can range from .04 for a clean polished surface to .80 for a black oxidized surface.[4] Choosing the wrong value could change the calculated value of radiated heat by a factor of 20. We have found that a value of .07 for clean copper lines is the proper value. Using a value of just .15 would increase the calculated power rating by 14%. The other factor, electrical conductivity, needs to be addressed. All copper is not the same. There are four types of copper tubing which are readily available today. Three of these are high conductivity copper with an electrical conductivity of 99% IACS to 101% IACS. These are CDA alloy numbers 101, 102 and 110. The fourth and most common is phosphorus deoxidized copper, CDA alloy number 122, and is commonly known as copper water tubing. Phosphorus deoxidized copper has an electrical conductivity of 85% IACS.[5] We already know which one is cheaper. In fact, high conductivity copper tubing costs 15% more than copper water tubing. The computer model tells us that if water tubing is used in place of high conductivity copper tubing, the average power rating will be reduced by 7% for both 50 ohm 75 ohm lines. Line losses will also be increased thereby reducing the amount of power delivered to the antenna. Dielectric uses high conductivity copper in all our transmission lines. Make sure your transmission line supplier does also.

NEW USES

Now that we have established what the power ratings are based on and are confident of the computer model, where do we go from here? We are now in a position to play a lot of "what if" games. Like, what if we painted the outside of the outer conductor with a high emissivity paint. We would want it to be a white color which would have the lowest solar absorptivity. Values of .85 for emissivity and .30 for solar absorptivity are attainable. Using a value of .85 for emissivity, the computer model tells us that the power rating can be increased by 28% for 50 ohm line and 23% for 75 ohm line with no increase in inner conductor temperature. What if, in addition to painting the outside of the outer conductor, we also improve the emissivity of the inside of the outer conductor and the outside of the inner conductor to a value of .85 without increasing losses. Here again, the model tells us that the power rating can be increased by a factor of 2.53 for 50 ohm line and by a factor of 2.38 for 75 ohm line. Painting the outside of the outer conductor to improve it's radiant heat transfer is indeed now feasible. A cost effective method for coating the inside surfaces to improve raditnt heat transfer between inner and outer conductors without increasing line losses has not yet been fully developed. This may not be too far off in the future.


CONDUCTOR TEMPERATURES VS POWER
3 1/8 50 OHM T/L CHANNEL 29
LEGEND
MEASURED
CALCULATED
INNER CONDUCTOR
OUTER CONDUCTOR
INNER TEMP DEG F
OUTER TEMP DEG F
73.5 DEG F AMBIENT TEMP
AVE POWER KW
340
320
300
280
260
240
220
200
180
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170
160
150
140
130
120

Fig. 2

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STRIPLINE TECHNOLOGY—FUNDAMENTALS AND APPLICATIONS IN HIGH-POWER TRANSMITTERS

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Abstract -- A stripline is a transmission line constructed using printed circuit techniques. Its propagation mode is similar to the transverse electromagnetic mode of coaxial transmission line. Because uniform transmission paths may be easily constructed, stripline construction is increasingly used for multiple parallel transmission of digital data and RF energy. Stripline is widely used in RF amplifiers, combiners and splitters because of its inherent simplicity, reliability, stability and lack of required maintenance.

INTRODUCTION

Television engineers work with many types of transmission lines. Coaxial cables are the most common, and waveguide is used for the transmission of microwave signals and some high-power UHF-TV applications. Less common, but becoming more widespread in RF transmission equipment, is a type of transmission line known as stripline.

The first part of this paper presents an overview of stripline theory. The second part presents practical applications of this technology in high-power television transmitters.

STRIPLINE FUNDAMENTALS

In its most common form, a stripline consists of a flat metal strip mounted above a ground plane as shown in Figure 1. Between the conductors is a dielectric or semiconductor; typically, it is an insulating printed circuit substrate.

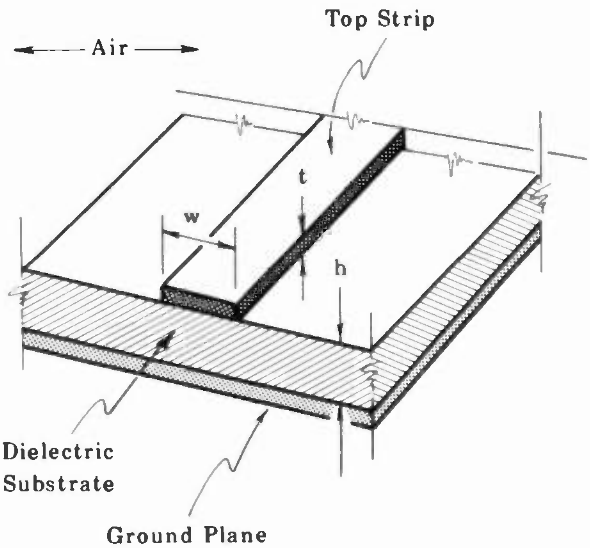

Figure 1. Stripline Geometry

Most transmission lines consist of parallel conductors. A signal is launched at one end and propagates to the other. A stripline employs the same electromagnetic concepts. The mode of propagation in a stripline is similar to the transverse electromagnetic (TEM) mode that is characteristic of coaxial transmission lines.

The analogy of stripline to coaxial line can be extended by making an imaginary conversion as shown in Figure 2. If a coaxial line is cut lengthwise and spread out flat, a stripline-like structure is created. Figure 2 also shows the relative electric and magnetic field distributions of coaxial lines and striplines. In stripline, the electric field lines are concentrated between the

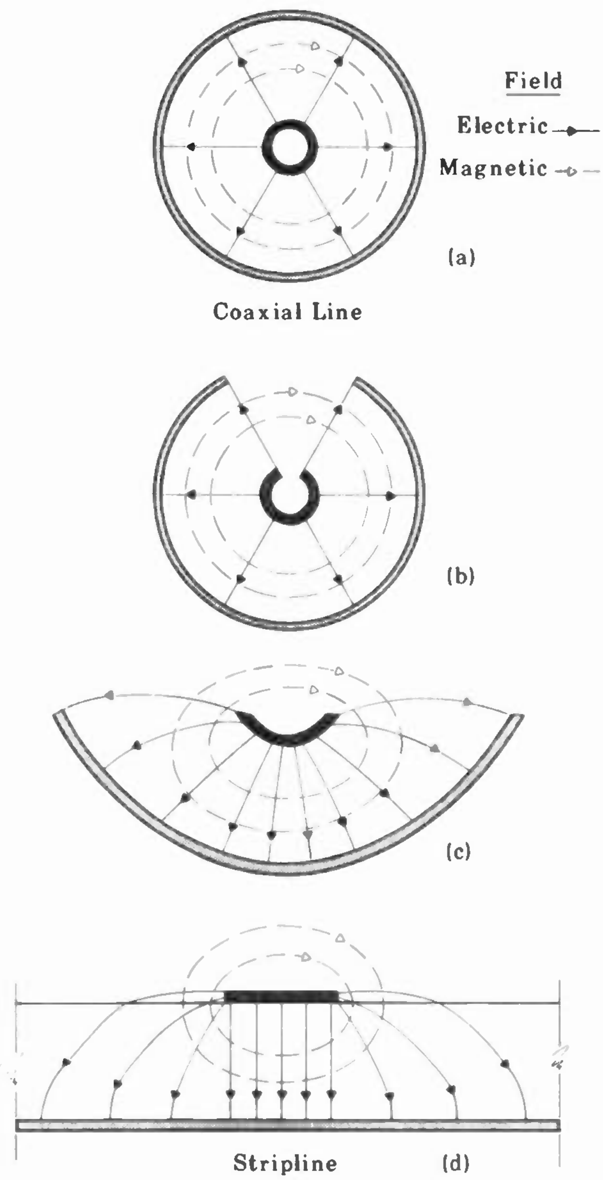

Figure 2.

Transformation from Coaxial Line to Stripline

conductors. The deviation from a true coaxial TEM mode is small and can be ignored for most circuit design applications, or taken into account through several numerical analysis and experimental techniques.

The characteristic impedance of stripline is a function of the strip width, strip thickness, strip distance from the ground plane, and the dielectric constant of the insulator. There are several equations and methods used to calculate the characteristic impedance of stripline by incorporating theory and empirical data. One expression is:

$$Z_0 = \frac{87}{\sqrt{\varepsilon_r + 1.41}} \ln\left(\frac{5.98h}{0.8w + t}\right)$$

where ε_r is the dielectric constant, h is the distance between the strip and the ground plane, w is the width of the strip, and t is the strip thickness. This expression is appropriate when h is less than $0.8w$. Typical values of the characteristic impedance of stripline vary from 50 to 150 ohms for ε_r = 2.9 to 5.23, t = 2.8 mils, w = 10 mils, and h = 8 to 67 mils (1 mil = 1/1000 inch). The dimensions can be any units as long as they are consistent within the expression.

Stripline circuits can be fabricated using printed circuit construction techniques. This allows for very uniform construction -- so uniform, that stripline techniques are used to meet the growing demands for higher speeds in high-speed computing and digital signal processing. These applications can range from parallel data paths on a circuit board connecting integrated circuits, to implementation on the chip itself to connect logic gates. In on-chip implementation, the stripline is simply etched on the semiconductor substrate. Care has to be taken to minimize radiation loss or interference due to nearby conductors. This is typically achieved in practice through shielding or by using a substrate with a high dielectric constant which concentrates the stripline fields to between the strip and the ground plane.

Another advantage of striplines is the simplicity of implementing combining and filtering circuits. Entire systems can simply be printed on a circuit board thus reducing the parts count. If it is necessary to mount active or passive devices, they can be soldered to the stripline without connectors or transitions. The open construction also makes it easy to modify the circuit after it has been constructed.

The raw material for stripline is usually provided in sheet form, and consists of a dielectric sheet with copper or aluminum laminated to both sides. Dielectrics used include glass-reinforced teflon, quartz-loaded teflon, polyolefin, polystyrene, and ceramic-filled resins.

There is no combination of conductors and dielectric that is optimum for all purposes. Trade-offs include dimensional stability with temperature and humidity, dielectric constant, thermal conductivity, structural strength, and chemical and mechanical processing constraints.

Stripline loss is a function of several factors, including frequency, geometry, and the electrical properties of the conductors and substrate. The dominant losses in stripline are dielectric loss in the substrate and ohmic skin loss in the strip conductor and the ground plane.

Power handling capacity is dependent on construction practice, ambient temperature, and thermal conductivity of the dielectric. Most striplines are enclosed, which can greatly affect power handling capability; enclosures can trap air and hinder cooling, or act as a conduit for airflow and increase cooling and power limits.

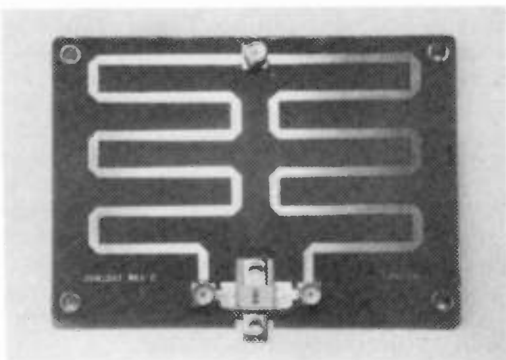

Figure 3.

Photograph of Wilkinson Hybrid Combiner

STRIPLINE APPLICATIONS IN HIGH-POWER TRANSMISSION

Stripline is widely used in RF amplifiers, combiners and splitters because of its inherent simplicity, reliability, stability, and lack of any maintenance requirement.

A Wilkinson hybrid combiner as shown in Figures 3 and 4 consists of two quarter-wavelength transmission lines and an isolation resistor. For a 50-ohm system, as shown, the quarter-wavelength lines have a characteristic impedance of 70.7 ohms which is the square root of the product of the source and load impedances. The isolation or balancing resistor is 100 ohms.

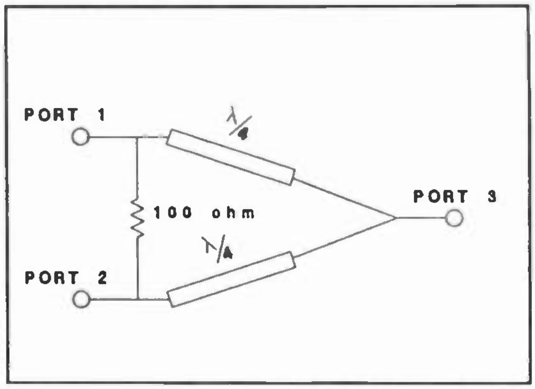

Figure 4.

Schematic of Wilkinson Hybrid Combiner

If two equal amplitude and in-phase signals are applied to ports 1 and 2, they will combine at port 3. The two quarter-wavelength lines operate as impedance transformers and step up the 50-ohm impedance of ports 1 and 2 to 100 ohms at port 3. Since the two lines are connected in parallel at this point, the output impedance becomes 50 ohms.

Also, because the signals at ports 1 and 2 are equal-amplitude and in-phase, there is no voltage differential across the resistor; no current flows, and no power is dissipated. Other than minor dielectric losses, all of the power from ports 1 and 2 flows to port 3.

As well as having the property of combining signals, the Wilkinson hybrid also isolates ports 1 and 2 from each other thus preventing any interaction between amplifiers connected to these ports. If a signal is applied to port 1 and no signal to port 2, power will flow from port 1 to port 3 through the quarter-wavelength line. Some of this power will also flow from port 3 to port 2 through the second quarter-wavelength line, undergoing a total phase shift of 180° in the process. Since the power at port 2 arriving from port 3 is 180° out of phase with the power from port 1 through the resistor, they cancel each other out.

A signal applied to port 1 will be attenuated by a factor of 100 or more before appearing at port 2. In other words, the isolation between the outputs of amplifiers connected to ports 1 and 2 will be at least 20 dB. This figure is easily achievable with this type of combiner.

With no signal on port 2, half of the signal applied to port 1 will be passed to port 3 and half will be dissipated in the resistor. For this reason, the resistor must be capable of dissipating one-quarter of the total input power.

The combiner shown in Figure 3 is used to combine two 125 W sync peak amplifiers. The total power dissipated in the resistor at black level with one input failed is 125 x .5 x .595 = 37 W. The resistor is rated at 75 W.

Figures 5 and 6 show a high-power stripline combiner used in LARCAN transmitters to combine the outputs of two equal-power amplifiers. This combiner is generally referred to as a 3 dB coupler. Because it is electrically symmetrical, it can be used either as a combiner or splitter.

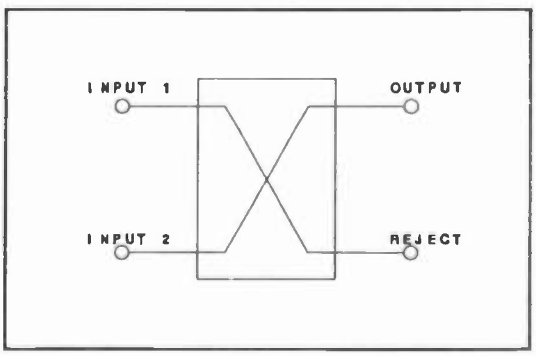

Figure 5. Schematic of High-Power Combiner

The coupler utilizes a one-quarter wavelength coupled section. Hence, the useful bandwidth is approximately one octave. The advantage of a coupler with this bandwidth is that only two are required to cover TV channels 2 to 13 inclusive. Coupling of the RF power signals is achieved through both capacitive and inductive coupling. When used as a combiner, the two equal-amplitude (90° phase offset) signals will combine to a single output port.

If one input signal is removed, the coupler acts as a splitter and half of the input power is fed to the output port and half to the terminated isolated port. For this reason, the termination must be sized to dissipate one-quarter of the total input power.

For the particular application discussed here, the spacing between the input ports on the coupler was determined by the physical spacing between the amplifier output ports. Because this spacing is less than a quarter wavelength, for the low-band combiner shown, the traces are folded as illustrated in Figure 6. The trace is one-quarter wavelength long at the design center frequency.

Mechanically, the two boards are supported along each edge with teflon support blocks. When assembled, the two halves overlay each other with a small air gap between the two parallel conductors. The input ports and the isolated output port use 1-5/8" 50-ohm connectors. The combined output is also 50 ohms and may be either 1-5/8" or 3-1/8" depending on the power handling requirements.

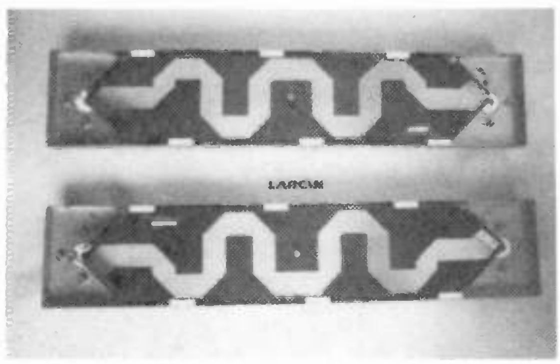

Figure 6. Photograph of High-Power Combiner

Brass angles are mounted both mechanically (rivets) and electrically (solder) to the end of each strip. To each of these angles, a brass output flange is mounted. These flanges provide both an area of capacitive tuning and a connection point for the external standard 1-5/8" or 3-1/8" output flange. A moveable grounded plate is fixed to the housing at each output port. These plates along with the brass flanges mentioned previously form an adjustable air gap capacitor for optimization of the unit during test.

Power handling capability was a major consideration in the coupler design. This was the primary reason for using the suspended substrate layout because it reduces the amount of lossy dielectric material within the coupler assembly. Since air is the only dielectric between the two coupled lines, dielectric losses are minimal.

In addition, the trace width of the stripline is sufficient to minimize conductor losses while still maintaining a reasonable enclosure size.

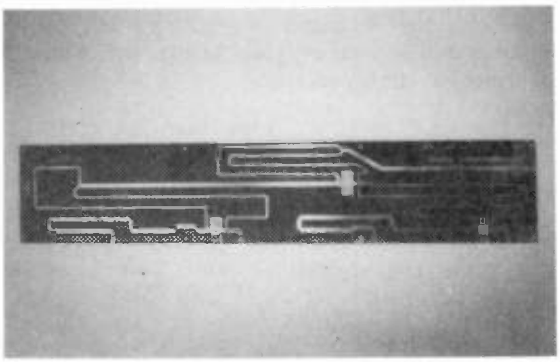

Figure 7. Photograph of 4-Way Combiner

Figures 7 and 8 show the 4-way combiner used on LARCAN 1 kW solid-state PA modules to combine the outputs of four 250 W amplifiers. The 4-way combiner comprises three Wilkinson hybrids as described previously.

Two amplifiers are combined in hybrid No. 1, and two in hybrid No. 2. The resulting two outputs are then further combined in hybrid No. 3 for a single 50-ohm output. 90° offsets were added to the combiner to accommodate the physical placements of the four amplifiers and to improve the output return loss.

A 4-way splitter with complementary 90° offsets and virtually identical to the combiner is located at the amplifier inputs. The splitter enables the four amplifiers to be fed from a single source.

Variations of the combiners and splitters described in this paper have been in use in LARCAN transmitters for approximately eight years and have proven to be reliable and stable.

SUMMARY

Though stripline differs in appearance from other transmission lines, it is electrically similar. The added advantages of the simplicity and convenience of stripline construction is making it more common in applications that require multiple paths of precise and reliable digital data and RF transmission. Stripline is used in RF amplifiers, combiners, and splitters. It has a record of reliable service in high-power transmitters.

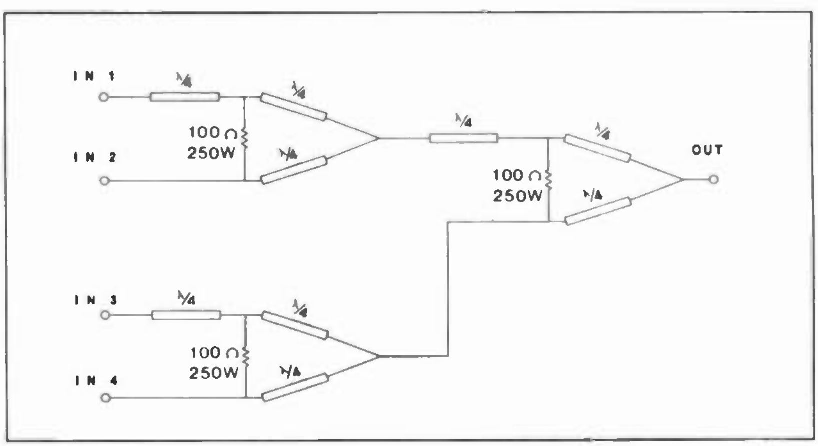

Figure 8. Schematic of 4-Way Combiner

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TELEVISION PRODUCTION AND POST PRODUCTION

Monday, April 15, 1991

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LARGE MULTI-CHANNEL WIRELESS MICROPHONE SYSTEMS

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INTRODUCTION

The use of wireless microphones has mushroomed in the past few years. This is due to advancements in technology, a trend towards greater mobility on stage, and the desire to control volume and equalization of individual performers. Consequently, installations where the number of wireless microphones, referred to as channels, being used simultaneously has increased dramatically. In the past, 6 channels seemed to be the limit. Now theatres and studios with large multi-channel systems, ten to thirty channels, are common. The largest installation to date is 44 channels being used in a theatre in Japan. Systems of this magnitude are a difficult engineering challenge. Careful planning, installation, operation, and maintenance are required.

FREQUENCIES

Manufacturers generally produce wireless microphones on very high frequencies (VHF) and ultra high frequencies (UHF) with specifications outlined by government agencies such as the Federal Communications Communications (FCC). Since the two frequency ranges have different associated wavelengths, they behave differently. The wavelength is inversely proportional to the frequency. Higher frequencies have shorter wavelengths. VHF frequencies (165-216 MHz) have a wavelength of approximately 2 meters. They exhibit good ability to bend around objects. UHF frequencies (450-960 MHz) have a wavelength of less than one meter. They have excellent reflective characteristics. They can travel through a long corridor, bouncing off the walls, losing very little energy. Due to its short wavelength, a UHF wave can sneak through small areas more easily. To take an extreme example, in a jail, the metal bars from a lattice or a "Faradays Cage" that will easily block a VHF frequency while a high UHF wave (950 MHz) is small enough, approximately 30 cm, to escape in between the bars. However, the shadowing effect is more critical in the UHF range. A small solid metal object could block a UHF wave while a VHF wave could probably bend around it.

On the practical side the UHF spectrum is less congested. VHF wireless microphones operate on frequencies used by TV channels 7-13. Frequencies should be chosen to avoid TV channels that are active in the city where the performance is taking place. This often presents a problem for a traveling performance company. Also computers and other machines emit frequencies that can cause problems in the VHF range. Furthermore, thousands of VHF systems are in use already and hundreds more are sold every month. All these units are beginning to crowd the VHF spectrum. Even though they are low power devices, the possibility of picking up someone elses performance on your system is increasing daily. The UHF range also has a larger share of the RF spectrum, about 500 MHz (450-952 MHz). UHF TV channels are less redundant. For example, only a few locations have channel 19 active. UHF stations are usually located on the outskirts of major cities, and generally have less transmitting power than the stations operating on the VHF channels. There is significantly less potential interference from machinery in the UHF range as well. However, UHF equipment is more expensive than VHF, with little difference in audio quality. It demands highly sophisticated RF design techniques with more stringent tolerances.

To summarize, if the system is to be installed in a fixed location, carefully chosen VHF frequencies are an economical choice. If the system is to be used by a traveling performance company or in a theatre that has saturated the VHF spectrum with wireless equipment, then UHF should be considered.

Frequency bands used by TV Channels

VHF

Channel	Frequency band
7	174 - 180 MHz
8	180 - 186
9	186 - 192
10	192 - 198
11	198 - 204
12	204 - 210
13	210 - 216

UHF

Channel	Frequency band
14	470 - 476 MHz
15	476 - 482
16	482 - 488
17	488 - 494
18	494 - 500
19	500 - 506
etc.	

TRANSMITTER CONSIDERATIONS

A radio frequency (RF) transmitter works like a miniature FM radio station. First, the audio signal of a microphone is subjected to some processing. Then, the processed signal modulates an oscillator, from which the carrier frequency is derived. The modulated carrier is radiated via the transmitter's antenna. This signal is picked up by a complementary receiver, via its antenna system, and is demodulated and processed back to the original audio signal.

Frequency Deviation

The modulation of the carrier frequency in a FM system greatly influences its audio quality. The greater the deviation, the better the high frequency response and the dynamic range. The trade off is that fewer channels can be used within a frequency range. However, since audio quality is usually the priority and the UHF spectrum has increased the number of available frequencies, wide deviation is most desirable.

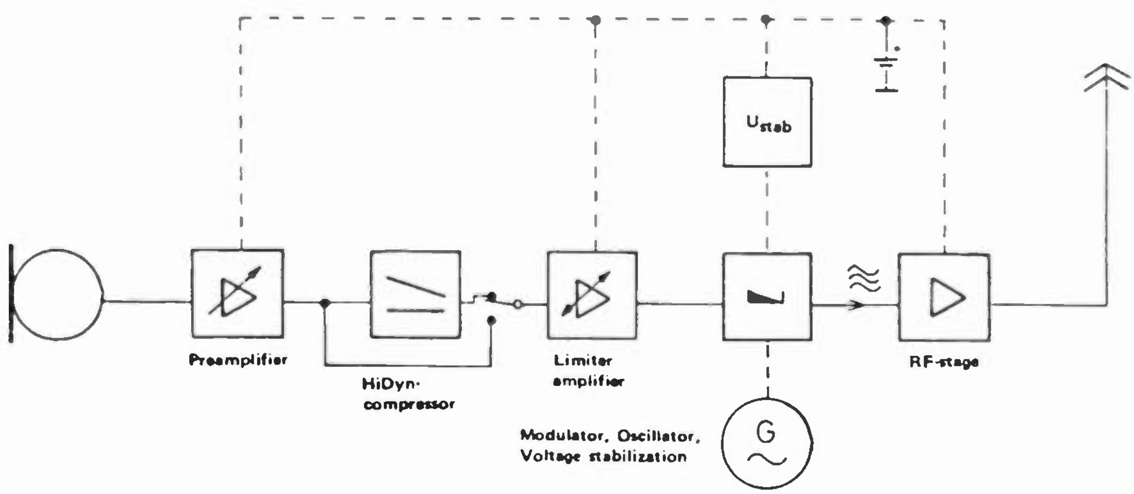

Figure 1. Block diagram of a transmitter

Power

Transmitter power is a rating of its potential RF signal strength. This specification is measured at the antenna output. The actual transmitted power is influenced by the efficiency of the antenna. Therefore, power specifications are of only limited use in assessing a transmitter's range, considering largely variable antenna conditions. Also, battery life is associated with RF output power. Increased power will reduce battery life with only a moderate increase in range.

DC-to-DC Converter

Transmitters should be designed to provide constant RF output power and frequency deviation throughout the event being staged. This can be achieved through the use of a DC-to-DC converter circuit. Such a circuit takes the decaying battery voltage as its input and regulates it to have a constant voltage output. Once the voltage of the batteries drops below a minimum level, the DC-to-DC converter shuts off, almost instantaneously. The result is a transmitter that is essentially either off or on. While it is on, the RF output power, frequency deviation, and other relevant specifications remain the same. Transmitters without regulation circuits, once the battery voltage begins to drop, will experience reduced range and the audio quality will start to deteriorate due to a degradation in the modulating signal.

Companding

Most wireless microphone systems use a companding (compressing/expanding) noise reduction system, similar to those used in recording studios and home stereo equipment, as well as a pre-emphasis/de-emphasis process to maximize signal-to-noise ratio, dynamic range, and transmission reliability. The transmitter pre-emphasizes, or boosts, the higher audio frequencies. The modulated signal is then compressed before being transmitted. This raises low audio levels sufficiently higher than the transmission noise and suppresses overmodulation. The receiver uses complementary expanding to restore the dynamic range and de-emphasis so that the overall response is linear.

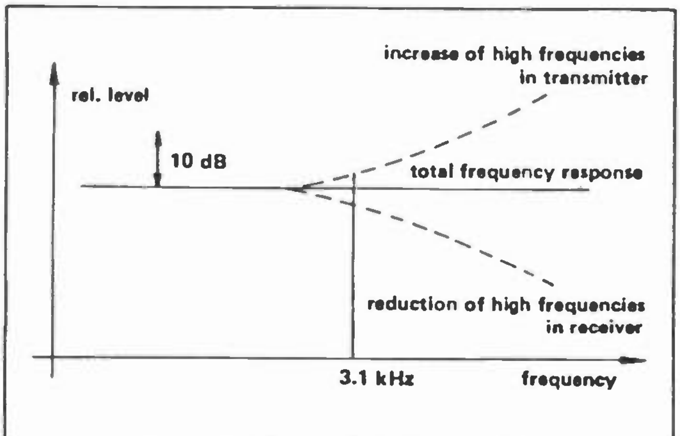

Figure 2. Preemphasis, deemphasis

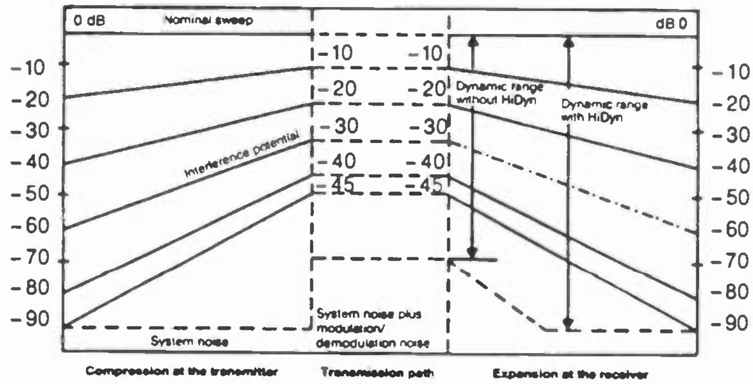

Figure 3. Companding diagram

Spurious Emissions

Apart from the wanted carrier frequency, transmitters also may radiate some unwanted frequencies known as spurious emissions. The carrier frequency is obtained by electronically multiplying a crystal frequency. A frequency is generated after each multiple. For example, a 25 MHz crystal is doubled to 50 MHz, which is doubled again to 100 MHz, which is doubled once more to the final carrier frequency of 200 MHz (8 times the original crystal frequency). However, spurious signals every 25 MHz might be generated. For large multi-channel systems these spurious frequencies cannot be ignored. They can be significantly reduced through elaborate filtering and contained by using a well constructed, RF "tight", metal housing for the transmitter. This metal casing should not have any slits since RF can leak through them. Therefore, it ought not be two half shells screwed together, it should be one molded piece. Small round holes in the casing are acceptable since RF cannot escape through them. They can be employed for access to adjustment locations on the circuit board. Also, an RF tight transmitter is less susceptible to outside interference. Despite this precaution, choosing a frequency for a new channel that falls directly on one of these possible spurious emissions should be avoided.

A metal housing is not only important for its shielding properties but also its durability. These devices usually experience much more abuse by actors and other talent than anyone ever predicts.

Antenna

A transmitter antenna should be tuned to its carrier frequency. There are generally two types, the long straight "whip" antenna and the coiled "rubber duck" antenna. The whip antenna is tuned by cutting it to 1/4 of the wavelength of the carrier frequency. The duck antenna is used on VHF transmitters to stifle objections to the length of a whip antenna necessary for the VHF range. It achieves its radiating efficiency along its shortened length over a much narrower frequency range. The coiling of the antenna wire concentrates its tuning elements. The tuning is sharply influenced by close proximity to conductors. On body pac transmitters, the duck antenna tends to be stiff and rest against the user's body. Since the human body is largely composed of water and salt, it is quite a good conductor and could easily detune the duck antenna. Therefore, the whip antenna is recommended for body pac transmitters. If a duck antenna must be used, it is recommended to bend it slightly so it does not rest against the performers body. UHF antennae are short enough that there are generally no objections to the ¼ wave whip antenna.

Handheld transmitters are often designed with their antenna incorporated on their circuit board under the outer housing. This design is not efficient because the performer's hand will absorb some of the radiated energy. It cannot be implemented with a metal, RF tight housing either.

RECEIVER CONSIDERATIONS

An ideal wireless microphone receiver would capture the carrier frequency of its corresponding transmitter and reject all other signals. Short of this unrealistic expectation, a receiver should be designed to capture its carrier, reject most other signals, and avoid mixing its carrier with the other signals it does pick up. When designing a multi-channel system, the "non-ideal" characteristics have to be overcome. These include widely varying RF signal levels, intermodulation, frequency spacing, and spurious oscillator frequencies.

RF Signal Level

Varying RF signal strength is mainly due to multi-path propagation, absorption and shadowing. These are familiar difficulties also experienced with car radios in cities.

Audible effects due to low RF signals, known as dropouts, can occur even at close range to the receiver, due to multi-path propagation. Some of the transmitted waves find a direct path to the receiver antenna and others are deflected off a wall or other object. The antenna detects the vector sum, magnitude and phase, of direct and deflected waves it receives at any particular instant. A deflected wave can diminish a direct wave if it has

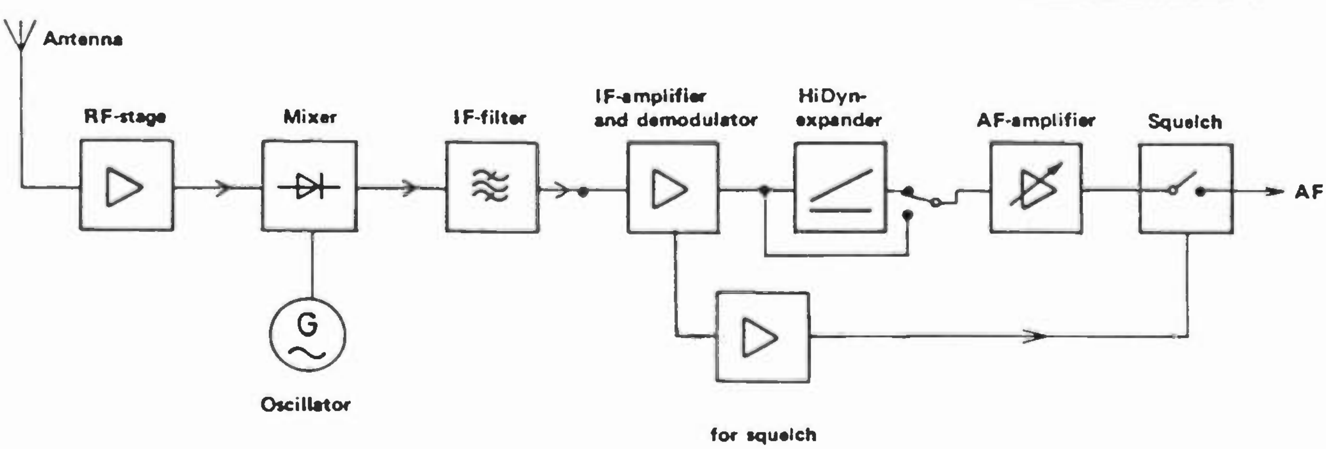

Figure 4. Block diagram of a receiver

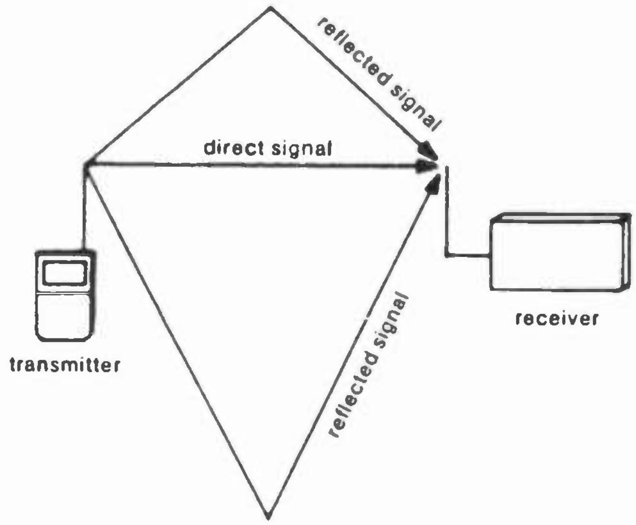

Figure 5. Multi-path propagation

different phase, resulting in an overall low signal. This difference in phase is due to the longer path a deflected wave travels between the transmitter and receiver antennae and any phase reversal occuring when it hits an object. Obviously, this phenomenon needs to be addressed in an indoor application. It is less critical outside.

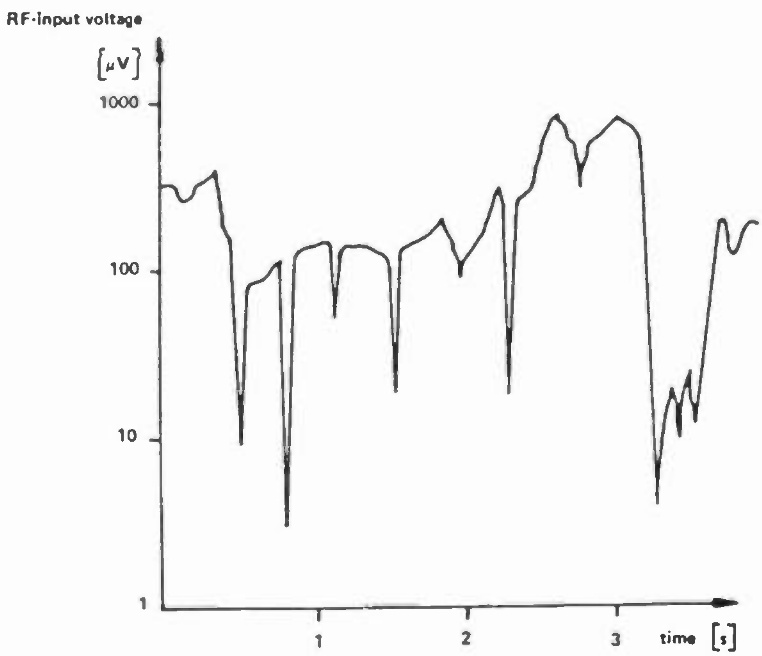

Figure 6. Antenna input voltage of a receiver when moving the transmitter.

Considering only the multi-path propagation effect, figure 6 shows a typical curve of the signal strength at a receiving antenna. The variation inside a building with reflecting walls is 40 dB or more.

RF energy can be absorbed by non-metallic objects resulting in low signal strength. As stated previously, the human body absorbs RF energy quite well. It is important to place antennae correctly to minimize this effect.

Shadowing occurs when a wave is blocked by a large obstacle between the transmitter and receiver antennae. This effect can be minimized by keeping the receiver antenna a distance of ½ a wavelength away from any large or metallic objects.

These problems are addressed by a diversity receiver. A diversity system is recommended even if only one channel is in operation. Large multi-channel systems are

Functional diagram of a diversity receiver

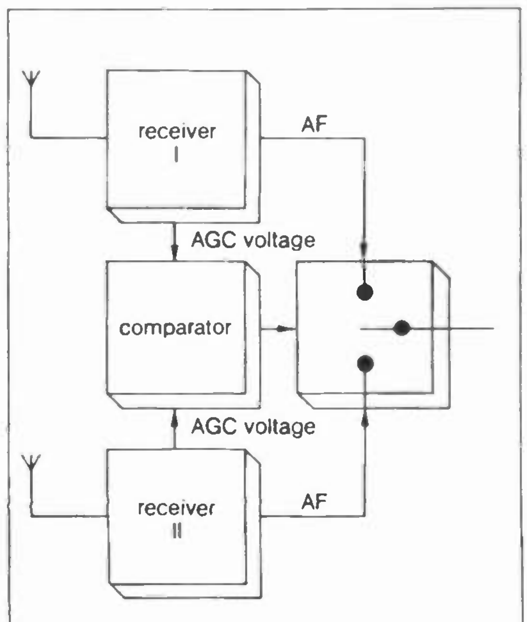

Figure 7.

only possible with diversity operation.

Comparing the different diversity systems, the "true" diversity has proven to be the most reliable design. This design has two independent receivers (usually incorporated within a single housing), each with its own antenna, with a logic switch between them. The logic switch constantly monitors the RF field strength as seen by

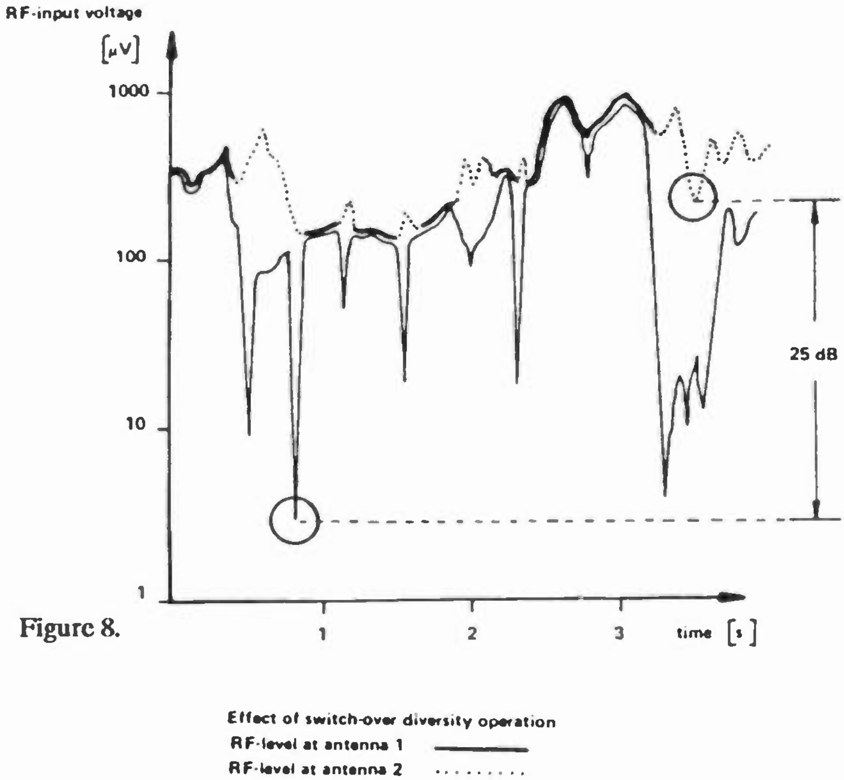

Figure 8.

each receiver. The receiver with the highest RF level is used for the audio output. This switching action can occur very fast and often, especially in UHF equipment. It is not audible in a well designed receiver. It is difficult to define an absolute value for the improvement when using this type of receiver. It can only be determined by statistical methods. Figure 8 shows the improvement when using a true diversity system. It has a similar effect of amplifying the wanted signal at least 25 dB.

Another method of diversity is "antenna" diversity. This incorporates a single receiver with multiple antennae, usually three. Using this method, the signals from each antenna are summed together. It assumes that if a weak signal is detected at one antenna, the sum of the other two will provide a sufficient signal. The problem is that both the phase and the amplitude of the signals are summed. Due to phase cancellation, the summed signal is often lower than the signal seen by a single antenna. Another problem is that each antenna needs an amplifier to keep it electrically independent from the others. These amplifiers can be a source of intermodulation.

A third technique is known as "phase" diversity. With this method, one receiver and two antennae are used. If the signal drops below a certain threshold the receiver switches the phase of one of the antennae. This method assumes that a low RF signal is due to multi-path propagation. However, it may be due to shadowing or absorption. Switching the phase may aggravate a multi-path propagation problem.

Intermodulation

Intermodulation is the result of two or more signals mixing together to produce a sum or difference signal. It is a common misconception that intermodulation is produced by the carrier frequencies mixing within the air. Intermodulation occurs within non-linear active components, such as transistors, exposed to strong RF input signals. This usually happens in the RF section of the receiver or in antenna amplifiers. In multi-channel operation, when several RF input signals exceed a certain level, the intermodulation products grow very quickly. There are different levels of intermodulation defined by the number of addition terms. Each addition term (f1, f2, etc.) represents a carrier frequency:

IM2 Products:
$f1 - f2 = IM2$
where $f1 <> f2$

IM3 Products
$f1 + f2 - f3 = IM3$
where $f1 <> f3$ and $f2 <> f3$

IM4 Products
$f1 + f2 - f3 - f4 = IM4$
where $f1 <> f3$, $f1 <> f4$
$f2 <> f3$, $f2 <> f4$

IM5 Products
$f1 + f2 + f3 - f4 - f5 = IM5$
where $f1 <> f4$, $f1 <> f5$
$f2 <> f4$, $f4 <> f5$
$f3 <> f4$, $f3 <> f5$
...etc.

Only the odd order intermodulation products need to be considered since the even ones are out of the frequency range of concern. Figure 9 shows typical intermodulation products caused by two strong input signals. The frequency of a new channel should be carefully selected

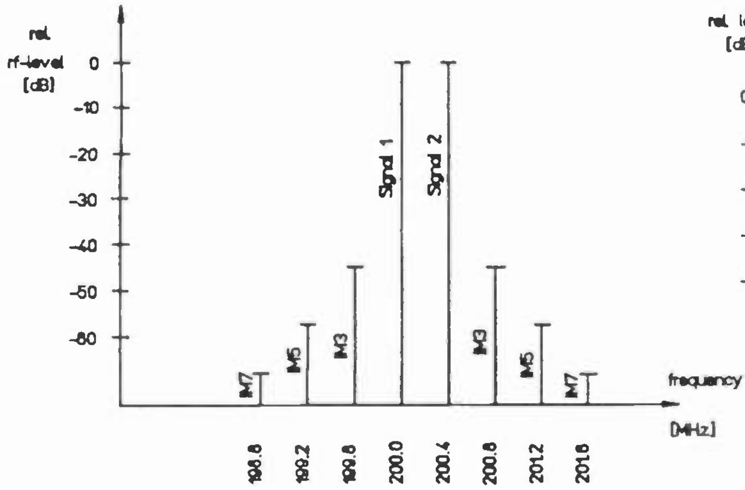

Figure 9. IM products of two strong signals

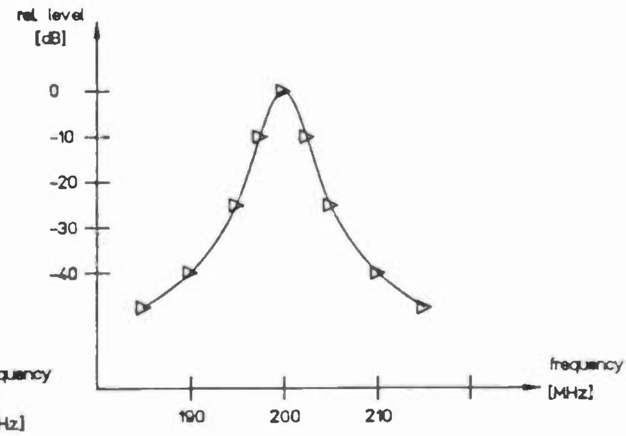

Figure 10. Filter curve of a 3 stage helical filter at the front end of the receiver.

to avoid intermodulation products of the other signals.

Equipment can be designed to minimize intermodulation. A specification known as intermodulation rejection is a measure of the RF input threshold before intermodulation occurs. For a well designed receiver, this specification will be 60 dB or greater. An intermodulation rejection of 60 dB means that intermodulation products are generated at input levels of approximately 1 mV. The highest quality multi-channel receivers presently available feature an intermodulation rejection of 80 dB.

Another important design feature is selective filters. Filtering out signals other than the wanted carrier frequency is no easy task. The filtering "window" should be as narrow as possible. This can be achieved through the use of helical filters in the first stage of the receiver. Figure 10 shows the curve of a third order helical filter of a modern VHF receiver. Strong input signals 5 MHz aside the receiving frequency are attenuated by at least 20 dB.

Present technology does not allow the use of highly tunable receivers in large multi-channel systems. The filtering window of this type of receiver has to be wide enough to accept all the frequencies it can be tuned to. This wide window is an invitation for unwanted signals to get into the receiver and cause intermodulation. A receiver with more than one frequency is acceptable as long as the difference between the highest and lowest frequency is only two or three MHz so that helical filters can still be employed.

Despite these precautions, frequency coordination must be done. Only the 3rd and 5th order intermodulation products need to be considered with most equipment. The higher odd ones are too weak to cause problems. If high quality receivers are used, having an intermodulation suppression of 60 dB or greater, only the 3rd order products need to be considered.

The distance between an intermodulation product and a carrier frequency should be kept to a maximum. A theoretical minimum safe distance can be determined by considering two criteria. First, an intermodulation product

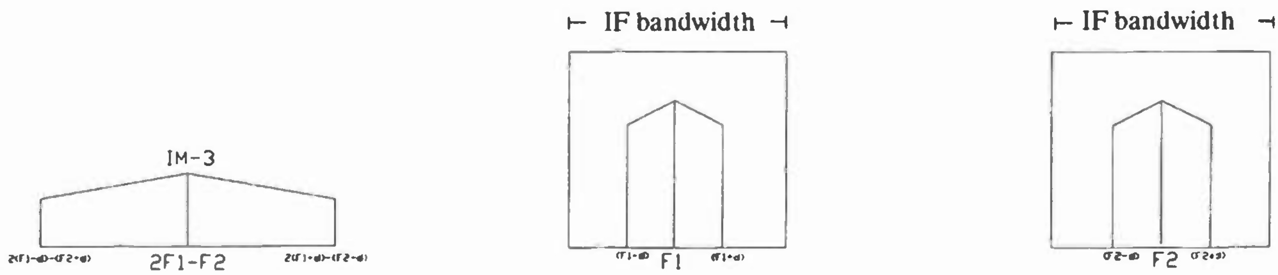

Figure 11. Two carriers with deviation=d and one of the generated IM3 products.

should not enter the final filtering stage, the intermediate frequency (IF) bandwidth, of the receiver. Secondly, since the carrier frequencies are being modulated, the bandwidth of an IM3 product, by nature of the algebra listed previously, is three times the bandwidth of the carriers. If full modulation is assumed, the bandwidth of an IM3 product is three times the maximum frequency deviation of the carriers. Therefore, the minimum safe distance regarding IM3 products is three times the maximum deviation of the transmitters plus half the IF bandwidth of the receiver. This is the theoretical ideal, however. Often, full modulation in the transmitters is not achieved. Therefore, the IM3 products would not be as wide. The practical minimum safe distance is often a debated subject. Nevertheless, it is recommended that IM-3 products should be 250 KHz away from any carrier frequency. If IM5 products are to be considered, they should be assumed to have a bandwidth of five times the maximum deviation of the transmitters.

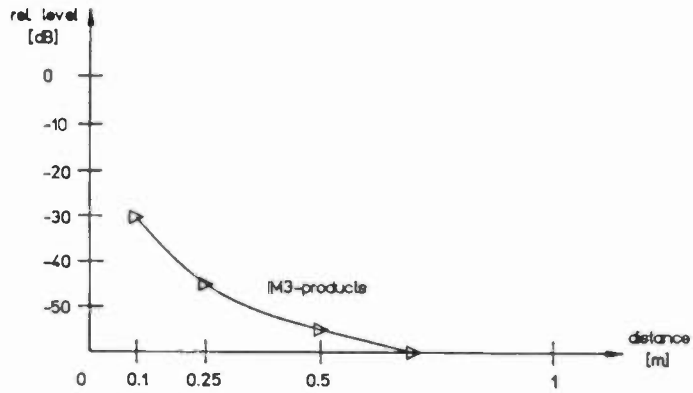

Figure 13. IM3 products generated by two body pac transmitters when approaching each other.

Intermodulation products are not only generated in receivers. Transmitters also have antennae which tend to pick up other signals. When these signals pass in a reverse fashion across the output filter of the transmitter, they are fed to a non-linear component: the output stage transistor. In this way, transmitters can generate intermodulation products themselves. Figure 12 shows the intermodulation products of two hand-held transmitters (30 mW each; 200 MHz; 200.4 MHz) used at a distance of 1 meter from each other. With body pac transmitters the problem becomes less critical since the antenna is close to the body. Figure 13 shows the transmitter intermodulation products of two body pac transmitters (30 mW each; 200 MHz; 200.4 MHz) when the distance is varied. This shows that actors with body pac transmitters can come rather close to each other without significant problems of transmitter intermodulation products. The situation changes dramatically, if several transmitters, still in operation, are put side by side on a desk. This mistake must be avoided. A highly selective output stage in the transmitter should be incorporated to minimize these problems.

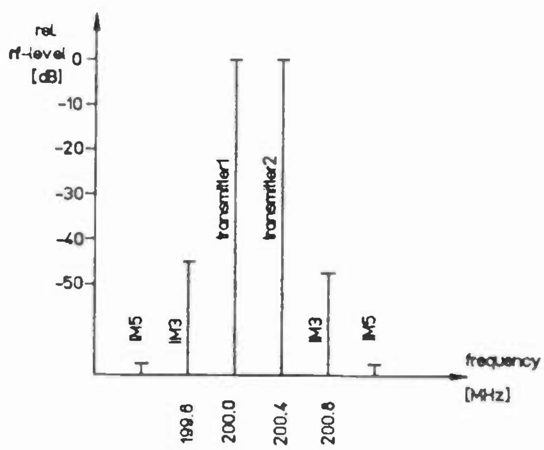

Figure 12. Intermodulation products generated by two handheld transmitters.

Frequency coordination can be extremely complex. It requires an appropriate computer program. For a six channel system, for instance, 90 IM3 products have to be taken into consideration. For twenty channels this figure grows to 3,800. As shown in figure 14, the necessary RF bandwidth rises exponentially as the number of

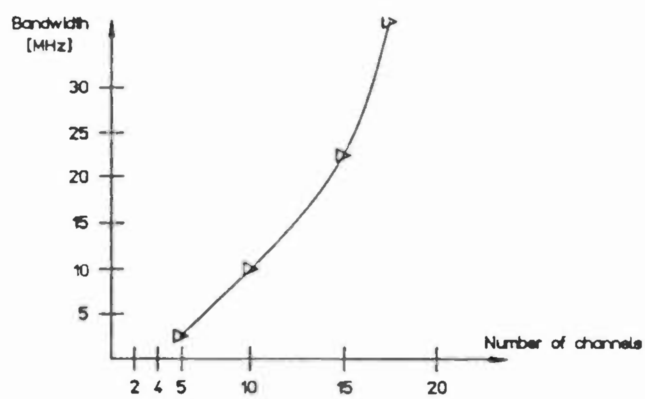

Figure 14. Necessary bandwidth of a multi-channel system if only IM3 products are taken into consideration.

channels is increased. This graph only considers IM3 products. The other constraints that need to be considered make the graph even more dramatic.

External disturbing sources such as TV transmitters, taxi services, police services, digital equipment, etc., also have to be taken into consideration. Fortunately, the screening effect of buildings is rather high (30 to 40 dB for VHF carriers). For indoor applications, this effect keeps strong outside signals at low levels. A significant problem can occur when poorly screened digital equipment is working in the same room. These wideband disturbing sources are able to interfere with all VHF channels. The only solution to this problem is to replace the poorly screened piece of equipment with a better one.

Spacing

In order to have a defined channel, without crosstalk and with an intermodulation safety gap, a minimum spacing of 300 KHz between carrier frequencies should be employed. A wider spacing is even more preferable since many receivers often exhibit desensitized input stages in the presence of closely spaced signals. However, caution should be used when linking receivers with widely spaced frequencies to a common set of antennae. The frequencies need to be within the bandwidth of the antennae.

Local Oscillator

Receivers contain one or two local oscillators (single conversion or double conversion). In most VHF systems it is 10.7 MHz below the carrier. A small part of the oscillator energy could be radiated via the antenna or via the housing. Although this energy is small it is not negligible. When the receivers are connected to each other through an antenna system, this potentially dangerous frequency will find access to the input stages. This must be considered in the computer program. The difference between two carriers should never be equal, or even close to this oscillator frequency. A safety margin of 200 KHz is recommended. Another related frequency, the image frequency, two times the local oscillator, should be avoided in the same way. To minimize this problem, high quality receivers apply a double screening. Inside an all-metal housing, hermetically sealed metal boxes contain the complete RF circuitry. This technique reduces the spurious emission by 20 dB.

Antennae

A good receiver antenna system is extremely important. There are several types of receiver antennae available. Similar to microphones, there are omnidirectional antennae and directional ones. There are far more omnidirectional antennae in use presently. However, in areas that are saturated with RF equipment, directional antennae become more attractive.

Omnidirectional antennae are generally tuned by cutting them ¼ wavelength of the operating frequency. This type includes the "rabbit ears" seen in the majority of systems. These are attached directly to the receiver. This is simply a monopole or Marconi type of antenna and is generally reliable.

A more sophisticated antenna is a remote ground plane antenna connected to the receiver by a coaxial cable. Besides having a main radial to pick up the signal, it has at least three others that form a virtual ground plane which protects the main radial from potentially interfering deflected waves bouncing off the closest large reflective surface, usually the floor. If the antenna is mounted from the ceiling, it should be turned upside down since the ceiling is more of a threat than the floor.

An antenna has a bandwidth. It is sensitive to the frequency that it is tuned to while it attenuates other signals. For a single receiver, it is desirable to have a very narrow bandwidth. For larger systems where several diversity receivers are linked to one set of antennae, it is necessary for the antenna to have a bandwidth that includes all the frequencies in operation. The bandwidth of an omnidirectional antenna can be broadened by increasing the diameter of its radials.

A directional antenna, similar to a cardioid microphone, is more sensitive to signals arriving from the front and attenuates signals from the rear. This is an excellent choice for a fixed installation where other nearby venues have wireless systems as well. An example is a theme park, especially if it has outside theatres. By carefully aiming these antennae, one can provide RF pick-up of the intended stage and reject the potentially interfering signals from other areas. This type of antenna is also tuned but generally has a broader bandwidth, an advantage with a large multi-channel system. However, correct installation is critical. They are larger than the omnidirectional type, need to be distanced farther from potentially blocking objects, and aimed in the correct direction. They can not be disassembled and neatly packed like omnidirectional ground plane antennae. These disadvantages are more pronounced with a VHF version. A UHF version is much more compact.

Polarization refers to the direction of the electric field of a transmitted wave. It is best to have the transmitter and

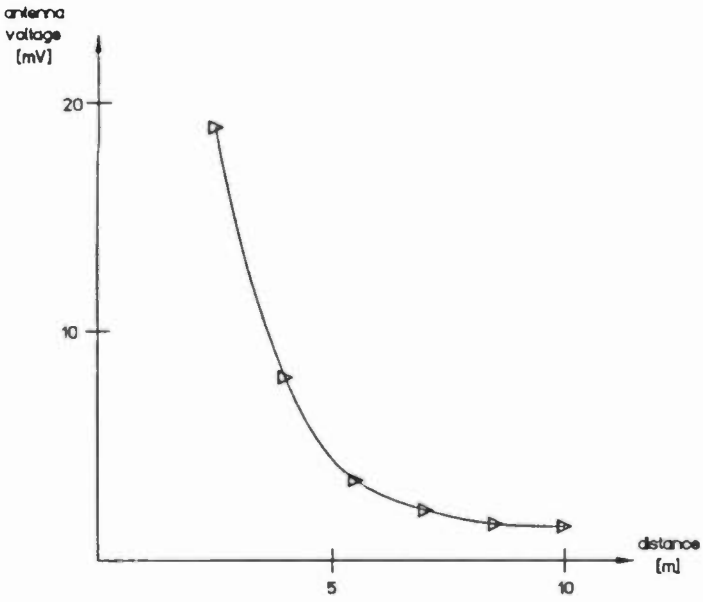

Figure 15. Antenna voltage as a body pac transmitter approaches it. (30mW, 200 MHz)

the receiver antennae polarized, which means oriented in the same direction, both horizontal or both vertical. However, reflected waves often change their polarization slightly. This is why the receiver antennae in a diversity system are often angled approximately 45 degrees. The worst condition, the transmitter antenna and the receiver antennae at a 90 degree angle to each other, should be avoided.

To prevent the receivers from getting unacceptably high input levels, the receiving antenna must be installed at a minimum distance to the transmitters. Figure 15 shows what happens, if a 30 mW bodypack transmitter in the 200 MHz range comes close to a receiver antenna. The receiving antennae should be positioned at a minimum distance of 6 meters (20 feet) from the transmitters. This condition is of high importance for good operation of large multi-channel systems.

__Splitter Systems__

Ideally, each diversity receiver should have its own set of antennae tuned to the frequency that it operates on. However, this is often cumbersome and unnecessary. One can still obtain optimum performance by operating several receivers from one set of antennae within the same frequency range. To accomplish this a splitter system needs to be used. However, signal loss between the antennae and the receivers needs to be considered.

The two major sources of signal attenuation are line loss and splitter loss. As a signal travels down a cable, some of its energy dissipates. The amount of the signal loss is directly proportional to the conductivity and the length of the cable, as well as the carrier frequencies traveling through it. Higher frequencies in the UHF range are attenuated more than VHF frequencies. Therefore, if long antenna cables are needed, low-loss cable or an in-line amplifier, or both is recommended. If amplification is to be used, usually 10 dB will be sufficient. Higher amplification invites stray signals to be picked up and can aggravate intermodulation. The amplifier should be positioned near the antenna to obtain the best signal-to-noise ratio. Splitter loss should

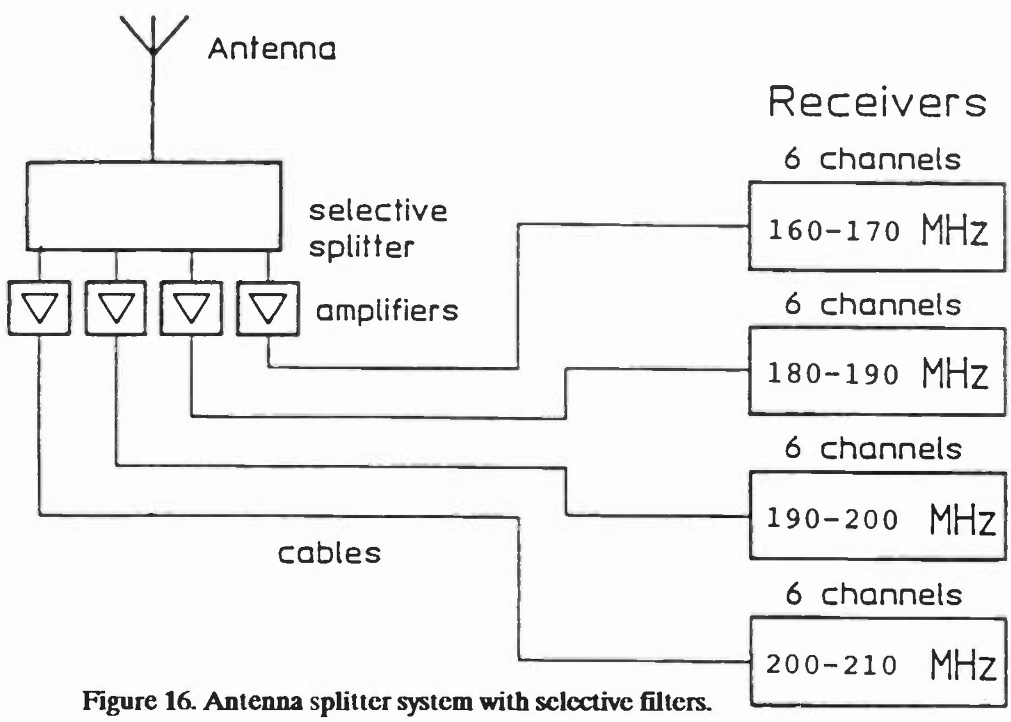

Figure 16. Antenna splitter system with selective filters.

also be addressed. Each receiver that is added to the antenna system requires another split to be made. Every time a split is made, some signal is lost. Therefore, an active splitter should be considered, especially if no previous amplification was used. An active splitter, as opposed to a passive one, is a powered device that incorporates amplification. Any amplifiers used must be of high quality, and should just compensate for the cable and splitter losses.

For additional security from interference, selective filters should be used in the splitter system. If an RF bandwidth of 40 MHz is available for a twenty-four channel system, the bandwidth can be divided into four subgroups of 10 MHz, and the twenty-four channels can also be divided into four groups of six channels. The subgroups can be separated from each other by highly selective RF-filters. The subgroups then become nearly independent of each other. In this way, any non-critical coordination violations between frequencies in different subgroups can be ignored.

CONCLUSION

Large multi-channel systems demand excellent planning, especially in the initial phase, and good technical support. Observing all the above mentioned items, perfect operation of a system can be guaranteed, even under difficult conditions.

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Graphs provided by:
Hans Kuhn, Sennheiser electronic KG, Wedemark Germany

TRENDS IN ELECTRONIC GRAPHICS EQUIPMENT

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ABSTRACT

Are there common themes in the progress of electronic graphics equipment? The technology has matured. The quality of graphics has improved, but broadcasters and their audiences have come to expect better quality. At any given performance level, the cost of the equipment is decreasing, but so are station capital budgets. What are the trends in electronic graphics equipment, and how can we, as broadcast engineers, take advantage of these trends in implementing graphic systems? This paper will explore these issues.

INTRODUCTION

The technology for electronic graphics in television is maturing, just has the technology for television itself has matured.

It used to be considered black magic and something unusual to be able to type characters on a keyboard and have them appear on the screen as block letters.

The state of the art has progressed, driven by developers working specifically for the broadcast market. Development of the first character generators, still stores, and paint systems were driven by needs of the major networks. The technology trickled down to local stations.

We went through a period over the last 20 years where electronic graphic systems for television had to be exotic, expensive, customized equipment. It had to be customized because very few systems needed to be produced. The equipment also tended to stretch the limits of commonly available electronics and computer technology. Exotic, customized equipment is, by definition, expensive. Now, we routinely generate, manipulate, and animate, full color, realistic images. As broadcasters, we do this every day. These tools, when used correctly, have enabled us to communicate more clearly, and with better aesthetic results.

OVERALL TRENDS

As we look at what is happening to all types of electronic graphics equipment, four major trends emerge. These are:

1. Software-based systems are emerging which run on off-the-shelf hardware platforms.
2. Networking is connecting electronic graphics systems with each other, and with other computer systems.
3. Mass storage cost has decreased, allowing more images to be practically stored.
4. The desktop publishing revolution is beginning to impact TV electronic graphics.

Standardized Hardware Platforms

Advances in computer and imaging technology, often in fields unrelated to broadcasting, make available off-the-shelf hardware platforms powerful enough to do electronic graphics. We're seeing software running on PCs, Macintosh, and Amiga Platforms that matches and exceeds what was done on the expensive, proprietary boxes.

Advances in applications for PCs and other standard platforms are not being driven by the broadcast industry, but by the electronic publishing, CAD CAM, and industrial imaging markets. These markets are much larger in terms of dollars and unit sales when compared to professional broadcasting.

The concept of running graphics software applications on standard hardware platforms also tends to blur the lines between different types of electronic graphics equipment. The same computer serves can serve as a character generator, still store, paint system, or 3D animation system, depending on the software. Features of the computer operating system could be used to bounce back and forth between functions and move images between applications. The hardware itself becomes non-specific to the type of application running on a graphics workstation.

Networking

Networking is another theme in the ongoing development of electronic graphics equipment. There is definitely a need for some kind of connection between electronic broadcast equipment, as well as to other computers in a broadcast plant.

In the dark ages when there was no connectivity between systems, images and text either could not be transferred, or had to be recaptured as video to transfer between one system and another. At a higher level of connectivity, data might be hand-carried on disks from one electronic graphic system to the next.

Now, the trend is toward interconnecting electronic graphic systems with each other via local area networks (LANs). What is also emerging are the benefits of extending the network to general purpose (non-electronic graphics) systems, such as newsroom computers.

There are various ways networking can be used in graphics applications. In the case of still stores and paint systems, pictures can be moved between nodes on the network. This is particularly useful for moving images between where they are being prepared, and where they are presented. The ability to prepare images in one location and have them show up somewhere else also is useful for character generators. This sort of networking between graphics equipment was first executed on a proprietary level by electronic graphics manufacturers - a connection between each manufacturer's family of equipment.

The concept of open architecture has forced the networking issue. Common standards now exist for data exchange on a network are now well established in the computer world. The use of standardized hardware platforms such as the PC, Mac, and Amiga for electronic graphics immediately leads to networked connectivity.

The manufacturers of electronic graphics equipment using proprietary hardware have responded by providing standardized network connections for their boxes. Often, standardized systems are used for data translation and connection to the network. The end result is that it doesn't matter how one connects to a network, as long as there is networking capability.

Network Media - Various network media are being used to connect graphic systems together:

Ethernet provides a 10 megabit-per-second data transfer rate, and is being used successfully for the exchange of full color still images between network nodes. Ethernet is a widely accepted networking standard which has been implemented on many different computer systems.

However, Ethernet is not the fastest (in terms of data transmission rate) way to interconnect electronic graphics systems, so some manufacturers are using SCSI bus, or serial composite digital video to speed up image data transmission. SCSI bus data transmission is becoming standardized in the general purpose computer world, but is inherently a short distance medium.(Fiber-optic SCSI extenders may circumvent the distance limitation). Serial composite digital video can move images at up to real-time speed, but is not a universal, general purpose data exchange medium. A serial video backbone has been used along side ethernet to speed up image transfer between network nodes.

Whatever the network medium, networking is part of the present and future of electronic graphics systems. When standard hardware is used, the type of networking can change and evolve independent of the other considerations.

Improvements in Mass Storage

With a typical size of one megabyte each, video images eat up mass storage. Improvements in mass data storage peripherals have been of direct benefit to electronic graphic systems. The cost per megabyte of storage has come down. 5 1/4" hard disks with 300 megabyte capacity are now commonly available. Magneto-optical read/write disk cartridges have become the de facto standard of convenience for removable image storage in paint systems.

Desktop Publishing Fallout

The desktop publishing revolution in the PC and Macintosh platforms has created a fallout of standards applicable to broadcast graphics.

The use of a page description language (PDL) for exchange of graphics layout information has several advantages which are beginning to be applied to TV graphic systems. Graphics engines which turn page description language into images have the intelligence necessary to eliminate the need for operator intervention in positioning and kerning text. CGs are beginning to appear with this capability.

With the kind of open data architecture already in place in the desktop publishing world, many kinds of image data could be imported into electronic graphic systems which accept the standard file formats. Some examples are:

Typefaces - The widespread use of desktop publishing has made available a variety of vector-based typography which can be scaled to any size without degradation.

Images - Various standard file formats for images exist, ranging in complexity from small monochrome clip art, to 32 bit full-color images in resolutions far exceeding broadcast television.

Databases - Databases which can be used to create images, such as maps, can be imported into electronic graphic systems.

SPECIFIC EQUIPMENT TRENDS

Character Generators

There are several basic trends in character generator systems now being produced:

Price/Performance - The performance which can be expected of character generator systems versus the acquisition cost is improving.

Quality of Typography - High quality, anti-aliased type derived from foundry fonts have come to be the norm in character generators. Kerning algorithms and kerning data are used to further enhance the quality of the typography.

Speed of Operation - Increased CPU power has sped up all operational functions. Many character generators are now capable of sizing type in real time, or close to real-time.

Internal Background Handling - Character generator displays can now incorporate complex backgrounds which can be captured from video, generated internally, or imported as data from electronic paint systems.

Real-time Animation - The variety of real time effects has improved, ranging from canned transitions, to manipulation of characters in 3D space.

How can we benefit from the improvement trend in character generators? The price for any given level of performance is coming down. For most broadcast use, speed of operation, quality of typography, and perhaps connectivity to other systems are of primary importance. The most fancy features, such as instant type sizing and real-time animation may not be relevant to daily use, yet adding these features to the mix of requirements could drastically increase the acquisition cost of the system. The major improvements in cost/performance are in the new systems, software based, which run on standardized hardware.

Electronic Paint Systems

Price/Performance - Like other electronic graphics equipment, the performance of paint systems is improving versus the acquisition cost.

When the first paint systems came out, the cost was upwards of $100,000. Four years ago, paint systems became available which had many of the features of the high-end systems. Paint systems in the $20,000 price range put the technology into the hands of many local stations who would not otherwise been able to afford any paint system at all.

Unfortunately, graphic designers shunned these lower cost systems because they exhibited sluggish response. Over the last two years there have been real cost breakthroughs in paint systems which are "designer-friendly". I would define the designer-friendly characteristics of paint systems to include speedy operation (especially brush functions), real-time cutout manipulation, high-quality typography, image manipulation tools, and fast image magnification (usually done in hardware).

Standardized hardware - In the future, the designer-friendly, high end features will become available in paint systems using standardized hardware, and at much lower cost than at present.

Networking - Paint systems will exist as peers in a networked graphics environment consisting of still stores, character generators, animation systems, and general purpose computers.

Electronic Still Stores

Picture Quality - Still stores used to degrade picture quality, but this is no longer the case. Now, still stores operating in the 4-field composite or the component domain have eliminated the flickering artifacts associated with early still stores. This kind of image quality is becoming a "must have" item in newly purchased systems.

Hardware Platforms - Like other graphics applications, still stores are now being designed around standardized hardware platforms which provide the basis for system control, image storage, and networking. It is possible to use standardized frame-buffer peripherals for still store video outputs, but currently available hardware is not quite up to the variety of transition effects to which we have become accustomed as broadcasters. Manufacturers get around this limitation by making their own frame buffer peripherals. I expect this situation to change at any time as commercially available frame buffers improve in response to market forces larger than broadcasting.

Networking - Access to a common pool of stills from multiple locations has been a feature of still stores for years. Present-day networking adds a new twist by allowing users to share and transmit stills between still store mainframes which are nodes on a local area network. The networking concept expands to allow stills to be shared with other graphics devices, such as paint systems. Another application for networking in still store systems is to allow still store database material (catalogs) to be conveyed to users of newsroom systems connected to the network. This level of connectivity could include information about the stills, or graphic representations of the images themselves.

3D / Animation Systems

Presence in broadcast facilities - 3D animation systems are not all that common in broadcast stations. There is not a daily requirement for this type of imagery. Specialized artistic talent is required to get the most out of 3D animation. 3D is still the province of high-end production and network facilities.

Will this change as the cost of 3D capability comes down? Possibly more 3D features will be incorporated into the electronic graphics systems used by broadcast stations. The key is to make these features useful from the graphic standpoint without specialized skills and a great deal of preparation time. In many ways, this is a repeat of what happened with electronic paint system technology. Although 3D capability may fall into the hands of many, the best work will still be done by those with specialized talents to apply it.

Standardized Hardware Platforms - 3D composition and rendering are now the domain of standardized hardware. Unix-based workstations predominate at the high end because of the computational power necessary for rendering frames in a timely manner. However, the PC, Macintosh, and Amiga run software for composition and rendering, at a tradeoff of speed and image complexity compared to the high-end workstations.

Data exchange between composition and rendering workstations - It is now possible to all composition work and testing on relatively low-cost equipment, such as PC and Macintosh platforms, and offload the rendering job to workstations. This can be done in a networked environment.

As the computational power available in low cost systems improves, it will become practical to do the rendering work on less expensive equipment, in less time.

CONCLUSION

Where are all of these trends leading?

1. Advances in electronic graphic systems for broadcasters will be driven by forces outside broadcasting.

2. New electronic graphics systems will be purely software-based, and will run on standard hardware platforms.

3. Graphic systems for broadcast will make increased use of page description languages.

4. Traditional CG, Paint, Still Store and 3D functions will merge into unified graphics workstations. The software for each function ma well originate from different developer.

5. Graphics workstations will network with each other and with other computer-based systems in the broadcast plant.

CHARACTER ANIMATION—THE MERGING OF TECHNOLOGY AND CREATIVITY

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While the subject of my talk today is character animation and what the latest generation of character generators are capable of, what I really want to discuss is the power of expectations.

Features and benefits are the mainstay of a product manager's usual assignments. I feel fortunate that I've spent the past three years working with a category of product that invites more expansive consideration than a mere recitation of specs, and definitely raises the ante on broadcasters' expectations of what equipment can provide.

Today, I'd like to present a quick overview of the colorful, twenty-five year history of character generators (CGs) and give you a preview of what creative powerhouses character animators will be in the 1990s. It's far more than people were expecting.

There has always been a need for the descriptive, written word to be displayed in television programs, commercials and news productions. Two key factors determine its application: the creative idea that sparks a need; and the technology available to implement it.

Today, we as broadcasters and post-facility people have a unique set of creative needs and applications for character generation. We all believe we know what's the possible, the tools available to us and how to use them to get the desired effect. Or do we?

The fact is, the creative ideas we have for character generation, special effects or anything else for that matter, are dictated by what we think is possible. And what we think is possible is determined by our technological frame of reference.

Back in the 1960s, the development of character generators was spurred by the acute need to identify people and places. At that time, it was perceived as adequate that the process of character generation did just that — provided static letters that sat on the television screen and functioned as simple I.D.s.

Yet even then, particularly in the world of commercial production, there was need to animate letters and symbols — to have those characters move, fly and change shape. Even with the benefit of hindsight, it's easy to understand why producers who required animation turned to film solutions, rather than relying on Videographs and Vidifonts. Video technology, character generation technology, were not yet evolved enough to address that expectation.

Yet, since the early days of television, directors used whatever means they had at their disposal to enhance the interest and communication value of messages that needed to be formulated as written words. It's my premise that, since words are inherently boring when compared to the visual impact of pictures, directors and viewers will continue to demand new and better ways of making the presentation of words more visually appealing. I think that an examination of the evolution of what have become to be known as character generators will illustrate my point.

THE EVOLUTION OF THE CHARACTER GENERATOR

Prior to use of the first character generator in 1967, art cards dominated broadcast environments. These state-of-the-art (then), intricate graphics were shot on camera and keyed over video. Rolls were innovatively devised by creating text on a continuous sheet of paper that was affixed to a drum that was sometimes as much as three feet in diameter. When the drum was turned, rolls were achieved. Roll speed was simply a function of how fast the drum was cranked. But as effective — and sometimes elegant — as these techniques were, they were also time- and labor-intensive.

Relate these capabilities to a real-time situation in a newsroom. It's 4:30 in the afternoon, 90 minutes before the early evening news. A producer has drawn up a list of the supers he needs for the six o'clock news: two anchors' names, the location of a major four-alarm fire, and information about tomorrow's weather. This list would be hand-carried from the newsroom to the artist. Working with black art cards, the artist would press type on the card, and chalk the board to flag where it belonged in the upcoming broadcast. The art cards were delivered to the floor director, who put them in front of a camera. At the appropriate time, a switcher would key the camera card over the video going out over the air.

The weather wouldn't come in until 6:15. There would always be a rush to get the characters on a spaghetti board, make sure the letters were all straight, and slam it on an easel 15 to 20 seconds before air time.

For credits, either permanent art cards were displayed, or a credit roll was created on a large typewriter and rolled through a drum or scroll machine in front of the camera.

Today, this scenario evokes feeling of nostalgia for the Golden Age of Television — or downright incredulity. But in the early 1960s, contrasting these rudimentary tools with what had gone before (essentially nothing) — well, people were amazed.

In these early days, character generation was more a technique than a specific product — directors used the tools available to present high quality, moving words to their audiences. As in any capitalistic marketing environment, entrepreneurial manufacturers set out to convert this "need," (as expressed by the ingenuity of the continuous sheet/moving drum/artcard approach to presenting messages) into useful, saleable products.

After the introduction of digital typesetting in 1965, it was the A.B. Dick company that first adapted character generation for television with its Videograph 990, first used in late 1967. System Resources Corporation (SAC) built a tape cassette drive, called a Titlefile, to be used as a memory for the 990. Like A.B. Dick, SAC became so enamored of the new technology that it adopted and then modified the 990 as its own machine called — when it hit the market in 1970 — the Chiron. Today, of course, SAC is called Chyron Corporation.

Around this time, CBS Labs came up with a third important development in electronic CGs. In 1971, CBS entered the broadcast CG market with the Vidifont, generally considered to be the first CG with what was then called graphic arts quality characters. In the Videograph 990, characters were formed of a crude matrix of dots. A capital 'E' could be perfectly formed of such dots, but a capital 'O' could not. The latter ended up looking like a rectangle, or a rectangle with tiny diagonal cutouts at the corners. In the Vidifont, an 'O' looked like an 'O'.

The secret of the Vidifont was the ability to refine and control switching times to within a few tens of billionths of a second. Each character was still comprised of a matrix of dots but now, instead of the matrix being five dots by seven or seven by nine, it might have been something like 25 by 35. And the Vidifont also introduced proportional spacing.

These first CGs provided the ability to type characters on the screen, correct or change characters and perform simple positioning such as center, left and right justify. While these devices represented a real advantage over art cards in time and labor savings, they were a step backward in terms of the quality and flexibility of their output (versus a real, live artist). Characters were typically blocky and font selections were limited.

Let's take these video typewriters back into that same newsroom: it's 90 minutes before the six o'clock news. Now, the list of program supers went directly to the video typewriter editor, eliminating the need to keep a talented artist on staff merely to create art cards for supers. The supers were created electronically and saved on magnetic storage disks. After calling them from storage, a switcher would simply key them on the air. In the same fashion, the weather would be typed up electronically, and could be instantaneously displayed on the air.

The benefits of video typewriters were several. Words could easily and electronically be corrected and changed; you didn't need a huge inventory of letreset letters to handle every combination of super imaginable. Now, every reporter who did a stand-up could get a super, not just Huntley and Brinkley; every piece of video could be identified by a location super; in addition to the weather forecast, you could broadcast current conditions and even a three-day forecast. In short, every station equipped with a video typewriter could do more functions, more easily, more quickly and more error-free.

And people were amazed.

The history of video CGs from 1971 to 1983 was much less dramatic than the introduction of the Videograph 990, the Chiron and the Vidifont. Other CGs became worthy of the designation "graphic arts quality," and more brands entered (and sometimes left) the market.

As next generation products were introduced, a broad range of capabilities evolved such as color (without requiring a separate matte generator), access to multiple fonts, edges, drop shadows, logo/image capture and crawls.

In the context of our six o'clock news show, this is when character generators really began to perform the function of informing, not simple identifying. Beyond reporters, now you could identify all the subjects within a segment; covering an emergency or disaster, you could list highlights of the story as the video rolled; and with the ability to perform crawls, you were capable of primitive animation purely driven by the video CG.

The early CGs had automated the necessary process of simple identification. Now we had more sophisticated machines that were production tools and communications devices in and of themselves.

Between the late 1970s and early 1980s, with the introduction of digital effects systems, a new creative wrinkle was added to the world of CGs. These new effects systems opened up whole new avenues of movement such as rolls and tilts in perspective, and ability to fly characters around the screen. This element of motion was a natural merger of character generator and effects features that provided more exotic animation capabilities than simple rolls and crawls. Again, as in the late 60s, creative directors turned to the available technology to develop a new and exciting visual communications.

During the past decade, the primary advancements in character generators has been in the enhancement of character quality, incredible increases in the range of colors, the availability of thousands of fonts, and the addition of logo and graphic creation. While the animation of characters has become more commonplace, animation capabilities heavily rely on the use of separate digital effects devices and even paint systems for many applications. While this is less expensive than the "classic" solution of film animation keyed on video, it ties up your equipment and, as a result, is very costly. Once again, directors and viewers themselves began "designing" the next generation of character generators.

THE FUTURE OF CHARACTER GENERATORS: CHARACTER ANIMATION

Professionals in television and commercial production have always needed characters to move — it's inherent to the medium. But historically, production people have not expected their CGs to deliver animation. So, in the past CGs have been built to fit the expectations that a CG is a video typewriter, incapable of delivering "pure," clean animation.

It's our contention that once character animation is easy, inexpensive and internal to a CG system, video professionals will do more of it, more creatively. Expectations will rise dramatically. Character animation will be to the 1990s what ADO effects were to the 1980s. In fact, observing the ADO system's increasing role in animating the output of standard character generators, spurred Ampex to combine character generator features and special effects technology to create ALEX — a true character animator.

For example, in the Ampex ALEX CG, animation capabilities are made possible by a new technological approach called pipeline architecture which, based on some of the ADO parallel processing techniques, is an alternative to the more common frame-buffer-based architectures.

Frame buffer-based character generator systems must make some compromises in either color depth, image sharpness or complexity to accommodate motion. While still images are antialiased to achieve maximum sharpness along video object edge transitions, frame buffers cannot always respond quickly enough to main full antialiasing when these objects move, causing edges to take on some amount of blur or stepping.

Frame buffer architectures begin to show deficiencies when called upon to display dynamic characteristics because the "dynamics" are based on how fast data can be written and moved within the buffer. Approximately 33 milliseconds are required to update a complete non-moving image in real-time video. If significant motion of symbols (that is, text characters, logos and other objects) occurs, the time for recalculation of an entire frame exceeds the 33 ms period. Therefore, to achieve motion, a compromise must be made, usually by dropping some or all antialiasing. Restrictions are also placed on the ability to overlap symbols dynamically in the frame buffer, again because of the speed at which calculations can be performed.

In response to these limitations, a unique "pipeline" approach was designed to maintain antialiasing during any motion. The graphics engine architecture uses an approach that avoids frame buffers by using a series of stages in a "pipeline" concept to regenerate the video output in each field. This type of design offers several advantages over the frame buffer system.

First, any symbol on the display may be moved, in real time and independently of any other symbol, because the display is regenerated for each and every field, 60x per second in NTSC and 50x per second in PAL. Characters and symbols, individually or as groups, can be animated in real-time along any arbitrarily defined motion path. While moving along a defined path, characters and symbols can be tumbled, flipped, squashed and so on. Another unique feature of this architecture is that all other antialiasing characteristics (up to

256 levels) are included in the regeneration process at all times.

Second, when symbols overlap other symbols, the pipeline system provides for identifying boundaries, defining the overlapped regions and maintaining antialiasing. The colors and key signal within these boundaries may also be controlled independently from the original characters creating the overlap.

The new "pipeline" has three graphic engines performing three different tasks: one engine selects the parameters or definition of the character (size and shape); one engine fills the character (color, blends, video, overlaps); and the third places the character in its proper location. The pipeline allows this process to be performed 60 times per second, so it is as smooth as the television medium itself. Every field is animated whether the objects on the screen are moved or not. Because the animation is always on, why not use it?

In other words, the unique pipeline architecture allows users to move more symbols or characters, in real-time, individually on the screen, in any direction, in every field, than any other CG. Now you can have the flexibility to do real-time animation without the use of external equipment, and still produce high quality graphics. Characters will remain fully anti-aliased, giving you sharp, clear characters.

Character animators represent the merging of a creative impulse with the latest technology to form a new class of products that will enable broadcasters and post-production facilities to again re-define the role of character generation in television.

In the post-production world, we expect that a real-time character animator will be a whole new source of income. Every post house has to have a CG to do titling, and they usually don't charge for it. Asking a client to pay for titling is like asking him to pay for the chair you give him to sit on during a meeting. But every post house who offers animation services charges for them. And imagine the economies when they're able to perform these services in real-time. For example, a post house in San Francisco, has been able to make money and at the same time reduce his charges to clients by two-thirds because of the savings he's realizing by doing real-time animation and freeing up other equipment.

For broadcasters, this new generation of CG can begin to relieve some of the conflict between the demand for immediacy and the desire for creativity. With real-time animation capabilities at hand, directors can create new, subject-specific animation on a "same day" basis — adding a new dimension to news and sports. And, with these new features being built upon basic, high quality character generation foundations, today's more fiscally aware broadcasters can provide both mandatory titling services while delving into a new and creative medium.

Within corporations and educational institutions, these new combination character generator/animators are likely to be a real boon in the 1990s. With effective communications quickly rising on corporate priority lists, video presentations in general and animation in particular are being looked at as the best way to increase audience attention in what has become a society inundated with thousands of messages on a daily basis. Using a real-time animation device, corporate and educational communicators can use the animation medium more frequently, thereby adding viewing appeal and increasing the communication value of hundreds of annual productions. And again, with capital spending coming under greater scrutiny, the fact that the animation machine provides all of the CG needs of any in-house facility will make it much easier to cost justify the addition of new, creative services.

In summary, character animation, already common through the use of a broad range of equipment, will become a new creative fashion in the 1990s as products become available. Real-time character animation will become far more prevalent in on-air broadcast and post-production applications. At broadcast stations and within corporations, animation will be done cost-effectively in-house. By not having to buy such services outside, their use will increase dramatically.

Post facilities can offer clients low-cost sophisticated motion path animations at a fraction of the cost of conventional methods. Assuredly, this will also increase the creative use of animation.

In the 80s, digital special effects changed the look of TV; in the 90s animation will be the fashion, achievable with character animators.

DESIGN CONSIDERATIONS FOR TODAY'S STILL STORE SYSTEMS

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Abstract - It is some ten years since still stores, using computer discs to record digital TV images, have been available as a practical tool for broadcasters. Over that time, there have been great advances in the technologies around which still stores are based, meanwhile the usage and importance of stills in TV production has grown immensely. Today's designer can take advantage of the technology and experience to establish a more effective role for still stores through the 1990s.

INTRODUCTION

Originally the idea of still stores was to take advantage of the fidelity and reliability of digital equipment for the presentation of stills on-air. Now, with advances in disc and integrated circuit technologies, it is possible to broaden this view and look at the more general use of stills in TV production. Whilst it is easy to identify the parts played by areas of specific hardware, it should not be forgotten that there is an ever growing amount of software needed to drive the system. Digital circuitary offers the ability to grow facilities onto a system base in a way that was never possible with analog technologies. Images can be passed from section to section without any impairment and timing signals for re-entry processes presents no special problems. Add to that the trend that yesterday's machine becomes today's board and tomorrow's IC, so immense possibilities can be realised economically.

Today's designer has the opportunity to re-cast the role of still stores taking into consideration the whole A-Z of stills production, storage and presentation. This paper outlines the chief changes in technology that have occurred since the arrival of still stores and then describes how these can be used to affect a new design in line with production trends in the 1990s.

New Technology

Still stores depend on computer disc drives for picture storage. Ten years ago typically platters were 14" across and the largest fixed drive capacities were around 100 MBytes. Today, development centers around 5¼" discs with capacities of 1,000 MBytes or more, resulting in a one hundred fold reduction in rack space for a given capacity. At the same time, data transfer rates in general have doubled to 2-3 MBytes/second.

During the 1980s, 80 MBytes 8" removable disc packs became widely used for the transfer and archive of images. Today, magneto optical technology offers over 600 MBytes on a single 5¼" platter.

Over the same period, VLSI technology has moved dynamic RAM chips, the major component for making the necessary framestore buffers, from 64K to 4 MBits. For our application, to store a 4:2:2 component picture used to take 128 RAMs, now it need take only 2.

With the concept of still stores and their basic structure well established, it has been possible to invest in application specific integrated circuits (ASICs) to compress large areas of common circuitary onto a single component. Besides greatly reducing the component count for a given task, these circuits generally work faster and consume far less power than their off the shelf equivalents.

With the new disc drives, RAMs, ASICs and other modern VLSI circuits to hand, one choice for the designer would be to simply reproduce the equivalent of what has been available over the last decade. This could be presented in a more efficient package, smaller, using less power and possibly with faster picture access. Such a system has its place, but it does not add to or enhance the possibilities for TV production. It reflects established design techniques and has no more need of mention here.

The Stills Chain

As already mentioned, the designer is now able to take a look at the whole process of stills in TV, from origination to air. Possibilities vary from, at one extreme, constructing one single integrated machine to do the lot, to dividing each task into separate units. In the final analysis, pragmatism must rule and boundaries be drawn as defined by the areas of responsibility and special expertise. Other considerations include the avoidance of clashes if the system has to be shared and, most important, security and reliability - especially for the on-air situation.

The stills chain starts with pictures originated in a number of ways, from cameras, grabbed from VTR or created in a Paintbox or other TV graphics system. From there, they can be entered into the still store. Prior to use, they will often be viewed and selected for assembly into a sequence or running order for replay into the program.

Coding Format

To maintain the picture in the best possible quality through to the program output, it is necessary to pay attention to its coding format at all stages. With most sources in component form, adopting 4:2:2 digital coding for the still store is appropriate. Being most directly compatible, it will retain the best quality, both into the store and for output of the frozen image.

In the past, many designers opted for 4FSC coding, taking advantage of its much reduced data requirement per picture, about half that of 4:2:2, and its compatibility with coded NTSC or PAL signals. Times have changed. The huge increase in disc and RAM capacities makes the economics of 4FSC unnecessary. So there is good reason for 4:2:2 coding to be universally adopted for stills. Using the rule of thumb that a 4:2:2 picture occupies 1 MByte on the disc (more or less according to filing systems), this translates to 500 - 1,000 pictures or more on the range of currently available drives - adequate for most on-line applications.

Styles of presentation dictate that not only picture but also, in some cases, key must be stored. In this way, irregular shaped objects can be cut-out and shown as floating graphics without having to be included in the familiar 4 x 3 rectangular boxes. For best results, the key should have the same resolution as the luminance information, so dictating a 4:2:2:4 recording format.

Although disc storage capacity is plentiful, there is still no need to waste it. 4:2:2:4 represents a 50% increase both in storage demands and, therefore in the time to access an image. Clearly, the video path through the system must allow for 4:2:2:4, so the RAM count must be increased from 2 to 3 per framestore. But for disc storage, economies can be made in two ways. First, the key should only be recorded if it is actually required.

Second, an examination of the use of floating graphics shows that they very often occupy a quarter or less of the screen area. So by storing only the used part it means in practice that more stills, rather than 30% less, can be stored in the 4:2:2:4 format.

Graphics Preparation

It is rare that pictures are output without some form of preparation or formatting into a given program style. Such work is carried out in the graphics department, usually with digital equipment. An examination of the work shows that a significant portion of it comes under the heading of "cut and paste". Given that a still store includes the disc, framestore and computer elements which are a necessary part of any graphics machine, maybe this is an appropriate area for expanding the capabilities of the still store. To do so involves the addition of a tablet and pen outside the machine, whilst inside the facilities can be efficiently realised with ASIC chips and of course, more software. The major functions required are to draw a stencil or key to define the cut-out area, manipulation of size, position and rotation, and to paste one object onto another to make composites and montages.

This is an example of how digital machines can offer new ways of achieving more varied results. Following the path of creating digital equivalents of existing analog equipment can prove expensive and gives digital hardware an "expensive" label. Good and carefully thought-out design, where attention has been paid to the whole process, not just a part of it, can show great benefits. Care must be taken, however, to recognise divisions of skills and operational demands. It may not be, for example, that it suites to have the still store devoted to graphics preparation as on-air time approaches. In which case, a separate dedicated graphics facility would be more appropriate.

An important aspect that rarely features in the specification, is that of speed. In cut and paste, re-sizing and rotation are demanding in terms of image processing, and the design should allow sufficient facilities for these transformations to be completed quickly. A yardstick would be to process a full picture in a second, similar to the normal speed of picture recall from disc, with smaller images taking less time. This implies dedicated hardware, rather than a software dependent process.

Management

Still stores have, in general, been operated with affordable rather than the actual required numbers of pictures on-line. So, the availability of high capacity disc drives able to store 1,000 or more pictures, is good news. To account for this increased capacity, the designer must include an efficient system for locating any required still. When only a hundred or so pictures were present, viewing of montage displays might enable a selection to be made in a reasonable time. But now with the much larger numbers, a more efficient method is necessary to prevent searches running into minutes - or longer.

A practical system which has already proved successful depends on the use of titles for all items. A single line of 80 characters has been found sufficient for most basic descriptions. This data is stored with the picture and can then be found by a keyword string search. With this method, a system could be expected to search through around 50 titles/sec. Even so, this becomes unacceptabley slow as the items list grows towards the 1,000 mark. Furthermore, a number of drives can be installed on one system and capacities show every sign of continuing the increases already seen over the last years. To ensure that the needs through the 1990s can be fulfilled, the management system should provide the means to complete all searches within a few seconds for up to 10,000 entries. This can be done using an index filing system where search times are more or less independent of the number of items filed.

Expansion

As the usage of still stores grow, it is important that the system can expand to meet new demands. This means interfaces must be built-in, or allowed for, to accommodate additional requirements. The SCSI bus is widely used for an extensive range of disc drives and enables the easy addition of extra capacity in the form of fixed magnetic, or removable magneto optical drives (up to eight devices on the bus). Data transfer is specified up to 4 MBytes/sec. Currently this enables expansion up to over 10 GBytes of storage to be available on-line.

For installations where there is more than one still store, it is often advantageous to have some form of direct interchange of pictures. SCSI can be used to join both machines and discs to form an effective network for stills (Fig 1). A network disc, accessible by all machines, is available for them to copy pictures to, or from, independently of each other. In operation this can be used as a 'post box' for picture transfer or as a general archive. Adding a magneto optical drive allows all networked machines access to off-line material.

Security of operation is essential for still stores, especially when used on-air. It is necessary for each machine on the network to retain its autonomy. In general it may not be too serious being held up for a second whilst someone else accesses the network disc, but the chance of such delays cannot be allowed when playing a still into a live program. This can be avoided by operating a dual bus system, one for the local discs, the other for the network. This design also brings increased flexibility by allowing up to eight local devices as well as a further eight on the network.

For further expansion, access should be allowed to a central data bank, comprising images and title data on a large scale, capable of handing the library needs of any larger TV station, with access for all graphics and stills machines. An ethernet interface can enable machines to be added to the facility. This gives easy system expansion and imposes no practical restrictions on the positioning of individual units.

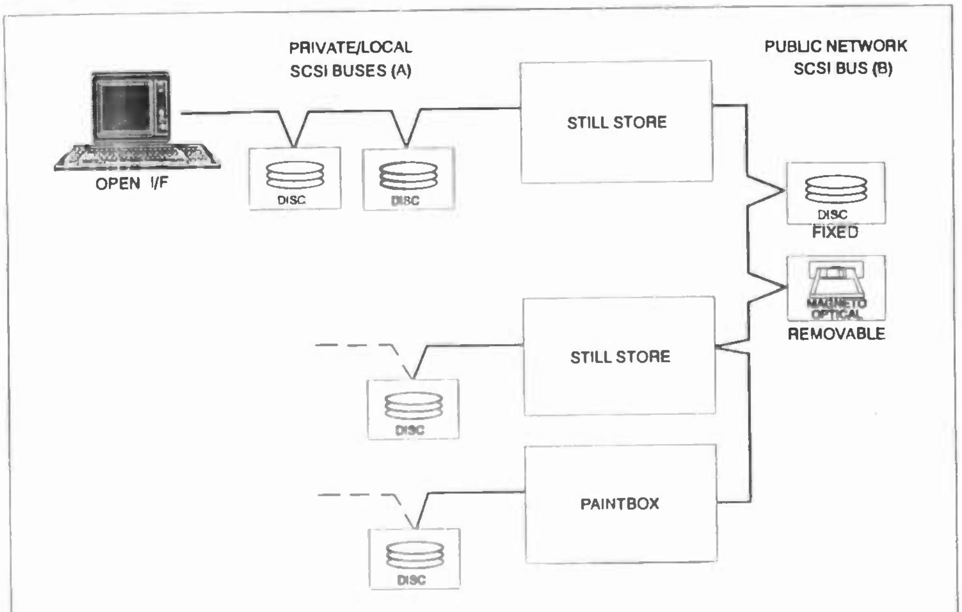

Figure 1 LOCAL AND NETWORK SCSI BUSES

Interfaces

Despite the expanded role now possible for today's still stores, it remains as important as ever to interface with a range of graphics and related equipment. It is logical that a graphics system such as a Paintbox, can be included on the network already described, so encompassing a broad range of activities, but not all stills requirements. To connect a broader range of systems, an open SCSI port can be provided. Here, the specific applications are at the user's discretion, but it might be expected that a PC would be connected for such functions as a browse station to view the contents of the store, or as an interface for long distance picture transfer. The port could make a direct link to a weather satellite terminal or to another computer graphics system. It could connect directly into a newsroom computer system to bring the journalist closer to the pictures for his story.

The implementation of individual applications may be beyond the scope of the still store designer, but the provision of the port is an important step towards making them possible.

Some emphasis has already been made on the subject of security. It is important that the open port activity is purely a background task causing no interruption to the stills operator's work.

It is most likely that images from such a diverse array of sources will not all be the same size. Therefore, the protocol of the SCSI port, as well as the file structure of removeable media, should include a message describing the file size. For best quality, the files can be held in their original form and the re-sizing hardware used to automatically convert them to the output standard of the still store. So 625 line pictures from Europe, 1125 or 1250 lines from HD - or any size, as well as the local 525 standard, can be intermingled and used freely without extra demands imposed on the operator.

Summary

Today it is possible for the designer to re-appraise the use of still stores in the light of a decade of rapid growth and change for stills usage. New technology gives the opportunity to affect ideas more efficiently than before.

A greater awareness of production needs shows that floating graphics and linear key signals should be catered for, both for presentation and possible origination with cut and paste. The provision of a wide range of interfaces gives the key to expansion for storage and networking. It also enables other devices to be connected for facilities beyond the scope of the still store itself and helps with the free exchange of images with a wide variety of sources. Plentiful storage capacity means management systems become a necessity and component formats can be applied universally.

Now there is the chance to realise wider uses for still stores. The new technology is a great boon, but is no substitute for good system architecture and software design applied with a view of the whole still's operation in TV production.

SECOND GENERATION DIGITAL NON-LINEAR EDITING IN THE POST PRODUCTION ENVIRONMENT

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Abstract: Digital technology continues to revolutionize the video production and post production processes. Now, second-generation digital non-linear editing systems are reaching the marketplace. This paper will describe this new technology and examine the impact that second-generation non-linear editing has on the creative and technical processes of generating an approval cut and then a final master program.

With the advent of JPEG-based compression, the impact of digital non-linear editing will no longer be confined to off-line editing. Compression will provide full resolution output directly from disk. This will open new markets, change existing markets, and in general greatly expand the potential user base that can create video-based programming.

BACKGROUND

The pace of change in film and video editing is increasing rapidly. Film editing remained unchanged for decades after the introduction of the upright Moviola. Later, flatbed editors provided improvements to the basic editing process, but the cut/splice approach was unchanged.

In the late 1960's broadcasters began to use videotape as a shooting and editing media. At first editing was done much like film, using magnetic filings to delineate frames for cutting and splicing. In the 1970's, the first helical-scan tape formats ended the practicality of splicing videotape, and electronic editing techniques emerged.

The electronic technique of copying from source to master forced video editing into a linear approach since the master could only be overwritten and no splicing was possible. This limitation has for two decades spurred entrepreneurs and engineers to develop and market a wide array of non-linear electronic editing systems.

Analog Non-Linear Editing

Analog non-linear editing actually predates timecode-based editing. The original CMX 600 built in the early 70's was a black-and-white, analog disk-based system. Final output required a method for auto assembly, and it was for this task that timecode was originated. While the disk technology of the time limited the success of the system, the timecode approach worked well and went on to become the standard for linear editing.

Tape-Based Non-Linear Editing

In the mid 80's, a range of tape-based analog non-linear editors such as the Montage and the Ediflex were introduced. These systems use multiple tape sources, each with identical footage to reduce access time. A computer positioned each deck at the right place and triggered the decks in sequence to simulate an edited program.

These and other tape-based systems provided the first practical non-linear solution, and they are still in use today, mainly for long-form programs.

Analog Disk Non-Linear Editing

At the same time, analog disk-based non-linear systems arrived. The EditDroid and the CMX 6000 both used 12" analog disks to provide relatively fast 1-second access to material. These systems provided a faster alternative to tape-based editors. But the disks required mastering on expensive equipment, thus limiting their acceptance.

Digital Disk Non-Linear Editing

In the late 80's, the first digital non-linear editing systems were introduced. Instead of storing images on analog media, systems such as the Avid/1 and the EMC capture video and audio digitally. By using high levels of compression for the digitized video signal, hours of footage could be stored on disk.

The use of digital technology brought the cost of fully configured systems to well under $100K; sales of non-linear editing systems began to increase rapidly in 1989 and 1990.

Image Quality

The first generation of digital systems addressed most of the barriers that slowed acceptance of digital non-linear editing. But the video compression process reduced image quality to levels that many editors found troubling. In particular, lip sync and eye movement on wide shots are not readily visible.

SECOND-GENERATION TECHNOLOGY

With non-linear editing now on a digital technology curve, editors and producers can expect extremely rapid advances that will change the editing landscape.

In this section, we'll cover the new technology being incorporated by second-generation digital non-linear editing systems.

Compression Techniques

First generation digital non-linear systems used compression techniques that operated directly on the pixels of the input image. For the most part the compression these systems offered came from subsampling of the incoming image, and from reduced sampling of the chroma signal, much like the D1 4:2:2 standard reduces digital data by sampling chrominance at half the rate of the luminance signal.

Now, extremely powerful chips that can run much more advanced image compression algorithms are available. These algorithms, known as DCT (discrete cosine transform) convert an image into frequency components, and encode those components. An inverse transform converts the frequency components back into an image.

A few years ago, a combined standards-making subgroup from ISO and CCITT formed to create a standard for color photo compression. This group is called the Joint Photographic Experts Group (JPEG). A previous standard for black-and-white text, graphics, and photos, called CCITT Group 3, is now the worldwide standard for FAX transmission, and is widely regarded as the reason for the huge success that FAX has enjoyed. Now JPEG is poised to cause a similar breakthrough that will revolutionize color image and video compression.

JPEG Chips

While the JPEG algorithm can be implemented in software, it is extremely complex and relatively slow. To provide real-time compression for video images, special chips have been developed specifically to perform this new algorithm. The first chip to reach the market is made by C-Cube, of San Jose, CA. A wide range of other chips are announced or planned by other vendors, including Intel, VLSI, JVC, and Thompson.

Delivering high image quality is not simply a matter of putting a JPEG chip into an existing system. High image quality requires a system designed with the specific needs of JPEG in mind. These issues will be discussed in detail in the next section.

DSP Processors

Digital signal processors (DSP) are revolutionizing signal processing in the same way that JPEG chips will revolutionize video processing. The most popular DSP chip in the audio arena is the Motorola 56001. This chip can perform over 250 complex calculations *per sample* of audio at 48 KHz. This means that capabilities that previously could be accomplished only on a full scale mixing board can now be done digitally, at low cost.

Multiple DSP Processors

As the use of DSPs has become more widespread, their cost has dropped dramatically. Advanced systems are now being built that use multiple DSPs to share the computational load. This means that not only are more sophisticated effects possible, but more channels can be handled in real time.

Phase Change Optical Disks

The optical disc will play a very central role in the development of digital media technology. The current technology has been on the market for a few years, and is based on a technique called magneto-optics. A laser is used to set bits on the disc, but slower magnetic techniques must be used to clear the bits. This means that the writing speeds are typically half to one-third as fast as reading speeds.

Now, a new technology is reaching the market that eliminates this drawback. Called phase-change, the new approach lets the laser set or clear bits. Thus reading and writing are at identical speeds. Equally important, the new technology provides 500 megabytes per side, compared to today's 300 megabytes per side. See Table 1.

Comparison of Storage Technologies

People are extremely attracted to optical discs because of their removability. However it is important to point out that magnetic disks have large advantages over optical discs in virtually every category except removability.

While a typical optical disc has an access time of 90 milliseconds (msec), a magnetic disk accesses in 16 msec. Access time on the magnetic disk is almost six times faster than the optical.

The comparison is similar for throughput. A SCSI magnetic disk can deliver sustained read/write data rates in the 1200K bytes/sec range. Magneto-optical discs typically deliver about 180K bytes/sec performance on writes, while phase change opticals can write at up to 450K bytes/sec. Even with the new optical technology, the difference in performance is over 3-1. See Table 1.

Disk Type	*Access (msec)*	*Read* ---(Kb/sec)---	*Write*
Magneto-optical	90	450	180
Phase-change optical	90	450	450
Magnetic	16	1200	1200
Magnetic (SCSI-2)	16	2000+	2000+

Table 1. Storage Performance Comparison

Interestingly, in the area of cost, magnetic disks are still the least expensive per megabyte of online storage. Magnetic disks cost one fourth as much as magneto-optical disks per online megabyte. See Table 2.

Disk Type	*Cost per megabyte*
Magneto-optical (300 MB disk drive with disk)	$18-20
Phase-change optical (500 MB disk drive with disk)	$8-10
Magnetic (1000 MB disk drive)	$5-6

Table 2. Storage Cost Comparison

SCSI-2

The Small Computer Systems Interface (SCSI) has become a standard disk interface. This standard makes it much easier for vendors to develop fast, sophisticated drives, that integrate easily into user systems.

Now, the original SCSI standard is being expanded into SCSI-2, which provides even faster performance with virtually no increase in cost.

USING JPEG COMPRESSION FOR HIGH IMAGE QUALITY

The use of JPEG for image compression means that image quality of digital non-linear systems will improve dramatically. But high quality JPEG images require high performance disks, as Table 3 shows.

High Quality JPEG Images Require New Hardware Architecture

The use of JPEG compression requires new hardware architectures. First-generation systems relied mainly on image subsampling for compression. This approach sacrifices resolution but reduces the data rate sent to the image coding stage (Figure 1). Second-generation systems will be able to handle much higher data rates, thereby reducing the need for image subsampling. This will increase picture quality (Figure 2).

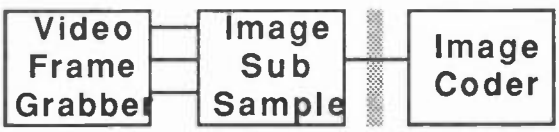

Figure 1. First-generation hardware architecture. The frame grabber output in first-generation systems is typically five bits for each of R,G and B. After image subsampling, the 128x96, 16-bit image represents a data rate of just 700K bytes/second. This data stream can be sent across a bus to a second board for further compression.

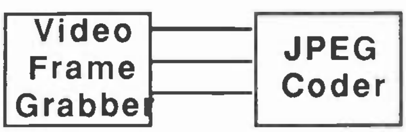

Figure 2. Second-generation hardware architecture. In second-generation systems, the output of the video frame grabber is fed directly to a JPEG image coder. The 640x480 image can be handled directly by the JPEG compression chip. However the data rate for full resolution 24-bit per pixel NTSC video is over 27 megabytes/second, which requires that the frame grabber and compression hardware be tightly integrated.

Variable-Length Frames

One of the main ways that JPEG provides high image quality and high compression is by varying the amount of storage used based on how complex an image is. A desert scene with a blue sky will take much less storage per frame than a brick house with a white picket fence and lots of leafy trees.

Variable-length frames provide an important challenge to the system designer, because now access to a particular frame in a file must be based on a stored index, not just by reading some fixed number of bytes from the beginning of the file.

Buffering

With data rates as high as 1200K bytes/sec for highest quality images, the JPEG-based system must incorporate large buffers to insure proper data throughput to the JPEG chip. Without these buffers, JPEG-based designs cannot deliver the sharp, clear pictures editors desire.

IMPLEMENTATION OF SECOND-GENERATION TECHNOLOGIES

The hardware described above is now becoming available to manufacturers. Integration of this new technology into a second generation of digital non-linear editors presents a major technical challenge, but provides important user benefits.

High Quality JPEG Compression

By using advanced JPEG compression hardware, second generation digital non-linear editing systems such as the Avid/1 will now integrate the digitizer, frame buffer, and JPEG chip into a single high performance architecture. This new hardware can provide full resolution compression, with image quality rivaling 3/4".

Input Image Size	*Input bytes/ frame*	*Output bytes/ frame*	*Data rate to disk (bytes/sec)*	*Image Quality*	*Usable disk type*	*Comment*
128x96 16 bits	24K	2K	60K	low	optical or magnetic	Resolution is not sufficient to see eye movement on wide shots. Direct output quality not sufficient for client approval.
256x192 16 bits	96K	6K	180K	moderate	optical or magnetic	Resolution sufficient for most offline editing. Eye movement clearly visible in wide shots. Direct output sufficient for client approval in most cases.
640x480 24 bits	900K	20K	600K	high	magnetic	Resolution sufficient for critical offline editing. Direct output sufficient for client approval in virtually all cases.
640x480 24 bits	900K	40K	1200K	3/4"	magnetic	Resolution virtually equal to 3/4" tape. Direct output often sufficient for use as master in not-for-broadcast programming.

Table 3. JPEG Compression Rates vs. Image Quality

Variable Picture Size JPEG

Some implementations of the JPEG hardware will let the user control the trade-off between image quality and disk usage. By choosing smaller picture sizes or higher levels of compression, the user can match the needs of their program to the available storage.

Multiple DSP Audio Subsystem

A new audio subsystem on the Avid provides two DSPs that work in parallel. The system can provide four channels of 44/48 KHz CD-quality audio in real time.

Digital Linear Keyer

A digital linear keyer that allows full-resolution anti-aliased graphics to overlayed directly on the video image is an important addition that is now available on second generation hardware. This means that high-quality type can be output directly from the system.

Software Implementation

The integration of new hardware advances means major improvements in the software that underlies any digital non-linear editing system. For example, the Avid/1's software has been updated to release 3.0, and provides a wide range of new features. These include JPEG compression, more tracks of sound, direct editing in the timeline, multi-camera features, slow-motion and integrated support for titling and graphics.

IMPACT ON THE CREATIVE PROCESS

Non-linear editing is impacting the creative process in profound ways. Second-generation digital systems will accelerate these trends.

Instant Feedback to New Ideas

Digital non-linear systems let editors and producers try new ideas extremely quickly. A show can be roughed out in minutes instead of hours, and the concepts compared. Systems that incorporate a highly-visual interface provide an easy way for editor and client to interact about a program.

Multiple Versions

Unlike film or video, a digital system can re-use video and audio media without limit, allowing multiple versions to be created and compared at very low cost, both in time and money.

With the ability of JPEG to output high quality images directly from the system, a client can walk away with a tape of all versions without waiting for an auto-assembly.

Online Features in Offline

Digital storage means that online-type effects can be computed by software, thus giving an offline editor much of the power of online. For example, digital dissolves easily simulate a 3-machine A/B-roll system. Wipes and ADO-style effects can also be simulated.

High-quality JPEG compression means that output taken directly from the system often provides sufficient resolution for client approval without the need to auto-assemble or online the program until agreement has been reached on all creative input.

IMPACT ON THE POST-PRODUCTION PROCESS

Non-linear editing is impacting the post-production process in profound ways. As second-generation digital systems expand the use of non-linear techniques, facility owners and editors will find that new approaches and careful thought will help them take advantage of this new technology.

Increased Throughput in Online

Non-linear offline editing means that clients will typically be more accurately prepared when they come to online. Their edit list will be accurate, and they will spend a much greater percentage of their online time dealing with special effects. Demand for creative online editors will likely increase.

Better Client Understanding

Highly graphical non-linear editors such as the Avid/1 make the editing process much more visible to the client. Instead of a person pushing a vast array of buttons on large consoles, the editor and client work together to create the program the client is looking for.

Higher Quality Programs

With non-linear editing, clients can be more exacting. Time constraints will no longer force clients to settle for something less than they wanted.

Cost Impact

While many facility owners have feared that non-linear offline editing would reduce revenues, experience over the last two years shows that clients typically have a specific budget in mind when they approach a program, and they most often end up getting a much higher quality result for the price they budgeted.

While some clients use the high performance of non-linear editing to simply cut time and therefore costs, the vast majority are working on programs that represent a significant investment in shooting and duplication, and choose to fine tune the program to a much higher level than they could before non-linear editing. Often they will come away with multiple versions, each more carefully crafted than the single version they got before non-linear editing.

Increased Market Size

Non-linear editing helps demystify the editing process, and delivers more results in less time. This will increase the range of clients that post production facilities can appeal to.

SUMMARY

Second-generation digital non-linear editors will greatly expand the acceptance of non-linear editing by improving image quality, increasing audio capabilities, and simplifying system use.

As new technology removes the barriers to an all-digital approach to editing, fundamental changes will occur in how programs are created and edited.

CBS DIGITAL VIDEOTAPE EDIT ROOMS IN THE HYBRID ANALOG/DIGITAL ENVIRONMENT

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Abstract-The expanding use of digital technology in television broadcasting is having a profound effect on videotape systems in television production, post production, and program distribution.

Digital videotape is finding increasing use in CBS post production applications due to the inherent advantages of digital audio and video recording in quality, multiple generation dubbing, and in the simplicity of cassette based operation.

During the conversion to all-digital plants however, the interface with existing analog equipment in hybrid facilities pose major systems design and compatibility problems.

This paper discusses the present CBS approach to digital videotape edit room design, employing D-2 format videotape machines, in this hybrid analog/digital environment.

Technical requirements, design criteria, and operational experience are presented for modern videotape edit rooms in the CBS Broadcast Center in New York. Based on this experience, plans for future CBS digital videotape editing rooms are also reviewed.

INTRODUCTION

Audio/Video recording and signal processing in the digital domain is widely seen as a panacea for post-production of television programs. While there are significant technical, operational, and economic advantages in the application of digital techniques, their integration into an existing plant where analog systems and equipment are in common use, pose significant problems. These problems are discussed, and possible solutions presented in the context of CBS network post-production plans.

MERITS OF DIGITAL RECORDING IN POST-PRODUCTION

Where multiple generations of recording and dubbing are required, digital techniques maintain the technical quality through many generations. When compared with one-inch reel to reel analog VTRs, the small size and light weight of the digital format cassette, together with the fact that it need not be rewound to remove it from the VTR, are important operational benefits leading to savings in time and material cost. For multi-channel audio, digital recording again offers superior quality through multi-generation dubbing.

DIGITAL FORMATS IN AN ANALOG ENVIRONMENT

Notwithstanding the merits of digital recording noted above, when the integration of digital formats with an existing analog plant is attempted some significant problems arise.

While one-inch analog VTRs will certainly be phased out, and replaced by digital equipment, the transition will occupy several years in practical and economic terms. The large archive of programs recorded in analog formats alone assures the need for one-inch analog VTRs to be retained for retrospective and post-production use.

It follows that edit suites must retain multi-format capability for some time to come, and they must interface readily with the network plant's routing systems, in addition to the digital islands of graphics, video effects, and character generators now in place.

DIGITAL SYSTEMS CONSIDERATIONS

Video

D-2 digital composite VTRs are incorporated in the Library Management Systems at CBS, and this format is used exclusively for the playback of commercial and promotional messages to the network, and to the local station, WCBS-TV. In the future all programming will be played to air from this system, using the D-2 VTRs. It is therefore planned to edit programs in the digital format, producing a cassette as an end product, ready for playback to air. This will obviate the need to transfer completed analog tapes to the digital format for broadcast.

In designing a digital composite edit suite, a key item is the switcher. Available switchers provide for inputs and outputs employing a parallel digital signal system using multi-core cables and 25-pin D connectors, while the cable length is limited to 30 meters. This constraint is acceptable within an edit room, but in a network environment, where interfacing with the Broadcast Origination Center, other studios and other edit rooms, is frequently required, it is unacceptable. Some parallel distribution amplifiers are available, but no efficient method of patching exists in this format. Patching, necessary for many operations and essential for maintenance operations, is time-consuming and costly when using such a complex connector.

No SMPTE standard exists for composite digital transmission, but Sony has developed a chip set to convert the D-2 parallel format to a serial format. When available, this converter will allow the use of longer cables and employ standard BNC connectors. Patching may then be effected easily using standard video jacks.

Audio

The current digital audio standard is AES-3, which calls for audio to be distributed on balanced twisted pair audio cables using XLR connectors. Each pair carries two audio channels. The D-2 VTR has two outputs, for channels 1,2, and for channels 3,4. The data rate is 3.072 Mbits/second, or a bandwidth approaching 5 MHz. The maximum cable length allowed is 30 meters.

The primary problem with the current standard is the limitation of cable length, and crosstalk and radiation are significant concerns. In addition, switching and distributing the AES signal requires special distribution amplifiers and switching equipment, which are as yet not available. A solution to this problem advanced by Rorden and Graham[1], is to normalize the digital audio signal to a common format in order that it may be treated like a video signal using available video cables and connectors. It is proposed to use an unbalanced 1 Vpp 75 Ohm signal to distribute the AES signal.

In one CBS edit suite we have used baluns to convert to unbalanced 75 Ohm coaxial cable. This allows longer cable runs and the use of standard video distribution amplifiers. An AES committee is working to modify the AES-3 standard to specify an unbalanced system.

CBS DESIGN REQUIREMENTS FOR A D-2 EDIT SUITE

Video

All interconnections and patching with D-2 VTRs should be accomplished using a serial digital bit stream, standard coaxial connectors and cable. Terminal equipment and distribution amplifiers are required for connection to multiple locations and patching.

Audio

The inherent limitations on cable length preclude the use of the AES-3 standard with twisted pair cabling. An unbalanced 1 Vpp 75 Ohm signal will be used to distribute the AES-3 signal. This will permit the two channels in each cable to be routed using standard coaxial cable and BNC connectors rather than the unshielded XLR connectors. It will also allow the signal to be routed using a single level of a routing switcher. While small and inexpensive format converters are available for this purpose, equipment manufacturers should be encouraged to provide the 75 Ohm output in their equipment.

Further, a multiplexing system is required to combine two AES audio streams into a single audio bit stream carrying four audio channels. In this way, the digital audio signal will be normalized to a common format, using existing video connectors and cable.

CONCLUSION

In a network environment, it is concluded that future edit suites must be designed around a serial digital transmission format, and that the construction of such edit suites should await the availability of the components and equipment required for this format.

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INTERNATIONAL TECHNICAL UPDATES AND AGENDAS

Monday, April 15, 1991

Moderator:
Charles Rush, NTIA, Washington, District of Columbia

HOW CCIR'S 1990-1994 AGENDA WILL AFFECT U.S. BROADCASTERS*
Walda Roseman
Director, Office of International Communications
FCC
Washington, District of Columbia

DAB IN THE CCIR—OVERVIEW OF THE INTERNATIONAL TECHNICAL BASIS FOR DIGITAL AUDIO BROADCASTING*
William Meintel
Datel Corporation
Arlington, Virginia

GEARING UP FOR THE SPECTRUM WAR: WARC-92
Julie E. Rones
Consultant
Washington, District of Columbia

EBU APPROACH TO BROADCAST STANDARDS: PRESENT AND FUTURE INITIATIVES AND U.S. IMPACT*
George Waters
EBU
Geneva, Switzerland

NHK INITIATIVES IN TECHNOLOGY DEVELOPMENT AND INTERNATIONAL IMPLEMENTATION*
Masao Sugimoto
NHK Laboratories
Tokyo, Japan

*Paper not available at the time of publication.

GEARING UP FOR THE SPECTRUM WAR: WARC-92

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Abstract- The WARC-92 will address worldwide frequency allocations for the accommodation of new services such as satellite sound and television broadcasting. As the WARC seeks to identify and carve out new spectrum territory for such purposes, existing users -- including **broadcasters** -- should note that a full-scale spectrum war with the potential for heavy casualties could be in the making. That the decisions taken at the WARC could directly affect broadcasters is a matter which broadcasters should understand. The WARC-92 process and the potential implications to broadcasters will be discussed.

INTRODUCTION

The International Telecommunication Union (ITU)[2] will convene a World Administrative Radio Conference in Spain (3 Feb.-5 March 1992) (WARC-92), for dealing with frequency allocations focusing upon certain parts of the spectrum[3] in contemplation of adding additional spectrum for High-Frequency Broadcasting (shortwave radio) and to address such matters as low-earth orbiting satellites (LEOs), new terrestrial and satellite-based mobile communications services, e.g., personal communications services and digital audio broadcasting (DAB) -- sound broadcasting from satellites; and new space services and satellite-delivered, Wide-RF Bandwidth high definition television (HDTV).

At the last general spectrum allocations conference held in 1979, ITU Administrations agreed to hold specialized WARCs during the 1980s and also agreed in principle to schedule other conferences in the 1990s for services not covered by the 1980 conferences. Hence, in light of the perceived need for a future conference to address those issues not dealt with during the 1980s, the supreme body of the ITU, during its 1989 meeting, authorized WARC-92. The ITU Administrative Council later met to establish the agenda, taking into account allocation issues relating to frequency allocations that were adopted at three prior ITU conferences: WARC for the Planning of the HF Bands Allocated to the Broadcasting Service, Second Session, 1987 (HFBC-87); WARC for the Mobile Services, 1987 (MOB-87); and WARC on the Use of the Geostationary Satellite Orbit and on the Planning of Space Services Utilizing It, Second Session, 1988 (ORB-88).

Federal Communications Commission Chairman, Alfred Sikes, believes WARC-92 "might well be the last significant allocation conference to be held before the 21st Century ... and that it presents a number of difficult challenges and choices [in order to encourage the provision of new technologies and services to the public, and that], the sound resolution of which is important to our country."[4]

The FCC is responsible for coordinating U.S. public and private sector WARC-92 preparatory efforts and for making and directing its recommendations to the State Department. In that regard, the FCC established a Notice of Inquiry and an Industry Advisory Committee (IAC) composed of 35 private and public sector members who advise the Commission on the FCC WARC-92 preparatory process.[5]

The National Telecommunications and Information Administration is responsible for coordinating the preparatory activities of affected government agencies through an Interdepartment Radio Advisory Committee (IRAC) and its Ad Hoc 206 interagency group, and directing those positions to the State Department.

The State Department bears the ultimate responsibility of formulating a U.S. position on the various U.S. proposals, and forms and directs a U.S. delegation that will advocate the positions at the Conference and during bilateral and multilateral meetings with other ITU Administrations and relevant bodies before the Conference begins.

If the Conference establishes an international agreement, the U.S. can abstain from the agreement in total, or it can adopt the agreement either in whole or in part.[6] After the State Department signs the agreement, Congress must elect to ratify that agreement in the form of a treaty.

Implementation of the treaty could entail Congress's directing the FCC to take necessary measures that may include frequency reallocation, displacement[7] and/or the sharing of frequency bands by different types of users.

WARC-92 IMPLICATIONS FOR BROADCASTERS

The National Association of Broadcasters, commenting in the Commission's WARC-92 proceeding, identified three specific areas of membership concern: (1) Broadcast-Satellite Service (BSS) (Sound) (BSS (Sound)), (2) HDTV BSS and (3) auxiliary broadcast service. NAB opposes allocating any spectrum for BSS (Sound). It believes there exist technical, economic and policy justifications to employ DAB in the U.S. through terrestrial means only and that the flexibility to consider these needs should remain open during the pendency of WARC-92 deliberations.[8] Additionally, NAB opposes allocating additional spectrum for HDTV BSS; and, urges that the FCC in its WARC-92 rule making continue protection of broadcast auxiliary spectrum used for live, on-air, mobile news gathering -- to counter proposed spectrum challenges to that service.

Domestic viewpoints, as thus far articulated in the FCC WARC-92 rule making on those and other relevant issues, will be discussed below; foreign proposals will generally not be discussed since most have not been formalized as of the date of this writing.

(1) BSS (Sound) Issue

The ITU Administrative Council adopted an agenda item to consider the allocation of frequency bands to the BSS and associated feeder links for BSS (Sound) in the range 500-3000 MHz, as indicated in Resolution 520 (WARC Orb-88). That agenda item included the accommodation of complementary terrestrial sound broadcasting uses within this allocation.[9] Res. 520 makes clear that the allocation in question is for vehicular and portable reception as well as for fixed receivers.

The FCC sought public comment upon whether there is a requirement for BSS(Sound) in the U.S.; what the projected spectrum needs of the service are in light of the fact that use of the UHF television band is not practical in the U.S. for BSS (Sound). (Upon subsequent reflection, the FCC decided to consider reallocating spectrum from the UHF-TV spectrum for BSS(Sound) since the service may have a "complementary" terrestrial broadcasting service.); whether any allocations for BSS should be exclusive or on a shared basis; and what other services or existing systems could be affected by reallocations for BSS (Sound), if specific bands are suggested.

The Voice of America opines that a need exists for BSS (Sound) allocations to support U.S. public diplomacy interests, that Europeans, including the Soviet Union, appear to be moving forward with this concept and that the U.S. will need to cooperate with many nations in order to provide international broadcasting services. VOA's ability to broadcast to other populations directly via satellite will depend upon reciprocal agreements in which other administrations will be given authority to broadcast to audiences in the U.S. in exchange for a U.S. authorization to broadcast into their territories.[10]

NAB, Utilities Telecommunications Council, Harris Corporation--Farinon Division and the Association of Maximum Service Telecasters feel there is no proven demand for this service, but that if the service is implemented that it should not be implemented in Region 2 and that service implemented in Europe or Asia must not create interference to current operations in Region 2.[11]

Aside from VOA, other U.S. BSS (Sound) proponents include National Public Radio[12], Satellite CD Radio, Inc. (SCD Radio), Afrispace, Inc. (Afrispace)(corporation proposes to establish a radio satellite service from the U.S. to Africa and the Middle East), Strothers Communications Inc. (Strothers), et al. Soon after the FCC began its comment period, Satellite CD Radio, Inc. and Strothers Communications, independently petitioned the FCC to establish a rule making for a domestic allocation for DAB. Consequently, the FCC established a rule making in a Notice of Inquiry in Gen Dkt. No. 90-357. This rule making to consider domestic DAB allocation issues is wholly separate and not contingent upon the WARC-92 NOI.[13]

Some studies have indicated that up to 100 MHz of spectrum would be required for BSS (Sound), including accommodation of terrestrial sound broadcasting within the same allocation.[14] BSS(Sound) proponents have advised the Commission that they believe 60 MHz of spectrum is required to meet domestic (40 MHz) and international (20 MHz) needs. Strothers recommends an additional 40 MHz of spectrum be allocated for a terrestrial DAB service.

The nature of a domestic satellite DAB allocation would entail a nationwide satellite delivery service. Some therefore believe that such a service would mean a concentration of ownership by the few and threats to localism. The FCC's WARC-92 Second NOI discusses three options, by frequency order without preference, in which to provide spectrum for BSS (Sound) service and sought comments on the proposals.

FCC Option One focuses on the UHF-TV band and is advanced by Strothers. Strothers does not propose any specific band. However, it has requested that 48 MHz be made available from within the band 225-2700. Strothers petitioned the Commission to establish a terrestrial DAB service in the U.S. and is an applicant for an experimental DAB system to operate in Washington, D.C. and Boston, Massachusetts. With respect to the range of frequencies offered under the Strother's proposal, the Association of Maximum Service Telecasters contends that 500-806 MHz UHF-TV band and the 1990-2110/2450-2483.5 MHz auxiliary broadcast bands ought be protected, whereas NAB asserts that the entire UHF-TV band at 470-806 be protected; and that a BSS (Sound) allocation in the band 728-788 MHz (UHF-TV channels 57-66) be left alone for advanced television service purposes.

FCC Option Two focuses on the 1435-1530 MHz band, as proposed by SCD Radio and Afrispace and one of the five bands mentioned by Strothers. This frequency band is heavily used for aeronautical mobile telemetering services by both the U.S. government and the private sector. The Commission believes that it is not feasible to share aeronautical mobile telemetering frequencies with BSS (Sound) or terrestrial broadcasting systems; and that any allocation to BSS (Sound) and a complementary terrestrial broadcasting service be made on an exclusive basis in the 1435-1530 MHz band.

The Commission would opt to reaccommodate the flight test and telemetry operations in the 2390-2420 MHz band, if it chose this option. This proposal has received considerable opposition from the Executive Branch and the Aerospace & Flight Test Radio Coordinating Council. Also, the Commission proposes to displace the fixed, radiolocation, amateur, and amateur satellite services currently operating in the band 2400-2420 MHz to above 2420 MHz. Also, proposed is that the Industrial Scientific and Medical Equipment band (2400-2500 MHz) be reduced to 2420-2480 MHz in order to accommodate the various shifts. (ISM devices encompass many different types of equipment ranging from microwave ovens to medical diathermy.)

Informally, Japan has opposed a worldwide allocation in the 1.4-1.5 GHz band based on a service it has recently implemented in the band.

SCD Radio proposes that every broadcast licensee would be able to operate on a local, 34 channel capacity, CD quality aural broadcast service by proposing a hybrid system where both satellite and terrestrial delivery methods would be used. It intends to make available a new 66 channel CD quality satellite radio service throughout the U.S. Concomitantly, SCD Radio proposes to launch two geostationary satellites at 103 and 121 degrees West Longitude. Each satellite would provide a total of 99 channels, 33 channels each in the Eastern, Western, and Central beams covering the U.S. The two satellites together would therefore provide a total of 66 channels to each of these three regions. Another 34 channels in each beam area would be provided by high-powered terrestrial transmitters to provide coverage in urban canyons. The feeder links (60 MHz of spectrum) would operate in the 27.5-29.5 GHz band, with the BSS (Sound) transmission in the 1470-1530 MHz band.

FCC Option Three proposes to allocate the 2390-2450 MHz band to the BSS (Sound) with a complementary terrestrial broadcasting service, vacating all other services except ISM equipment. The federal government makes limited use of this band for airport surveillance radars. The band is also allocated to the amateur and amateur satellite The IAC has expressed a preference for BSS (Sound) FCC Option Two frequency bands.

(2) HDTV BSS Issue

Res. No. 521 (COM 5/3) of ORB-88 called for a future conference to consider allocations for wide RF-band HDTV broadcasting by satellite, including the necessary feederlinks, preferably on a worldwide basis.[15]

A worldwide allocation for HDTV BSS is believed to be driven by the European countries whose current plan offers sufficient constraints with respect to establishing an HDTV BSS. WARC-77 assigned five channels for each of Region 1's services areas (Europe, Africa, the Soviet Union, the Middle East and Mongolia) and established a Plan of frequency and orbital position assignments in the BSS allocations at 11.7-12.5 GHz. Also, WARC-77 assigned four channels in Region 3 (Asia [except the Soviet Union and Mongolia], Australia, and the countries of the S.W. Pacific). Region 3 was given a Plan of frequency and orbital position assignments in the BSS allocations at 11.7-12.2 GHz. The BSS Plan for Region 2 (the Americas, the Caribbean and Greenland) established at a 1983 regional WARC established a Plan assigning frequencies to 38-Region 2 countries in the 12.2-12.7 GHz band for BSS service. The band is divided into 32 channels, each 24 MHz wide.[16]

That the bands have already been planned on a worldwide basis for the BSS and provide for 27 MHz wide channels in Regions 1 and 3 and 24 MHz wide channels in Region 2 has given rise to FCC public comment that the channels in this band would not be wide enough to provide for a wide RF-band HDTV service.[17]

Although many frequency band options have been and are being studied by the FCC during the ongoing proceeding,[18] it has announced that both the 12 GHz and 17.3-17.7 GHz[19] bands offer some promise for a future HDTV BSS and should therefore be considered. Tentatively, the Commission concludes that the current 12 GHz band is most appropriate for HDTV BSS and that the 17 GHz band will also provide an alternative allocation for those countries with assignments in the 12 GHz BSS plans that might have difficulty in implementing those assignments in an HDTV format.

Several commenters question the need for any broadcasting via satellite; others state that HDTV can be provided within the existing allocations at 12 GHz; and some envision a new service that will require spectrum separate from that allocated to the DBS at 12 GHz. The Commission will not take a position on this until it determines the nature of any HDTV BSS after examining the proper place of this service vis-a-vis the planned DBS service at 12 GHz.

(3) Broadcast Auxiliary Service Issues
Challenges by new mobile services proponents have given rise to attacks on the Broadcast Auxiliary Service. At issue are the 900 MHz band (advanced wireless personal communications, i.e., cordless telephones (CT-2)); and 2 GHz band (personal communications networks (PCNs), etc.) bands. NAB comments provide that the Commission should take into account the extensive congestion of the 900 MHz and 2 GHz bands used for Broadcast Auxiliary Services, on the basis that congestion is a critical component in any plan to make those frequencies available for mobile services.

Proponents of providing additional mobile spectrum in the 1700-2450 MHz band point to the considerable emphasis in Europe and Japan for allocations within this band to support new mobile services; e.g.., PCNs, Digital European Cordless Telephone, and Future Public Land Mobile Telecommunications Services.

Another challenge concerns a Radio Determination Satellite Service matter involving the 2483.5-2500 MHz band. The TV Broadcast Auxiliary Service is at 2.450-2.484 GHz. The Conference agenda includes Res. 708 which invites studies to obtain more precise results concerning the conditions of sharing in the 2483.5-2500 MHz bands between RDSS and other terrestrial services. Among other things, constraints on the terrestrial services to protect RDSS and the MSS are being contemplated. MOB-87 adopted RDSS satellite power flux density limits in the 2483.5-2500 MHz band as sharing criteria to protect existing terrestrial services. However, no criteria were adopted to protect the RDSS services from terrestrial services.

(4) Other Issues

(A) General Satellite Service Proposal advocates a reallocation of frequency bands above 20 GHz from the FSS to a GSS, which would permit fixed-, broadcasting-, and mobile-satellite operations in the same frequency band, using NASA's Advanced Technology Satellite program at 20/30 GHz. The broadcasting aspect of the proposal is not clear.

(B) HF Broadcasting Service Conference agenda item calls for an increase amount of HF spectrum allocated exclusively for the Broadcasting Service. Aviation, telephone companies, utilities, oil and gas and public safety HF operations stand to be displaced in the spectrum remaining to the Fixed and Mobile Services.[20]

What's Next?

The Commission is currently evaluating comments in the 1st and 2nd NOI and will issue a 3rd NOI to address limited, yet undetermined, issues. The 3rd NOI could include matters pertaining to the NAB DAB spectrum study and also discussion on any formal expressions from other Administrations. The State Department will formulate a U.S. position after sorting out any disagreements between and among the government and private sector proposals, taking technical matters, telecommunications policy, foreign policy, defense, trade and national security considerations into account.

CONCLUSION

Because a frequency allocations conference by definition is a change event, someone's spectrum territory will be invaded since spectrum is a limited resource. As the new services and technologies seek to invade, WARC-92 promises to be a full-scale spectrum war on both the domestic and the international fronts. Yet, who gets nuked in order to accomplish worldwide and domestic spectrum upheaval will be determined at the close of the Conference, and to an extent beforehand at the FCC level in the various rule makings proposing new services.

ENDNOTES

1. Licensed in Pennsylvania only.

2.

2. The International Telecommunication Union is the international organization responsible for the regulation and planning of telecommunications worldwide, for the establishment of equipment and systems operating standards, for the coordination and dissemination of information required for the planning and operation of telecommunications services and for the promotion of and contribution to the development of telecommunications and the related infrastructures. The ITU is a specialized United Nations agency with a membership of 164 countries.

3. WARC-92 will focus upon three particular frequency spectrum areas: (a) 3-30 MHz (High Frequencies (HF)): allocations for the HF broadcasting service for international broadcasting; (b) 0.5-3 GHz (Ultra High Frequencies): allocations for the Mobile Service for terrestrial use in the frequency range 1-3 GHz, the Mobile-Satellite Service in the frequency range 1-3 GHz, the Broadcasting-Satellite Service (Sound) for aural broadcasting in the frequency range 500-3000 MHz and for associated feeder links, the space research and space operation services in the 2 GHz frequency range and etc.; and (c) 11-35 GHz (Super High Frequencies): allocation for the BSS for Wide-RF Bandwidth HDTV in the frequency range 11.7-23 GHz and for associated feeder links, the new space services in the frequency range above 20 GHz, etc.

4. Sikes, Remarks at the Washington Annenberg Program Conference on the 1992 World Administrative Radio Conference (Nov. 5, 1990).

5. An Inquiry Relating to Preparation for the International Telecommunication Union World Administrative Radio Conference for Dealing with Frequency Allocations in Certain Parts of the Spectrum, Notice of Inquiry in Gen. Dkt. No. 89-554 (adopted, Nov. 28, 1989; released, Dec. 13, 1989)(1st NOI or NOI).

6. Although WARC-92 will consider worldwide change in the global telecommunications environment in all three of the ITU Regions (Region 1: Europe, Africa, the Middle East, the USSR and Mongolia; Region 2: The Americas, the Caribbean and Greenland; and Region 3: Asia (except the Soviet Union and Mongolia), Australia, and the countries of the S.W. Pacific), should the U.S. believe that it would not be in our best interest to accept a worldwide allocation for a particular service, it can opt against such an allocation and uphold its present status quo domestic allocation scheme.

7. U.S. terrestrial-fixed microwave interests who suffered displacement resulting from an earlier WARC in order to accommodate the Direct Broadcast Satellite Service are vigorously attempting to ward off another displacement at the hands of this particular WARC.

8. In the same proceeding, the Association for Maximum Service Telecasters opposes any allocation to BSS (Sound) for stand-alone terrestrial digital radio services in the UHF television band. (Citation omitted.)

9. NAB believes that a "complementary" terrestrial service permits establishment of terrestrial-only DAB systems in frequency bands that may differ among the regions of the world. It urges against a narrow interpretation of the word, opining that the word "complementary", in the context of hybrid system -- where both satellite and terrestrial delivery methods would be used --, recognizes that BSS (Sound) satellites would require installation of extensive terrestrial repeater networks to enable the system to function adequately in urban area. (Citation omitted.)

10. NAB strenuously opposes VOA's reciprocal approach on the basis that it would result in a means for foreign governments to digitally broadcast news, opinion, propaganda and perhaps even advertiser supported programming direct to the radio listeners of the U.S.

11. The Americas, the Caribbean and Greenland.

12. National Public Radio wants a 20% set-aside for public radio of any new spectrum allocated for DAB. Public radio stations now occupy about 7% of the FM band. Another 13% of band is believed to be needed to provide simulcast channel for each existing AM/FM broadcasters. Communications Daily at 3 (Dec. 15, 1990).

13. However, NAB urged the Commission to consider and resolve the major policy, service and implementation issues and requirements for DAB in Gen. Dkt. No. 90-357, prior to reaching a determination of BSS (Sound) spectrum requirements or preferred frequency bands in the FCC's WARC-92 proceeding.

14. NAB plans to introduce an NAB initiated study of spectrum requirements for DAB in the DAB proceeding. That study shall focus upon use of the technology developed by the Eureka Project No. 147 consortium; terrestrial DAB application; extending DAB technology to existing AM and FM stations; and locations in the spectrum where terrestrial DAB technology can be implemented. The study will presume that available DAB facilities will be segregated into an appropriate number of classes of facilities, in approximate proportion to the existing classes of AM/FM broadcast stations, and will analyze total spectrum requirements, frequency bands up to 2500 MHz, and compare these requirements with those proposed for domestic and international satellite digital broadcasting technologies.

15. Although it noted that certain types of HDTV could be provided in the currently planned 11.7-12.7 GHz band, the resolution cited the 12.7-23 GHz frequency range as the appropriate band from which to select a worldwide allocation. The Res. also called for further studies on the suitability of the 12 GHz band for wide RF-band HDTV, without prejudice to the existing plans. The ITU Administrative Council clarified that the agenda for WARC-92 would include consideration of the studies on HDTV below 12.7 GHz.

16. IAC document IWG-3 No.20.

17. The IAC believes that the term "wide-RF bandwidth HDTV" will become a misnomer because the quality associated with that term will be achievable in present channel bandwidths, and that "Second Generation HDTV" is more appropriate (IAC document IWG-3 No. 20).

18. Aside from the 12 and 17 GHz bands, the FCC reviewed the following frequency band options for HDTV BSS:

19.7-20.2 GHz; 21.4-22.0 GHz; and 22.5-23 GHz.*

*
- Japan seems to be the only country to have specific plans for using this band for HDTV satellite broadcasting.
- U.S. terrestrial fixed system interests staunchly opposed use of this band for HDTV BSS because of their current use of the band. Users in this band were recently displaced from the 12 GHz band to make room for the DBS terrestrial service licensee. The FCC is reluctant to require another displacement of this service at this time.

(For further discussion see, 2nd NOI, paras. 124-149.)

19. It is believed that the FCC is advocating 17 GHz as a fall-back position in order to achieve Regional consistency since Canada might be leaning in this direction. However, the Commission notes that two potential problems with the 17 GHz proposal exist: the possible need for mobile or transportable feederlinks for the 12 GHz BSS to provide for news gathering and coverage of sporting events and modifications to the worldwide secondary radiolocation allocations might be required. Second NOI at para. 144.

20. For further discussion see, An Inquiry Relating to Preparation for the International Telecommunication Union World Administrative Radio Conference for Dealing with Frequency Allocations in Certain Parts of the Spectrum, Second Notice of Inquiry (2nd NOI), Gen. Dkt. No. 89-554, paras. 124-149.

BROADCAST AUXILIARY AND SATELLITE SYSTEMS

Monday, April 15, 1991

Moderator:
Carl Girod, PBS, Alexandria, Virginia

AN OBJECT-ORIENTED NETWORK AUTOMATION SYSTEM
W.J. Spurlin
The Christian Science Monitor
Boston, Massachusetts

ADVANCED TECHNOLOGIES BRING IMPROVED PERFORMANCE TO MICROWAVE ENG TRANSMITTERS AND RECEIVERS
Dr. John B. Payne, III
Nucomm, Inc.
Hackettstown, New Jersey

A REVIEW OF SCRAMBLING TECHNOLOGY FOR BROADCAST APPLICATIONS
Stan R. Moote
Leitch Video International, Inc.
Toronto, Ontario, Canada

A LOW COST VIDEO UPLINK FOR BROADCASTERS
Anthony D. Campbell and Khalid Khan
Andrew Corporation
Richardson, Texas

ENHANCEMENTS IN VOICE COMMUNICATIONS FOR SATELLITE NEWS GATHERING
James Brink and Daryl Hunter
GTE Spacenet Corporation
McLean, Virginia

AN OBJECT-ORIENTED NETWORK AUTOMATION SYSTEM

W.J. Spurlin
The Christian Science Monitor
Boston, Massachusetts

ABSTRACT

An automated, news and public affairs radio network is described. As many as ten different simultaneous programs are fed on a routine basis to FM, AM and short wave stations by satellite via computerized audio router. The capacity exists to rout sixty outputs simultaneously. DAT tapes are randomly accessed. Cost vs. performance requirement of audio tape formats in the automated environment are analyzed, along with hardware and software maintenance and installation. The reliability of the DAT - based system is asserted to be better than that of the cart/reel-to-reel system. Object-oriented programming techniques allow new machine control modules and automation modes to be easily and flexibly added, without the need to reverify or debug software. From two years experience with the above system, generalized hardware and software specifications for radio network automation are developed.

THE CHRISTIAN SCIENCE MONITOR RADIO NETWORK

The Christian Science Monitor is known in the United States for its daily radio news programs - Early Edition and Daily Monitor Radio, distributed from the Monitor via the American Public Radio satellite network. The Monitor also operates a world wide short wave network, with transmitting stations in Maine(WCSN), South Carolina(WSHB, two transmitters), and the Pacific island of Saipan(KHBI, two transmitters). The short wave stations operate around the clock. (Fig. 1)

In 1988, as the Monitor planned the expansion of its short wave and domestic networks, it became apparent that some form of program automation would be necessary. As many as ten different, simultaneous programs, sometimes in different languages, are originated from the Monitor's Boston studios and transmitted to the short wave stations and to American Public Radio affiliates, as well as to WQTV, channel 68, Boston, and local Boston radio stations that carry the Monitor's programing.

It is further required to generate and automatically rout internal signals for monitoring and production purposes.

After reviewing various program automation systems available, it was decided to develop a system in-house, using the considerable engineering resources of the Monitor's Broadcast Engineering and Electronic Systems departments.

The inflexibility of existing network automation systems meant that an off the shelf system could not be purchased. A custom-built system would have to be purchased from the outside.

The serious penalty accepted by staying in house is, of course, the lack of outside support for the system. Outweighing this penalty are the advantages of total control over hardware and the possibility of designing software from a modular, extensible, object-oriented framework.

CURRRENTLY AVAILABLE AUTOMATION TECHNOLOGY

Existing radio automation systems fall into three broad categories:

Local Radio Station Automation.

Typically music and music-rotation oriented, one output signal is controlled. The design goal of the automation reduces itself to a solution of Turing's famous artificial intelligence game in favor of the machine. The automation system is considered to be perfect if its output cannot be distinguished from that of of a human DJ. [1]

Running from personal computer platforms, usually under MS-DOS, these systems are descendants of pre-personal computer systems based on the DEC PDP-8. The audio hardware controlled is usually playback-only, 1/4' tape, although in the past few years various digital audio systems have appeared which move the playback function to DAT or VCR.[2] In these newer systems the digital audio equipment is sometimes made to emulate older, NAB cart equipment, assuring a degree of compatibility between old software and newer audio hardware.

Local radio station automation systems are unsuitable for network operations because of their inability to control multiple, simultaneous output and cueing events.

Network Minicomputer Automation

The computer platform in Canada and the U. S. has generally been a Data General mini. Bus-oriented, audio switchers with about forty audio or video inputs and twelve (monaural or stereo) outputs are controlled in synchronization with playback and record functions at the network headend. The architecture of the switchers is derived from that of video switchers used in television production. Although attempts have been made to extend the automatic control of machines to stations on the network (i. e., Netcue), these have been somewhat less than fully successful.

The source code of these mini-controlled systems is written in assembly language, and has been maintained by software patches requiring extensive debugging after re-assembly. As technology has changed (i. e., by the introduction of digital audio), the capabilities of the automation have not kept pace.[3], [4], [5]

Time-bus, Dedicated Module Arrangements.

These are the oldest radio automation systems, dating back to the 1950's. A typical arrangement (used today, for example, by Associated Press Radio in Washington) feeds time-of-day signals (either SMPTE time code or binary-coded decimal signals derived from time code) to numerous machine-control modules located around the radio plant. Each machine-control module is dedicated to a given audio machine and can be set to fire at a given time of day, usually by thumbwheel switches. While characterized by simplicity and ease of maintenance (no software is involved), the number of machine control modules can grow very rapidly. There is further, the need to reconfigure a module, or dedicate more than one module, if a machine has to perform more than one event per day. Such systems are characterized by lack of flexibility and capability.

In the case of network automation, "It might be argued that ... the best strategy for the machine may possibly be something other than imitation of the behavior of a man", to quote Turing. In the case of single-operator automation, the capabilities required (i. e., simultaneous playback and cueing of dozens of events) are far beyond the human. Here the design goal includes a substantial reduction in staff from the unautomated level.

THE MONITOR'S DESIGN GOALS

DAT

An early decision was reached to use DAT for most of the on-air and production recording functions. This decision was reached by comparing the cost, size, timing characteristics and ease of computer control of the competing formats.

The superior fidelity of DAT was not a consideration.

The less than ideal timing characteristics of analog machines have been responsible for discrepancies in the length of radio programs as played back for air. DAT's timing accuracy is about ten times superior. A 1/4" reel to reel machine is ten times the size of, and costs two to six times as much, as a DAT machine. Reel to reel would have required the use of 25 Hz tones and foil sensing tape, unnecessary with DAT.

Computer control of reel to reel and cart machines is extremely limited, compared with the random accessibility of up to 99 cuts on a single DAT tape.

In 1988 a number of high end consumer DAT machines based on the Sony DTC-1000 became available. These shared the same remote control format and most of the same circuit boards as the Sony PCM-2500, a professional machine. Complete computer remote control of these consumer machines, it was soon demonstrated, could be accomplished by making a few simple modifications to them.

Time of Day

As the head of the network, it was originally specified that the timing of any program transmitted should not deviate more than 500 ms from National Bureau of Standards time. This specification has since been narrowed to 200 ms. Pre-roll is sometimes required to meet the specification. The CMOS clock of the IBM PC AT is set by automatic dialup.

Automation of Production

From an early stage it was decided to off-load some of the routine tasks of production (recording of backup masters; tone sequences prior to uplink; routing of live audio to uplink, etc.) to the automation system.

Integration of Audio Router

The general-purpose audio router has the advantage, in the news production plant, of making any audio source available to the audio operator in any studio (the Monitor runs twelve production studios). It was decided that flexibility demanded complete integration of the router into the automation system. A stereo, 50 in by 30 out, configuration was adopted. A Utah Scientific two plane router is used. Because the router is completely controllable under software, the number of automated outputs can be effectively doubled (from 30 to 60) for monaural production and transmission, which constitutes the bulk of the Monitor's product.

Machine Control

Machine control was assigned entirely to industrial process control cards designed for the I/O bus of the PC AT. These are relay, open-collector, optoisolated or serial I/O cards.

Day Orientation.

The simplest form of operation seems to be one in which a floppy disk for the day's events is inserted at 00:00 UTC each day. Floppy disks are cheap, reliable, easy to maintain, back up and easy to move from computer to computer. The 24- hour orientation allows easy organization, maintenance and data entry. The virtual disk capacity of MS-DOS is used to limit disk access under normal operation to the retrieve or save event list operation. We have not experienced a floppy disk failure under normal operation.

Editing, Cueing and Multitasking.

The system is interrupt driven at two levels. While word processor-like editing of the day's event list (i. e., data entry) takes place in the foreground, the system continues to take all necessary control action in the background.

Backup and Manual Control.

A complete backup computer is provided, including interface modules. The cost of the backup is about $2500. In the event of a total system failure a certain amount of manual control is possible through the control panels of the router and the front-panel controls of the machines. However, the network often requires 10 takes per second, an impossibility under manual control. If an emergency announcement or special live news insert is required, the necessary data entry can be accomplished in about one minute.

Maintainability.

In order to have some hope of maintaining the system's source code over a long period of time, the choice of higher-level languages for development reduced itself to Pascal or C, since these are the languages most widely used in engineering. Both have powerful object-oriented extensions. (The advantages of object-oriented languages in broadcast automation are discussed below.) Because of Pascal's preeminent status as a teaching language, it was felt that the average entry-level BSEE who might be expected to maintain the program would be most likely to know Pascal, hence its selection as the development language.

IMPLEMENTATION

Twelve DAT machines, four reel to reel, eight cart machines, and a fifty in by sixty out (mono) audio router are controlled by an IBM PC-AT class computer.(fig. 2)

Time checks referred back to the National bureau of standards are made once an hour. In plants not co-located with television, an automatic call could be made once a day.

The day's event list may be edited while complete control continues in the background. Approximately 1600 events per day are taken.

It is sometimes convenient to use the backup machine for editing purposes. The day's floppy disk is not required for normal operation, since most disk access is to virtual disk. Hence, the day's floppy disk may be moved from computer to computer for maintenance, editing, etc.

PERFORMANCE

Extensive software/hardware test modules have been developed which allow a confident prediction of the failure rate of the system before changes or additions are made. Testing has revealed the need to install closed-loop status provisions in some instances. However, closed-loop verification has been found to be unnecessary for router control.

The overall failure rate of the system, (failures attributable to hardware or software non-performance related to transmission discrepancies) is about 1 in 10,000. In practice it is sometimes difficult to distinguish between operator error and machine failure. A double program ID number recorded to a single point on DAT tape, for example, may cause a failure to cue, yet be reported as machine failure by the operator.

The failure rate of a properly maintained DAT machine (failure to cue) is not measurable in a reasonable amount of time with the techniques available to us. Our test modules will issue 20,000 cue commands per day, during which time no failures are typically reported.

The system is relatively easy to use. With the outbreak of war in January of this year, sweeping format changes were made in the Monitor's programing as events developed, requiring operators to enter data into the day's event list on an ongoing basis, often only an hour or two ahead of the format changes. While this data entry was going on, the system continued to perform flawlessly, issuing all necessary commands in the background.

EXTENSIBLITY

The requirements of The Christian Science Monitor do not, at the moment, include a need for either data logging or commercial insertion. These functions may have to be added in the near future. End - of - event triggering, segues and fades are capabilities appropriate to music programing that may be required.

There will be a need in the very near future to integrate Panasonic and Sony DAT machines using an RS-422 SMPTE 9-pin interface into the system. This integration will allow a simpler command bus structure, more access by the software to the status of the DAT machine, and user read/write capability of subcode information.

Data structures have already been extended, and a future requirement may be to extend these further to accommodate greater ease of entry of mnemonics and control assignments.

Any change to the automation system implies its possible growth or extension. In the past, network automation systems have been plagued by the extremely high cost of implementing a substantial change in user interface, machine control or control and logging capabilities. [4]

Extensibility and encapsulation are inherent properties of object-oriented programs. Our programs are assigned to a team of programmers, each of whom is responsible for a module or modules. With the exception of interface between modules and certain high-level object definitions, the contents of a module need not be known, except to the programmer working on that module.

A DAT machine module may use either a parallel or serial interface to control the DAT machine and determine its status. Only the programmer responsible for the module need know if a parallel or serial format is being employed.

Extensibility implies that, once a machine-control or user interface software object is defined, that the object may be extended to control another class of machines or add capabilities to the interface without knowledge of the source code of the parent module, greatly simplifying the task of the maintenance programmer.[6]

By encapsulation, a fixed data structure that limited the software to a maximum of 2048 events per day was replaced by a dynamic data structure with no particular upward bound on the number of events. This change was accomplished without affecting any of the other modules in the program (i. e., machine control, editing), and without requiring general program debugging.

RELIABILITY

The reliability of an automation system is largely determined by its software. For many years it has been known that structured higher level languages make the task of software verification easier than with their less structured counterparts.

Earlier automation systems, even the most advanced, were subject to a host of serious operational problems related to the unverifiability of the software.

In one very widely used local system of about 1978, running on the PDP-8 and controlling hundreds of cart machines and a dozen or so reel to reel machines, any character entered at the keyboard that was not part of the command set would throw the whole system into the middle of next week.

It has been shown in recent years that proof of performance of software is intimately connected with the use of object-oriented programming techniques as an extension of the structured techniques of an earlier generation of programming languages.[7]

CONCLUSION

A network radio automation system, to be truly useful, must be extensible and maintainable. On site maintenance implies the existence of a new class of maintenance engineer, whose familiarity with the fundamentals of computer programming is at least equal to his/her knowledge of audio engineering and signal theory. Object-oriented programming allows field software maintenance without a need for the maintenance programmer to have complete knowledge of the source code.

Modular, object-oriented code allows the addition of new production and transmission subsystems as these become available (i. e., tapeless audio workstations), as well as the tailoring of capabilities to the needs of the network(i. e., commercial vs. non-commercial).

The reliability of such a system has been proven by its two years' use at the Christian Science Monitor.

My thanks to Matt Adams and Michael Lamelza, without whose work this paper would not have been possible.

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[5] Wahl, Bruce, "NPR's Netcue System", *Broadcast Engineering*, March, 1983.

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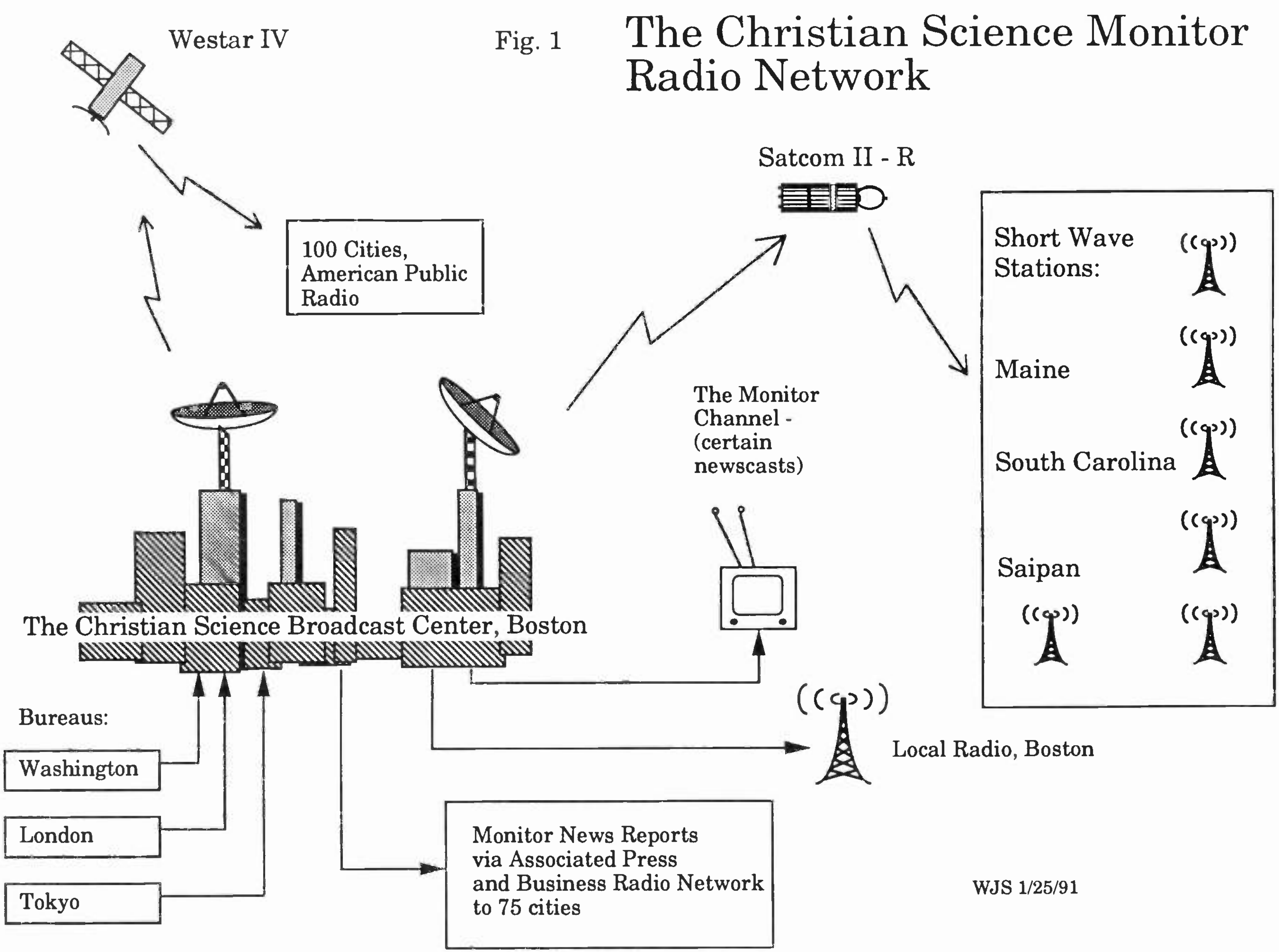
[7] Hoare, C. A. R., "Mathematics of Programming", *Byte*, Vol. 11, No. 8, August, 1986.


Westar IV
Fig. 1
The Christian Science Monitor Radio Network
Satcom II - R
100 Cities, American Public Radio
Short Wave Stations:
Maine
South Carolina
Saipan
The Monitor Channel - (certain newscasts)
The Christian Science Broadcast Center, Boston
Bureaus:
Washington
London
Tokyo
Local Radio, Boston
Monitor News Reports via Associated Press and Business Radio Network to 75 cities
WJS 1/25/91

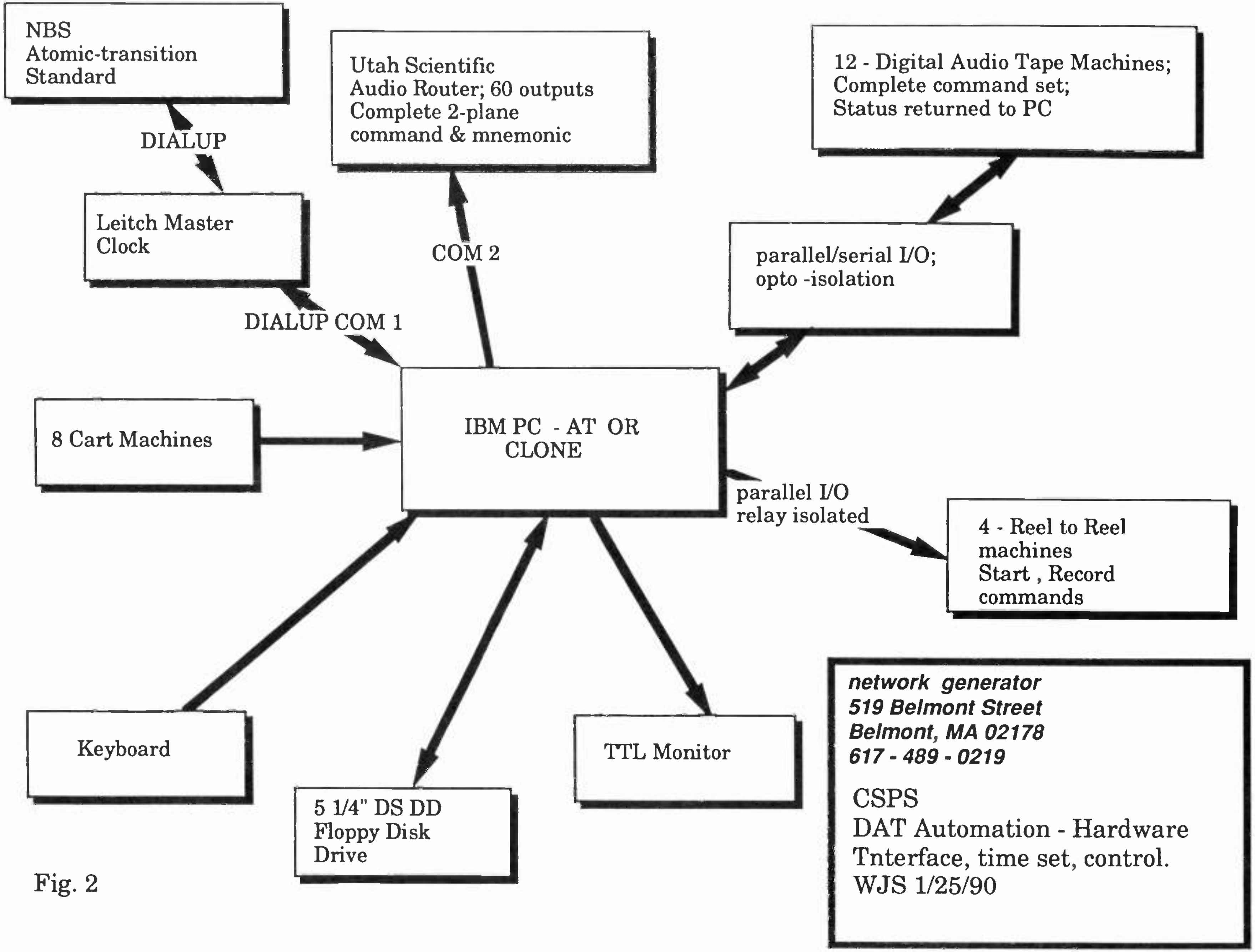

NBS
Atomic-transition
Standard
DIALUP
Leitch Master
Clock
DIALUP COM 1
Utah Scientific
Audio Router; 60 outputs
Complete 2-plane
command & mnemonic
COM 2
12 - Digital Audio Tape Machines;
Complete command set;
Status returned to PC
parallel/serial I/O;
opto -isolation
8 Cart Machines
IBM PC - AT OR
CLONE
parallel I/O
relay isolated
4 - Reel to Reel
machines
Start , Record
commands
Keyboard
5 1/4" DS DD
Floppy Disk
Drive
TTL Monitor
network generator
519 Belmont Street
Belmont, MA 02178
617 - 489 - 0219
CSPS
DAT Automation - Hardware
Tnterface, time set, control.
WJS 1/25/90

Fig. 2

ADVANCED TECHNOLOGIES BRING IMPROVED PERFORMANCE TO MICROWAVE ENG TRANSMITTERS AND RECEIVERS

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ABSTRACT

Microwave transmitters and receivers used for ENG have remained relatively unchanged for many years. However, thanks to surface mount technology, Erasable Programmable Logic Devices (EPLD), microwave silicon and gallium arsenide semiconductors, and computer aided design, today it is possible to design systems that not only perform better, but are much smaller yet include more desirable features. This article describes some of these improvements and how they are made.

INTRODUCTION

ENG transmitters and receivers utilize a broad combination of components, from RF and microwave small-signal and power devices to digital logic and active and passive power supply components. Much of the technology embodied in these devices has been enhanced, sometimes by orders of magnitude, over the last decade. Yet ENG transmitters and receivers themselves have remained relatively unchanged.

Nevertheless, some of today's microwave systems fully exploit the advances in these technologies, and deliver better performance in a smaller package than their predecessors. They also provide features that simply could not have been implemented only a few years ago.

The trend in the semiconductor industry has always been to provide more functionality in smaller spaces without trade-offs in any major performance criterion. In this area technology has delivered even more than it has promised. Today's systems generally offer more of everything in less space than one of anything did before.

One of the most important semiconductor devices is the function-specific Erasable Programmable Logic Device (EPLD), a custom chip that orchestrates many disparate functions that previously required their own space. Another is the microwave power device, which device manufacturers are steadily improving in both power output and efficiency. Small-signal receiving transistors have gained in performance over the years, yielding better lower noise figures over wider bandwidths.

Surface mount technology too contributes to reductions in cost, parts count, board size, and reliability, and is a standard feature of the most sophisticated electronic systems. Of course, it would be much more difficult to build such complex systems without the help of computer-aided design, which today produces excellent results well into the microwave region.

These technologies have allowed the incorporation of many new functions into the serene world of ENG transmitters and receivers. These functions include SMPTE color bar generator, programmable ID generator with memory, RS-170A gen-lockable synchronizer, multiple band operation from 2 to 13 GHz, single AC-DC power supply that operates from 100 to 260 VAC without jumper change as well as from 11 to 32 VDC, and many others.

CONTRIBUTORS TO INCREASED PERFORMANCE

While the overall design of the transmitter or receiver is extremely important, the technologies upon which it is based and the design and manufacturing techniques that bring it to life, are equally important.

Surface Mount Technology

This manufacturing technique improves performance in nearly every area. Size and power consumption of the overall system are reduced, and printed circuit board area declines by a factor of 2:1 to 4:1. Volume shrinks by 10:1 and power consumption by 100:1. Frequency response performance can be improved from 3:1 to 5:1 and assembly time can be reduced by 5:1 to 10:1. Overall reliability is improved as well. It is safe to say no other manufacturing technique contributes more to the overall improvement of an ENG transmitter or receiver than surface mounting.

Erasable Programmable Logic Devices (EPLDs)

EPLDs are custom designed multi-function ICs that can be designed and simulated by an engineer on a microcomputer. A blank "PROM like" PLD IC can be burned-in using the computer, erased, and reprogrammed. One type of EPLD IC can be programmed to perform many different functions so only one blank IC need be stocked. The result is low inventory and just-in-time delivery of programmed ICs. Revisions to the design can be made easily without making inventory obsolete or by changing the design.

Microwave Silicon and Gallium Arsenide Technology

Watts of power can easily be produced above 23 GHz with today's solid-state devices, and with higher efficiencies as well. Thanks to 40 years of design experience and the advent of microwave CAD, most of the black magic in microwave design has been removed. Amplifiers can be computer-designed to work the first time, and there need be virtually no tweaking. These factors alone produce units with improved reliability, lower cost, and smaller size.

In addition, multiple functions--oscillator, low noise amplifier, mixer and preamplifier--can be designed into single "supercomponents", so cost is significantly reduced. CAD allows designers to incorporate better performing components, and more accurately accommodate the electrical, mechanical and heat transfer properties of the system.

Computer-aided Design

Like so many areas of the industrial world, microwave technology too has benefited from the power of the computer. Today it is possible to design a complete system in the same amount of time formerly required to construct just the fundamental design of one of its elements. While CAD is generally associated only with the digital world, it is a staple too of microwave design.

Linear as well as nonlinear analysis of microwave components and circuits is routinely conducted on minicomputer workstations, with good results into the low millimeter wavelengths. This capability gives the ENG transmitter and receiver designer a huge advantage over the previous method -- tweaking.

THE BENEFITS TO ENG TRANSMITTERS AND RECEIVERS

With lower power consumption and higher-frequency operation, new ICs are being designed that combine many more functions on a chip. As a result, the basic transmitter and receiver can be made smaller and lighter, and more functions or enhancements can be incorporated to improve overall versatility.

Color Bar and ID Generation

A typical color bar generator of only a year ago required about 306 in.^{3} (8.5 in. x 12 in. x 3 in.). Today a gen-lockable SMPTE color bar generator with programmable 16 character ID can be designed into less than 16.5 in.^{3} (3 in. x 8.5 in. x 0.65 in.), a reduction of 18:1. This feature can easily be incorporated into today's ENG transmitter (see Fig. 1).

The small color bar generator can be incorporated because of surface mounted components and function-specific EPLDs. Not long ago the industry marveled at its ability to have a custom IC designed for as little as $20,000 and in production 6 months later for $10 apiece. With EPLD technology, engineers can design and program a custom IC on a microcomputer in a few hours.

The EPLD is like an EPROM. Once the design has been entered into the computer and its operation fully simulated, a basic EPLD IC is burned-in. The designer can now connect the IC into a circuit and evaluate its actual performance. If a change is required in its operation or design, the EPLD can be erased and reprogrammed quickly. For example, in a color bar generator designed by Nucomm, a single EPLD device generates the majority of the color bars. During its initial testing, several minor flaws where observed that needed to be changed. With a CAD system, several patterns were observed and the correct one chosen without ever touching a soldering iron.

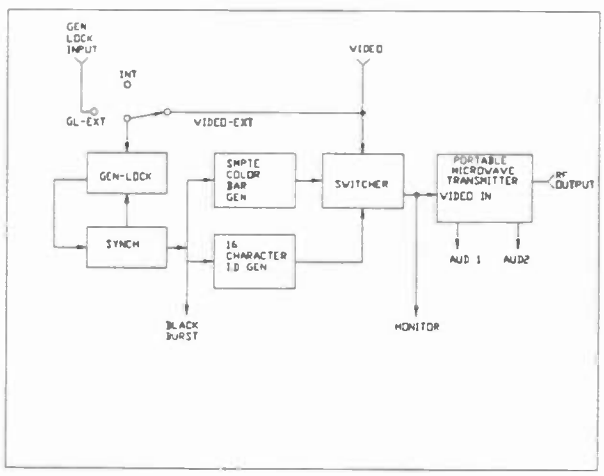

Fig. 1 Color bar generator, 16 character ID generator, and gen-lock circuitry

Synchronization Circuitry

A small oven to stabilize the crystal oscillator in the color bar generator, and the extraction of the black burst waveform, provide the basic elements of a synchronizer. If the television camera can be synchronized, the next step is to synchronize the color bars and the black burst to an incoming video signal from the camera. Using custom ICs and EPLDs, all of these features can be incorporated into a portable ENG transmitter without increasing its size or power drain. This eliminates two pieces of expensive test equipment.

Video-Activated Color Bar Control

As circuits get smaller, less expensive and consume less power, it is possible to incorporate a video-activated remote on/off control into the transmitter (see Fig. 2). The transmitter operates in remote standby with no video present and in full operation by applying a video signal. When video is removed, the transmitter automatically returns to standby.

Upon the application of black video, the transmitter turns on with color bars present. Black video can be produced by placing a cover over the lens for cameras without a black video feature. Upon the application of video, the transmitter automatically switches off the bars and turns on the regular video.

Smaller Footprint Systems

Today's microwave transmitters and receivers are much smaller and more efficient, thanks to silicon and GaAs technology. The frequency synthesizer of only a few years ago occupied 64 $in.^3$ and consumed 17 W. Today that same synthesizer is built in 9 $in.^3$ and consumes less than 5 W. Power amplifier design has been significantly improved by the use of computer design programs such as Touchstone and SuperCompact. In the past, microwave components have been one of the largest contributors to equipment failures. As designs become smaller and simpler, reliability has improved significantly.

More Efficient Power Supplies

Using state-of-the-art switching power supplies, size and weight of microwave transmitters and receivers can be reduced and overall efficiency improved. Efficiencies of 80 to 90 percent are possible today, which means there is less heating when operating from AC and longer battery life when operating from DC.

Power supplies can now be built that operate over the full input voltage range from 100 to 260 VAC, eliminating the possibility of burnouts caused by the wrong cable or incorrectly marked outlets. An input voltage range of

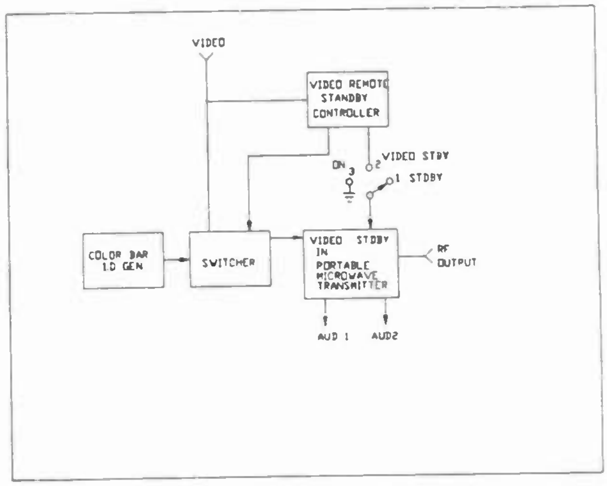

Fig. 2 Video remote standby controller

11 to 32 VDC can also be realized, which makes it possible to operate the transmitter from standard 12 VDC locations as well as helicopters that use a 24 VDC supply.

Fewer Demands on ENG Truck Space

All of these advances produce a transmitter that reduces demands on the host vehicle. In the past, three rack spaces (5.25 in.) have been required to accommodate the transmitter and power supply for the mast-mounted amplifier. Today, both transmitter and mast-mounted amplifier and power supply fit in 2 rack spaces (3.5 in.). A built-in color bar generator and synchronizer yield additional space savings.

Multiple Bands in Single Units

Broadband, compact frequency synthesizers and power amplifiers can now be built to cover multiple ENG bands. Several companies now offer receivers that cover the 2, 2.5, 6, 7, 12 and 13 GHz bands, and transmitters that cover 2 and 2.5 GHz or 6 and 7 GHz, or 12 and 13 GHz. However, it is now possible to incorporate all of these bands in a single ENG transmitter without increasing the unit's overall size.

70 MHz IF Input to Portable Transmitter

Although most portable as well as fixed receivers have 70 MHz IF output connectors, most portable ENG transmitters do not accept 70 MHz inputs. The ability to accept a 70 MHz IF input is desirable when using a receiver and transmitter back to back as repeaters. Using custom IC chips and surface mount technology, a very small, high-performance 70 MHz discriminator can be incorporated in the transmitter (see Fig. 3). The modulation and demodulation process takes place within the transmitter where discriminator and modulator characteristics can be matched for optimum performance without video degradation.

This approach has several advantages in ENG applications over the traditional upconverter technique usually employed. In most portable ENG receivers the frequency accuracy of the 70 MHz IF may not meet the FCC requirement for frequency stability. By demodulating and then remodulating, this frequency inaccuracy is removed. In addition, group delay error introduced by sending the video or composite video signal over a distance is minimized when using a 70 MHz IF.

Fiber Optic Camera-to-Transmitter Interconnects

There are several fiber optics systems on the market for transmitting video and audio between camera and ENG van. Most of these systems require an interface box at both ends of the fiber cable. Depending on the number of video and audio program and communication channels, one system may be desirable over another. However, a very simple system can be implemented between

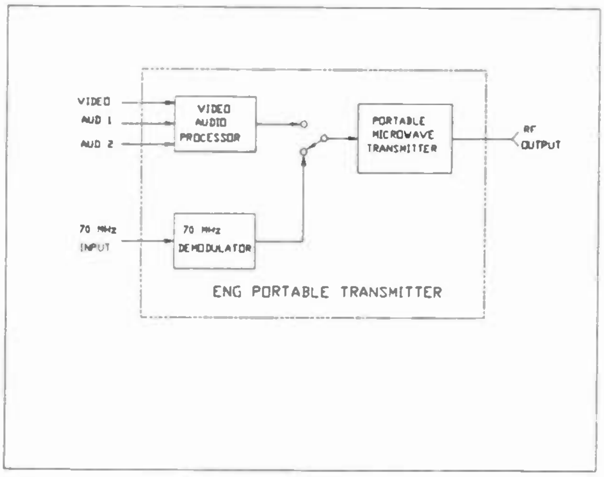

Fig. 3 Portable transmitter with 70 MHz input

the camera and the ENG transmitter (see Fig. 4).

A typical modulator with two audio subcarriers and video processor measures about 3 in. x 4 in. x 2.5 in. with connectors. The module is located at the camera. The composite video signal modulates a simple low-power LED, the output of which is coupled into a fiber optic connector. A length of fiber optic cable connects the camera to the ENG transmitter. The other end of the cable at the ENG transmitter is demodulated by an optical detector. This detected signal is the composite video.

Since the composite signal modulates a light beam, no video distortion will occur caused by its transmission down the optical cable. The composite output of the

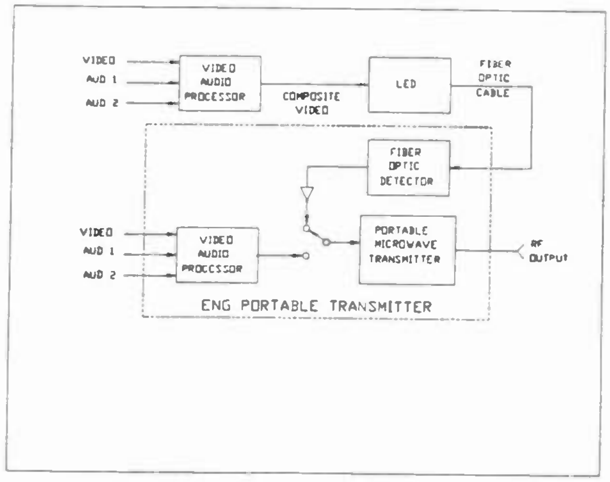

Fig. 4 Fiber optic interconnect from camera to ENG transmitter

detector can now be used to modulate the ENG transmitter through its HF input connector. Since the optical detector is built into a fiber cable connector, the connector can be mounted on the ENG portable transmitter and connected directly inside the unit through an input switch. The result is a compact self-contained fiber optic camera interconnect system that requires only a small module at the camera end.

If the ENG transmitter is equipped with the remote-activated video controller and the color bar generator, the detected fiber composite signal can be used to switch the transmitter from its standby mode to its "on" mode. With the transmitter on, color bars or program video can be selected made remotely as discussed earlier.

SUMMARY

The ENG transmitter, once an all-too-familiar sight, is changing rapidly. It is becoming smaller and lighter, incorporates many of the features found only in outboard units not long ago, and generally performs more reliably.

Additional improvements will be made as technology allows, such as two-way communication on a single transmit/receive link and remote control features delivered on additional subcarriers or via the vertical component of the signal.

A REVIEW OF SCRAMBLING TECHNOLOGY FOR BROADCAST APPLICATIONS

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Abstract - Scrambling is the most effective way of combatting satellite piracy of 'backhauls' and network television signals.

However, scrambled signals must be given special consideration since they are to be transmitted through existing terrestrial and satellite networks, while still preserving the utmost quality and security.

This paper reviews the advantages and disadvantages of the various analog and digital techniques of scrambling. The synchronizing information, finite bandwidth restriction, characteristic non-linearity and noise of a typical link are discussed.

An ideal system is a complete video, audio, and data scrambling product. A close look is taken at the various system configurations as well as the implementation.

Why Scramble?

Scrambling is often discouraged. The complexities of signal transmission alone cause enough problems without having to scramble.

The main reason a broadcaster scrambles is to stop unauthorized viewing of broadcasts. For example, when a major live event is covered, the remote crew covers the event and beams back the signal, over 'backhauls' to network master control. Master control inserts advertising and network promotions before beaming the signal (network feed) to its affiliates. The local affiliates then insert their local material before the final transmission to the consumer. In some cases, live events are 'blacked out' in an area so that local affiliates do not receive a particular event.

Anyone with a satellite TVRO could receive either the 'backhaul' or the network feed. The only way to protect against this is to scramble both network and backhaul feeds.

Scramble Audio/Video

Video Scrambling

Why not just scramble the audio and forget the video? Will people watch a picture without sound? Most definitely! Major sporting events have very little sound associated when viewed in person. Many people will not watch a political event without sound, but certainly fights, ball games, and horse races don't need sound.

Audio Scrambling

Many sporting events are only blacked out on television, not radio. For these events, audio scrambling is not important. Election black outs are, of course, on both television and radio, and audio scrambling here is very important.

What the Broadcaster needs

The broadcaster needs to protect his network and local advertisers. Any TVRO owner would prefer to watch a clearer and commercial free satellite picture on the 'backhaul' or the network feed rather than from off-air or cable. To prevent this, the broadcasters must scramble.

Scrambling for the broadcaster presents different challenges than scrambling for Pay-TV, whether cable or DBS. Unfortunately, broadcasters do not always have direct access to their own uplink and downlink. The signal may travel through several paths, each maintained by various companies, before reaching the affiliate's master control. This can include numerous microwave, satellite, copper land lines, fiber transmissions, and various types of both uplink and downlink equipment. All these paths have finite bandwidth and distortions, but have been tailored to handle baseband standard audio and video.

If scrambling is desired, signal compatibility is necessary. It must be possible to monitor signal quality throughout the transmission path without a descrambler.

Methods of Scrambling

Various methods of scrambling must be examined individually:

Sync suppression/elimination or complete video inversion is commonly used in cable systems. Most video transmission equipment relies on the presence of sync for a DC reference, clamping, and sometimes AGC. Energy dispersal units used in C band exciters need sync to vertically lock the dispersal waveform that is added to the video before satellite transmission. The receiver needs sync to remove the dispersal waveform to trigger its clamping circuitry. Also, standard video monitoring equipment cannot be used.

Video inversion of the active horizontal portion only, preserving sync and inverting whites (100IRE) to blacks (7.5IRE): This method is only secure when the lines are randomly inverted, which directly relates to the doubling of link non-linearities. For example, a link having a differential gain of 3% would double to 6% due to the random inversion.

Line reversal, again, must be random to be secure, and low frequency link distortions, such as line tilt, will appear to double.

Conversion from NTSC to a different format for transmission: If standard sync is maintained, most format conversions that preserve the NTSC signal, once decoded, have an increase in bandwidth. Not all transmission paths can be easily modified to support the increase in bandwidth. In most cases, this change in bandwidth requirement will exceed the available bandwidth typically used for audio or control sub-carriers. Additionally, link performance parameters have been designed for NTSC signals, not other formats. Again, standard monitoring equipment cannot be used.

Line segmentation, or cutting a horizontal scan line into segments and then swapping the segments, also increases bandwidth. Consider a linear ramp. The ramp starts at blanking (black), increases to white, then returns to blanking level. When this line is segmented into two halves, and the two halves are swapped, the result is a sawtooth, with the first segment starting at grey. These new transitions at the beginning and end of the line now have zero rise time (or infinite bandwidth). Once transmitted through a limited spectrum, these edges will become sloped and will ring. When the signal is reconstructed, these sloped and ringing edges will occur as artifacts. Also, any line time tilt will result in an abrupt level shift where the two segments meet.

Product Design Goals

The scrambling process that the broadcaster requires must adhere as closely to NTSC as possible in order to incur the least number of artifacts once transmitted over various paths. Standard monitoring equipment should be able to identify transmission faults throughout the transmission path, whether the signal is scrambled or not, and without a descrambler.

The scrambling method selected must make the video signal unintelligible when viewed on a picture monitor. If audio is scrambled, it must also be unintelligible.

The system must not allow unauthorized access, therefore must be secure, and the decoders must be individually addressable.

Audio must be timed (lip sync) to video after the scrambling process.

The system must be compact in size to be practical at remote sites.

Advanced Technology

The ideal system uses line dispersal to achieve scrambling. The picture is scrambled vertically. This means that full field test signals do not appear to be scrambled (except for the mismatch in color interlace, Figure 1). The natural break of active video during the horizontal blanking interval is ideal for line dispersal. Full sync and burst is maintained, and there are no bandwidth restrictions. Basically, if a good quality picture goes through the link, so will a line-dispersed scrambled signal.

Most link distortions generate the same artifacts, whether scrambled or not, with one exception. Unacceptable low frequency distortions, such as

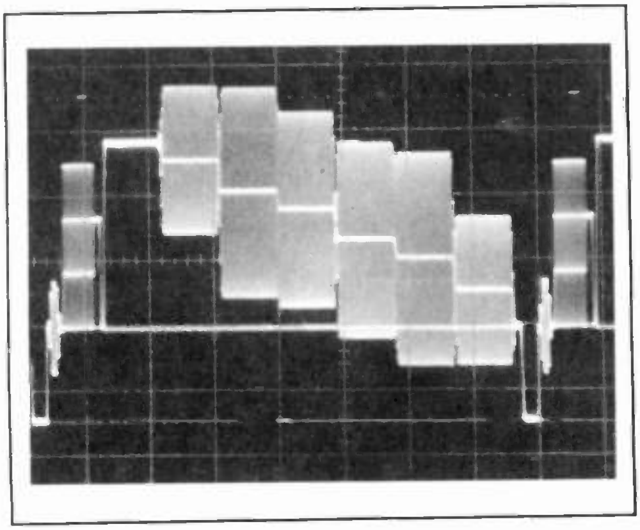

Figure 1: Full field scrambled color bars

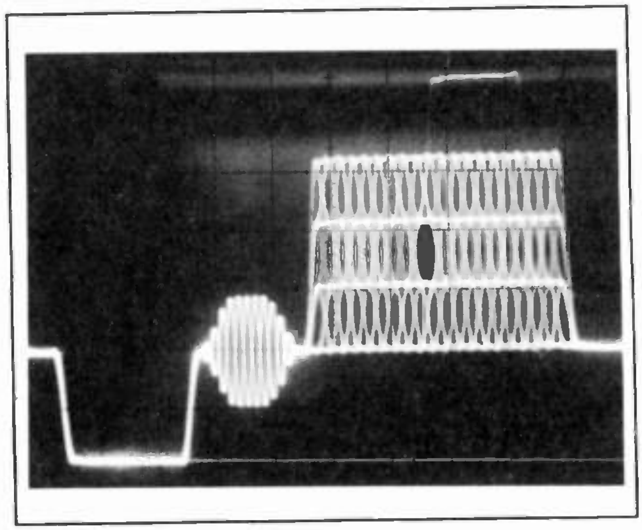

Figure 2: Audio PAM data on back porch

vertical tilt or APL-dependent clamp errors, cause significant errors on line dispersal scrambling systems, where with standard video the fault shows gracefully. Fortunately, there are few links that exhibit this fault, and they are easily recognized. This system has a tilt corrector option that digitally corrects these errors and dramatically improves basic link performance.

The compact scrambler (2 RU), splits the active picture area into two blocks of 120 lines. The lines are randomly intermixed within these blocks, representing 120! (factorial), or approximately, 10 to the power of 200, different ways of arranging these lines. To add to the complexity, the scrambling pattern randomly changes twice each field (120 times per second).

This leaves the vertical interval, lines 10 to 21, not scrambled and clear for VITS, VIR, and control information. The VITS on the scrambled signal can be analyzed on standard monitoring equipment, and control information can be used.

To achieve good quality and secure audio scrambling, this process must be done digitally. Unfortunately, baseband audio lines will not pass scrambled full bandwidth digital audio, so it was decided to transmit the scrambled digital audio along with the video. Video sync pulses are shortened, equivalent to equalization pulses (2 us), and pulse amplitude modulation (PAM) of the audio is placed on the back porch between color burst and active video (Figure 2). Since most transmission paths only use sync tip clamping, and not back porch clamping, this is acceptable.

Two channels (stereo) of audio are digitized using adaptive delta modulation (ADM) digital audio encoding sampled at 220 KHZ.

Traditionally audio is digitized generating an absolute value, pulse code modulation (PCM). Delta modulation produces digital data indicating the change of the analog signal. This significantly reduces the bit rate necessary to transmit digital audio. When there are transmission errors, the effect is less severe. ADM errors are very low in amplitude and typically sound like distortion within high frequencies, whereas PCM errors are very loud crackles and pops creating the need for heavy error correction.

The descrambler (1 RU) has been designed for permanent operation. There are relay contacts for both audio and video. The descrambler stays in bypass until the system control data is sensed. Audio and video are then descrambled if authorized, otherwise black without audio, and new sync and burst are inserted in the output as per RS-170A.

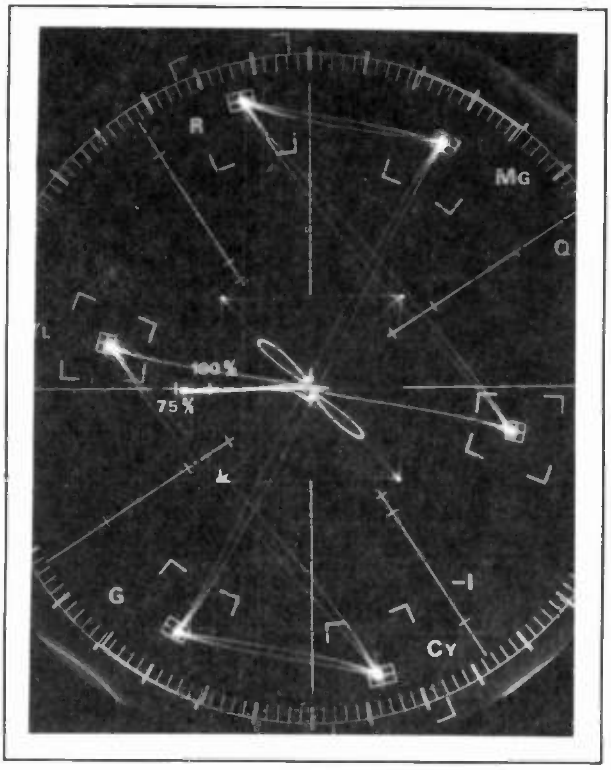

Figure 3: Scrambled full field color bars.

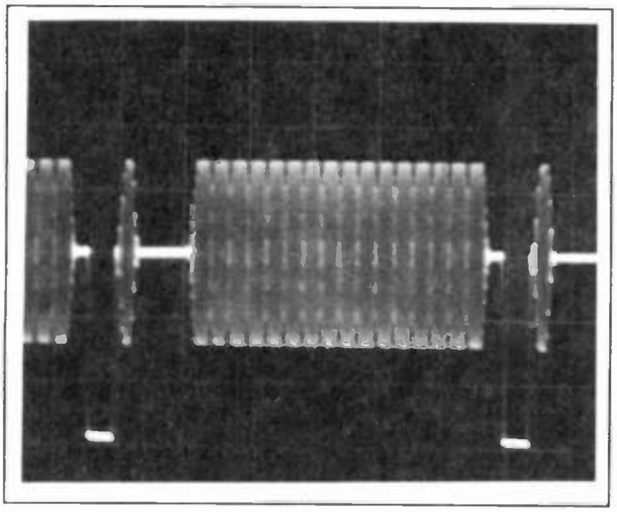

Figure 4: QPSK Data Packets on Post & Pre Equalizing Pulses

Control Data

Once the picture is scrambled, an authorized descrambler must have the information necessary for picture reconstruction. This control information is encoded and transmitted. NOTE: QPSK control data as quadrature phase modulation (QPSK) on the color sub-carrier (Figure 3 & 4). Packets of data are placed on the post and pre-equalizing pulses. This control data does not affect sync strippers throughout the transmission path, since they have color sub-carrier notch filters.

Additional control information is also sent on this QPSK carrier: 9600 baud data channel, descrambler authorization data stream, basic timing data, on screen text channel, GPI control, and re-keying (security) information.

Security

It is pointless to scramble if the scrambling method or system is not secure. This is the purpose for using various random scrambling patterns. Having so many patterns changing rapidly makes it practically impossible to reconstruct the picture.

The control data stream is encrypted within the scrambler, using a mathematical transformation which changes data into apparent random noise with the use of a KEY. By using the same KEY and the inverse transformation, the encrypted data stream can be restored. Only the correct KEY can restore the data to its original form. The ViewGuard system uses a 56 bit KEY which translates into 10 to the power of 16 different possible KEYS, making KEY searching for the purpose of system compromise inconceivable. Only the correct KEY can restore the data stream, therefore the security of the system is a direct function of the distribution of the KEY.

More than 100 KEYS are super-encrypted and stored in a tamper-proof medium within the descrambler. The scrambler can revise the KEYS within each decoder by sending a new set of super-encrypted KEYS over the QPSK data link. KEYS must be revised one descrambler at a time. If a descrambler is stolen, it will miss out on the re-keying process.

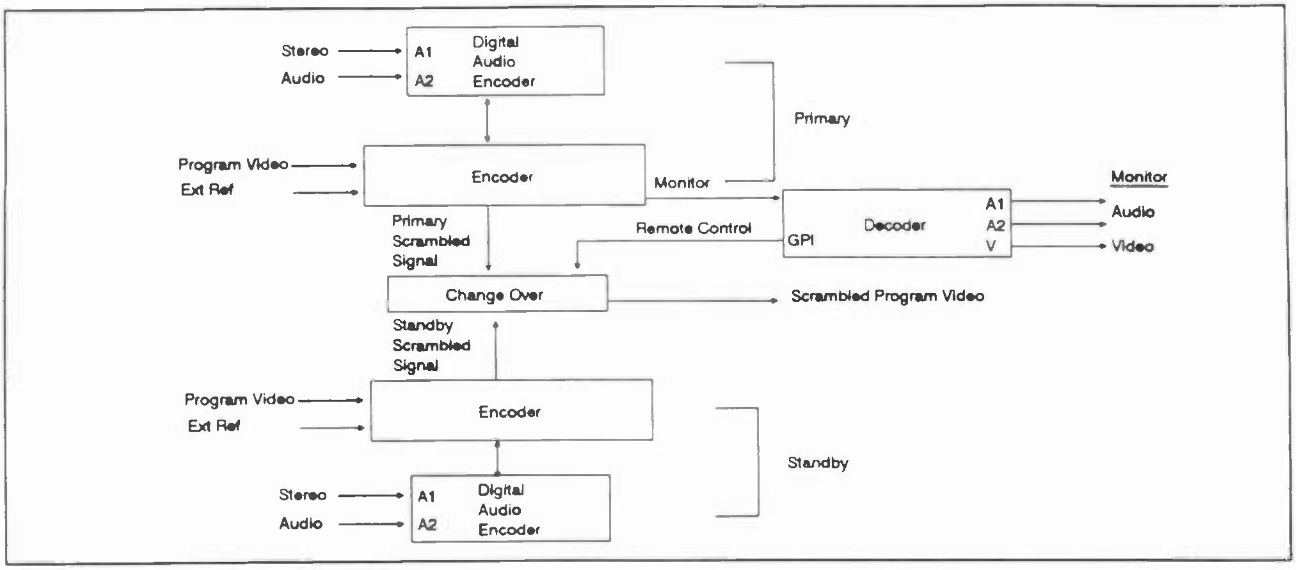

Figure 5: Redundant Scrambler Site

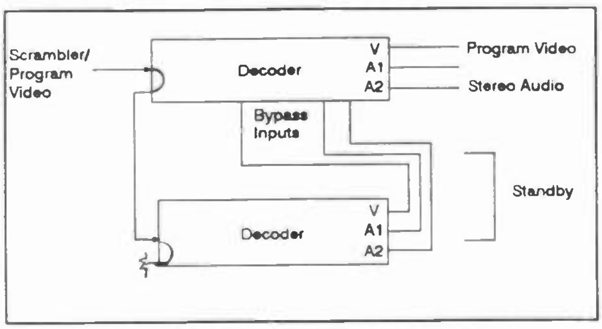

Figure 6: Redundant Descrambler Site

Redundancy
The system has been designed to allow complete redundancy at both the scrambler and descrambler sites. The back-up system can be set up with automatic change over control or two units can be parallelled for a manual switch over.

For automatic operation, two scramblers, one changeover and a monitor decoder are connected, as shown in Figure 5, at the scrambler site. If the monitor decoder senses a failure, the decoder's GPI relay forces the changeover to switch to the hot standby scrambler. Figure 6 illustrates a redundant descrambler site. When the primary descrambler is powered off or fails, it provides a relay bypass for the hot standby descrambler's outputs.

Control
Most 'backhaul' feeds are received by a single site. Once a feed is scrambled it is necessary to address or enable the correct descramblers. In this case only a few descramblers need to be enabled. The scrambler unit allows addressing for up to forty descramblers with local DIP switch control. A modem interface on the scrambler along with PC software allows limited descrambler control for up to seven thousand decoders. For even larger systems such as network feeds, a control computer is necessary. The encoder computer supports up to 15 scramblers via a high speed RS-485 Interconnect and up to 50,000 descramblers. This provides complete control of authorization, rekeying, multi-tier control and a comprehensive scheduler. Modem control via the encoder computer is also available.

Quality Control
The broadcaster is very concerned about quality of all the network feeds. The system is equivalent to passing the video through two good quality frame synchronizers. Typically all feeds are passed through a frame synchronizer before being transmitted. This is not necessary with the unit since a frame synchronizer is contained within the scrambler. This built in frame sync not only sets up an approximate video delay to match the audio delay of the Digital Audio encoder, but also verifies that the program feed is free of all non-synchronous switches before the video is scrambled. Otherwise, if a scrambled

feed sees a non-synchronous switch, both the scrambler and descrambler must not only achieve video lock but also descramble lock. This relock can take several frames and during the re-locking process, control data is corrupted.

The system uses high quality 8 bit circuitry sampling at 14.3MHz. When more than two scrambler/descrambler passes are necessary, a 10 bit scrambler may be desirable.

Conclusions

Following thorough analysis of the short comings of the common video, audio and data encryption techniques, and in consideration of the current and future broadcasters requirements, the design goals of the Leitch ViewGuard® system have called for a scrambling system that is clean, NTSC compatible, addressable, compact and of uncompromising security.

The development of this product has surpassed these goals, further offering unique methods of redundancy and control as well as uncompromised video quality.

ViewGuard® is a trademark of Leitch Video Ltd.

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A LOW COST VIDEO UPLINK FOR BROADCASTERS

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Abstract- Large aperture antennas equipped with high power TWT or Klystron amplifiers have been considered as a key to transmitting a quality video signal to a satellite in the 1980's. Improvements in satellite performance, as well as ground communications equipment, now make it possible to implement smaller and less costly satellite earth terminals. A system utilizing a 3.7 Metre Ku-Band antenna and a 75 Watt amplifier is capable of performance levels equivalent to the large systems of the past. This paper presents the design parameters and physical requirements for such a system. A brief discussion on the currently available required equipment is presented.

INTRODUCTION

The use of Ku-Band satellites to distribute video and audio signals is well accepted in the broadcast industry. Because of the possibility of rain induced fading there was an initial reluctance to employ Ku-Band as a means of distribution. While rain fades have not gone away, their severity and duration have become better understood and accepted. Since one of the primary uses of Ku-Band has been the collection of news events, Satellite News Gathering; SNG, the transient nature of this format is not commonly adversely affected by rain.

The initial high cost of the equipment for a Ku-Band satellite earth station has prevented many organizations from acquiring this ability. Recent advances in satellite and earth station equipment now make it possible to provide basic uplink capability for far less than previously considered. This technology and equipment has been proven to operate easily and reliably.

SATELLITE COMMUNICATIONS SYSTEM BASICS

At this point we must cover the basic operation of a satellite system. In broadcast applications the satellite of interest is almost always in a geosynchronous orbit. This allows the satellite to appear stationary over a particular point on the earth.

A satellite system consists of three basic subsystems. A transmit earth station provides the source of the information. The transmitter conditions the video and audio, modulates the information on an FM carrier,

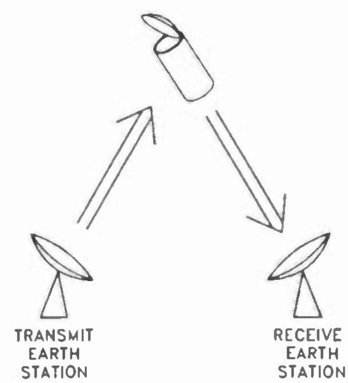

Figure 1
Typical Satellite System

upconverts the information and amplifies it to a level necessary to saturate the satellite with sufficient energy. This energy is transmitted through approximately 22,300 miles to the satellite. The satellite then receives, conditions, amplifies and retransmits the signal on a frequency offset by 2300 MHz, domestic Ku-Band. The signal again travels 22,300 miles to be received by the earth station. The receive system amplifies the signal with an antenna and low noise amplifier; LNB, downconverts, demodulates and conditions the audio and video signal.

As with all operational communications systems, the information we put into the transmitter is very seldom exactly what is received. Noise generated by the electronic equipment and transmission through space causes this degradation.

SATELLITE PARAMETERS

To explain the operation of this system, the parameters of a typical high-performance Ku-Band satellite must be defined. The performance of a satellite can be basically defined by three parameters. It must be noted that the value of these parameters will vary with the "foot print" of the satellite.

Saturation Flux Density

The Saturation Flux Density, SFD, is a measure of the sensitivity of the receivers of the satellite. Expressed in dBW/m^2 this parameter determines the amount of earth station power that is required to saturate the satellite. Typical values range from -95 to -70 dBW/m^2. The lower the number the less power required from the earth station.

Effective Isotropic Radiated Power

The Effective Isotropic Radiated Power, $EIRP_{sat}$, describes the RF power output. In general the larger this number the better the earth station system performance will be. Typical values are 33 to 45 dBW.

Satellite Gain/Temperature

The Satellite Gain/Temperature, SG/T, defines the noise performance of the receivers in the satellite. The greater this number, the better the performance. This parameter, expressed in dB/K, typically varies from -13 to 5 dB/K. The SG/T is related to the SFD. The better the SG/T the better the SFD.

EARTH STATION PARAMETERS

In our satellite system the earth station equipment also contributes to the overall system performance. To calculate the overall system performance a number of factors are used to determine the receive video signal/noise ratio. This is primarily determined by the total carrier/noise ratio of the system, C/N_{total}. The C/N_{total} is determined by the various C/N contributions from the uplink, downlink, interference and adjacent satellites.

Uplink Power

The Uplink Power, $EIRP_{up}$, is a measure of the total power output of the system. The required $EIRP_{up}$ is determined by the SFD, spreading loss and transponder backoff. The waveguide loss and branching/combining losses must also be accounted for to determine the high power amplifier power required.

Uplink Carrier/Noise Ratio

The uplink carrier/noise ratio, C/N_{up}, is a function of the SFD, satellite G/T, uplink frequency and IF noise bandwidth of the receive equipment.

Receive Earth Station G/T

The primary measurement of the receive system performance is the receive earth station G/T, G/T_{down}. The factors contributing to the G/T_{down} are the receive gain of the antenna and the system noise temperature. The system noise temperature is the addition of all the noise contribution including the antenna, LNB, VSWR noise and interface waveguide noise.

Downlink C/N

As mentioned before one of the contributors to the total system carrier/noise is the Downlink C/N, C/N_{down}. The noise contribution includes pathloss, satellite EIRP, G/T_{down} and the receive system IF noise bandwidth.

System Video Performance

The receive signal to noise ratio is determined primarily by the C/N_{final} and a number of receive parameters including receive IF bandwidth, baseband frequency and video deviation.

The C/N_{final} is effected by the C/N_{down}, C/N_{up}, C/N_{terr}, C/N_{adj} and C/N_{xpol}. In a Ku-Band satellite system the C/N_{terr} is not a factor. This is contribution due to terrestrial interference. The contribution due to adjacent satellites, C/N_{adj} is an important factor. The C/N_{xpol} due to antenna cross polarization can also become an important factor when the feed used on the receive antenna does not deliver adequate performance. In high quality systems of the five C/N contributions mentioned above only the C/N_{down} and the C/N_{up} contribute significantly to the total C/N ratio.

CONVENTIONAL EARTH STATION CONFIGURATION

A typical current satellite uplink system consists of the following equipment:

5.6 M Antenna
100 ft Interconnecting Waveguide
300 Watt TWTA
Video Exciter
Receive Equipment

This equipment can cost in excess of $200,000 depending on the exact configuration. The major disadvantage to the system is its physical configuration. The amplifier is mounted in the shelter or building. This requires that the high power RF be routed though the 100 foot of waveguide. This waveguide exhibits a loss of approximately 5 dB. The expensive 300 Watt amplifier now only provides about 80 watts at the base of the antenna. In most case this is sufficient power to operate over many satellites. If this power is not enough several steps can be taken. The first is to provide a higher power amplifier or phase combine two amplifiers. This configuration is very popular but will add at least $50,000 to the system cost. The other option is to position the antenna closer to the RF equipment. This may require the purchase of an equipment shelter.

LOW COST VIDEO UPLINK

As discussed above the major drawbacks to conventional Ku-Band uplink systems are the high cost and the necessity to locate the high power amplifiers very near the antenna. The Low Cost Video Uplink addresses and solves both of these problems without greatly reducing the functionality of the system. Four factors have made this system possible.

Improved Performance Satellites

Many of the recently launched Ku-Band satellites now have SFD's that are low enough to allow earth station with lower EIRP's to operate. The current series of GSTAR® satellites have a maximum SFD of -92 dBW/m^2 in the continental US. The current GE K series satellites have a maximum SFD of -91 dBW/m^2.

Environmentally Protected TWTA's

Several vendors are now producing TWTA's that are designed to operate in an exposed outdoor environment. While these units are in the 50 - 100 Watt power range, this is sufficient to generate the required EIRP in most cases. This development allows the power generating source to be mounted very near the feed of the antenna eliminating the loss and expense of a long waveguide run.

New Exciter Configurations

New exciters are available that allow the upconverter to be located separately from the modulator and conditioning equipment. With the two devices interconnected at a lower frequency the units may be easily separated by distances greater that 500 feet without effecting system performance. This fact and the hub mounted TWTA allow this system greater flexibility in the locating of the antenna.

Improved LNB Technology

Great strides have been made in the improvement of LNB noise temperature. Less than five years ago a Ku-Band LNB with a noise temperature of 215 K was considered very good. Today devices with noise temperatures of 75 K or less are available at half of the price. This improvement can contribute up to 2.5 dB improvement in receive signal-to-noise ratio.

These four factors combined allow for some very dramatic changes in the configuration of Ku-Band uplink systems. These changes reduce the cost and maintain a high performance level.

System Configuration

Integrating the above equipment can result in a Ku-Band earth station system for less than $100,000. The system is shown in Figure 2. The system utilizes the following basic equipment:

3.7 M Antenna, 2 Port R/T Feed
75 W Hub Mounted TWTA
Upconverter, Hub Mounted
LNB, 90 K
Video Exciter
Video Receiver

With this configuration the system will have an EIRP of 70.5 dBW. Options exist for the motorization, anti-ice and feed configuration for the antenna. The system is fully compliant will FCC rules and includes the required Automatic Transmitter Identification System, ATIS.

System Performance

As noted above the basis system has an EIRP of 70.5 Dbw. This is sufficient for the system to operate on the GSTAR® and K series of Ku-Band satellites both in the full and half transponder modes over a majority of the country. When additional power is required, either a higher power amplifier or a larger antenna can be supplied. When operated over these satellites high quality audio and video performance can be expected.

At times satellite operators place, via ground command, attenuators or pads, in the satellite transponders. This is done to decrease the SFD. This decrease has two effects. The first is that the system requires additional uplink power. The positive result of this is that this increases the uplink carrier/noise ratio. Since in most systems this is the limiting factor in the performance an increase will have a net result of increasing the receive signal to noise ratio.

System Diagram

A simplified diagram of the system is shown in Figure 2. Please make note of the ability to separate the outdoor antenna mounted electronics from the indoor equipment by up to 700 ft. This greatly enhances the installation flexibility and reduces the system cost by using coaxial cable instead of waveguide to interconnect the equipment.

SYSTEM APPLICATIONS AND COST

Systems of this type and design are ideally suited for operation as part of a network of uplinks. The network, such as a state association of broadcasters, can contract with a satellite provider to get the most favorable pricing on satellite transponder time and ensure the proper satellite parameters. Other particular areas of interest include business television systems, news gathering organizations and government agencies involved in the interactive training of personnel. As mentioned previously the system can be provided for less than $100,000 in its basic configuration. Options are available that can both increase the performance and versatility. But as with most things the price will also increase.

This system, because of its simplicity, lends itself very well to be operated by groups or individuals who would not normally be capable of operating a conventional uplink system.

System Options

The system described in this paper is a very basic system. A number of options are available to enhance and expand the system.

Larger Antenna - A larger antenna will increase the uplink system EIRP and generally improve system performance. Options exist for 4.6 Metre, 5.6 Metre and 7.6 Metre antennas.

Antenna Motorization - While the system is intended primarily for fixed operation on a specified satellite, conditions may exist where motorization is required. Applications may include transmission to inclined orbit satellites are included.

Anti-Ice - When antennas are located in

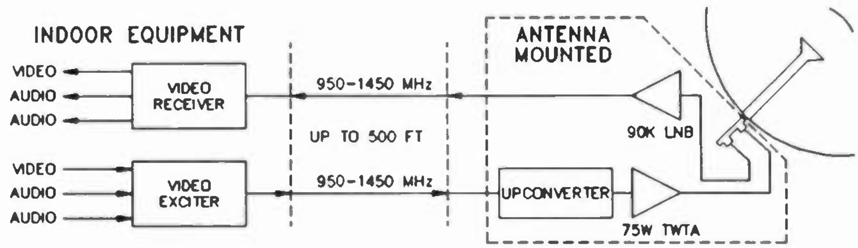

Figure 2
Typical Low Cost Video Uplink Configuration

climates conducive to icing, either full or half reflector anti-icing can be provided.

Multi-Port Feed - The basic system is supplied with a 2 port feed system. This allows the system to transmit on one port and receive on the other. If it is desired to increase the versatility of the system either 3 or 4 port feed can be supplied.

Performance Calculations

As with any complex communications system it is very difficult to predict the performance without the exact specification of all of the variables. Below is an example of how the system would perform if operated from Dallas, Texas over the GSTAR® I satellite utilizing the CONUS, Continental United States Beam.

Satellite Parameters

Satellite EIRP	4.2	dBW
Satellite G/T	1.3	dB/K
SFD	-94.4	dBW/m^2

Earth Station Parameters

Available EIRP	70.5	dBW
LNB Noise Temperature	90.0	K
Receive G/T	30.1	dB/K
Maximum Video Deviation	13.8	MHz p-p
Required EIRP	67.9	dBW
EIRP Margin	2.5	dB
Expected Video Signal/Noise	54.7	dB

Conclusion

A system such as described above can provide a low cost solution to organizations that need Ku-Band uplink capabilities but could not afford the cost of a conventional system. Although the system does have limitations, these are few and are easily off-set by the general ease of operation and affordable cost.

Acknowledgements

As with any project of this nature many people assisted in the formulation and verification of the concept. I would like to particularly thank Khalid Khan, formerly of Andrew Corporation, for the initial theoretical calculations. Ahmed Elkassih and Jim Purull for the actual over the satellite measurements and Tom McCutcheon for the system engineering of the project.

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ENHANCEMENTS IN VOICE COMMUNICATIONS FOR SATELLITE NEWS GATHERING

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This paper outlines the importance of the voice communications portion of a satellite news gathering (SNG) vehicle.The GTE Spacenet demand assigned multiple access (DAMA)-based voice communications is the system in use in a majority of the SNG vehicles operating today. There is a discussion of completed system enhancements that were driven by the unique SNG operating environment, such as the new design of the RF and IF distribution portion of the system, and the steps involved in developing the new equipment by using in-house design and engineering talent. Futher, planned improvements and the corresponding engineering problems inherent with the use of new microwave packaging techniques and digital signal processing techniques, for increased reliability and size reduction and expansion into international SNG are also discussed.

SATELLITE NEWS GATHERING AND VOICE COMMUNICATIONS

Satellite news gathering is a key recent development working in concert with innovations such as the telephone which is shrinking our perception of the size of the world. The telephone is as important to the broadcaster at a remote location as it is to the same person at his home station. Satellite news gathering is also an integral part of the new broadcasters pool of resources.

Recent developments have presented the broadcaster with the unique challenge of providing live broadcasts, not only in their immediate broadcasting area, but also from places very remote and out of line-of-sight microwave transmission distance. The viewing public has come to expect and rely on live broadcasts to show them the news as it is happening. The scale and location of a news event like the deployment of forces during operation Desert Shield greatly emphasized the need for simple telephone-like voice connection to places in the world outside the home news bureau. The voice communications capability is essential to a successful video news shot.

As the broadcasters region of responsibity increases, the reporter and film crews are taken further and further away from areas with reliable, public switched telephone network (PSTN) service. This is particularly evident in rural areas and many overseas locations. Other technologies such as radio telephone and cellular are limited by their need to be within a few miles of a transmission or repeater site. Presently cellular coverage is available in only a portion of the United States, usually around heavily populated areas that have sufficient PSTN access. The utility of providing voice communications using the wide transmission area of most satellite news gathering transponders is self-evident.

Since 1986, the GTE Spacenet demand assigned multiple access (DAMA)-based SNG voice system has been the predominant system in use by satellite news gathering organizations. This network, in place on four GTE satellites, provides the SNG community with reliable public switched telephone access to the world from any location in the United States. This system provides the user with multiple channels of single channel per carrier (SCPC) audio/voice communications using the same satellite that is used for the video feed. Referencing figure 1, the user in the field can communicate over the PSTN from anywhere in the continental United States through the satellite's voice hub.

The master network controller at the voice hub recognizes a call request from the system in the field, commonly called the site, and dynamically assigns a channel

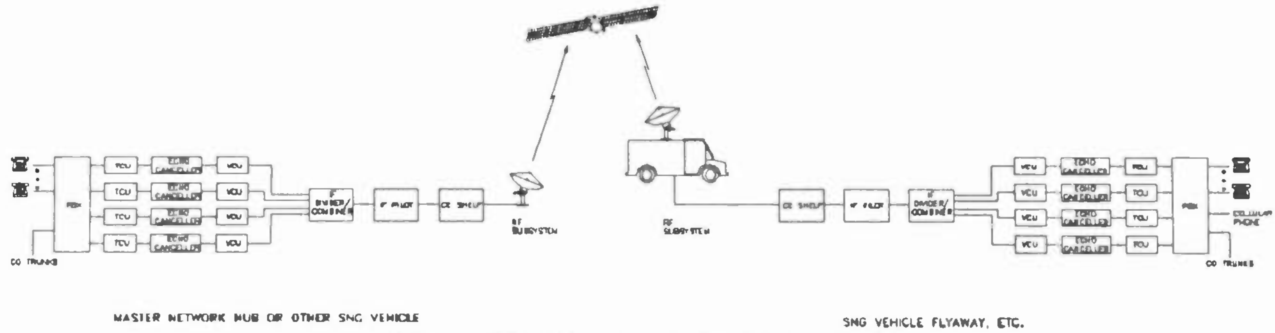


Fig. 1: SNG voice communications network

for the call. The site voice channel unit tunes to the transmit and receive frequencies assigned to that channel and the call is connected to the final destination through the hub private branch exchange. A home station can also contact the crew in the field to coordinate a feed by dialing the number for the respective satellite voice hub and entering the unique site access code. Full duplex voice communications can take place between the called and calling parties independent of the video and interrupt feedback (IFB) transmissions and without operator intervention.

An important feature of this type of system is the separation of the video and voice communications throughout the conversion process. It was found that a narrowband voice system operating at band edge with the video carrier generated significant intermodulation products. Also, the small amount of bandwidth available to voice communications limited the number of voice channels, usually to two, that each SNG operation could use. The answer was to separate the two systems giving the voice system its own baseband to Ku-band converters as well as its own low power solid state power amplifier (SSPA). This scheme has advantages in improving reliability and quality as well as availability since communications will not be lost in the event of a high power amplifier (HPA) failure. It also cuts down on wear and tear of the main HPA, since only the SSPA is powered during voice communications. Redundancy is provided in the voice system by supplying power amplification through the HPA in case of SSPA failure.

A system was found that provided the separation of the video and voice systems as well as call placement without operator intervention. The earliest systems, however, were installed with little or no modifications for operation in a mobile environment.

THE SATELLITE NEWS GATHERING ENVIRONMENT

There is very little equipment that is designed specifically for the transportable environment of satellite news gathering. Today, the broadcaster has to consider size, weight, and environmental requirements when purchasing broadcasting equipment for news gathering. Demands of the voice system on all three of these requirements must be a minimum since it is competing against necessarily large and heavy equipment such as tape machines and video monitors. Other factors affecting equipment operation are: temperature and humidity extremes, AC voltage fluctuations typical of generator operation, electromagnetic interference from the close proximity of RF equipment, and shock and vibration. With these conditions in mind, the redesign of the RF conversion portion of the voice system was launched with the goal of improving the reliability, cost, required, space, and performance under a broad range of temperature conditions.

UP/DOWN CONVERTER REDESIGN AND MANUFACTURING

A. Background

The News Express Communicator's^{sm} progenitor was developed to supply telephone connectivity to fixed, remote locations such as small, rural villages or oil rigs that had no conventional telephone service. These applications were mainly fixed stations of four or more channels and the system design naturally did not address the requirements of todays compact transportable or fly-away installations. Its design also predates the commercial emergence of efficient digital voice encoding techniques and the small inexpensive radio frequency terminals (RFTs) commonly used in modern very small aperture terminals (VSATs).

After the decision was made to move the system into a SNG application, some early SNG trucks used an equipment configuration that was essentially unchanged from the fixed version. This version had a seperate indoor and large outdoor unit (ODU). These first few systems demonstrated the value of the design and prompted a request for configuration improvements. Since then, most have installed a second generation that was commonly called the VCP-1. There were major changes made to the channel cards and conversion hardware especially in relation to the ODU.

An up/downconverter was designed to replace the large ODU in SNG applications.

The converter became the familiar 3.5 inch conversion electronics (CE) shelf now in use in the majority of SNG vehicles. The CE shelf was designed to work as the fixed frequency Ku to L band converter while the original pilot module remained intact as is. The pilot module acts as a L band to 70 Mhz converter and provides automatic frequency control (AFC), by locking onto a stable system pilot signal. Automatic frequency control is required because of the tight frequency tolerances of the FM demodulator. Rather than implement AFC on a per channel basis it was decided to place the AFC circuitry inside the up/down conversion process instead of in each demodulator. This reduced the cost and complexity of the individual voice channels and also allowed the manufacturer to spread the cost of the AFC circuitry over all of the voice channels installed at a particular site. For this reason, an off-the-shelf up/down converter could not be used.

GTE investigated the possibility of using a smaller, more compact version of the CE shelf. However, after an extensive search and the production of four prototypes, GTE undertook in-house redesigning and manufacturing of the component to achieve the desired performance specifications as well as an overall size and cost reduction.

The resulting system is compared to the original in Figure 2.

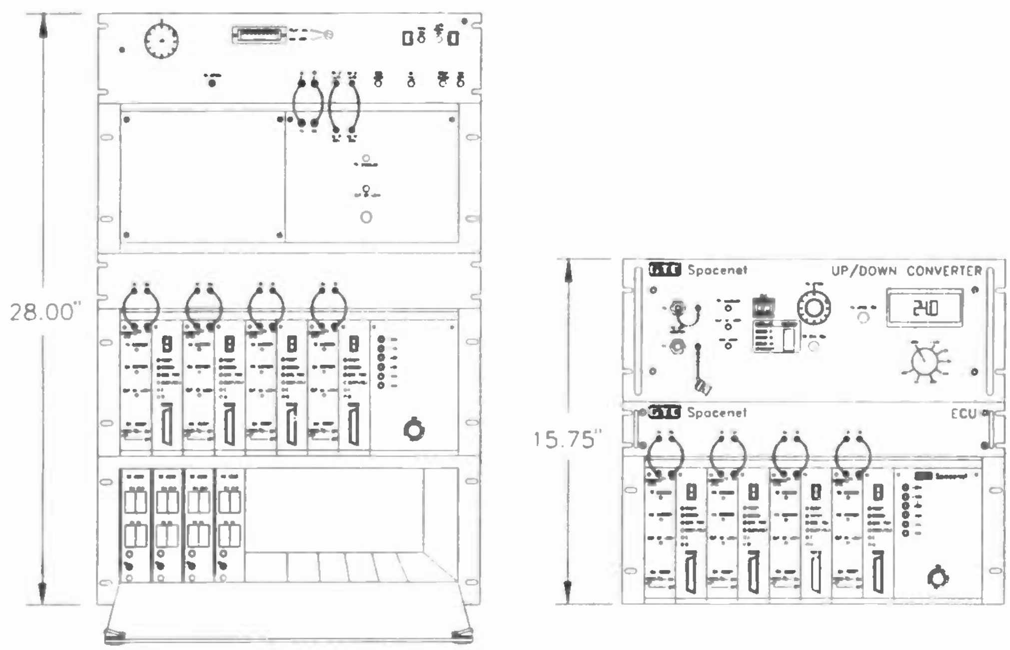


Fig. 2: Original and redesigned systems

B. Redesign and Manufacturing

The scope of the work was such that a plan was laid out in 5 stages; initial design, prototype, testing, redesign, and manufacturing. The time frame for this process encompassed an 8 month period, an unusually small amount of time for this type of effort.

The design tasks involved taking the separate CE and IF pilot chassis and repackaging them into what was later called the up/down converter (UDC). The main printed circuit board in the IF Pilot Unit was simply rehoused with other RF, IF and electronic components into the UDC. Doing this repackaging design in-house provided the opportunity to incorporate SNG specific requirements into the new equipment. For example, faster warm-up and user tunable (voltage controlled) crystal oscillators to allow for multiple satellite use.. Experience gained from talking to broadcasters in the field prompted the inclusion of such things as an improved cooling system for both the UDC components and the voice channel cards themselves. The high heat, tight space environment of the SNG vehicle created new demands on the equipment. The design of both the UDC and the echo canceller/combiner unit (ECU) were done keeping the high stress SNG environment in mind. Incorporating a small, 23.5 CFM fan along with vents in the ECU chassis produced excellent results by bringing the temperature of the voice channel cards down 30° C. This minor change significantly decreased the risk of heat related failures in the voice packages.

Further development of the main power supply of the voice system provides protection from the primary AC power source. Special sensing circuits added to the power supply provide overvoltage, undervoltage, and overcurrent protection. Integrated circuits, such as the main voice channel electrically erasable PROMs, are very susceptible to these power abnormalities and can be seriously damaged by them.

A working prototype was built from which standard chassis fabrication and assembly drawings were created along with a bill of materials. Preliminary test data was taken and some resulting redesign effort yielded the final product approved for manufacturing.

C. Testing

Testing was performed throughout the development stages. Every main component and cable assembly was pre-tested either by the manufacturer of that component or GTE to ensure it met the stated specification. A test plan was developed from which the assembled UDC's and ECU's could be tested before delivery to the customer. The testing was divided into the following stages:

a. Physical check. The overall inspection of the device under test (DUT) to confirm proper assembly. Wire connections are checked for proper solder connections and routing and the RF connections are checked for tightness and routing.

b. DC voltage testing. Before power is applied to the main boards and components, a check is conducted on the DC power connections to confirm correct operational values.

c. RF/block and level testing and alignment. The DUT is fully powered up with all external signals applied and a series of tests including transmit and receive conversion gain, 1 dB compression point, frequency response, and return loss. The DUT is aligned and adjusted to a set of SNG operational specifications.

d. Environmental testing. Units are placed into an environmental chamber and are cooled and heated to a point below and above the specified operating temperatures. Proper operation is verified at each point.

e. On-air testing. This final step tests the DUT on each of the SNG satellites. Final alignment is completed and test calls are placed. A UDC and ECU are integrated with the other system components and shipped to the user.

DIGITAL AND INTERNATIONAL SYSTEMS

All the widely used systems in use today for voice communications in the SNG market are analog and frequency modulated based. Analog technologies have the advantage of being well-proven and readily available. There are a number of applications within SNG where an analog based system is preferred while in others, a digitally based system may be required. As the technology progresses more and more, systems may incorporate digital technologies in whole or in part. Digital components and voice encoding techniques are becoming more efficient and in many cases surpass the quality of competitive analog methods.

A. Digital Voice Systems

During the past ten years there have been many improvements in digital voice coding techniques, and these have reduced the required transmission bit rates while at the same time retaining or actually improving the perceived voice quality.

Where pulse code modulation (PCM) or Deltamodulation required 64 kbps to 32 kbps for toll quality voice circuits, many of the voice coding schemes made possible by new digital signal processors (DSP) require as little as 14 kbps for toll quality circuits while orderwire grade circuits require less than 7 kbps. DSP based designs allow for considerable cost and space savings while at the same time increase the overall system performance. Many of the functional blocks that were implemented separately such as echo cancellation, DTMF tone detection, and audio AGC can now be implemented directly by the same DSP used to process the voice signal for speech compression. DSP based designs have the added benefit of being able to change the entire design by simply changing the software program; no expensive printed circuit board redesigns are necessary. All these advantages aside, how does a digital voice system compare with analog voice systems for the task at hand; efficient voice communications by satellite for the SNG vehicle?

Figure 3 shows the audio signal to noise (S/N) versus the required end-to-end carrier to noise density (C/No) for a typical News Express Communicator FM SCPC voice channel and a 14 kbps digital SCPC voice channel using quadrature phase shift keying (QPSK) and rate 1/2 forward error correction (FEC).

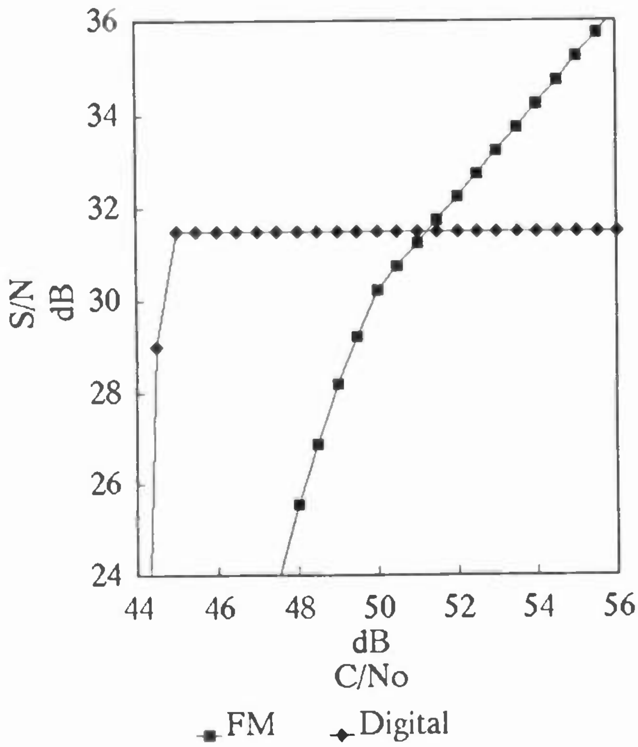


Fig. 3: Analog and digital audio S/N vs. C/No

The FM SCPC channel's S/N varies at a one for one ratio up to the threshold of the FM demodulator at approximately 50 dB C/No and then falls off rapidly. The digital channel's S/N remains constant as long as the bit error rate (BER) is less than 1×10^{-3} and drops to zero as the BER approaches 1×10^{-2}. On the figure this occurs at a C/No of roughly 45 dB. Even after allowing for the more gradual S/N falloff of the FM channel there is still nearly 3 dB of power savings when using the digital channel. Because most of the SNG vehicle's voice traffic is from the vehicle to our hub, GTE Spacenet has tested several four channel multichannel per carrier (MCPC) digital voice multiplexers. MCPC digital voice has several advantages over FM SCPC for both the satellite operator and the SNG vehicle operator if the network uses a star topology. The RF power and bandwidth required to transmit the same four voice channels is lower so the satellite operator is able to fit more voice channels on the satellite. See figure 4. Since the SNG vehicle operator generally requires two to four voice circuits the required HPA size is much smaller, less than 1 watt versus the 5 watts required for the FM SCPC News Express Communicator system. This translates directly into a significant cost savings. There is also a space savings, but the overall HPA assembly size is mostly determined by the filter/diplexer and waveguide switching which are included in the HPA package.

In summary, there are many good reasons for SNG voice communications systems to move towards digital voice transmission systems:

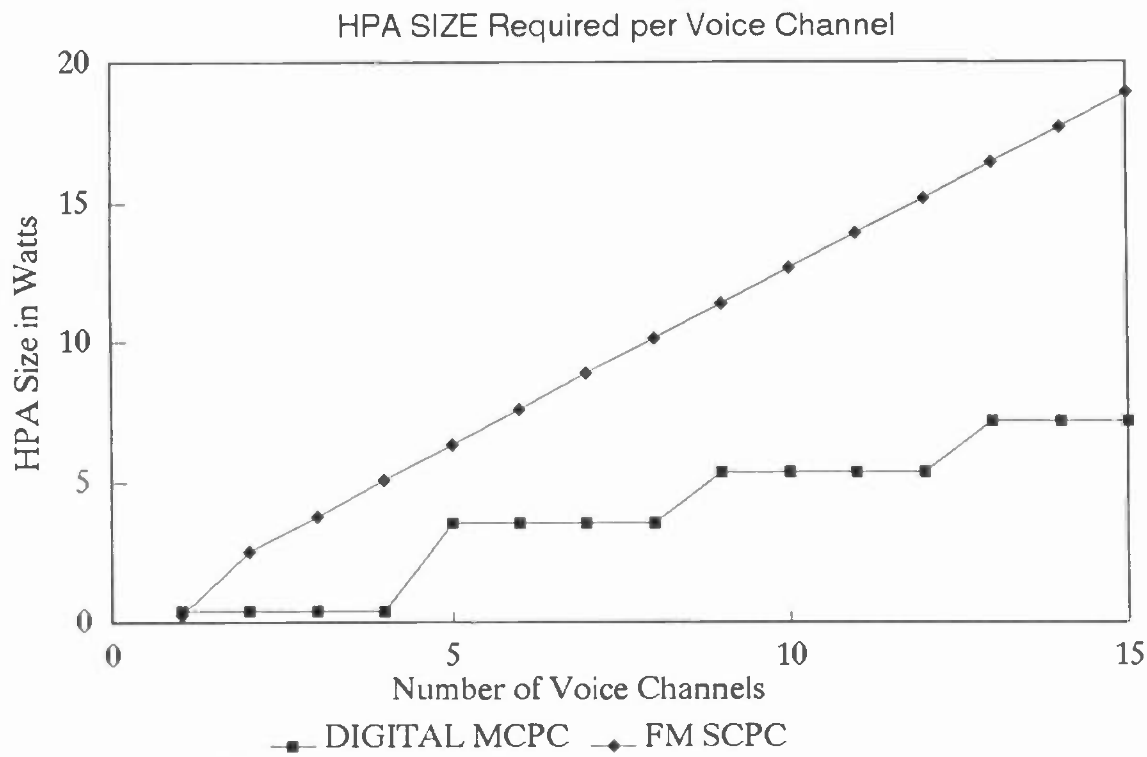
a. Multiplexing and switching is much easier to do digitally. Multiplexing even as few as 24 voice channels using analog frequency division multiplexing (FDM) often required as much as a rack of equipment and this equipment was difficult to align and maintain. Digital audio and data can be combined onto one carrier using the same mutltiplexer, thus reducing required SSPA power and intermodulation products.

b. Digital components are not subject to drift or parts tolerance problems. Digital systems are therefore much less costly for the manufacturer to produce, and simpler for the field technician to align and maintain. Designers are incorporating more built in test and self diagnostic features into new products.

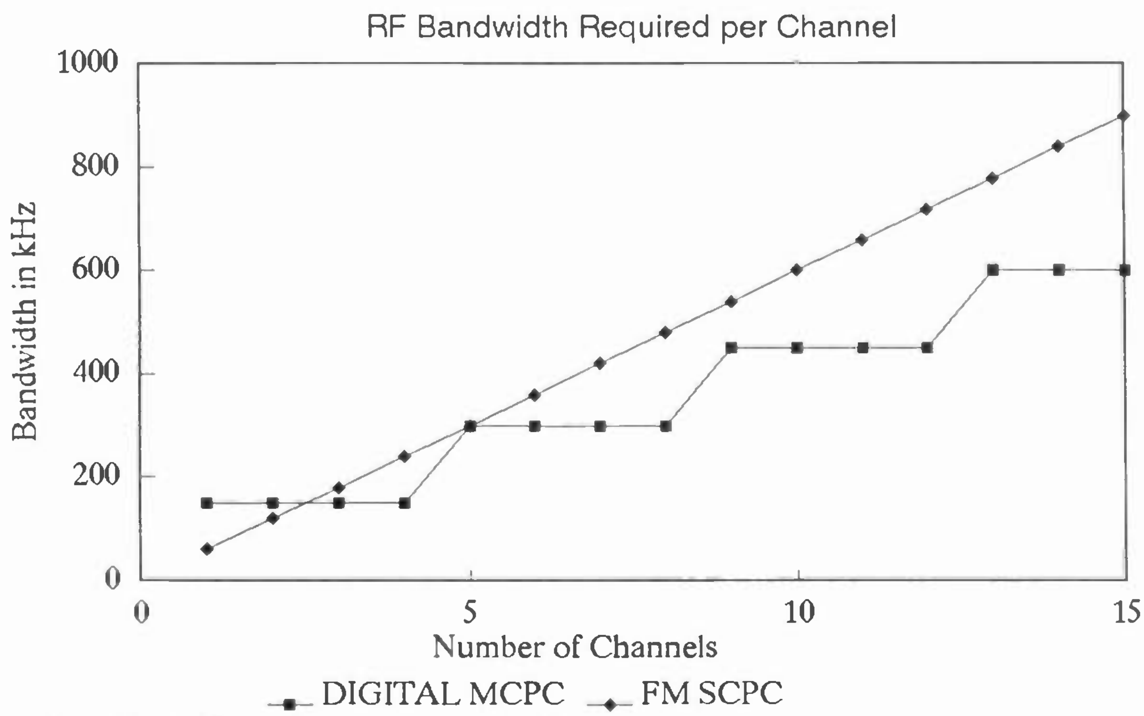
c. Inexpensive high performance DSP microprocessors have made possible significant reductions in the required bit transmission rate.

d. Encryption. Voice security can be provided through encryption.

e. Forward error correction. Digital based systems have the added benefit of offering error correction, which helps compensate for Ku band weather fades.



Multicarrier HPA backoff = 7.5 dB



Digital MCPC carrier spacing = 75 kHz
FM SCPC carrier spacing = 30 kHz

Fig. 4: Analog versus digital voice system comparison

f. Configurable data rates. Bandwidth can be divided based on the number of voice channels needed. This allows the user to tailor his system for different requirements.

B. International Voice System

The use of SNG is undergoing strong expansion overseas especially in Europe. A voice system designed for use for international SNG must be able to work with many different satellite configurations. These include regional beam use on INTELSAT transponders, various Ku receive and transmit bands, dissimilar telephone trunk interfaces, and varying AC power voltage levels and connector configurations.

A voice system for use over international satellites, as well as foreign domestic satellites, is being developed by GTE Spacenet. The prototype system as depicted in figure 5 will be functionally similar to the News Express Communicator system but will be able to operate over any satellite. GTE Spacenet examined the possibility of using the existing News Express Communicator for international service in order to save domestic users as much additional investment as possible. However, the existing News Express Communicator cannot be used for the international SNG system without extensive modification. The DAMA software requires that both the master and remote sites receive their own transmitted signal from the satellite, and because of the International satellite beam configuration, this simply is not possible. Another deciding factor was the fact that in order to use the News Express Communicator, new up/down converters would have to be built for each different satellite frequency band or one to cover all the possible bands.

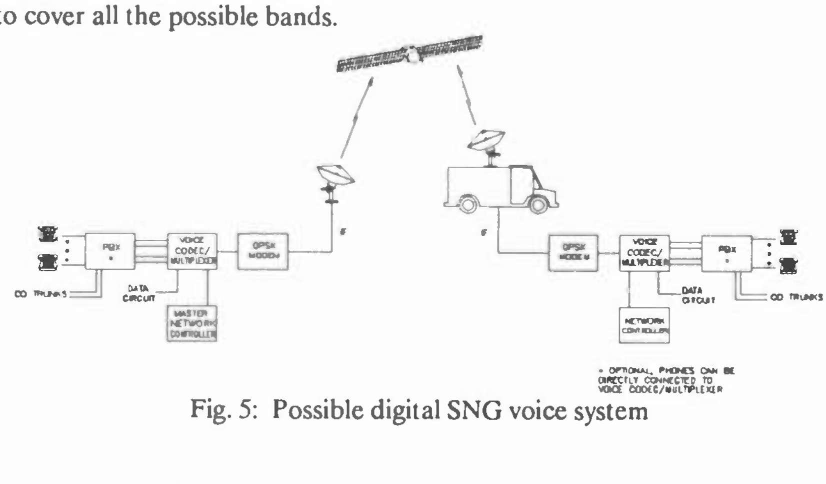


Fig. 5: Possible digital SNG voice system

Given these problems weighed against the clear advantages of digital voice, GTE is investigating a digital voice MCPC system. Because the data modems typically have AFC built in, a specially designed up/down converter will not be required. This will allow the system to use the inexpensive RFTs developed for the VSAT marketplace. The system is also designed so that neither the master nor the remote site have to see their own transmitted signal from the satellite. This will allow the broadcaster to operate the system over virtually any satellite band/beam configuration.

Conclusion

The importance of a reliable and supportable voice communications system for SNG applications is essential. The general lack of equipment specifically designed for SNG requirements will at times create the need for SNG service providers to react to this shortfall by developing their own equipment. Innovations such as digitally coded voice and microwave integrated circuitry allow a designer to produce a product ideally suited for the unique SNG environment.

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TELEVISION AUTOMATION

Monday, April 15, 1991

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COMPUTER TECHNOLOGY APPLIED TO OFF-SITE REMOTE TRANSMITTER OPERATION*
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THE USE OF ETHERNET FOR BROADCAST FACILITY CONTROL
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DESIGN FOR REAL-TIME NETWORK AUTOMATION AT NHK
Michael R. Fuqua and Suresh K. Gursahaney
IBM Corporation
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NHK
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ADVANCES IN CART MACHINE CONTROL SYSTEMS*
Raymond Baldock
Odetics Broadcast
Wayne, New Jersey

SMART CARTS: IMPROVING STATION EFFICIENCY WITHOUT BREAKING THE BANK
William F. Carpenter
Ampex Corporation
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A CRITICAL LOOK AT CAMERA ROBOTICS*
David Philips
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ROBOTICS . . . THE CAPITOL HILL PROJECT
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AN INNOVATIVE INTELLIGENT REMOTE CONTROL SYSTEM*
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*Paper not available at the time of publication.

THE USE OF ETHERNET FOR BROADCAST FACILITY CONTROL

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Abstract - Local Area Networks are extremely common to the computer industry, and exist in nearly every place of business. As the broadcast industry begins to expand the role of the computer in the station, there is a natural desire to utilize the local area network as the medium for control. The basic concepts of the use of Ethernet for this purpose are discussed.

INTRODUCTION

Ethernet is the most popular local area network installed today. It is used by all sizes and types of computer systems, from the smallest of personal computers to the largest of mainframes. This popularity, along with the large number of readily available products and expertise, makes it an ideal network for mission critical applications.

Certainly the concept of interconnection of control signals over a network is not new. ESBus, multidrop, and other such methods do compromise a network. However, the physical cabling and connection methods offer unparalleled flexibility for the initial installation and later modification.

There are a number of real-time, reliability, and maintenance considerations for any network and Ethernet is no different. The technology has evolved, however, to the point where all engineers must consider this medium for transport of control signals as well as other data.

The in-depth technical details of Ethernet are a matter for study and there is a considerable amount of information available on the subject. What is most needed is some practical applications for this technology and an understanding of the capabilities and limitations in the broadcast environment.

AN OVERVIEW OF THE BASICS OF ETHERNET

Ethernet - A Few Facts

Ethernet was developed as a joint project between XEROX, Digital Equipment Corporation and Intel, based on work done by XEROX in the mid-1970's. A great deal of the final product was arrived at through a series of trials and tribulations, which are well beyond our scope here.

The original designers formed the results of innumerable experiments into a specification which could be used as a basis for technology development and manufacture of network-based devices.

Ethernet operates at a theoretical bandwidth of 10 Million bits per second (Mbps). This theoretical limit is rarely reached when there is more than a few users of the network. The network uses a method known as Carrier Sense Multiple Access with Collision Detection (CSMA/CD). This method is essentially analogous to a room full of people each allowed to speak one at a time. As each finishes another may begin. If two people speak at once, they both stop, wait a random period of time, and then begin again. This is the mechanism used to arbitrate access to the network. As you can see, there is no single owner of the network.

Ethernet is the network itself, and is not connected with the type of use. The name Ethernet is often mistakenly interchanged with the protocol it carries, such as DECNet, NOVELL, TCP/IP and others. Again, it does not matter to a telephone what language the subscriber may use. As one can see from the analogies drawn here, Ethernet can often provide networking services to several computer systems or other devices who may not necessarily share the same protocol, eliminating the need for costly duplicate networks in a facility.

Coax - A Familiar Cable

Most of us are concerned with the physical aspects of an Ethernet installation. This is fairly easy to understand since Ethernet operates over a 50 ohm terminated cable. This cable may take the form of a larger, less flexible cable, or the more popular and inexpensive RG-58C/U cable, commonly known as ThinWire, ThinNet, or CheaperNet. Each medium has a set of benefits and detriments, but the methods of wiring and attaching connectors is familiar to most maintenance staffs in the broadcast arena.

More recently, products are available which allow Ethernet networks to operate over unshielded twisted-pair wiring and fiber optics as well as 18 Ghz. interoffice wireless LANs. There has been a rush of products into the twisted-pair (sometimes referred to as 10-BASE-T) wiring marketplace, since virtually every installation in the world is already wired!

The physical cable may be configured to form networks of great lengths, depending on the type of cable selected. The use of satellite, microwave, fiber, and other technologies has extended the capability of Ethernet almost indefinitely.

Ethernet is a bus structured network, which means the network must be one logical piece of cable from end to end. As such, loops and branches are not allowed. There are some slight modifications for 10-BASE-T, but the basic principles still apply.

There are a large number of devices manufactured by several well-known vendors which provide the ability to concentrate connections to the network, isolate traffic to certain portions of the network, and to extend the physical dimensions of the network as a whole. These devices are field proven and provide the network designer with an incredible array of tools to allow an Ethernet-based system to fit nearly any physical constraint. A simple diagram of an extended Ethernet is shown in Figure 1.

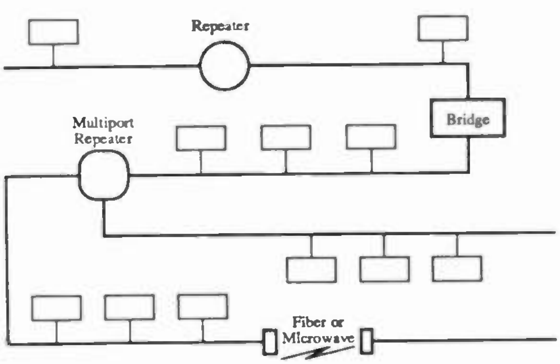


Figure 1
Extended Ethernet

CONSIDERATIONS FOR THE USE OF ETHERNET FOR CONTROL

When considering any network for use in a mission critical control environment, such as a broadcast facility, there are a number of factors which must be considered. Among these are:

- Reliability
- Serviceability
- Flexibility and Extensibility
- Multivendor support
- Accessibility

With proper construction techniques and attention to detail, Ethernet networks are extremely reliable. In general, the failure of a single component will not affect the operation of the rest of the network. The maturity of the technology is also a contributing factor to the reliability of an Ethernet network.

Each segment of the Ethernet is considered a passive element, and operates without any power or signals other than those provided by the transceivers themselves. The failure of a repeater which may be used to connect two separate segments of the network will still allow each segment to operate independently.

Failures on the network may often be traced by isolating the downed segment and inspecting devices on the network until the problem is found. Ethernet transceivers, which provide the interface to the physical cable, may be replaced without interrupting the network itself.

There are many good monitoring and diagnostic tools available for Ethernet networks, again a sign of its maturity as a networking scheme. Most Ethernet problems are well within the scope of the maintenance staff of a broadcast facility.

The flexibility of Ethernet allows simple access directly to the devices for which control is desired. The various cabling schemes allow Ethernet to be distributed on an as-needed basis, without the need to change the facility layout to suit the network.

Ethernet is currently supported as a network option by virtually every computer manufacturer. While in some instances the protocols vary, multivendor interoperability is a reality without resorting to slower standards such as RS-232. Through the use of standardized protocols, dissimilar systems may share data and resources with very little trouble, and again, on the same physical cable.

Accessibility is an important criteria when control signal timing is critical. This is the case when frame accurate pulses are desired for machine control, or when return tallies must be acted upon in a timely manner. CSMA networks are inherently unpredictable, in that the amount of time to completely and correctly send a message on the network is a function of the number of users and the amount of data to be transferred in a given interval of time. This requires the protocols to be used over the network to minimize the effect of other traffic. For the timing required in the broadcast market, the advantages available through software and hardware are able to provide satisfactory service.

As we have discovered, Ethernet meets the fundamental requirements for a network in a control environment. The technology is proven and relatively inexpensive, while meeting all of the needs of the broadcast market.

USING THE ETHERNET FOR CONTROL

Since most broadcast devices do not interface directly to the Ethernet network, some form of gateway is required. Certainly most every computer can provide this function, but a careful choice must be made of its memory and processor speed to provide the necessary response time.

In many cases, it is not practical to use standard computer products due to their relatively high costs and limitations of the operating system. Few operating systems are tailored for use in a real time control environment.

Further, much of the broadcast equipment in use today is operated by a simple series of closures and hardware tallies, which were not originally designed be interfaced to the high speed network.

An Approach to the Problem

To provide these functions, Utah Scientific developed specialized hardware to act as the gateway between the device to be controlled and the Ethernet. Known collectively as the Ethernet Machine Control System, the unit consists of a dedicated 68000-series processor, an Ethernet port which may be configured for several different types of cabling, and plug-in boards which provide interfaces to relay, serial, and parallel devices. A block diagram of the EMC product is shown in Figure 2.

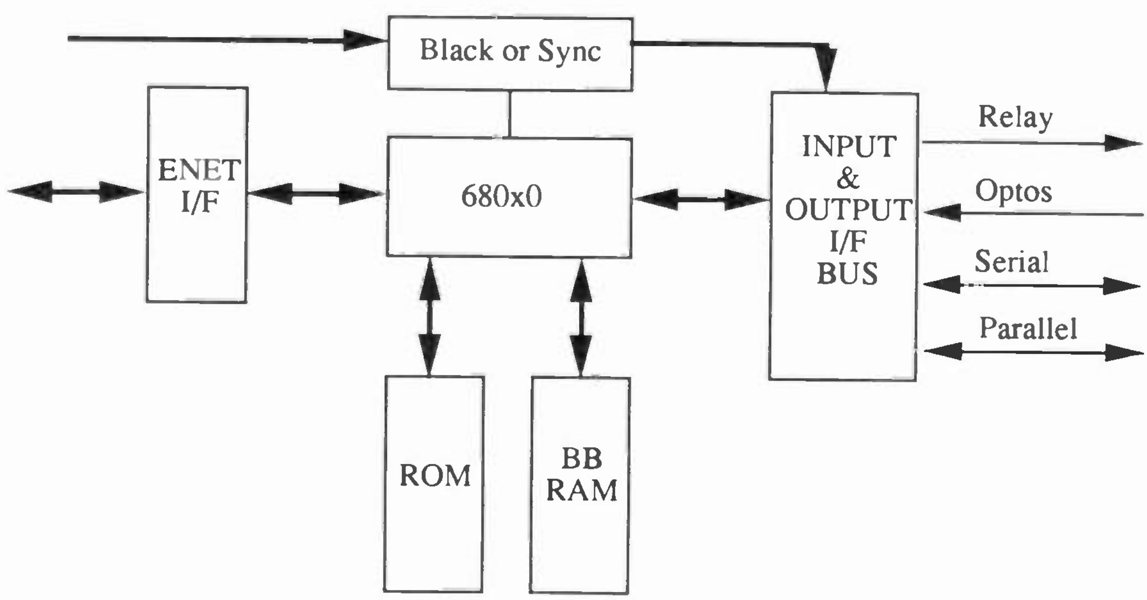
The EMC products are fully programmable, which allow the EMC to host not only applications for standard VTR's, carts, and other devices, but to host custom designed applications for any number of specific circumstances.

The EMC products employ a standard network protocol known as Transmission Control Protocol/Internet Protocol (TCP/IP). The use of a standard protocol is important so it may pass transparently through standard repeaters and bridges, and may be detected by standard Ethernet traffic monitoring devices. The use of a standard protocol will also reduce the development time when it is necessary to integrate with other computer systems, since no specialized networking software is required.

The operating system used by the EMC has been optimized for real-time response. Ethernet traffic handling is given the highest priority so delay within the EMC is minimized. Further, when control paths are required over the network, the EMCs involved will establish a network connection in advance of the requirement for messaging. This again reduces the overall time required to pass a message since a path is already available. Supervision of the connection allows quick identification of a failure, and is also a method to guarantee the reception of the message by the destination EMC.

For specific critical applications, the EMC contains a sync

Figure 2
Ethernet Machine Control
Block Diagram



or black input, so commands may be queued to execute during a vertical interval. This external reference augments the synchronization information available over the Ethernet and is not subject to variable access delay. It is a fact there are certain time-critical applications where Ethernet is not suitable. Under extremely tight timing constraints or when Ethernet traffic is unknown in type or quantity, it is not the best choice. However, for most normal applications in the broadcast arena, Ethernet can serve with a careful choice of Ethernet interface devices which have been optimized to provide the best response time possible.

Through the use of the parallel, relay, and serial boards, we have an effective gateway to the Ethernet. We can use the boards as either input or output devices, so we can easily build control panels for remote VTRs, status displays for studio tallies, and master control to machine connections. Effectively, we can connect to the Ethernet those devices which were never designed to be controlled in such a manner.

A Typical Application

A typical application for this type of control system is found in the Master Control environment. For example, the Utah Scientific MC-502 master control is a popular choice for station control. Through the development of an interface to the EMC system, this panel can now control all of the remote devices over a single cable. Figure 3 diagrams a typical machine control system over Ethernet.

When power is first applied to the system, the EMC which is dedicated to the master control establishes a connection to each remote device. This connection provides immediate status on the panel of the controlled device.

When any command is issued from the Master Control, the message is passed to its EMC, which passes it to the receiving EMC, who translates it into the programmed command, whether it be serial, parallel, or relay. Any return status is handled in the reverse manner.

Since one EMC may handle many machines depending on the computer and network requirements, EMC-based systems may grow to very large sizes.

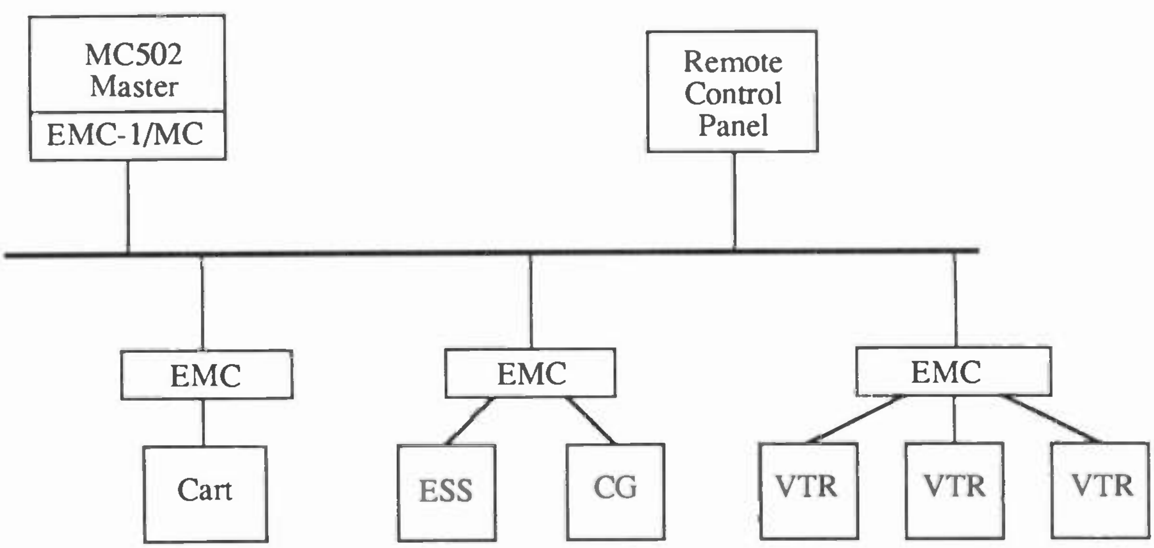
Large Multi-vendor Systems

There are innumerable examples of large, multi-vendor computer installations sharing a single Ethernet network. One must look somewhat closer, however, to find such examples in the broadcast industry.

Utah Scientific is currently working on just such an example. The diagram shown in Figure 4 provides some idea of the scope of the system. The system contains proprietary switching hardware, personal computers, Digital Equipment VAX computers, and EMC computers operating on the same Ethernet. The Ethernet supports machine control, master control automation, and interface to the host system as well several tallies. There are NOVELL, LAT and TCP/IP protocols all sharing the same network.

While it is not appropriate to discuss the details of the system here, it is easy to envision the use of Ethernet and multivendor interoperability in the broadcast control market.

Figure 3
Simple Machine Control System



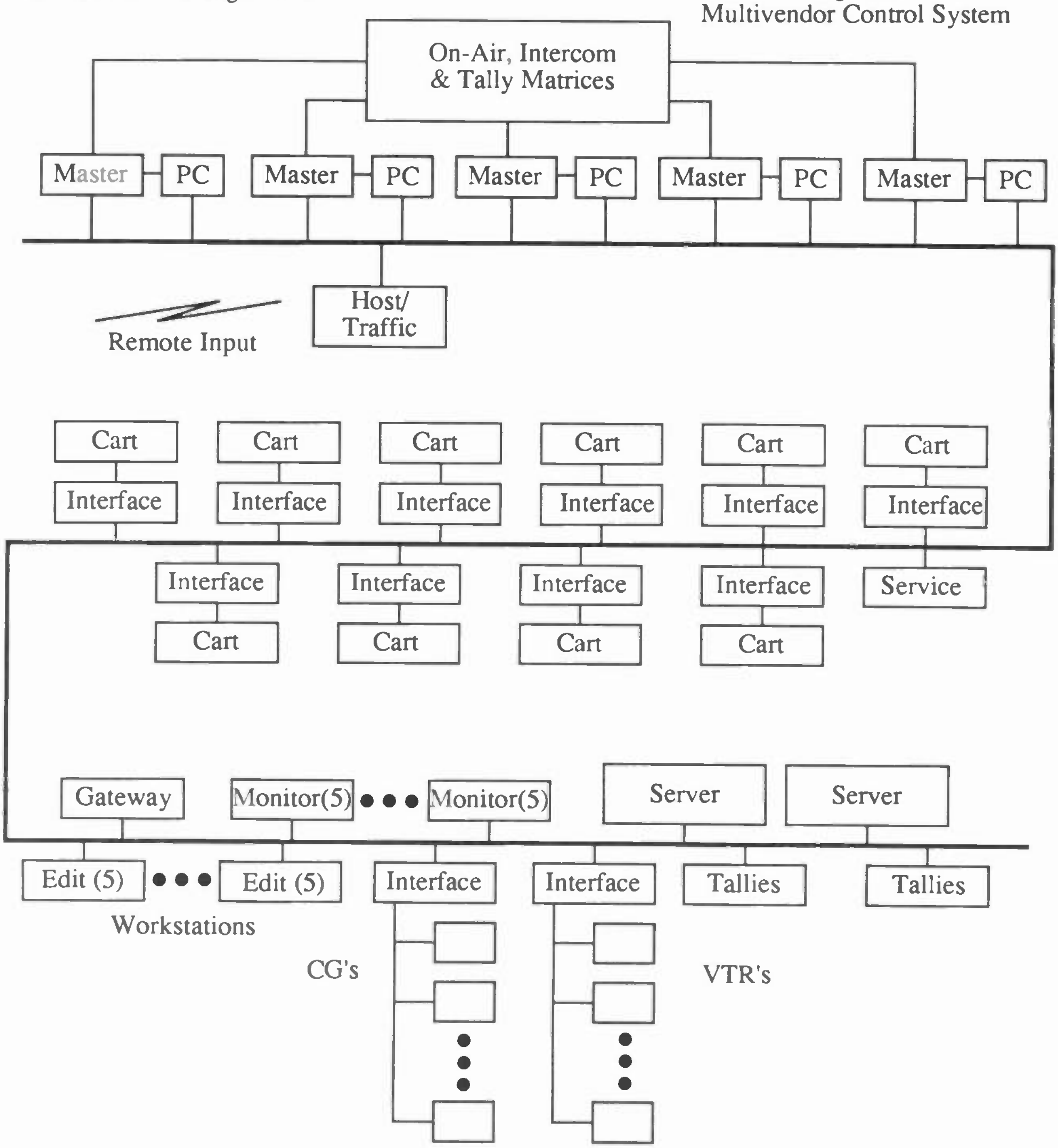
SUMMARY

We've shown that Ethernet can be an effective medium for providing control signals for the broadcast market. With only a small amount of imagination, we can see a total integration between our computer systems and our broadcast machines.

Over time, more and more devices will be directly connected and controlled over a network, a trend which has already been identified in the latest generation of multicassette and switching devices.

When selecting an Ethernet-based system, the use of standard protocols, programmable controllers, and configurable interfaces are likely to prove out to be the best choices to begin the migration to a more powerful control system. The flexibility of the network will allow a gradual transition with a minimum of difficulty.

Figure 4
Multivendor Control System



DESIGN FOR REAL-TIME NETWORK AUTOMATION AT NHK

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Abstract -Advances in microcomputer technology have paved the way for the introduction of real-time control systems in environments typically outfitted with large, complex systems. The implementation of a network automation system at Nippon Hoso Kyokai (NHK) in Japan originated the use of microcomputers and local area networks to automate an environment previously supported by redundant mainframe systems. The inauguration of such technology supports the microcomputer's ability to meet functional, performance, and availability conditions required for complete network automation.

INTRODUCTION

The development of a real-time network automation system at Nippon Hoso Kyokai (NHK) was performed by NHK in collaboration with International Business Machines (IBM) Corporation. NHK, the largest broadcasting company in Japan, is a public broadcasting servicer that supports three regions in Japan via four television channels and three radio channels.

A staunch supporter of new technology, NHK has had a history of continually investing in the latest computer and broadcasting equipment in order to provide the highest quality of service to their customers. NHK inaugurated use of the first network automation system in the late 1960's when they installed two, redundant IBM 1800 Data Acquisition and Control System (DACS) units to control and monitor their program scheduling. In 1987, NHK began a joint effort with IBM to develop a low-cost network automation system that would not only control and monitor program scheduling but also handle all aspects of on-air operation and the program production operation.

NHK enlisted the aid of IBM's Systems Integration Division (formerly Federal Systems Division), the group that had joint development responsibility for the original 1800 DACS implementation in the 1960's with IBM-Japan and NHK. In the new implementation, IBM-SID was tasked with architecting the system and developing the software for the real-time control components of the system. Development of the broadcasting device specific software and associated hardware communication to the target devices, such as Video Tape Recorders (VTRs) and Audio Tape Recorders (ATRs), was contracted to IBM Japan, NEC and Toshiba. The copyright and property of the system expressed hereafter belongs to NHK.

SYSTEM REQUIREMENTS

NHK's primary objective was to replace and enhance their current network automation system with a more reliable, low-cost alternative. Of critical importance was to develop a comprehensive fault-tolerant system, whereby the system would be self correcting and would automatically activate backup devices in cases of hardware failure. Moreover, the same architecture had to support both the on-air operations and the program production operation.

To meet their objective, NHK outlined a number of system requirements that were key to maintaining its high quality operation. Many of these requirements took into consideration both the viewing customer as well as the operating personnel. The ability for the entire system to maintain an on-air look was one such requirement. Simply put, the goal was for the entire system to have a Mean-Time Between Failure (MTBF) of over five years. This requirement implied that individual components within the system could fail, but backup measures had to automatically bypass the component in order to guarantee that no channel went off the air. In such cases, the system had to recover within 1 second of actual component failure.

Of utmost importance to NHK was the requirement for the system to be designed for future growth. NHK's plans for future expansion included supporting additional channels and increasing program schedule activity. The architecture clearly had to support increased workload and additional hardware. Moreover, the system had to accommodate new advances in technology as they became available.

ARCHITECTURE CONCEPT

The system requirements outlined by NHK and the new advances in computer technology eventually originated a solution that included Local Area Networks (LAN) and microcomputers. It was determined that complete separation of the channels would ensure the highest system reliability. Thus, independent channels were isolated onto their own LANs with each LAN bridged to a common LAN that controlled devices common to all channels. The isolation of program production control and on-air control was also of critical importance to reduce risk to the on-air system. Figure 1 depicts the final architecture.

Dual LANs were added to further improve the system reliability in case of a LAN failure. Redundant messages would be transmitted on each LAN to guarantee reception. The receiving devices would accept both messages, compare them for accuracy, and then select one for processing. Missing messages would be logged and tracked to identify any spurious LAN errors.

A hot-standby concept was employed to provide redundancy for the microcomputers. The primary microcomputer called the "Sender" would process commands and transmit messages that would control the broadcasting devices. Secondary microcomputers called "Listeners" would perform the same processing of commands as the Sender but save the messages for comparison with messages issued by the Sender. The concept allowed the Sender and Listener to remain in constant lock step until the Sender failed at which time the Listener would automatically takeover.

Messages to the broadcasting devices would be processed in real-time, thus placing the burden of precise execution of the program schedule on the Sender and Listener. The broadcasting device controllers would act as protocol converters, converting messages transmitted by the Sender in real-time to commands recognizable by the broadcasting devices. Millisecond accuracy was required to ensure that screen "black outs" would not be discernible to the human eye. Moreover, to eliminate any chance of a single point of failure, redundant controllers were utilized with algorithms designed to select which controller would actually transmit the commands to the broadcasting devices.

To guarantee complete synchronization between the microcomputers, an interface to an external clock source was required. The external clock would be to synchronized with the national time and employ its own redundant fail-safe to ensure accurate timing.

The final component of the architecture involved an external scheduler that was to operate on a mainframe or minicomputer. The external scheduler would accept program schedules from a user, select available independent and common devices, and periodically transmit the detailed schedule to the appropriate Sender and Listener for the channel. In case of program schedule changes, the external scheduler would transmit program schedule updates immediately to the Sender and Listener. Offloading such activity to a mainframe or minicomputer would allow for more efficient operation of the Senders and Listeners.

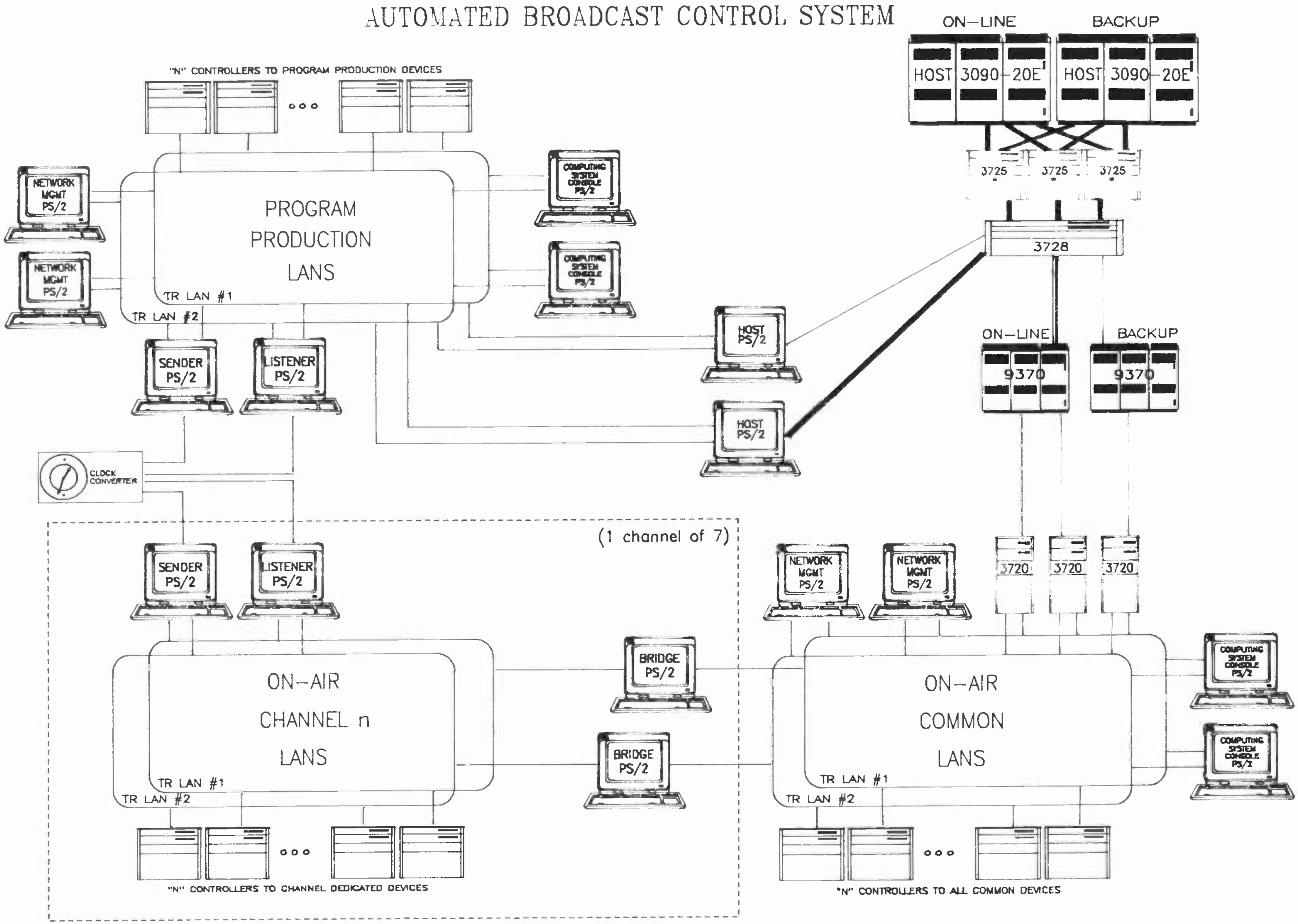

Figure 1. Automated Network Control System

DESIGN

During the design phase, the system was separated into two functional areas, Program Production and On-Air Control. Selections were then made on the physical elements (hardware and software) to meet the requirements of each area and all functions were separated and assigned to a physical element. Figure 2 shows the final breakdown of the system.

Separation of the primary functions of Program Production and On-Air Control provided physical separation of the NHK automated broadcasting schedule into two distinct schedules. The On-Air Control function would provide automated broadcasting of scheduled events for the seven on-air NHK channels, while the Program Production function would handle automated control of scheduled events for programs in production.

Although the functions of On-Air Control and Program Production were distinct, they contained similar composite functions. Each was divided into two lower level functions, Event Management Services and Local Management Services.

The "Event Manager" provided all the functions of the 1800 DACS. The Event Manager would generate instructions that control assigned device controllers using information from host systems. These instructions or commands would be generated in real-time within the timing constraints specified by the host systems. The Event Manager would also handle asynchronous data from the device controller, such as device responses, application and error data, and asynchronous (real-time) commands from the host systems. The Event Management Services functions would operate on the Senders and Listeners.

Backup capability for the critical Event Manager was provided by the Listener. The Sender would be the operational Event Manager; the Listener would monitor the actions of the Sender without actually sending any messages or commands to the device controllers. The Listener would maintain an exact copy of the Sender tables. If the Listener detected a failure in the Sender, the Listener would "take over" as the new Sender. Multiple Listeners could be configured for each Sender to allow for several levels of redundancy.

The Local Management Services functions would incorporate those miscellaneous functions outside the scope of the Event Manager Services. These services included network management, operator status display, operator system control, and any other services that required direct access to the Event Managers. Local Management Services would operate on operator consoles called Computing System Consoles.

Local Management Services would monitor the Event Managers (Senders/Listeners) and obtain status information for display to an operator. Status and error information from the controllers would be sent to the Event Manager as part of normal manager-controller interaction. It could also selectively monitor LAN data traffic. As the Event Manager processed information, it would forward data to the Computing System Console for consolidation and display. In the event of a component failure, the Event Manager would continue to operate without interruption.

Monitoring of the entire system would be isolated to Computing System Console operating on the common LAN. The Computing System Console would monitor each channel simultaneously displaying general status of the channel and error conditions as they would occur. Controlled startup and shutdown of individual devices could be performed as well as forced takeovers of Senders. Multiple Computing System Consoles could be installed to allow separation of the monitoring activity by operator personnel.

The design of such a system necessitated the use of high performance equipment with state-of-the-art software. However, growth and expandability constraints required the hardware to be readily available and part of a strategic hardware platform. Standard hardware and software available in the marketplace was appropriate in most cases. However, the precise synchronization of the microcomputers required additional hardware adapters.

To achieve the millisecond accuracy required for the day-to-day operation of the system, the software programs needed to operate in real memory. Thus, microcomputers needed to be outfitted with sufficient memory to allow both the operating

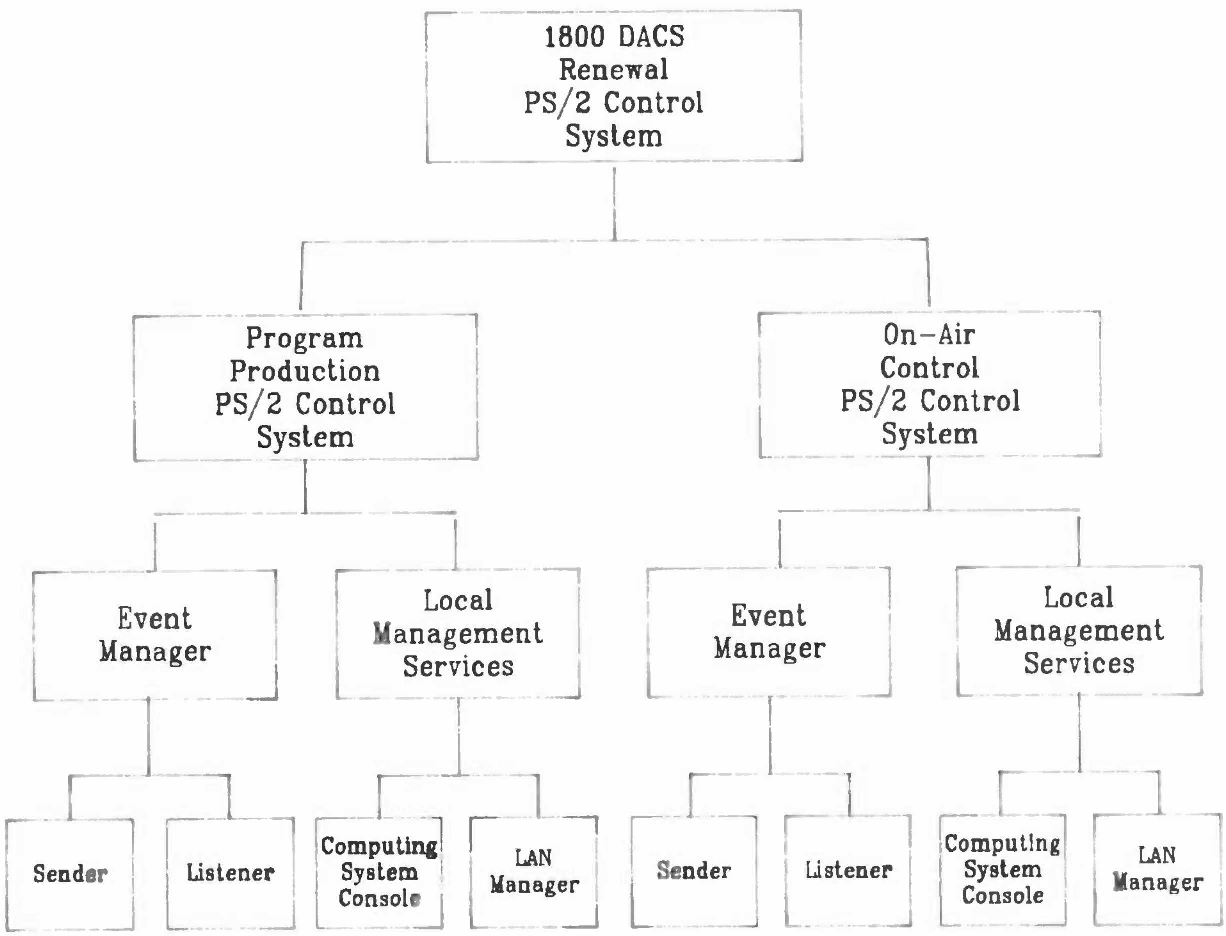

Figure 2. Function Tree

system and scheduling software to be memory resident at all times.

The simultaneous, independent activities to be performed by the Sender and Listener obviated the need for a multi-tasking environment that could support simultaneous execution and processing of instructions. Special timeslicing and prioritization of tasks would be employed to guarantee the operation of critical tasks.

Synchronization of Sender and Listeners would require the use of an external clock that constantly synchronized the hardware clocks on the microcomputers. Moreover, the microsecond clocking accuracy required to support millisecond response by the system necessitated the use of a special clock devices for the microcomputers that utilized independent CPUs.

Communication between the Sender and Listener was of critical importance to ensure proper operation of the "hot-standby" concept. Contrary to popular thinking, it was determined that the Sender rather than the Listener would be the primary decision-maker in case of conflicts. Thus, in cases of a Sender or Listener failure, arbitration between the Sender and Listener would be performed with the Sender making the final determination of the system health status. If the Sender determined that one or more of its components had failed, it would contact the Listener and instruct it to takeover. In case of communication failure between the Sender and Listener, the Listener would automatically takeover after a predetermined timeout.

SYSTEM IMPLEMENTATION

The design of such a highly reliable system obviated the need for high-performance hardware and software.

IBM PS/2 Model 80-071's were selected as the microcomputer of choice. Equipped with 10-12 Megabytes of real memory, IBM 4 Megabits per second Token-Ring adapters for LAN communications, and IBM Real-Time Co-processors for independent clocking, the microcomputers met all the hardware specifications. IBM's Operating System/2 Extended Edition was selected as the operating system because of its timeslicing and multi-tasking flexibility. Sender/Listener code was developed using IBM C/2.

Implemention of the Sender/Listener code was finally completed with 60,000 lines of C code. IBM OS/2 Presentation Manager was used for the user interface because of its standard window interface and user-friendliness. Development of the system consisted of a system architecture phase, high and low level design phase, synchronization study phase, system modelling phase, coding phase, and extended integration and system test phases. The methodology used in the development process is depicted in Figure 3. The entire development was completed in two years.

The system architecture phase of the project was performed in 4 months. Total system requirements were examined to determine the appropriate environment to be used. Analysis of the previous 1800 DACS implementation was performed to understand the limitations of the system and holes in the architecture. In the end, a number of architectures were examined before the microcomputer, LAN-based solution was selected.

The design phase of the development was accomplished in 8 months encompassing such activities as producing the high-level design, performing a synchronization study, modelling the system, and producing the low-level design. In reality, the high-level design was constantly updated throughout the design process. Results from the synchronization study and system modelling phases were used to refine the design to meet the system requirements. The synchronization study attempted to verify that the operating algorithms and system recovery algorithms proposed would perform in a real-time environment. System modelling verified that the millisecond accuracy required could be achieved using the available technology.

Because of the complexities of code development for a multitasking environment, actual implementation of the Sender and Listener code was achieved in 6 months. At the time, IBM's OS/2 Extended Edition 1.1 was in Beta Test and operating system bugs were common. Implementation of the Sender/Listener code and Computing System Console user interface code was performed by 10 developers.

The final phase of the development, integration and system testing, was completed in 6 months. Only limited unit testing of the system components could be performed during the coding phase due to the complex interrelationships of the 13 tasks involved. Thus, the burden of the testing was handled during integration test. System test of the final system verified that all the system requirements were met and performance of the system met all the design requirements.

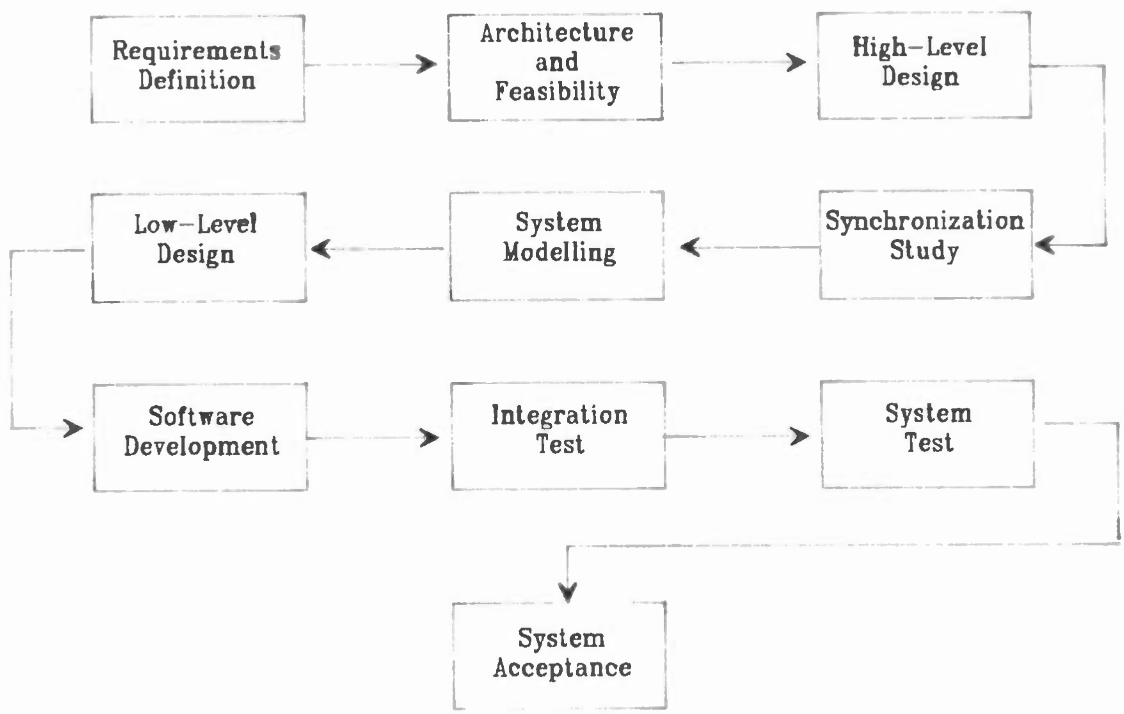

Figure 3. Development Methodology

NHK SYSTEM STATUS

In 1990, NHK implemented the network automation system for their Program Production operations. NHK is currently preparing the software to be incorporated in their On-Air operations this year. The implementation of the On-Air software will also introduce the concept of real time updates.

WHAT'S NEXT?

The use of the new technology available today has already shown performance improvements in NHK's implementation of the system. Moreover, an analysis of recent advancements, such as the Intel i486 processor and IBM's 16 Megabit-per-second Token-Ring adapter, has shown that additional significant performance improvements can be made. The introduction of SCSI high-performance drives and low memory OS/2 offerings could also eliminate the need for microcomputers equipped with large amounts of memory.

Flexibility of the architecture and the standardization of computer technology will allow the real-time network control system to be extended to different environments. With the availability of time-critical Reduced Instruction Set Computers (RISC), the system could be implemented to perform at even faster transaction rates. Utilization of the common Ethernet technology will allow the system to interface with other network systems such as back-office and billing operations. Migration to the UNIX or AIX operating system will permit standardization of the software to operate on a number of different computer systems.

SUMMARY

The evolution of computer technology offers the broadcast industry new opportunities for automated networks. The NHK implementation is just an example of how automation has evolved from the mainframe to the microcomputer. With vision and the proper design methodology, microcomputers could be the solution to low-cost network automation in the future.

REFERENCES

- *Operating System/2 is a registered trademark of IBM Corporation and Microsoft Corporation.*
- *OS/2 is a registered trademark of IBM Corporation and Microsoft Corporation.*
- *Presentation Manager is a registered trademark of IBM Corporation.*
- *AIX is a registered trademark of IBM Corporation.*
- *UNIX is a registered trademark of American Telephone and Telegraph.*
- *PS/2 is a registered trademark of IBM Corporation.*
- *i486 is a registered trademark of Intel Corporation.*

SMART CARTS: IMPROVING STATION EFFICIENCY WITHOUT BREAKING THE BANK

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Failing banks have been front-page news for the past year or so, but in the broadcast business we have been living with tighter and tighter financial resources long before any S & Ls went under. In today's increasingly competitive broadcast market it is imperative that chief engineers and operation managers use their time most efficiently in order to reduce overhead.

One of my basic premises is that efficient human involvement is the most important asset to a station's operations. This is because both manpower and resources are at an extreme premium today; this is certainly the situation at any station I'm familiar with. I contend that the new generation of cart systems is a central solution to this problem, and that the move toward increased station automation, especially in the area of automated cassette systems, is a key factor to improving profitability.

Depending upon their implementation, some cart playback systems can even serve as intelligent controllers for existing station systems. This significantly broadens the role of cart systems while adding valuable new automation capabilities without necessitating the purchase of expensive new systems. To select an automated system most effectively, it is critical to understand the key technical challenges that the automated cart playback system faces, and how the system's design will affect both performance and overhead costs.

I identify these key challenges in terms of: storage flexibility, conflict resolution, program flexibility, and quality of playback.

STORAGE FLEXIBILITY

Storing events on tape has contributed to labor-intensive activity throughout the short history of equipment that automates the presentation of commercials to air. The most primitive method was simply stacking a bunch of tapes on top of flatbed machines and threading them while a second machine played its spot to air. Later products actually impeded the efficiency of the station, since the tape decks were slanted to reduce overall size and then there was nowhere to stack the tapes.

The ACR-25 and TCR-100 were the first machines that allowed you to store a minimum number of events on-line and contributed to genuine labor-savings. The next step was the Betacart, which increased the cassette storage capacity to 40. And in some smaller markets the cassette sequencer, which requires significant manual intervention, contributed to labor savings and continues to do so in those areas where they are used to this day.

All of these systems involve off-line storage of a large portion of stations' libraries. The process of transporting cassettes from a separate library and then loading them into the on-air presentation system is one of the prime overhead costs that an automated system is intended to reduce.

The enormity of these overhead costs can be calculated when you take into consideration that in today's network affiliate station, a typical library might contain between 1,500 and 2,500 individual elements counting commercials, IDs, promos, opens, closes, bumpers and public service announcements. Since a copy of each element is normally kept in an off-line library, the space for such a library becomes a major consideration. At the same time, the investment in cassettes, ranging from $15 to $50 depending upon the format, represents a major investment in media (Figure 1). And if proper organization and storage of that many cassettes is difficult for the affiliate, consider an independent TV station. The count of video elements needed by an independent can realistically number 50% to 100% more than that of the affiliate broadcaster.

Clearly, on-line spot storage density is one of the primary benefits of the modern cart playback system. To maximize storage capacity, manufacturers are increasingly designing multi-event systems — systems that store multiple spots of varying lengths and even longer program segments on a single cassette. Multiple event systems offer important advantages, allowing random access to a vast on-line library (typically more than 10,000 30-second spots). This capacity is in dramatic contrast to that of the machines cited above, which are single event per cassette.

Because more than one spot can reside on a single cassette, critics of multi-event storage are quick to point out the purported Achilles heel of multi-event systems — playlist conflicts, an issue that I will address in detail later in this paper. They champion the simplicity and ease-of-access of the single-event system. But the limitations of the single-event design, which lead to highly inefficient use of the cassette size, runaway media costs and compounded by labor in handling, are severe.

	Analog Single	Digital Single	Digital Multi
Library Size	2,000	2,000	2,000
Cassettes required	2,000	2,000	50
Cassette playtime	5 min	6 min	32 min
Cost per cassette*	$15	$15	$50
Elements per cassette	1	1	42
Total library cost	$30,000	$30,000	$2,500
Cost per element	$15.00	$15.00	$1.19

* Average market prices.

Figure 1. Single vs. Multiple Event Cost Comparsion

In terms of media cost alone, the arithmetic is impressive. Based on D-2 cassettes of 32-minute lengths (as can be specified for the ACR-225 system) a maximum of forty-two 30-second spots can be stored on one cassette (with each segment requiring an overhead of 15 seconds). If a cassette costs $50.00, the price of storage for each of the 42 segments is $1.19, a dramatic reduction in cost from the single cut per cassette approach.

Extrapolate these figures to apply to savings in library space. For all those who have had to wrestle with the costs of commercial real estate, storing one cassette vs. 42 is a very appealing proposition.

In terms of the labor required to move cassettes around, with multi-event machine we have seen as much as a seven to one reduction in labor needs. A specific example is WHDH in Boston that replaced TCR-100s with Ampex ACR-225s. The TCR-100s required no less than three operators per 24-hour day (one per eight-hour shift), seven days a week (168 man hours). The two ACR-225s require just one operator assigned five hours a day, five days a week (25 man hours). This 85% reduction in labor enables a single operator to do the job of dubbing new elements into the system, and deleting obsolete spots and promos, while still leaving time open for other tasks. Typical dubbing requirements are between 15 and 50 elements per day for this type of network affiliate in the highly competitive sixth market.

CONFLICT RESOLUTION

All automated cart systems have some unavoidable time limitations or conflicts. Two conditions form the basis of most irreconcilable playlist conflicts. First, on multiple event per cassette systems, two or more events are on the same cassette, but reside physically at different locations. This makes it impossible to play them back-to-back as specified by the playlist. Second, short duration events are below the system's minimum cycle time for back-to-back operation. The minimum duration factor varies from manufacturer to manufacturer and is predetermined by a transport's acceleration, threading and cueing speed and the system's robotic speed.

Solutions used by some systems to reduce potential conflicts include making duplicate copies of events on other cassettes and/or instituting a strict organizational structure to the on-line library. Both of these approaches effectively increase labor and media costs and are therefore relatively counterproductive.

Ampex takes a different approach to the problem with a software utility resident in the ACR-225 called AutoResolve™. AutoResolve software implements automatic detection and correction of playlist conflicts without operator intervention or adjustment to the playlist.

Prior to going on-air, the program compares the selected playlist with the content of the on-line library. During the process, it automatically constructs a schedule and thereby identifies any spots that cannot be executed as specified. Next, it pre-cues the events and automatically uses an operator-designated "work" cassette to create a buffer copy of the spots presenting conflicts (Figure 2). This is done during the time when the system is "standing by" while local or network programming is "on-air." The net result is the station regains unrestricted access to any portion of the on-line library at any time.

The AutoResolve feature can also recognize and resolve a more sophisticated conflict where the creation of a single-

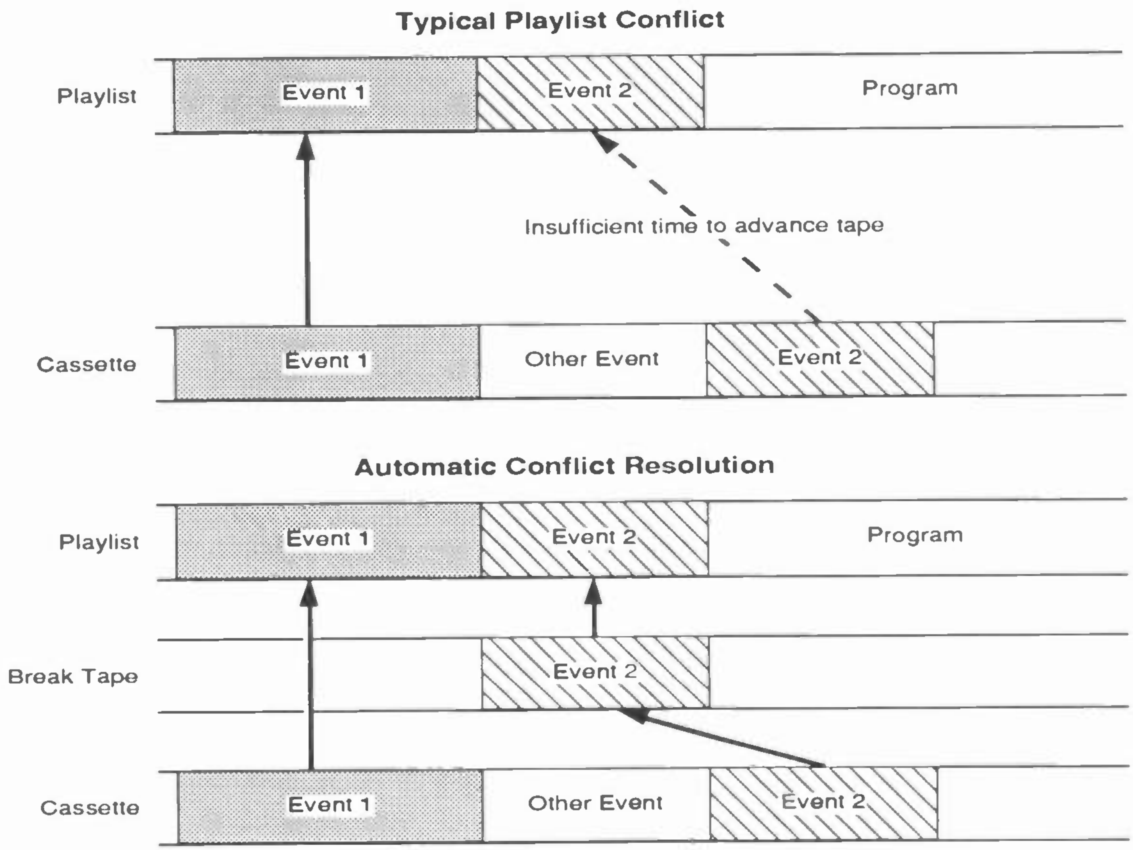

Figure 2: Conflict Resolution

element buffer tape is insufficient. In some instances, the system will elect to assemble edit a series of events representing a portion or all of a station break. This situation comes into play when neither enough time nor tape transports are available to create and play discrete "dubs." With this "assemble edit" feature, the ACR-225 is capable of assembling break tapes so they roll as one piece. A good automated cassette system should be able to perform both the simple and complex dubbing functions.

The practicality of creating entire break sequences on the buffer tape is also determined by the operation at a particular station. If last-minute changes to playlists are common, then the approach to making longer sequences containing complete station breaks is possibly wasted effort.

Different approaches to conflict resolution may be used in different situations. Sometimes, it's practical to check an entire day's playlist for conflicts. Others may find it more suitable to resolve a portion of the playlist, such as the length of time of one operator's shift. For a typical day's playlist used in most stations, only a few minutes of standby time will probably be needed to create all buffer copies necessary to resolve the conflicts.

The Ampex ACR-225 is able to provide its conflict resolution capabilities due to the unique software design of its scheduling and validation routines. Also, integral to the conflict resolution system is its unique transport design, which delivers exceptionally fast cueing.

PROGRAM FLEXIBILITY

It is very important that an automated cassette system have a good traffic system interface, since this is another factor that will minimize labor requirements. The driving source of a program automation system is the station log or playlist, typically prepared by the traffic department. The log for a given period of time may be introduced to the automation unit in one of several ways. It can be provided as a printed list, which an operator must enter into the computer (for the second time). Although this approach can be inexpensive, it is time consuming and various errors may be introduced, both in the traffic department and in the entry procedure. More efficient methods — and ones permitted by the Ampex automated cassette system — include importing a file into the system by means of a floppy disk or using a direct local area network, such as a thin net version of the Ethernet concept, linking the traffic department to the automation computer.

Either of these methods means information is entered into a computer only once. Using a network approach offers advantages over the disk, because bi-directional communication is available between the traffic and the cassette systems. This allows access to the system database by the traffic department for review of element availability.

Once the playlist is actually received by the automated cassette system, and begins to be acted upon, an acute and common need for program flexibility becomes apparent. Because of the very nature of station program material — news, fires, airplane crashes, 911, and late arriving commercials — accommodating last-minute changes in the playlist is essential. All broadcasters from time to time need to change PSAs and promos to commercials. If the order of commercial presentation has to be changed — even within seconds of airtime — the playlist must be altered without disrupting the system's on-air presentation. Once the change has been made, the system must verify its capability to satisfy the new schedule.

Last minute changes in a playlist represent a potential hazard under any condition. In the Ampex software, a utility called "Edit Active," along with AutoResolve conflict resolution software, provides the operator with a means to manually modify the previously scheduled events.

In normal operation, a cursor moves through the menu on the system monitor, highlighting each event as it occurs. Edit Active introduces a second cursor, which can be scrolled through the available elements to select and initiate any single event edits. This action activates the conflict resolution software and immediate operator feedback is given if the machine cannot fulfill the desired changes. It will also activate the automatic buffer copy dubbing previously mentioned, if this is required.

Another hallmark of program flexibility is a cassette system's ability to run program material from an external VTR. This ability to control multiple external machines from the various tape formats available today permits users the flexibility to expand the automation capabilities by using the cassette system to play spots or short segments, and using external VTRs to play long program material.

This demonstrates how an automated cassette system can become the central control point for an entire day's programming driven by the system's playlist. And you are making significant advances in automation while utilizing — and amortizing — existing broadcast systems.

QUALITY OF PLAYBACK

All who have been working with tape media for decades won't need to be convinced that the playback quality of commercials is a paramount concern. Quality is dependent on the format used, the number of recorded generations and wear on the playback cassette. Obviously, I represent a company that has a large investment in the D-2 composite digital format — a format we invented — and contend that the best possible automated cassette system will be a digital system. With the digital format, you can be assured that you will maintain the same quality level of all the video sources in your station.

In the digital domain, you will get a subjectively better picture but, even more importantly, when you begin to encounter wear and drop-outs you can do a better job with digital. You can predict when you're going to have a problem with digital, and possibly have no disturbances to air. In terms of tape usage, with digital you get better quality tape playback over a greater number of passes. We've had stations tell us that they will see "excessive degradation" after 800 to 1200 passes in the analog world. After 7000 passes in digital, they saw no signs of degradation.

If elements have to be copied to buffer cassettes to resolve time conflicts within an analog system, the quality loss of video dubs may be noticeable during playback. Our AutoResolve capability only makes sense because you can be reassured that the quality of the copy you're making is as good as the original, a true "clone" — which you can be assured of with a digital copy.

The digital format also delivers easy handling of four digital audio tracks, which is important if your station has any involvement in stereo or Secondary Audio Programming.

With analog, people have been making do with error concealment methods, which can only cover drop-outs with approximations of the lost material. The digital format provides powerful error correction, which yields a perfect mathematical replacement of missing information. Particularly with automated cassette systems, we are finding that customers need little persuasion that the future of tape formats is digital.

CONCLUSIONS

In closing, I'd like to make a few comments about how I see automated cassette systems developing over the short term. We have seen most manufacturers embrace some type of multi-event future software program to meet the economy values stated earlier in this paper.

I would also like to refer to the title of this paper, which promises improved station efficiency without breaking the bank. Granted, automating a station's commercial operation is not inexpensive, but times are such that the investment of capital dollars can be a wiser investment than the ongoing cost of labor in many applications. And, I hope that I've made the point that by buying the right cart system, you can deploy your current resources even more efficiently, with better control of your traffic system, and better use of existing people and equipment.

ROBOTICS...THE CAPITOL HILL PROJECT

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Abstract-During 1989, the CBS Television Network was interested in exploring various possible locations for the Network's first robotic camera installation. This location would serve as the Network's test-bed for future robotic installations.

The Washington News Bureau became the selected location. To satisfy time critical operational requirements and the potential economic advantages, a remotely-controlled robotic system was installed at CBS's Senate office booth in the Capitol. From this location, a correspondent could broadcast almost instantaneously without having to wait for an ENG camera crew.

While this project was underway, additional applications for camera robotics were identified. A seven camera system was installed in the Bureau's two studios and newsroom.

This paper describes the operational objectives, local constraints, and the engineering implementation of the Capitol project. Subsequently, due to the international political situation related to the Gulf crisis, a system identical to the Capitol installation was also installed at the Pentagon in the CBS defense correspondent's office.

INTRODUCTION

In 1987, CBS re-examined the status of studio robotics development. At the outset it became clear that practical systems at an acceptable price would have to come from broadcast-oriented manufacturers. However, it was not until a year later that the technology offered all the features CBS required of a robotic camera system.

In determining the first location for a robotic installation, a number of technical, operational, and financial considerations were taken into account. CBS wanted to find a location that would afford the best opportunity to evaluate current technology by utilizing robotics within a variety of program formats. Such location was found at the Washington News Bureau.

The Washington News Bureau provides daily inserts into CBS THIS MORNING and accounts for up to 75% of the programming for THE CBS EVENING NEWS with DAN RATHER. NIGHTWATCH, a late night magazine-type news program, is recorded and composited at the Bureau and FACE THE NATION with LESLEY STAHL originates live every Sunday.

In addition, throughout the city at various government buildings, the Bureau maintains correspondent offices for daily news coverage, and it was decided to use these locations for the first application of camera robotics.

In the past, ENG camera crews would be deployed on an as-needed basis to each of these locations to permit the correspondents to file stories, either live or taped. During busy periods when the reports from these locations were frequent and sometimes simultaneous, the availability of crews became a problem. Freelance crews would be hired, if available, or a staff crew would be sent to the location directly from another assignment. In the case of late breaking stories, any delays caused by staffing constraints usually reduced any editorial competitive edge.

The installation of a remotely controlled pan/tilt camera in one of these locations would provide a solution both operationally and economically desirable. Operationally, if control of all vital camera functions could be done back at the Bureau, then the dispatchment of an ENG crew to that particular location would no longer be necessary. Control of the system could be handled by staff already on duty within the central technical area.

Two specific areas offered an opportunity to introduce robotics into news operations. The first one was one of the off premise correspondent offices and the second Studio One, the larger of the two studios in the Bureau.

THE CAPITOL HILL PROJECT

CBS News selected the first location for such an installation at the Senate Gallery CBS correspondent office in the Capitol. By eliminating the need for a crew at this location, news manpower could be reassigned to other locations.

The operational objectives which the system would have to meet were supplied to CBS Engineering who then designed and implemented the system to meet them.

Based on the operational objectives, Engineering decided that the system should be capable of controlling the camera itself. In other words, in addition to the expected tilt, pan, zoom and focus functions, an operator at the Bureau had to be able to control other camera functions such as iris and black level.

The size of the office imposed its own requirements and constraints. The office is so small that camera crews would typically shoot from the hallway through the door in order to gain the desired camera angle. For an unmanned operation, it was out of the question for the camera to be mounted outside of the office. In order to experiment with different camera positions and to determine the best solution to this problem, a full scale mock-up of the office was constructed in New York. After several ideas were explored, it was decided to mount the camera on a bracket in a position that would not interfere with the staff's ability to work; yet afford a good field of view for a camera shot (Fig. 1).

To assure reliable, maintenance-free operation, CBS chose a CCD camera. Size constraints narrowed the field of possible candidates even further.

Following a laboratory evalution, the camera finally selected was the Sony DXC-750. The main advantage of this camera was its extremely small head, measuring only 3"x3"x4", and weighing just 1.3 lbs. The camera employs the same CCD sensors as a standard ENG-type BVP-7 camera and thus produces broadcast quality pictures.

Among the several manufacturers that were considered for the robotic system, EPO, represented in the USA by AF Associates, was chosen. Their equipment was found to be best suited to CBS's needs. Especially attractive was the compact size of the pan/tilt head and associated electronics.

The size of the remote control panel was also a factor. Space within the Central Control area at the Bureau, where the panel was to be located, was also at a premium. Since the system would not be used continuously, the control panel was installed in a collapsible roll around cart, which could be stored under the central operating console while not in use.

Figure 1. The Senate Office Installation

The EPO control system can communicate with up to eight cameras located at various locations. In addition to the four standard control parameters (tilt, pan, zoom, and focus), the EPO system allows for up to eight additional remotely controlled functions--four analog and four digital. For our application these functions were assigned in the following manner.

The analog functions included remote control of manual iris, master black level, and independent level control of two microphones. The digital functions were remote control of technical power (including lighting and delayed turn-on of auto black balance), automatic/manual iris selection, auto white balance on/off and the selection of color bars and tone. Since the system allowed for only four controls, black balance was included with technical power. When the system is turned on, a separate circuit activates the black balance. After a few seconds delay, the camera goes into its normal operating mode.

Another important consideration in the system design was to provide automatic dialing. Modems (9600 baud) were installed at both ends of the system to allow remote control over telephone lines between the Bureau and the Senate office. The telephone number is stored in memory of the camera control unit by pressing a single push-button, the number is automatically dialed and the connection established. Once this is done, the power-on function is manually initiated at the bureau and the system is ready for use. If during usage, the telephone line is accidentally disconnected, the system continues to operate and stays positioned on its current shot. This feature provides some protection against failures during live coverage. The correspondent has a repeat monitor to see him or herself on the outgoing feed and stay properly framed.

Figure 2. The Robotic Room

All necessary interfaces between the robotics system and the camera were designed by CBS Engineering and implemented by both AF Associates and CBS. The system has been in operation for approximately one year. It proved to be particularly useful during the first days of Operation Desert Storm, when it was used by a variety of on-air talent with almost no advance notice. As mentioned earlier, an identical system was subsequently installed at the Pentagon. This system has proved invaluable due to the high volume of material originating from there, during the Gulf crisis. This has especially been the case where feeds were originating from the Pentagon at various times of the day for many different broadcasts.

THE STUDIO PROJECT

While the Capitol Hill project provided CBS with an opportunity to test field robotics, studio applications still needed to be addressed. As discussed before, the Bureau's programming variations best fit the Network's need for a suitable test environment. Studio One was selected for this installation. While this installation was in design stage it was decided to expand the system to include all of the Bureau's cameras.

Specific operational constraints had to be taken into account during the design stage. The first requirement was that the system and indeed the cameras and their pan/tilt heads could operate both robotically and manually. In the manual mode, the pedestal and camera head performance had to closely emulate that of a traditional studio camera. The second constraint was that the conversion from robotic to manual and vice versa, could be done quickly and easily.

Another operational requirement was to be able to control the cameras from two different locations. For full studio operation with multiple cameras, a robotic room (Fig. 2) was set up to provide adequate monitoring for the operator. However, for daily one camera stand-ups, it didn't make sense to assign staff solely for this purpose. The solution was to provide another control point located in Central Control. This area is staffed full-time and could activate the system when necessary. This second control position was assembled in a roll around console which could be positioned in such a way as to be able to utilize existing monitoring (Fig. 3). Any camera in the system can be individually assigned to either one of the control panels via an assignment selector. The selector is located in Central Control.

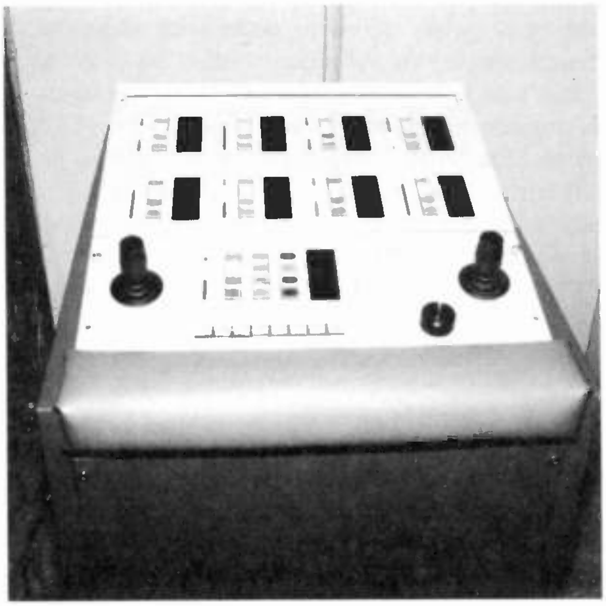

Figure 3. The Roll Around Console

The last requirement was that the pedestals in Studio One be truckable. The studio normally has four cameras and services four shows. Due to space limitations and the set placement, the need for X-Y movement on the pedestals was not of immediate concern. However, the pedestals had to be truckable in the manual mode. This meant that the pedestal drive mechanism had to be easily disengaged and the pedestals had to be equipped with a way to easily truck them manually.

The vendor selected for the entire plant project was Vinten. The six studio cameras utilize Vinten MH-240 heads while in the newsroom a Panasonic WV-F250 ENG camera is mounted on a Vinten MH-150 mid-size head. Technical modifications included improvements to the manual height control on the Z-axis pedestals. The changes allowed for smooth manual operation. Another modification involved adding a standard steer/crab ring on the Z-axis pedestals. On delivery, the pedestals came with tiller steering which was unacceptable for manual operation and trucking moves. Vinten engineered these pedestals with standard rings. Subsequently, an additional modification was added which allowed the height drive mechanism to be mechanically uncoupled from the pedestal's vertical column to facilitate easy and smooth manual vertical moves.

The studio system has been in use since May, 1990 and includes four robotic cameras in Studio One, two robotic cameras in Studio Two (flash studio), and one robotic camera in the Newsroom for its stand-up position. Two of the cameras in Studio One have the remote Z-axis height control feature.

CONCLUSION

The first two camera robotic installations proved that robotic technology has advanced to a level where, for some applications, it could be comfortably introduced into day-to-day activities of News operations without imposing any significant restrictions on production.

From an operating and financial standpoint, the system allows for better utilization of limited resources, thus allowing staff to be assigned to areas where human involvement is essential.

ADVANCES IN FM SYSTEM DESIGN

Monday, April 15, 1991

Moderator:
Charles T. Morgan, Susquehanna Radio Corporation, York, Pennsylvania

NRSC FM SUBCOMMITTEE REPORT*
Wes Whiddon
Group W
Houston, Texas

AN "N + 1" COMPATIBLE FM EXCITER*
G.W. Collins
Harris Corporation, Broadcast Division
Quincy, Illinois

ADVANCES IN TECHNIQUES FOR AIRBORNE ANTENNA PATTERN MEASUREMENTS
Harrison J. Klein, P.E.
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RDS IN THE UNITED STATES—A REVIEW OF 1990 AND PLANS FOR 1991
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*Paper not available at the time of publication.

ADVANCES IN TECHNIQUES FOR AIRBORNE ANTENNA PATTERN MEASUREMENTS

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Abstract

Advances in computer and navigation technology have made it possible to significantly improve the accuracy and reduce the cost of airborne antenna pattern measurements, replacing the chart recorders and visual tracking that were formerly used. The paper discusses new techniques that can be used to perform airborne measurements of broadcast transmitting antennas, and describes an equipment package that was developed for this purpose. Suggestions for improvement are also provided.

INTRODUCTION

Airborne measurements of antenna radiation patterns can be useful in many different circumstances. In AM, airborne techniques can be substituted for tedious ground-based field strength measurements, speeding and simplifying the iterative process of tuning a complex directional array. At VHF and higher frequencies, airborne measurements are the only way to avoid the terrain effects and local obstructions that make it impossible to determine the patterns of installed antennas by ground observation. In FM, airborne techniques allow stations to determine the effects of towers on side-mounted antennas. In TV, airborne techniques can be used to determine not only the horizontal plane pattern, but also the true gain and beam tilt at both visual carrier and color subcarrier frequencies, important performance parameters that are not always supplied by the antenna manufacturer.

PROBLEMS WITH PREVIOUS AIRBORNE MEASUREMENT TECHNIQUES

Lack of positional information

To make accurate airborne field strength measurements, there are two requirements relating to aircraft position. First, the position of the aircraft must be accurately known. If the aircraft is flying a one-mile radius around an FM antenna, even a 0.1-mile error in position may become a 1 dB error in field strength or a 6-degree error in azimuth. The previous techniques have been incapable of correcting such errors.

Second, the aircraft must fly the proper course. If measurements are needed along a particular radial of an AM directional antenna system, the aircraft must fly close to that radial. There is no mathematical procedure that can reconstruct the field strength on that radial if the aircraft was not on it.

Lack of accurate knowledge and control of aircraft position has been a principal difficulty with prior airborne measurement techniques. The measurement aircraft has traditionally followed a ground track laid out on a topographic map, usually a USGS quadrangle. Aircraft position has been only as accurate as the pilot was able to fly and the flight engineer was able to estimate with reference to the map and visual landmarks.

In many areas the landmarks needed for the traditional system do not exist. Even where landmarks exist, flying a precise ground track under crosswind conditions is difficult for the best of pilots, and errors are inevitable in judging when one is directly over a point on the ground that may be 2,000 feet below.

Cumbersome measurement equipment

Receiving antennas. A number of different antenna types have been used in the past for airborne measurements. One "consultant" merely holds a conventional AM field strength meter in the helicopter cockpit and records the readings by hand during flight; no credence can be placed on such data.

A better technique has been to mount a measurement loop on a pole and to suspend it below a helicopter. This requires some mechanism to lower the antenna after takeoff and to raise it again before landing. Since these installations are invariably temporary, they are often distinguished by an abundance of duct tape and bungee cords and an absence of FAA approval.

The antennas used are generally directional, either a modified lid antenna from an AM field strength meter or the dipole antenna from a VHF or UHF field strength meter. All require the flight engineer to manually orient the antenna properly with respect to the station. If a circular course is being flown in a crosswind, the proper orientation of the antenna must continuously vary, either requiring constant readjustment or resulting in measurement error.

Data recorders. The traditional data recorder for airborne measurements has been a chart recorder. Depending on the model used, they have been bulky, heavy, and messy (many an engineer has returned with red hands and clothes from Esterline-Angus ink). Their main problem, though, is the difficult and time-consuming data analysis they require. Yard after yard of paper must be examined. The geographical checkpoints, which are usually marked directly on the recorder paper during flight, must be deciphered. The data must be manually converted to a numerical format before they can be analyzed. This process is clearly best done by a computer.

The combination of cumbersome antennas and data recorders has kept the cost of airborne measurements high. In most cases, helicopters were required, at a cost several times that of fixed-wing aircraft. The antenna mechanism had to be redesigned for each aircraft installation. At least two persons, the pilot and an engineer, were required in the cockpit, and often an additional engineer was needed to navigate while the other operated the equipment and reoriented the antenna. Finally, data analysis was always slow.

IMPROVED MEASUREMENT AND NAVIGATION EQUIPMENT

An equipment package was developed to improve the airborne measurement process. The package consists of the following components:

- custom antennas for medium-wave and VHF
- navigation equipment, including receivers for LORAN-C and GPS (Global Positioning System) and an encoding altimeter, to provide guidance for the pilot and a three-dimensional record of aircraft position
- a computer-based data acquisition system that records field strength and aircraft position and drives a display system to guide the pilot along the desired flight path
- a data analysis system that determines the pattern of the antenna being measured
- a presentation system that produces data and figures ready for publication in a report.

The initial package was developed for the common Cessna 210 airplane. It was designed to be installed on a rented airplane without requiring any permanent modifications of the aircraft. FAA certification was obtained for the installation, making it completely legal and permitting the package to be moved easily from one rented 210 to another with minimal additional FAA involvement.

A second package is currently under development for a company-owned airplane. As described below, this will permit more extensive integration of the measurement equipment with the aircraft flight systems and will improve measurement accuracy still further.

AIRBORNE ANTENNAS

Medium-wave measurement antenna

For medium-wave AM measurements, a receiving antenna was developed with elements in each of three orthogonal planes. Each element consists of an inductive loop wound on a ferrite core and resonated with an appropriate capacitance. Use of an inductive loop minimizes the interaction of the antenna with the airframe, because the loop is sensitive to the magnetic field of the received signal which suffers little distortion due to the airframe. The ferrite core and the tuning serve to increase the antenna's gain. The absolute gain at a particular frequency is not critical because the antenna is calibrated as described below.

The three antenna elements are sealed in a plastic assembly that protects them from damage and maintains the proper orthogonal orientation. Figure 1 is a diagram of the antenna

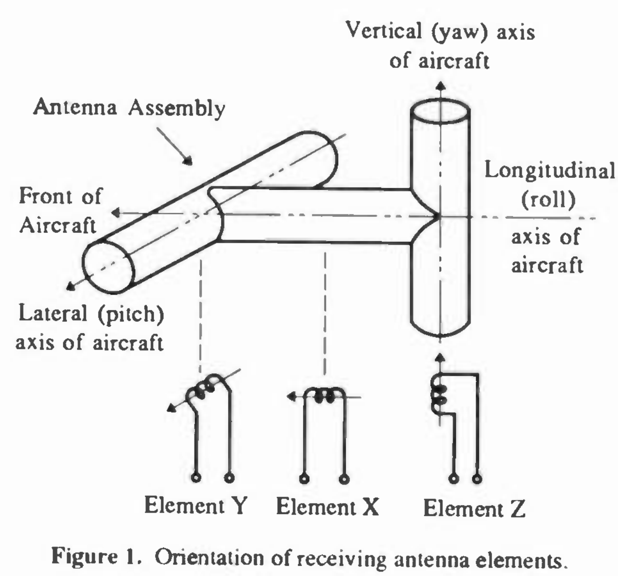

Figure 1. Orientation of receiving antenna elements.

assembly showing the orientation of the elements that extract the X, Y, and Z magnetic components of the radiated field.

Aircraft mount. The receiving antenna was mounted on a replacement baggage door for the Cessna 210. The 210 baggage door is an ideal location for such an antenna. It is an almost perfectly vertical surface. It is located on the left side of the fuselage behind the wing but in front of the tail, so it is has a clear view of the transmitting antenna on the counterclockwise orbits preferred by pilots. And it can be installed and removed without permanently altering the airframe.

A steel frame was designed and fabricated to strengthen the door and provide a stable mount for the measurement and GPS antennas. The mount is used with both medium-wave and VHF measurement antennas. The measurement antenna assembly slides into the frame and is held in place with hardware. The antenna leads exit the assembly inside the airplane.

Installing the antenna assembly on the airplane is simple: the hinge pin of the original baggage door is pulled, the original door is removed, and the door with the antenna assembly replaces it. Braces inside the airplane lock the door assembly into place; the hinge pin and latch on the door itself are not needed for structural integrity. Figure 2 is a photograph of the medium-wave antenna assembly as mounted on the Cessna 210 (the egg-shaped GPS antenna is also shown).

Figure 2. Medium-wave measurement antenna.

Antenna calibration. Calibration involved placing the antenna assembly in a uniform known field, as measured on a conventional AM field strength meter, measuring the antenna pattern by rotating the assembly, and determining for each of the three elements the three-dimensional antenna correction factors to convert from RF voltage at the antenna terminals to field strength in millivolts per meter. A local radio station near the frequency of interest was used as the reference field source.

Prior to calibration at a specific frequency, a comparison was made of the antenna assembly by itself and when mounted on the airframe to verify that there was minimal interaction. The principal effect of the airframe was a slight reduction in gain of the "Y" element when it was mounted on the door; mounting the door on the airframe had negligible additional effect.

Figure 3 shows the measured relative field pattern of a typical antenna element mounted on the baggage door. The

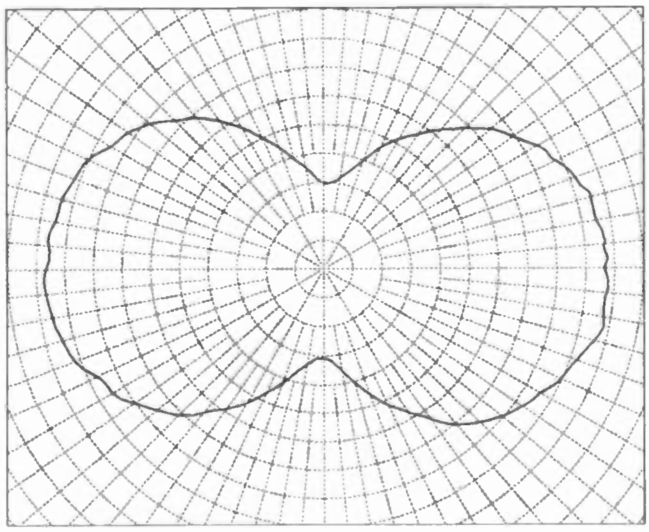

Figure 3. Pattern of typical medium-wave antenna element.

pattern is essentially cosinusoidal indicating minimal interaction with the aircraft structure. Because the three antenna elements are orthogonal and exhibit cosinusoidal patterns, the full-field field strength independent of aircraft orientation can be simply obtained by a root-sum-square of the three orthogonal components. This eliminates the need for the engineer to manually orient an antenna in the direction of the station.

VHF measurement antenna

A crossed-dipole antenna was developed for use at VHF, with one dipole oriented horizontally and the other vertically. Crossed dipoles facilitate the measurement of circularly or elliptically polarized antennas because they permit horizontally and vertically polarized signals to be measured simultaneously; in the past, circularly polarized antennas often required two sets of measurements. The two elements of each receiving dipole are connected to the balanced side of a balun transformer mounted at the center of the antenna assembly. A coaxial cable is connected to the unbalanced side of each transformer and the cables exit the assembly inside the airplane.

The VHF antenna uses the same mounting configuration as the medium-wave antenna. Figure 4 is a photograph of the VHF antenna mounted on the door. Because dipoles are

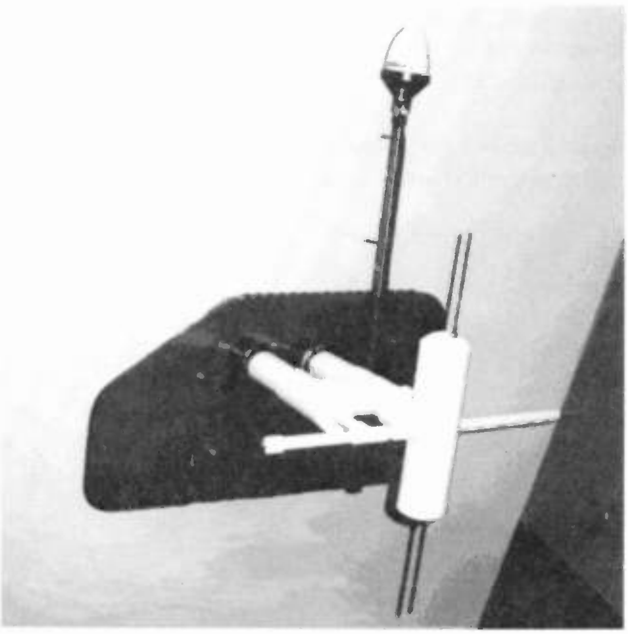

Figure 4. VHF measurement antenna.

used as the elements, the antenna is less directional than the yagis sometimes used for airborne measurements, making the orientation of the aircraft less critical. However, the antenna is not omnidirectional and the airframe has a greater effect on its directional characteristics than it does on the medium-wave antenna.

Antenna calibration. The VHF antenna was calibrated with an RF signal generator fed to a rotatable dipole antenna which produced a horizontally or vertically polarized field as desired. The generator and dipole were mounted on a framework that could be rolled to selected bearings and distances relative to the airplane.

The relative pattern of the measurement antenna was determined by mounting it on the airplane and moving it to an area of the airport that was free from structures and other aircraft. As the generator/dipole was moved to each bearing, the voltage at the measurement antenna terminals was recorded. The absolute gain of the antenna was determined by measuring the field at a particular location produced by the generator/dipole, then placing the airplane antenna at that same location and calculating the appropriate antenna correction factor.

The antenna exhibited a directionality of approximately ±2 dB over a ±10 degree range of bearings relative to broadside of the airplane, and ±5 dB over ±45 degrees. An error of as much as 5 dB in the measured field strength is unsatisfactory; even 2 dB is undesirable given the much better accuracy in the rest of the measurement system. Discussed below are methods that were used to reduce the actual error in early tests of the system and proposed enhancements to the system that will minimize the error still further.

NAVIGATION AND DATA ACQUISITION

Navigation equipment

The equipment package, which is self-contained and easily installed in the airplane, includes navigation receivers for both LORAN-C and GPS to provide backup horizontal positioning information. Each system has its strengths and weaknesses. By including both LORAN-C and GPS receivers in the equipment package, and continuously recording indications from both systems, the advantages of each can be realized and their disadvantages avoided.

LORAN-C. LORAN-C is a proven system which is available 24 hours a day throughout the world. Its published accuracy is ±60 feet which, if achieved, would be sufficient for many measurement projects.

Unfortunately, LORAN-C has not worked well in our experience. First, LORAN-C was intended mainly for maritime navigation; reception may not be reliable in inland areas. Second, accuracy may be substantially degraded depending on the relative geometry of the transmitters in the particular LORAN-C chain being used. The results of one airborne measurement project indicated that LORAN-C was accurate to only ±400 feet. Third, LORAN-C must be calibrated for the area in use. Shipping a LORAN-C receiver across the continent and using it without recalibration may result in errors of several miles in the absolute latitude and longitude readings it produces (although the relative changes in lat/long with respect to a starting point will remain accurate). Finally, LORAN-C is a two-dimensional system; it offers no altitude indications.

GPS. The Global Positioning System is a navigation dream almost come true. When all of the GPS NAVSTAR satellites have been placed in orbit, GPS will be continuously available, will provide three-dimensional position information and, depending on the receiving configuration, will have an accuracy of a few feet.

However, the full potential of GPS has not yet been realized. Only a portion of the proposed NAVSTAR satellites are in orbit, so GPS is available only part of the day. The available time changes daily, making it difficult to do detailed advance planning of a measurement program. The U.S. Defense Department has reserved the right to implement the "selective availability" provision of GPS, which consists of intentionally degrading, in the interest of national security, the accuracy of non-military GPS receivers. When S/A is implemented, errors may be several hundred feet, which would be unacceptable in a project to measure, for example, a high-gain UHF antenna with a beamwidth of only a few degrees. Methods to overcome S/A limitations are described later.

Encoding altimeter. Because altitude indications are not provided by LORAN-C and are not yet reliable from GPS, the navigation system uses an encoding aircraft altimeter. The altimeter was modified to improve its accuracy to ±17 feet. However, any pressure altimeter suffers from an inherent limitation due to its dependence on a standard air pressure gradient for accuracy. The altimeter is calibrated by referencing its indication to a known elevation, usually that of the airport. However, if the measurements are made at several thousand feet above ground and the pressure gradient is non-standard, substantial elevation errors may occur. Visual calibration is normally done using the transmitting antenna being measured as a reference.

Display system. The computer used for data acquisition, described below, also generates signals to drive a display system to guide the pilot along the desired circular or radial flight path. The geographical coordinates of the antenna being measured are entered into the computer as well as the desired distance from the antenna for circular paths or the bearing for radial paths. As the airplane's latitude and longitude are received from the navigation receivers, the computer calculates the error in position and drives a left–right indicator for the pilot.

Data acquisition

Field strength measurements are taken with conventional Potomac FIM-41 or FIM-71 field strength meters. If necessary, the system can use two meters set to different scales, which provides a dynamic range of up to 80 dB with a resolution of better than 0.3 dB. The meter outputs are fed to separate analog-to-digital converters to be read by the data acquisition computer. Each A/D converter was calibrated by feeding known levels from a signal generator to each field strength meter and recording the output of the converter. By using these data, along with the calibration data on the measurement antennas, the field strength meter output(s) can be converted to millivolts per meter.

A portable computer is used to control the system and to record all necessary data. The computer records position information, field strength, time, and event markers (described below). The position information from the navigation units is recorded quasi-simultaneously with each set of field strength measurements. Meter inputs from the three medium-wave or two VHF antenna elements are continuously switched at intervals approximating the settling time of the field strength meters.

Figure 5 shows the navigation and data acquisition system installed in the right rear seat of a Cessna 210. The computer

Figure 5. Navigation and data acquisition system.

is raised to a convenient operating position for the flight engineer. Power for the system is obtained from the cigarette lighter socket of the airplane.

MEASUREMENT PROCEDURES

The medium-wave system has been used for both circular and radial measurements of directional AM arrays. The VHF system has been used for circular measurements of side-mounted and top-mounted, circularly polarized FM antennas. Circular orbits of 1 to 1.5 miles were made, as well as radial measurements out to 10 miles from the AM arrays. Several runs along each path are made during all measurement programs to provide assurance of repeatability.

The navigation equipment must be calibrated on each day of measurement to account for variations due to propagation conditions, satellite positions, and barometric pressure changes. Calibration is performed by making low-level flights over the center of the antenna and recording the position as indicated by each navigation system. Subsequent position reports use these measured center points as a reference so that the true location of each field strength measurement with respect to the center of the array can be determined.

Additional position accuracy is obtained through the use of event markers. At appropriate known locations during each run, such as when crossing a known road on a circular run or when crossing the antenna on a radial run, a marker is inserted into the data. These markers provide additional reference points throughout each measurement period.

Correction for VHF antenna pattern errors. In early tests of the VHF measurement system, the pilot's attempts to maintain a precise circular flight path with reference to the left–right indicator produced wildly varying airplane headings of as much as ±60 degrees from tangency. This, in turn, caused significant errors in the measured field strength due to the directionality of the VHF receiving antenna.

It was recognized that, because the intended flight path was in the main beam of a low-gain FM antenna, errors in distance from the antenna could be corrected after the fact simply by multiplying the measured field strength by the actual distance from the antenna as recorded by the navigation system. However, variations in heading could *not* be corrected, because aircraft heading was not recorded. Therefore, the pilot was instructed to maintain a fixed attitude relative to the station, even if that meant a lack of precision in maintaining a circular flight path. This was a good working approach to the immediate problem, but a better long-term solution is needed.

DATA ANALYSIS

The measurement data are analyzed by computer. The program first converts the raw field strength data to units of mV/m, based on the analog-to-digital converter and antenna calibrations described above. The position data are then analyzed to determine the location of each field strength sample relative to the center of the measured antenna.

No pilot can always maintain a perfectly circular or radial flight path. To eliminate anomalous data, the computer program filters the field strength data from multiple runs over the same path to eliminate data from those locations that depart substantially from the desired flight path. Field strength data are converted to unattenuated field strength at one kilometer by multiplying by the slant distance from the antenna in kilometers. For AM measurements, these are the desired units. For VHF, these data are further converted to equivalent effective radiated power in dBk or kilowatts, which are generally the desired units.

The data remaining after the above filtering are then analyzed and averaged to produce the final radiation patterns. The circular patterns are determined at two-degree azimuth increments around the array. At each two-degree azimuth the remaining data from each run are examined to find the closest two bracketing data points. If the two data points are within 20 degrees in azimuth, they are interpolated to yield a field strength at the desired azimuth for that run. If the closest two data points bracketing the desired azimuth are more than 20 degrees apart, no field strength from that run is used. The interpolated data at each two-degree azimuth from multiple runs is then averaged to yield a final field strength or equivalent power at each azimuth.

TYPICAL MEASUREMENT RESULTS

Figures 6 and 7 show examples of circular and radial flight paths, respectively, about an AM array. Each small "+"

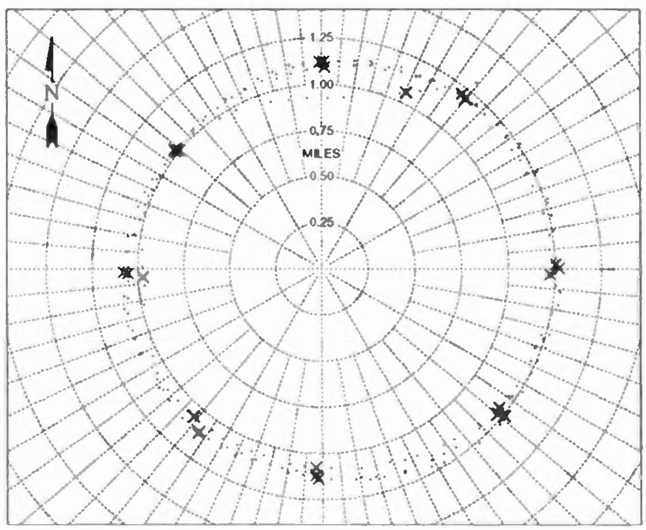

Figure 6. Sample circular flight path.

indicates the location of one data sample. On the circular runs, event markers "X" were recorded along the line of towers (39° and 219°), broadside to the line of towers (129° and 309°), and at the four compass points (N, S, E, W). The marker at 25° marked the beginning of the run, before a stabilized circular path had been established. On the radial

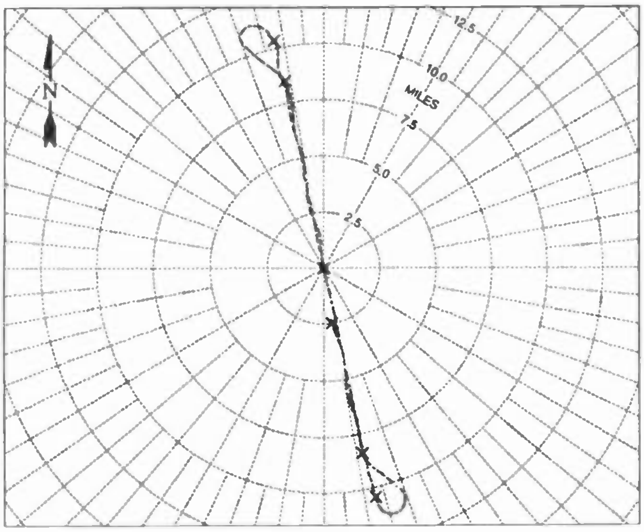

Figure 7. Sample radial flight path.

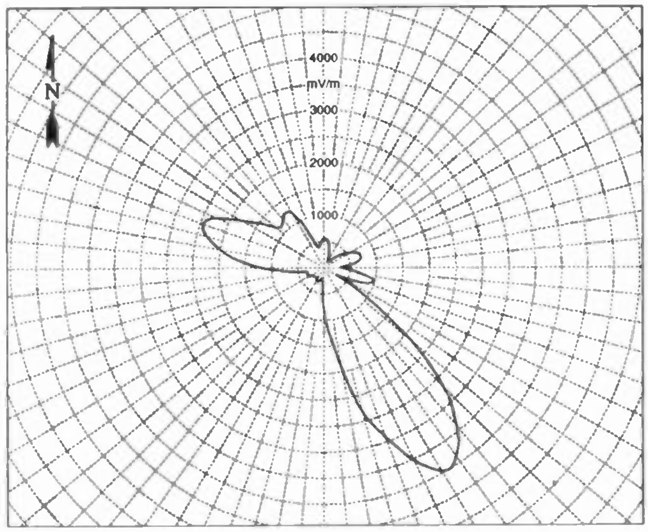

Figure 9. Measured radiation pattern of AM antenna.

runs, event markers were recorded at the beginning of the run, at the start of the course-reversal turns, when the airplane was established on the correct course, and over the center of the array. The figures show that the system was able to achieve good repeatability.

Figure 8 shows field strength data recorded over several circular paths around an AM array. Excellent repeatability is indicated; note the pattern nulls of each run.

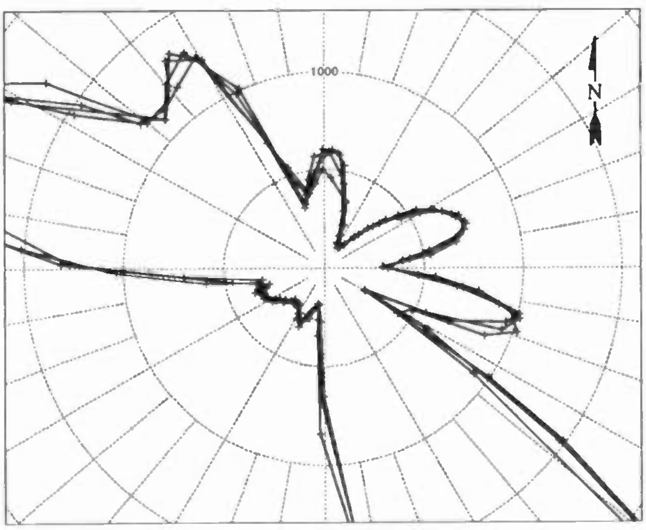

Figure 8. Repeatability of field strength data.

The results of a set of AM and FM measurements are shown in Figures 9 and 10, respectively. The AM measured pattern correlates almost perfectly with its calculated pattern. The FM pattern has the characteristics one would expect of a side-mounted antenna on a large cross-section tower.

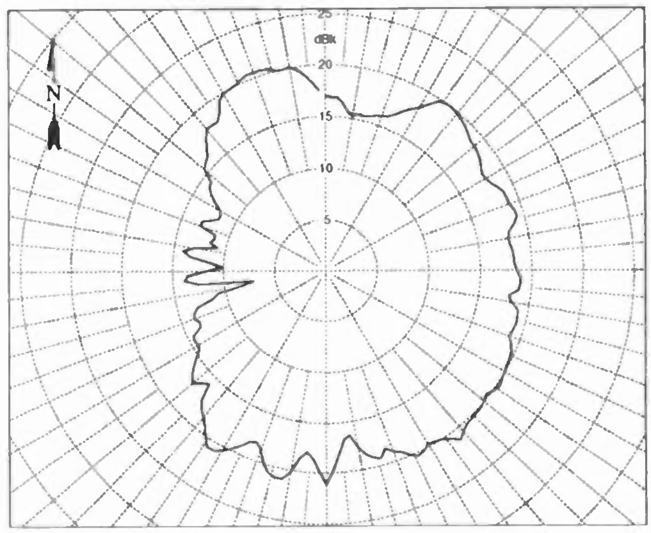

Figure 10. Measured radiation pattern of FM antenna.

Figures 6–10 were produced directly by the computer. They are PostScript files, ready for inclusion in a report or presentation.

AREAS FOR FUTURE IMPROVEMENT

Simplify the piloting requirements

Maintaining the desired flight path for airborne measurements is a demanding task. As a result, at least two people, and in congested areas three, have been required in the airplane: the pilot, a navigator to direct the pilot and watch for traffic, and an engineer to operate the equipment. Even with the pilot devoting full attention, excessive deviation from the desired course sometimes occurs.

An improved system is under consideration that would couple the computer to the airplane's autopilot. Similar to autopilot operation while tracking a navigation aid, the computer would control the course and altitude more accurately than could an unaided pilot. The pilot would be able to devote more attention to other duties including, perhaps, operation of the measurement system. The long-range goal is to have the airplane and measurement system operated by one pilot-engineer.

Reduce VHF/UHF antenna directionality

Although it was possible to compensate for the directionality of the VHF measurement antenna as described above, this is not always possible. For example, if it is desired to measure the pattern of a high-gain antenna at a –2 degree elevation angle, the precise altitude and radius to place the airplane at –2 degrees must be flown. The pilot must be permitted to change heading to compensate for any crosswind. Any directionality in the measurement antenna will produce errors in the measured field strength.

Two approaches are under consideration to reduce the effects of directionality. First, an improved crossed dipole antenna is being developed for installation in the left wingtip of the airplane. It is expected that the increased distance from the fuselage will produce a more cosinusoidal pattern. Second, in conjunction with the autopilot integration discussed above, the computer could record an indication of the airplane's heading along with each field strength reading. During data analysis, the bearing of the airplane with respect to the station could be reconstructed, and the measured field strength could be corrected for the relative field pattern of the measurement antenna.

Implement differential GPS navigation

As the GPS system matures, it will clearly be the navigation method of choice. However, errors of several hundred feet under selective availability are unacceptable. Even without S/A, errors of ±80 feet or so are typical and are larger than desired for measurements of high-gain UHF antennas.

The improved navigation system under consideration includes "differential" GPS capability. With differential GPS, a ground station continuously monitors the same GPS satellites used by the airborne receiver. Since the location and elevation of the ground station are known, the airborne measurements can be post-processed to cancel out any satellite errors present. Using this method, accuracies of better than ±15 feet are achievable.

Obtain FCC approval for airborne AM measurements

The field strength measurements in an AM proof-of-performance are some of the most time-consuming and expensive engineering requirements in broadcasting today. AM continues to require the most antenna engineering in broadcasting yet AM stations are often financially weak. The result is deterioration, out of tolerance operation, and interference.

The AM measurements performed with the system described in this paper have produced in one day a more accurate indication of the true directional antenna pattern than a complete FCC proof would have after weeks of work. To demonstrate the validity of this technique and, hopefully, to obtain FCC approval for its routine use, a complete set of airborne measurements must be made in parallel with a conventional full proof-of-performance and the results compared. The author's firm intends to undertake such a project shortly with the cooperation of a client station.

CONCLUSION

The equipment and techniques described in this paper can be used to improve the efficiency, accuracy, and usefulness of airborne measurements of antenna radiation patterns. Despite the current limitations of both LORAN-C and GPS, the benefits of simultaneously recording aircraft position and field strength have been demonstrated. The omnidirectional medium-wave measurement antenna has eliminated the need for manual orientation of a loop. VHF measurements have been made satisfactorily and an improved VHF/UHF antenna is contemplated. With continued development, the ability to cost-effectively gain a complete understanding of antenna performance can only improve.

RDS IN THE UNITED STATES— A REVIEW OF 1990 AND PLANS FOR 1991

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Abstract

Radio Data System (RDS) which is already fully implemented throughout Europe, is coming to North America. 1990 was the pivotal year when RDS was demonstrated to U.S. broadcasters at the 1990 National Association of Broadcasters Convention in Atlanta, GA. During 1990, the adoption of Standards for RDS in the United States was discussed by the NRSC (National Radio Systems Committee), and its members, the NAB and EIA. The Standards process is proceeding quickly with a target of presenting the North American RDS Standards at the 1991 NAB Convention in Las Vegas in April 1991.

Background

RDS (Radio Data System) was developed in the mid 1980's in Europe as an auxiliary service for broadcasters. Developed by and for broadcasters, RDS technology incorporates many elements which are very important to American broadcasters, including:

1 - Display of call letters or slogan on home and car radios. (PSN)

2 - Searching for radio stations stations by program format. (PTY)

3 - The use of the alternate frequency technology for translators or simulcasts. (AF)

4 - RDS paging. (RP)

5 - Emergency alerting. (EBS)

6 - A housekeeping function for broadcasters including telemetry and remote control. (IH)

Since January 1990, the first commercial RDS signal in the United States has been broadcast by WHTZ-FM (Z-100) 100.3 MHz in New York. That signal emanating from the Empire State Building has been on the air for more than one year in the toughest radio environment in the country. The New York "loudness wars" have proven that RDS is compatable with all formats and all types of processing.

In June of 1990, WYTZ-FM Chicago, owned by Capitol Cities/ABC has been on the air with RDS. Delco Electronics has assisted four stations in the Kokomo and

Indianapolis area to begin broadcasting RDS, including:

1 - WZWZ-FM Kokomo, IN.
2 - WKHY-FM Lafayette, IN.
3 - WSHW-FM Frankfort, IN.
4 - WLHN-FM, Anderson, IN

Delco is in the process of assisting 5 stations in Detroit to begin RDS services, including:

1 - WWWW-FM Detroit, MI
2 - WJOI-FM Detroit, MI
3 - WLLZ-FM Detroit, MI
4 - WHYT-FM Detroit, MI
5 - WKQI-FM Detroit, MI

The first emergency warning and alerting system utilizing RDS technology is going into Jefferson County, Texas (Port Arthur and Beaumont). 12 radio stations (3 AM and 9 FM), 3 TV stations, and 2 TCI cable headends will be incorporated in a county-wide emergency alerting and warning system for use by the petrochemical industry, and for severe weather alerts during hurricanes and tornadoes.

AXCESS Corp. of Metairie, Louisiana, has developed a new alphanumeric RDS pager at a very affordable price. This pager, which will be available early 1991, will allow broadcasters to utilize portions of their RDS signal for commercial paging activities. While most broadcasters will not enter the paging business on their own, they can lease this capacity to a local paging company. The system will support tens of thousands of RDS pagers. This capability allows paging companies to increase their paging capacity without adding new transmitters and antennas, and provide revenue to broadcasters.

U.S. RDS Standards

In Europe, the RDS technology has been standardized under the auspices of the European Broadcasting Union (EBU) and CENELEC. This Standard RDS No. EN50067 is widely distributed and describes the standards for RDS transmission. In the United States it is proposed to adopt the bulk of this CENELEC Standard as the North American Standard with certain editorial changes, including:

1 - PI Codes.
2 - PTY Codes.
3 - Code Sequences.
4 - Injection Levels.

Also included in the U.S. Standard will be the Group 9-A National Warning System (EBS) and the Group 8A Traffic Message Channel (TMC).

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)

SPB 482

Proposed enhancements of the EBU on CENELEC EN 50067 (RDS)

Method for linking RDS programme services
Usage codes of Block 3 of Type 1A groups
Specification of extended country codes
Coding structure allocated to emergency broadcast systems

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

R·D·S
RADIO DATA SYSTEM

PI CODES

Several scenarios were suggested to assign PI codes via geographical areas. This would have necessitated some group or agency, such as the NAB or SBE, assign PI codes to individual stations. A more practical solution seems to be the use of call letters as the basis for creating hexadecimal PI codes. Formulas were developed which will allow stations to determine their own PI codes from their call letters.

If a station was not transmitting alternate frequencies, it would take a default PI code of 1001 Hexadecimal. If a station had a 2 letter call such as KYW, a specail group of PI codes would be assigned. Special PI codes would be assigned to network stations such as National Public Radio.

METHOD: Call Letters mapping to PI code.
NOTE: Call letters or slogan to be displayed by receiver are sent using the PS (program service) data.

A) If a program is unique, transmitted on only one frequency, set the PI code = 1001

B) If a program is carried on more than one frequency (translators or simulcast), a unique PI code needs to be assigned with the following formulas:

1) Assign decimal values to last 3 letters of call letters:

LETTER	DECIMAL VALUE	LETTER	DECIMAL VALUE
A	0	N	13
B	1	O	14
C	2	P	15
D	3	Q	16
E	4	R	17
F	5	S	18
G	6	T	19
H	7	U	20
I	8	V	21
J	9	W	22
K	10	X	23
L	11	Y	24
M	12	Z	25

2) Assign weighted value according to call letter's position and add together to obtain a *DECIMAL* value for last 3 letters.

K	3rd letter position	2nd letter pos.	1st letter pos.
W	3rd letter position	2nd letter pos.	1st letter pos.

3rd letter position x 676
\+ 2nd letter position x 26
\+ 1st letter position
decimal value for 3 letters = *DECIMAL*

3) If station begins with K, HEX (*DECIMAL* +4097) (value obtained above + 4097) to obtain four digit PI code. However, if station begins with W, HEX (*DECIMAL* +21673) to obtain four digit PI code.

IF K...	HEX [*DECIMAL* +4097]	= FOUR DIGIT PI CODE
IF W...	HEX [*DECIMAL* + 21673]	= FOUR DIGIT PI CODE

EXAMPLES OF ASSIGNING PI CODES FROM CALL LETTERS:

STATION1: KGTB

G = 6 X 676	= 4056
T = 19 X 26	= 494
B = 1	= 1
	= 4551

SINCE STATION BEGINS WITH K: 4551 + 4097 = 8648 = STATION DECIMAL NUMBER HEX [8648] = 21C8 = KGTB'S PI CODE

PTY Codes

The PTY or Program Type Codes would identify the format(s) of radio stations such that new generations of RDS radios could select radio stations by program format. There are 31 possible PTY codes with PTY 30 and 31 reserved for emergency and alerting purposes. The remaining 29 PTY Code definitions will need to be agreed upon to specify program formats in existence today with room for modification in the future.

Two schools of thought exist on PTY codes. One says that PTY Codes should provide narrow definitions for broadcasters such as Soft Rock, Oldies Rock, Hard Rock, Adult Rock, Classic Rock, etc. This would allow listeners to accurately zero in on their music preference.

The other school of thought says that PTY definitions should be broad such that all stations fit into a few large categories and the listeners can differentiate after sampling the stations within that category to find the ones that they like best. Broad categories such as Rock, Classical, Easy, Country, Ethnic programming, etc. could be proposed. While this scheme would expose listeners to many stations, it may not satisfy the listener's desire to zero in on the particular type of music they want.

Several PTY proposals have been put forward to the NRSC (see Appendix). It will be up to the broadcasters to finalize this selection so that receiver manufacturers can proceed to incorporate this format search feature into their receivers.

PTY TABLE

FORMAT	8 CHARACTER DISPLAY
1. Classical	CLASSICL
2. Country	COUNTRY
3. (Drama)	(DRAMA)
4. Easy Listening, Beautiful Music	EASY
5. Educational	EDUCATE
6. Ethnic - Spanish, Greek,...	ETHNIC
7. Folk Music	FOLK
8. Jazz, New Age	JAZZ
9. Middle Of the Road, Variety	M_O_R
10. News, Sports, Weather	NEWS
11. (Nostalgia - Big Band)	(NOSTALG)
12. NPR (National Public Radio)	N_P_R
13. Oldies	OLDIES
14. Religious	RELIGION
15. AOR/Classic Rock/Rock	ROCK
16. Soft Rock, Light Rock, Adult Contemp.	SOFT_RCK
17. Sports	SPORTS
18. (Talk)	(TALK)
19. Top 40/CHR	TOP_40
20. Urban/Black/Soul/R&B/Dance	URBAN
21 - 29. Spare	
30. Emergency	ALERT!
31. Emergency	ALERT!

Code Sequence

Within RDS there are up to 16 different data groups which can be defined. How often they are transmitted depends on the importance of the information contained in each group. Certain types of information such as PTY, PI, and PSN are transmitted frequently to ensure listeners quick access to station information. In addition, the alternate frequencies of radio stations (translators or simulcast), as well as clock time, are also transmitted frequently.

If a station uses paging, a 7A Group would be transmitted as often as required to ensure a reasonable access time through the paging terminals. The same would be true for the traffic message channel (TMC) in the 8A Group which would send digital information which would be converted into spoken words by a voice synthesizer in the radio. The 9A Group used for emergency alerting would normally be transmitted very infrequently (once every hour) except when an emergency situation exists. At that time, the 9A Groups would take priority over all other transmitted data, and would be sent through the system with its full capacity to quickly discharge the emergency information as required.

A code sequence has been proposed to the NRSC (see Appendix) which would accommodate the broadcasters' needs. In addition, a dynamic code sequence technology has been developed by Rohde & Schwarz, one of the major manufacturers of the RDS encoders wherein the code sequence will change based on the minute to minute needs of the information to be transmitted. Priorities would be assigned to such information as emergency alerting, traffic information, paging, call letter display, etc.

Emergency Alerting and Warning

Within the RDS system Standards is an alerting and warning system called NWS or EBS. This system, called WARI in Germany and SAGE I in the United States, is actuated by using the 9A and 1A Groups of the RDS system to carry a variety of control and command signals. All RDS car radio will contain the PTY 31 function for emergency alerting which is controlled by this system. The system will also actuate siren systems, page emergency workers and send digital messages to electronic signs. This system will electronically capture all mass media simultaneously (AM, FM, TV, cable, SMATV, MUSAK) etc, to transmit emergency messages over their main channel. In addition, the system provides alphanumeric data to TV and cable facilities so that the emergency message which is being transmitted audibly will also appear as a full screen visual text presentation for the hearing impaired. The RDS based system could also automate the emergency broadcast system during a transition from today's analog EBS technology to an RDS based system.

The Federal Communications Commission has begun work on an EBS Notice of Inquiry (NOI) which is expected out during the first quarter of 1991. This Notice of Inquiry will address the existing EBS system and look at various technological options and developments that have occurred in the 30 years since the current EBS was adopted. Indications are that the Commission will look favorably towards new technologies which will eliminate many of the shortcomings of the existing EBS system.

A possible infrastructure for an RDS based national emergency alerting and warning system using the over 300 RDS equipped National Public Radio (NPR) stations as has been proposed to the FCC. Since NPR already has a nation-wide satellite delivery system in place with up links from 22 sites and down links to all 300+ stations, NPR could provide a national emergency network almost immediately. In the areas where NPR stations did not exist, or had insufficient coverage, low cost down links (less than $2,000) would be provided to stations which would be the primary alerting stations for that area.

With the RDS based emergency alerting technology, audio and alaphanumeric emergency messages would be recalled from digital storage or originated live from an emergency operating center. These messages would be sent via an RPU frequency in the 450 or 455 MHz band to one or more radio stations in an area for over-the-air transmission. In addition, the second RPU channel from the emergency operating center to the radio station(s) would carry 1200 band encrypted X.25 packet data which is used to command the RDS encoder at the radio station(s) for emergency alerting functions.

All mass media facilities, AM and FM, TV and cable would have special control boxes in their studios which would electronically self tune to the primary (CPCS-1) station in their area. If an emergency message was about to be transmitted, an advance warning would be sent to all mass media over the RDS digital subcarrier. The message would be seen on an alphanumeric display in front of the board operator or in master control at a TV or cable facility. The message could say "TORNADO WARNING IN 30 SECONDS". If the station operator took no further action, their facility would be "captured" for an audio and/or video transmission of the tornado warning. If a station did not want to broadcast the emergency message, it would have an "inhibit switch" at its control location so the station could "drop out" of the alerting network. If a station threw the inhibit switch, a digital data packet would be sent to the actuation center indicating which station(s) were not transmitting the emergency alert. If one of the stations to drop out of the network was the primary or CPCS-1 station, all specially equipped frequency agile receivers in schools, hotels, homes, airports and other mass media facilities would electronically retune to the designated secondary, teritary or fourth stations in the area which would be carrying the emergency alert.

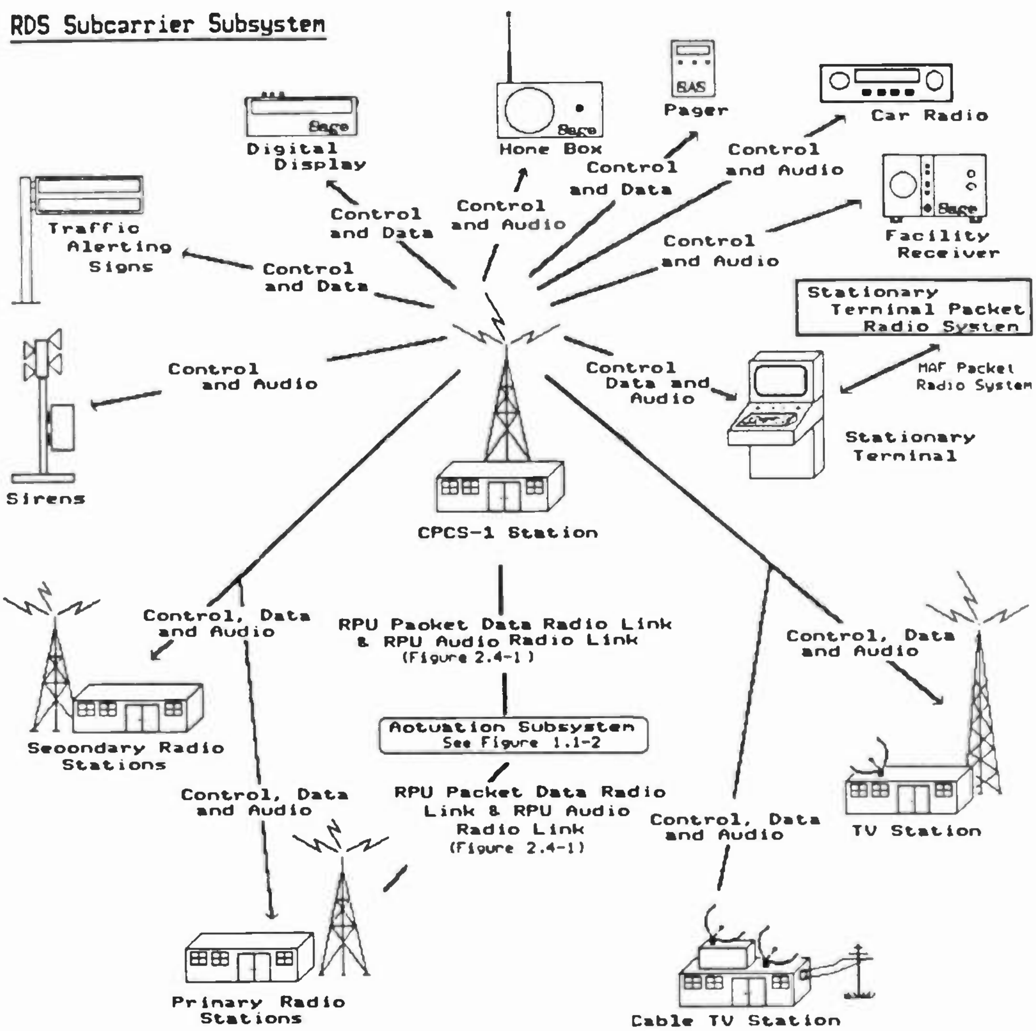
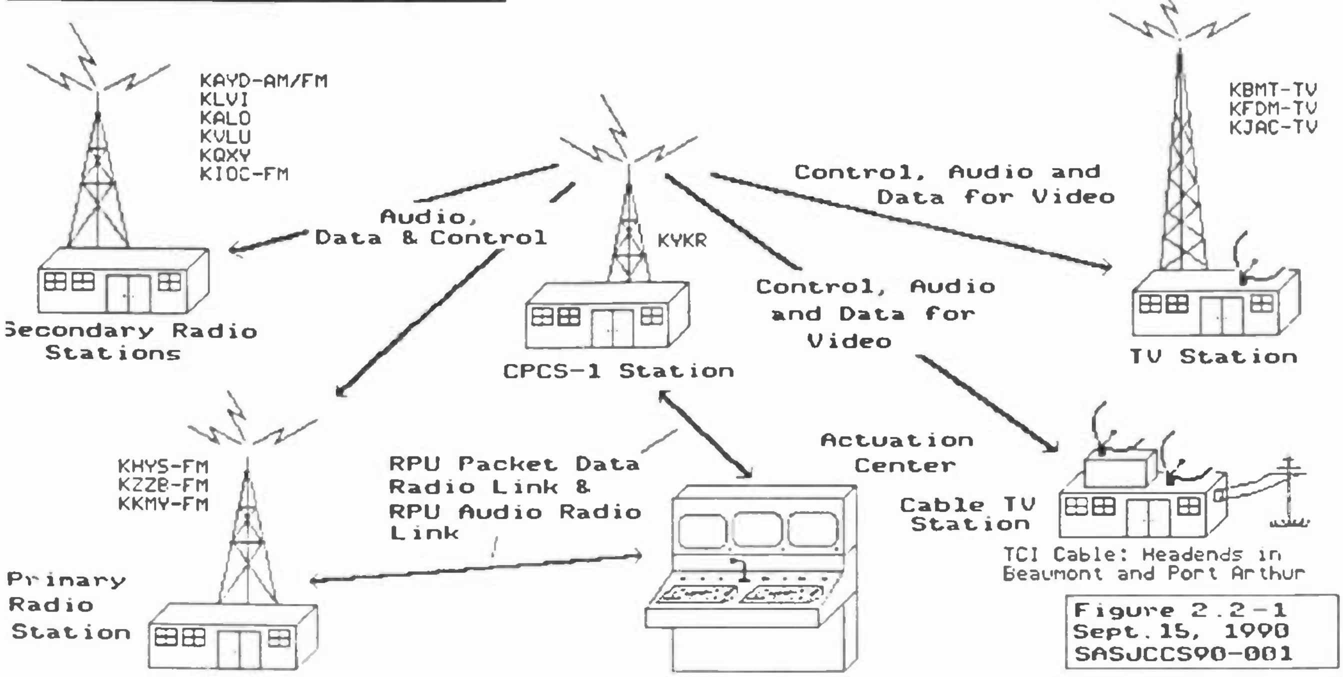


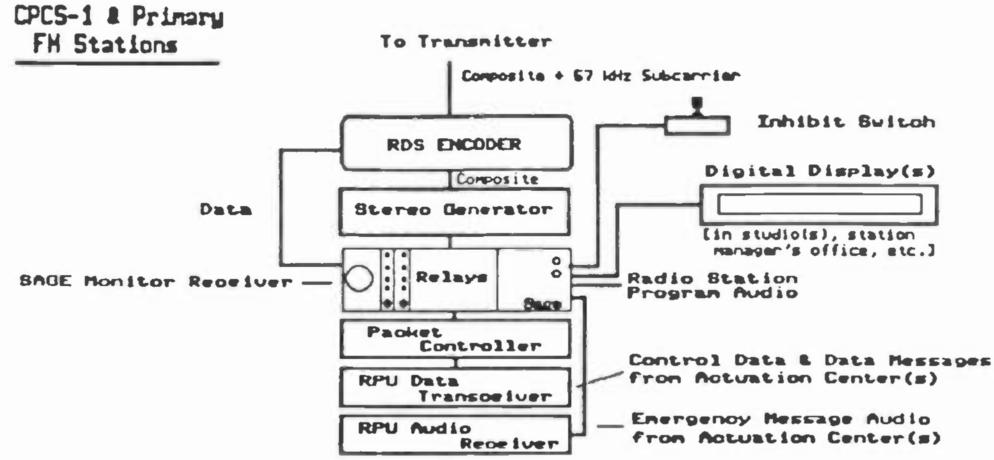
Figure 2.4-3
Sept. 15, 1990
SASJCCS90-001

SAGE I Transmission Subsystem

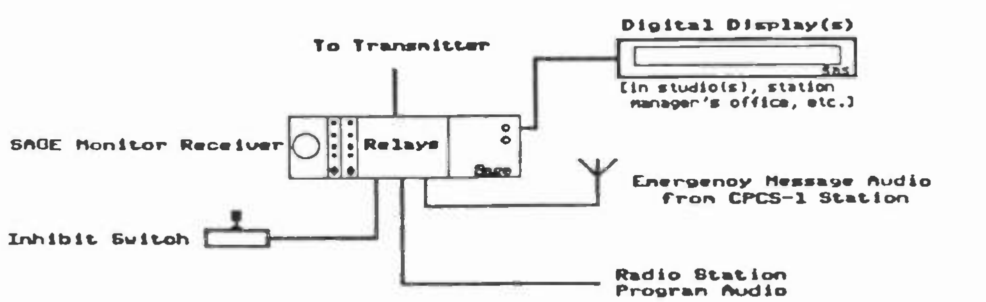


JEFFERSON COUNTY, TEXAS
SAGE I SYSTEM

CPCS-1 & Primary FM Stations



Secondary Radio Stations



Conclusion

RDS has made excellent progress in its move across the Atlantic. 1991 should see more RDS stations on the air and more RDS receivers in the market. RDS paging and emergency alerting will be but two of the many RDS features we should see move into the main stream of the U.S. market.

ACKNOWLEDGMENTS

The author would like to thank Terry Beale of Delco Electronics, Matt Straeb of Rohde & Schwarz, Robert Adams of Axcess, Hans Duckeck of Blaupunkt, and Ditmar Kopitz of EBU for their valuable assistance. I would also like to thank Stan Salek of the NAB and Tom Mock of the EIA for their help and guidance in RDS Standards Process in the United States. The author would also like to thank Michael Starling of National Public Radio and David Reeves WHTZ-FM, New York, for their continuing support of RDS.

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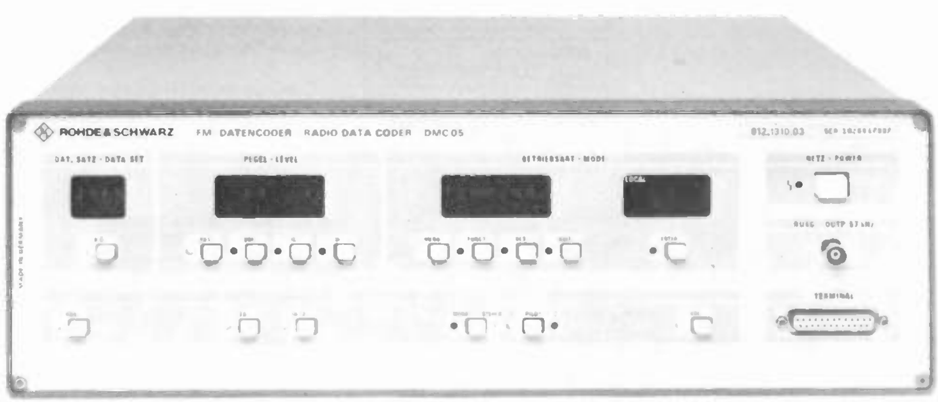

ROHDE & SCHWARZ RDS ENCODER

DELCO RDS RADIO FOR NORTH AMERICA

PROPOSED RDS GROUP SEQUENCE

0 — 2 sec

	0A		0A	3A		0A		0A			6A	1A	0A		0A	3A		0A		0A			1A
Every minute																							4A
Every 12 hours		9A												9A									
Normal run		2A			0A		2A		0A	0A				2A			0A		2A		0A	0A	
Before + after traffic message		2A			15B		2A		15B	15B				2A			15B		2A		15B	15B	
Emergency		9A			9A		9A		9A	9A				9A			9A		9A		9A	9A	
Paging (Peak)		2A or 7A			7A		2A or 7A		7A	7A				2A or 7A			7A		2A or 7A		7A	7A	
TMC (Peak)		2A			8A		2A		8A	8A				2A			8A		2A		8A	8A	

BROADCAST/AERONAUTICAL COMPATIBILITY: AIRWAVES AND AIRSPACE, WHO GOVERNS WHICH?

Tuesday, April 16, 1991

Moderator:
John F. X. Browne, John F.X. Browne, P.C., Bloomfield Hills, Michigan

BROADCAST/AERONAUTICAL COMPATIBILITY ISSUES AND STATUS*
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AN UPDATE ON THE FAA ELECTROMAGNETIC INTERFERENCE MODEL
William P. Suffa, P.E.
Lahm, Suffa & Cavell, Inc.
Fairfax, Virginia

FAA PERSPECTIVES ON PROTECTING THE NATIONAL AIRSPACE*
David F. Morse and Gerald Markey
FAA
Washington, District of Columbia

FCC/FAA COORDINATION*
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Chief, Audio Services Division
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*Paper not available at the time of publication.

AN UPDATE ON THE FAA ELECTROMAGNETIC INTERFERENCE MODEL

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Over the last three years or so, there has been a great deal of interest in the Federal Aviation Administration actions concerning electromagnetic interference to aviation systems. The interest stems from the impact of a computer model employed by the FAA to conduct studies of potential FM broadcast interference to VHF aviation receivers.

This paper will present some new technical information concerning the model and its development, as well as update the information provided in an earlier paper.

Introduction

Electromagnetic compatibility will be a term frequently heard in the future. A prime focus for broadcasters in the next few years will be the compatibility between different radio services. This will be true regardless of whether the transmission method is AM, FM, TV or digital.

Not surprisingly, the Federal Aviation Administration (FAA) has exerted its authority in an attempt to control interference to its aviation radio systems, claiming that such interference represents a hazard to air transportation. While interference is a technical issue, engineers must understand that solutions to such problems in today's environment often involve political influence as well as technical expertise (some might describe the politics as being <u>more</u> important). The issues surrounding the FAA interference model are both technical and political.

This paper will address technical issues, with an eye towards the politics lurking in the background. The following topics will be presented:

- a short background discussion
- the FAA receiver testing method
- the FAA validation methods
- improvements made to date
- remaining problems
- what the future holds

and

- political considerations.

For additional information, and additional reading on this topic, the reader is directed to a paper presented in the National Association of Broadcasters 1989 Broadcast Engineering Conference Proceedings. Much of the technical reference material is contained in that paper.

Background

Until recently, compatibility between radio transmission systems operating in various parts of the radio spectrum has only been of isolated concern. With today's crowded spectrum and the demand for radio frequencies being high, compatibility matters are an increasing concern.

It has long been recognized, for example, that there is potential for interference problems between high power broadcast facilities and lower power communications and navigation systems. Joint industry committees in the 1960's and 1970's dealt with approaches which could be used to minimize prospective interference between FM broadcast stations and air navigation systems operating in the 108 to 118 MHz frequency bands. The approach taken by the FAA was to relocate or modify the navigation aid, where possible to accommodate the broadcast proposal. Recently, however, the FAA has placed the burden on the "newcomer"; that is, the proponent of new or modified broadcast facilities.

While it seems as though interference between radio systems would be a technical matter easily resolved by engineers, the case at hand has become a highly political turf war between various interest groups.

The FAA, for its part, seeks to ensure the safety of air travelers and protection of its facilities, while the FCC and communications industry groups want to have the most flexibility possible for private and commercial radio transmission facilities. Both goals should be capable of being achieved - if there is cooperation between the parties. In the past two years, some progress has been made towards a dialogue between the two federal agencies. A wide gap continues to exist, however.

Until 1987 the FCC was the sole federal agency responsible for allotment of non-governmental radio spectrum and the establish of technical operating criteria to minimize interference. In Public Law 100-223, the FAA was given explicit authority to evaluate proposed radio transmission facilities for interference to aviation systems, and to issue determinations as to whether these proposals would constitute a hazard to the safety of air travel. In adopting the law, Congress recognized the need for coordination in the technical analysis conducted, and directed the FAA to cooperate with the FCC in the implementation of the regulations. (It is important to point out that the FAA determinations are <u>advisory</u> only; the FCC may permit construction if it finds the proposal in the public interest.)

Since the enactment of PL 100-223, the FAA has moved quickly to conduct evaluations of FM station proposals. Initially, those studies were performed by hand, using the Venn diagram techniques. Later, a computerized evaluation model was implemented, resulting in rejection of the vast majority of FM construction proposals. In response to complaints, the FAA and its contractors refined the model, leading to the version in use today.

While the model and its development are central to the current debate (and to this paper), the larger issue is one of compatibility. To broadcast engineers, there are spectrum allocation and usage problems not only between the FM band and the aviation band, but also between land mobile and UHF television, between educational FM and VHF television, between high power broadcast facilities and federal land mobile operations, and, of course, the infamous broadcast remote pickup band and NASA's shuttle operating frequencies.

One possible move towards a solution would be a <u>national</u> set of priorities for spectrum usage. Such a move, political in nature, is not foreseen to occur anytime in the near future.

A brief review of the operation of the FAA EMI model is in order.

There are three primary interference concerns to the FAA. The first is adjacent channel interference (splatter), the second is overload (blanketing) and the third is intermodulation. There is some relationship between overload and intermodulation; if a receiver is close to or in overload, there is a greater likelihood of intermodulation interference being created.

For airborne receivers, such interference is a concern when the aircraft is operating in an area where the signals of the desired or protected navigation aid is expected to be received. Ordinarily, aircraft travel along specific, narrow paths when making use of a navigation aid. A much wider area, called a Frequency Protected Service Volume (FPSV), is the area within which the FAA protects the navigation aid from interference. The FPSV assumes different shapes depending on the function of the navigation aid. Similarly, there are protected coverage areas for air to ground voice and data communication channels.

Immediately above the FM broadcast band, from 108 to 118 MHz, is a series of frequencies used by the FAA to provide navigation information to aircraft. These navigation aids were the first priority for FAA protection and analysis computer programs.

The current FAA interference model establishes a point grid over a vertical or horizontal "slice" of the service volume corresponding to the navigation aid being protected. At each of these grid points, the free space signal strength is computed for the proponent station. All other stations which could (when combined with the proponent) create an intermodulation product on or near the navigation frequency are similarly computed. The signal strengths for each pertinent station are summed at each point, with the sum compared to a computed interference threshold value. This threshold value is determined by the frequency relationship of the FM and navigation signals. If the threshold is exceeded, the model indicates that interference is caused at that point. The process is then repeated for all pertinent station combinations.

Receiver Testing

In developing the Airspace technical model, the FAA contracted with Ohio University to conduct receiver testing, software modeling and validation. Additional technical data was drawn from Canadian tests and FAA tests.

The Ohio University testing used the "black box" approach. In this type of testing, the desired and undesired signals are applied to the receiver, and the output response is measured.

For receiver testing, a simulated localizer (or navigation) signal, at the minimum protected signal level of -86 dBm, was introduced at the receiver antenna terminals, along with FM signals at appropriate frequencies that intermodulation interference would be expected to occur. The FM signal levels were varied and the output indicator was observed. Figure 1 is a basic block diagram of the receiver test bed.

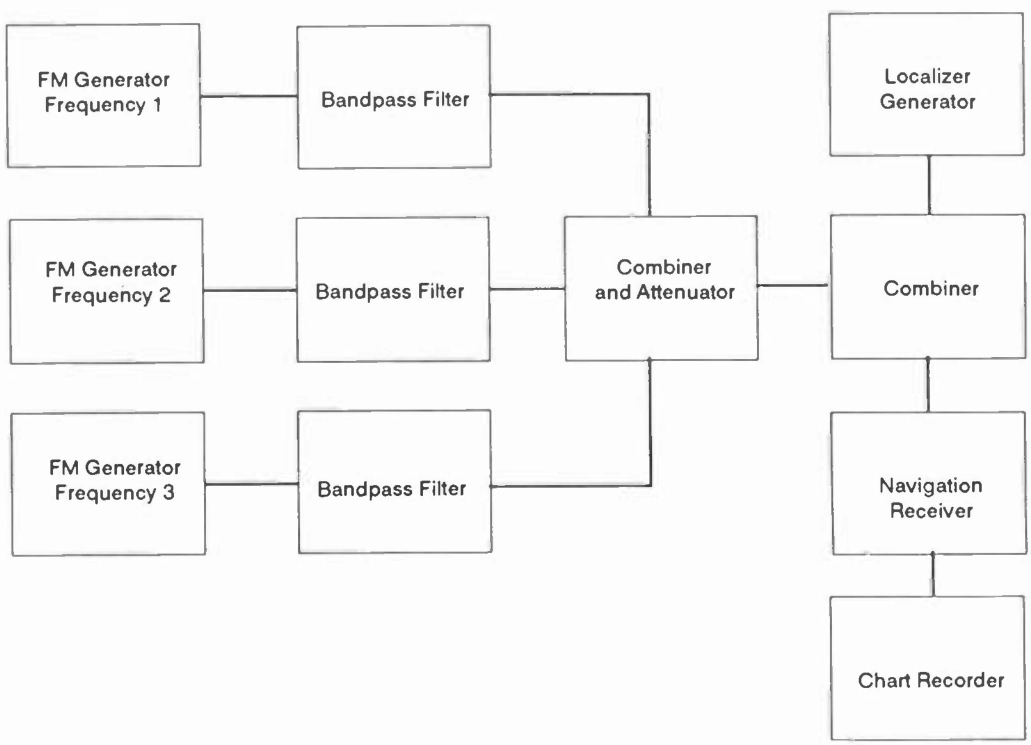
In navigation receivers, the output is a panel indicator which tells the pilot which direction to fly to remain on course; for test purposes, the effects of the FM signals were observed on this panel meter (the course deviation indicator, or CDI). The FM signals were advanced equally until the maximum allowable perturbation in the CDI was observed. The FM signal levels were recorded and later analyzed for the entire body of receivers tested.

In the initial tests, all FM signals were maintained at equal signal levels. Likewise, the navigation signal level was maintained at the lowest signal level permitted at any point in the service volume. And only a limited number of FM frequency combinations were tested.

In the "real world", FM signals vary substantially. Unless the stations are operating from a common site, with identical antenna systems, it is unlikely that any two stations will be of equal signal strength at any given point. If one signal is sufficiently weak, it is unlikely that it will be a contributing factor to interference. The signal strength of a navigation aid varies, too, depending on the aircraft location.

Subsequent tests by Ohio University indicated that a minimum FM signal level of approximately -86 dBm was required to created the same intermodulation effect observed for equal signal levels.

Black box testing is acceptable to determine the effects of outside influences on a constructed system, the results are not useful for identifying the reasons for inadequate performance, nor is sufficient data gathered to find improvements.



Test Arrangement for Intermodulation Performance

Figure 1

None of the FAA sponsored testing was designed to obtain data on receiver filters or otherwise improve receiver performance.

There are indications from the Ohio University researchers that their research shows some of the more expensive ("better") receivers do not reject interference as well as some of the less expensive ones. A plausible explanation is that these receivers are more sensitive; as the front end sensitivity improves, the intermodulation and overload performance decreases. An interesting experiment would be to determine how close the receivers are to overload when intermodulation interference occurs. In today's crowded environment, the utility of this improved receiver sensitivity must be questioned.

The Ohio University researchers developed mathematical equations to linearly fit the frequency data points to a straight line curve. This curve, the log frequency product, is independent of the signal strength of each individual station. It is claimed by the researchers and the FAA technical staff that this is an accurate representation of what actually occurs in a receiver. The data seen by the author to date is insufficient to support that conclusion. The latest version of the model has revised mathematical relationships, based on multiple straight line fits, which are claimed to more accurately represent receiver performance.

It is important to note at this juncture that all receiver tests have been conducted under FAA contract with Ohio University. The situation virtually begs for an independent investigation and verification of these tests. Yet no one has stepped forward to finance or conduct independent tests.

Validation

In its current form, the model calculates the free space signal strength from all pertinent facilities without regard to terrain obstructions, fresnel clearances, or the navigation aid radiation pattern. The model can make use of digitized elevation patterns for various FM antennas; however, the distribution version of the program does not allow for calculations using horizontal plane FM antenna radiation patterns.

Free space attenuation generally applies only when fresnel zone clearance is obtained above terrain and obstacles. While the NGDC terrain data is commonly available in forms suitable for use on a desktop computer, such data is not incorporated in the FAA analysis.

The agency and its contractor claim to have verified the model in flight tests. These validations used actual broadcast signals and simulated localizer signals. The outcome, not surprisingly in light of use of a simulated localizer signal, showed correlation to the predicted interference.

The troubling aspect of this validation is that "fringe" cases appear to have been overlooked. The descriptions of such flight testing indicate that only stronger signals were used. In addition, no consideration is made for the variation which will occur in the actual localizer signal as that aircraft approaches the runway (or navigation aid). A review of the literature makes no accounting for possible ground reflections of either the FM or navaid signals.

As with receiver testing, there has been no attempt to independently verify the results. Some limited field testing may be undertaken by at least one broadcast licensee within 1991, but no concerted effort has been made to expend the necessary funds to complete comprehensive independent verification of the model.

Improvements to Date

In the course of discussions between the FAA, FCC and some outside participants, Ohio University reviewed and improved the model. These improvements have resulted in a substantial decrease in the number of "hazard determinations" for proposed FM facilities. The more significant changes will be described below.

Additional testing apparently revealed that there was a minimum FM signal level for intermodulation contribution. Version 4.0 of the model incorporates a limitation of -86 dBm for the weakest contributor to an intermodulation product. This serves to reduce the impact of "far away" proposals.

Version 4 also has been modified so that intermodulation interference is not predicted at any location where overload interference is computed to occur. Additionally, "near overload" points are not omitted, leading to some ambiguity in the results. This too has reduced the interference findings, particularly in those areas where a large, multiple use FM site is located near the service volume.

The most substantial improvements deal with the developed thresholds for 2 and 3 frequency products. Additional receiver measurements were made, and the threshold equations were modified to increase the threshold for higher frequency products (i.e. FM stations at the high end of the band).

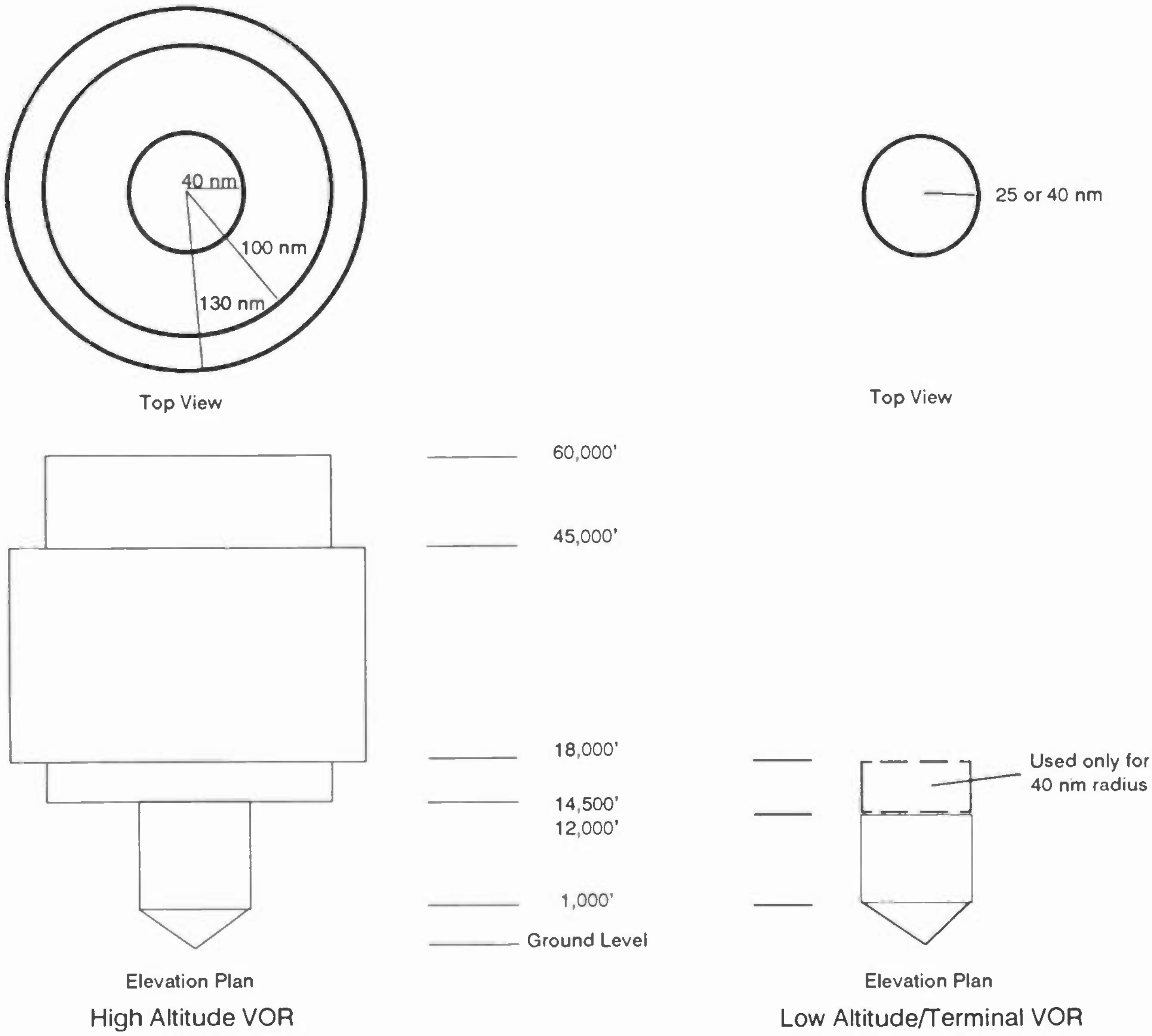
Other more minor modifications were made, including limitations for "beyond the horizon" FM signals. These changes are not as significant as those listed above.

Remaining Problems

The larger, more global problems remain, however. The most significant concern validation, margins of error, and exceptions for cases where a facility is proven not to cause interference. At the minimum, independent testing and verification are required, along with an examination of receiver improvements.

Enhanced safety with minimum impact on private facilities can be established, as Canadian authorities have done. In the Canadian system, the actual navigation aid is tested following construction of a nearby FM station. If the tests show no out-of-tolerance conditions, the FM station is certified for operation. This would be a reasonable compromise between the need for aviation safety and other radio service. Most broadcasters are likely to be willing to pay for such testing if their desired sites can be used. The FAA, however, is reluctant to implement such a process.

Other areas requiring examination include the improvement which can be obtained through use of receiver filters, use of median test results and a final error factor in developing the criteria (as opposed to use of the worst three receivers at each test frequency), the impact of aircraft antenna directionality in "real world" cases (where the FM signals arrive from different directions), the use of terrain data and fresnel attenuation, and the impact of variations in the navigation aid signal levels.



VOR Protected Service Volumes

Figure 2

Future Considerations

At some point in the near future, it is anticipated that the FAA will begin using this model for analysis of VHF Omni Range (VOR) type navigation aids. These navigation aids cover a much larger FPSV, as depicted in Figure 2. Within the hundreds of square miles of area covered by the FPSV, aircraft typically use only a small fraction. This has the potential for eliminating any change whatsoever in FM station facilities.

The FAA staff has also made it clear that they intend, in the future, to analyze FM, television, and AM stations for possible effects on an array of aviation related facilities. These may include navigation markers, communications channels, LORAN, and microwave links.

Political Consideration

No one wants to be against aviation safety. But there are ways of approaching technical issues so that safety is enhanced, while permitting operation of broadcast facilities.

Both the broadcast and aviation industries have a vested interest in their own facilities. And each will not hesitate to take advantage of politics to advance their cause. Each Federal agency responsible for regulating these interests responds to its constituents. And politicians respond to the safety concerns over the interests of thousands.

Technical issues are best solved by scientific means. A concerted effort is needed by the broadcast (and private communications) industry to conduct research and find those technical solutions. The industry should also make an effort to seek out contacts within the aviation industry to develop consensus solutions. As with many other matters, a politician presented with consensus opinion from industry is likely to develop policies based on that consensus.

Broadcast engineers who think that such politics are unnecessary may find themselves sadly mistaken.

Conclusions

Although some progress has been made towards refinement of the FAA's interference modeling program, significant work is still required. Some investment in time and money is required on the part of broadcasters to provide new research into the compatibility problems. Lacking investment, it is likely that the FAA and aviation interests will continue to see the broadcasters as being unconcerned.

The broadcast industry must learn from the past, and begin the political and negotiation process now, to ensure that technically accurate standards are adopted.

References

FAA Spectrum Management Regulations and Procedures Manual

Technical Reference Manual to the Airspace Analysis Technical Model

Reference material for FAA/FCC/Ohio University meeting on the development of the Airspace Model (Spring 1990)

DIGITAL AUDIO BROADCASTING—SYSTEM CONCEPTS

Tuesday, April 16

Moderator:
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BROADCASTING SYSTEMS CONCEPTS FOR DIGITAL SOUND
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SUBJECTIVE ASSESSMENTS ON LOW BIT-RATE AUDIO CODECS
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*Paper not available at the time of publication.

BROADCASTING SYSTEMS CONCEPTS FOR DIGITAL SOUND

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Abstract- *A new generation of sound broadcasting systems is needed to: a) improve the audio quality delivered to the public to render it competitive with Compact Disc and DAT, and b) to provide a high level of service reliability to vehicular and portable receivers which are now becoming the major means of receiving radio services. Technical studies were carried out in Canada to determine the most appropriate systems concepts for the delivery of digital sound broadcasting to the public, considering the latest developments in the fields of audio source coding, channel coding and digital modulation. Several systems concepts for both terrestrial and satellite broadcasting are discussed. A new service/system concept is presented; the mixed terrestrial/satellite approach which enables terrestrial broadcasting as well as satellite broadcasting to operate in the same frequency band so that both services can be received by the same consumer receiver. Optimum frequency bands and service bandwidth considerations are also discussed in the light of the up-coming ITU Conference, WARC-92, which is to allocate a frequency band for this new service.*

1. INTRODUCTION

Since the advent of radio broadcasting at the beginning of this century, some events took place that somewhat changed the original nature of this service. Since the start of commercial radio broadcasting in North-America in 1920 with the launch of stations XWA in Montreal and KDKA in Pittsburg, both the service needs and the technology have evolved considerably. Following the simple 'galena' receivers, tube receivers provided somewhat better sensitivity and sound quality. Later, transistor receivers allowed size reduction and improvement in reliability. Today, integrated circuits allowed palm size receivers and more complex features such as PLL and digital tuning.

In the 50's, the advent of FM radio broadcasting started to change the environment with the provision of stereophony and improved audio quality. The advent of car radios had no apparent effect at the time but has now a far reaching impact on the type of service that needs to be delivered. Although AM and FM broadcasting had been intended and planned only for fixed reception, with time, vehicular and portable reception have developed to a point where a large fraction of the audience now receives radio programs in these modes. Further, the public expects better audio quality from their receivers as indicated by the tendency to incorporate hi-fi systems in cars and by the increased penetration of 'Walkman' type receivers capable of providing very high sound quality. In addition, since a sizeable audience is travelling outside the main city centres, there is an increasing need to provide continuity of service outside cities, along highways and in rural areas.

New technologies are being developed that will be able to respond to these changing service and coverage requirements by: a) providing better audio quality, b) be especially designed for vehicular and portable reception and c) be available from terrestrial and satellite transmitters. As a consequence, application of these technologies can drastically improve the service quality and reliability and may lead to a major change to the radio broadcasting infrastructure as we know it today.

2. SERVICE OBJECTIVES

The expectation of the listeners for high quality sound reproduction has increased considerably in the last few years. The enormous advances in the performance of domestic 'hi-fi' equipment, culminating in the extensive use of compact discs, have conditioned many people to expect sound quality greater than that available from current radio broadcasting. Most people in urban areas now expect high quality stereo sound, even from portable and vehicular (car) radios. Even in more remote areas, similar expectations often exist, owing to the wide availability of good quality cassette recorders and compact disc players. There is therefore a need for a new sound broadcasting service which could offer high quality sound comparable to compact disc audio quality. In such a case, the level of quality should be established for the most demanding listening condition (i.e. home hi-fi listening environment), while comfortable listening in noisier environments such as in a car would be achieved, according to the taste of the listener, by a local control of the dynamic range in the receiver rather than by the current practice where companding and limiting are performed at the transmitter.

There is also a need for such a new sound broadcasting service to provide high-reliability service to fixed, vehicular and portable low-cost receivers equipped with simple omni-directional antennas. High quality reception should be secured over any kind of environment, even in dense urban areas and mountainous regions, both characterized by shadowing and multipath conditions. The new broadcasting service should be able to accommodate the coverage, programming and commercial requirements of local radio stations as well as the regional and national requirements of public broadcasters over wider coverage areas. For these reasons, terrestrial and satellite systems should be considered. The service should be planned in such a way that the consumer does not have to buy two different receivers (or add-on converters) to receive both terrestrial and satellite services.

Since most of the costs of such a new service would be born by the consumer through his purchase of a new receiver, it is critical to the success of this new service to make sure that the cost of the receiver is kept low (even if its complexity may be very high) through volume production secured by the early establishment of a transmission standard at the regional and preferably at the international level. Commonality of current sound broadcast formats around the world is a given for international travellers carrying their portable radio receivers abroad; it would be a step in the wrong direction if different regional standards were established.

The new service should also be spectrum efficient in terms of bit/hertz or stereophonic programs per MHz, and also in terms of frequency re-use within a given frequency allocation. It should also be flexible to adapt to new trends in broadcasting and programming, such as narrowcasting and additional data services.

3. NEW TECHNOLOGIES

Sound broadcasting has been based, so far, on analog modulations (e.g. AM,FM) but with the increased use of digital technology throughout the sound reproduction chain (i.e. from the studio to the consumer with the CD's), digital technologies are poised to over-take the last analog link in the chain: broadcasting. The development of very effective digital audio compression schemes, channel coding and modulation algorithms and the likely cost reduction of digital technology through volume production have now made such technology accessible to sound broadcasting.

Advanced source coding schemes which can reduce the bit rate of a CD quality monophonic channel to about 120 kbit/s have been developed. The International Standards Organisation (ISO) is in the process of establishing a standard for recorded media based on these coding schemes. This means that this technology will become available at low cost and could be used in a new generation of radio receivers.

Advanced digital channel coding and modulation techniques have been developed to improve the service reliability to portable and vehicular receivers, and to reduce the required transmit power without any sizeable increase in spectrum requirement. This is done through the multiplexing of a number of stereo programs (typically 12) and the time and frequency interleaving of this information in a relatively wide channel (e.g. 3.5 MHz). For the first time, these advanced techniques can use multipath echoes constructively, resulting in near-perfect reception, even under severe multipath conditions.

One system has been developed in Europe under the Eureka-147 project and has been demonstrated in Europe and in Canada and is being demonstrated in Las Vegas during this NAB 1991 Convention[1]. In this system, the information to be transmitted is split into a large number of elementary narrow-band sub-channels which are modulated (DQPSK) in a way to maximize spectrum use (orthogonal frequency-division multiplex : OFDM). Information is distributed among these sub-channels and also time interleaved to randomize the effect of static and dynamic selective fading. A guard interval is inserted between transmitted symbols to remove inter-symbol interference caused by echoes. A Fast Fourier Transform is used to process the large number of carriers in both the modulation and demodulation sub-systems. Convolutional coding (C-OFDM) is used in conjunction with a Viterbi maximum-likelihood decoding algorithm to correct the randomized errors.

Another system is under development in the United States[2]. This system uses dynamic frequency assignment over 12 sub-channels (dynamic SCPC) and time interleaving to counter static and dynamic frequency selective fading. Adaptive equalisation is used in each sub-channel to remove inter-symbol interference caused by multipath. Other systems have been proposed for digital sound broadcasting but those are not applicable to the development of the systems concepts, as given in the following section, since they do not make constructive use of echoes created by multipath.

4. BROADCASTING SYSTEMS CONCEPTS

A number of systems options are possible for the broadcasting of digital sound to the public[3]. At the service planning stage, and even more so at the early frequency allocation stage, all these options should be clearly defined and included so that maximum flexibility is ensured for the service implementation phase. The extent of implementation of these various options will then depend mainly on the market demand. Both terrestrial and satellite transmissions are considered feasible and desirable for this new service and the consumer receiver should have access to both. Various systems options are described and discussed below.

4.1 Terrestrial broadcasting (sound BS)

In the future, terrestrial broadcasting will continue to address the same service needs, that is local broadcasting typically covering a city. In digital sound broadcasting, the need to multiplex a number of stereo programs to mitigate the propagation problems will confirm the current tendency to aggregate a number of FM broadcasters on the same transmission facilities. The definition of coverage would be somewhat different than in the case of analog transmission because of the rather abrupt failure mode of digital modulation and the resulting muting of the signal compared to the 'still intelligible' nature of the reception during signal fades in the analog case. The fact that noise and interference have the same effect on the received signal will lead to interesting trade-offs in planning the service. Since the transmitted signal will be in digital form, addition of extra features such as ancillary data services for display of station identification, program information, traffic information, etc., will be easy.

Traditional methods of planning for terrestrial broadcasting have used an availability criterion which requires 50% of the locations within the coverage area to meet the quality objective for at least 50% of the time. It is likely that for digital sound broadcasting with its more rapid transition to outage, this criterion will not be sufficient. Such criterion also needs to be related to vehicular reception. The perception of quality, which the occupants of a vehicle will have is more related to the time for which the service is disturbed than to the percentage of locations.

4.2 Terrestrial broadcasting with gap-fillers and coverage extenders (improved BS)

The coverage provided by a terrestrial transmitter can be improved by the use of repeaters. This is currently done by re-broadcast stations using different frequencies. These repeaters need therefore to be considered in frequency allocation plans and consume spectrum that cannot be used by other nearby stations. With the advanced digital modulation schemes described above, these repeaters can re-use the frequency of the main transmitter, therefore not requiring more bandwidth to extend the coverage of the station. These repeaters will in fact generate active echoes which will be constructively used in the receivers, as if they were multipath echoes. Certain basic rules need to be followed, however, to make sure that the active echoes produced by these repeaters will be received within the time window where the advanced demodulation schemes can use them constructively, otherwise, they would become destructive. This translates into a required directivity of the re-transmit antenna away from the main broadcast station as a function of distance of the repeater from this main station. The power of these repeaters can be very low, in the order of a few watts, depending on the size of the area to be covered.

A number of these on-channel repeaters could be used around a main broadcast station to either improve the coverage by filling the shadowed areas produced by large buildings, tunnels, valleys, etc. (gap-fillers). For a given main station power securing the given grade of service to, say, 90% of locations and 90% of the time: F(90,90), the use of gap-fillers could improve this quality of service to F(99,99) with no increase in the main transmitter power. The main advantage is that the coverage can be improved in time by progressively adding gap-fillers.

It is well known that the last kilometres at the edge of the coverage are the most expensive to cover in terms of main transmitter power, a service extension can be realized by keeping the power of the main broadcast station constant and progressively adding repeaters extending the coverage beyond the limits reached by the main transmitter (coverage extenders). These extenders actually provide a sharper gain profile at the edge of the coverage area, hence a smaller co-channel separation distance and a higher spectrum efficiency. An even sharper discrimination profile could be produced through the use of highly directional transmitting antennas.

4.3 Satellite Broadcasting (Sound BSS)

The concept of satellite sound broadcasting has been under consideration by the CCIR for over twenty-five years and is well described in Report 955[4]. Service areas are covered by satellite beams extending over wide coverage areas. The size of these beams determines the dimension of the satellite transmit antenna. The transmission power at the satellite has to be large enough to provide adequate service to fixed, portable and vehicular receivers on the earth. A propagation margin needs to be included in the link calculation to cover signal attenuation, blockage and frequency selective fading due to multipath mainly in urban environment (e.g. 15 dB). The use of advanced modulation techniques with time and frequency interleaving which can counter selective fading will allow for a reduction in satellite transmit power (e.g. 5 dB). The car receiving antenna, which has to be omni-directional at least in the horizontal plane, provides for a rather limited gain (e.g. 5 dB at best). The advanced digital modulation techniques described in section 3 were in fact originally developed for satellite broadcasting in order to improve the service performance and reduce the required satellite power.

4.4 Sound BSS with terrestrial gap-fillers (hybrid BSS)

This hybrid BSS concept is a special application of the new advanced digital modulation schemes which make constructive use of echoes and is similar to the improved BS concept[5]. In this case, active echoes are deliberately introduced by on-channel terrestrial repeaters (gap-fillers) to fill the shadowed areas produced in the satellite beam coverage by large buildings, tunnels, valleys, etc. As long as these active echoes are within a certain delay range, the advanced digital modulation schemes can use them constructively. This translates into a restricted gap-filler transmit power (typically a few Watts) radiating over a small area size so that the satellite coverage beyond this small area is not affected by the presence of a too high level destructive active echo. The satellite coverage can therefore be improved and this can warrant a reduction in the urban propagation margin to typically the level needed to cover rural areas (e.g. 5 dB).

4.5 Terrestrial and satellite sound broadcasting using the same frequency band (mixed BS/BSS)

The concept of a 'mixed' terrestrial/satellite sound broadcasting service is based on the use of the same frequency band by both terrestrial and satellite broadcasting services. This can improve the spectrum use by allowing these two closely related broadcasting services to coordinate their service development rather than attempting sharing of the frequency band by totally unrelated services. The assumption is that the same transmission format would be used for terrestrial and satellite broadcasting and that, with the car receiving antenna which has to be near omni-directional, the receiver would capture the emissions from either terrestrial or satellite transmitters with equivalent performance.

All channels not allocated to BSS in a given service area could be used for terrestrial broadcasting in this same area subject to the usual co-channel re-use separation distance dictated by the robustness of the transmission format to co-channel interference and adjacent channel separation distances dictated by filter rejection in the receivers. Certain precautions will need to be exercised in implementing such mixed terrestrial/satellite broadcasting service where the edge of coverage of a terrestrial system is located near the edge of a co-channel satellite coverage area.

Such re-use of the satellite channels of other countries for terrestrial broadcasting will improve the spectrum usage as well as provide a flexible way by which a service could evolve from strictly local terrestrial broadcasting to mixed terrestrial/satellite service when wide area national services are added through satellite coverage. For example, commercial (local based) broadcasters would continue to use terrestrial transmitters while public (coverage oriented) broadcasters could use a combination of local stations and satellite transmitters to achieve their mandate. This could also evolve from a national (or even supra-national) service carrying national interest programming and/or specialised services over satellite to a mixed satellite/terrestrial service when it makes economic sense to add local terrestrial broadcasting. This could also be attractive for future implementation of specialty commercial services over satellite for national coverage when the receivers have reached a high level of penetration. This mixed implementation of terrestrial and satellite sound broadcasting is particularly of interest to large countries such as Canada and the United-States where the population is unevenly distributed and where local based coverage would never result in a full-country coverage.

4.6 Terrestrial and satellite sound broadcasting with gap-fillers and coverage extenders (mixed hybrid BSS/BS)

The mixed terrestrial/satellite service concept can also be augmented with the use of terrestrial on-channel gap-fillers to improve the coverage of the terrestrial as well as the satellite services. The use of these gap-fillers can also result in a reduction of the satellite power as well as a reduction of the power of the main transmitter for the terrestrial service. Coverage extenders can also be used to extend the coverage of terrestrial stations and possibly create a sharper discrimination profile towards the country using the same frequency for satellite reception, therefore allowing closer spacing between the terrestrial service and the border of the adjacent country where the frequency is used for sound BSS. Hence, this concept leads to an even more efficient and flexible use of the spectrum. It is assumed here that the transmission format will be the same for both terrestrial and satellite transmissions to allow the use of a common receiver and that this format is based on advanced modulation techniques which allow constructive use of passive as well as active echoes.

5. CHOICE OF A FREQUENCY BAND FOR DIGITAL SOUND BROADCASTING

For terrestrial broadcasting of digital sound, a range going from the current VHF (88-108 MHz) band to a few GHz is considered. For satellite broadcasting, the initial frequency range was originally identified by the CCIR as going from 0.5 GHz to 2 GHz and then extended to 3 GHz by the WARC-88. Since then, technical studies on this question have been conducted in Canada in the context of the CCIR[6].

5.1 Terrestrial broadcasting aspects

The extent of the frequency range for terrestrial broadcasting is dictated at the lower end by the presence of man-made noise which increases rapidly with a decrease in frequency[7]. This becomes even more important if very low noise figures are proposed for the front end of the new generation of receivers (e.g. 4 dB). The advantage of such front end design in terms of reducing the transmitter power would tend to be lost at the low end of the frequency range. The required transmission power would normally decrease with frequency except that the level of man-made noise, especially in urban environment, increases faster than the increase in effective receiving antenna aperture (i.e. $\lambda^2/4\pi$). This results in a net increase in the required transmission power at the lower end of the frequency range (see Table 1).

The upper end of the range is restricted by the increase of the required transmitter power caused by the reduction in the effective receiving antenna aperture which decreases by a factor of 6 dB/octave with an increase in frequency. This cannot be compensated for by an increase in antenna gain since the receiving antenna located on a vehicle needs to be omni-directional in the horizontal plane and needs to have a rather broad beam in the vertical plane leading typically to a maximum gain of 5 dB independent of the frequency. Furthermore, the required transmitter power will need to be increased to compensate for additional propagation losses caused by a reduction in amount of diffraction at higher frequencies.

5.2 Satellite broadcasting and mixed BS/BSS aspects

The extent of the frequency range for sound BSS is dictated at the lower end by the increase in the size of the satellite transmitting antenna to produce coverage beams of small sizes (e.g. 1°). The diameter of the transmitting antenna is inversely proportional to the frequency and soon becomes impractical, considering current and near future satellite busses and launchers. In the case of the mixed use of the same band by both terrestrial and satellite broadcasting, the lower frequency limit would still be dictated by the restriction on the satellite antenna size since much lower frequencies could be used for the terrestrial service. However, since this mixed BS/BSS concept allows more extended frequency re-use, satellite beam overspill into adjacent countries using the same frequency terrestrially would be less of a problem. Furthermore, the need for small satellite beams to cover small countries would be reduced by the possibility for such small countries to implement this new sound broadcasting service by terrestrial means, making more economic sense in these cases. Larger satellite beams (e.g. 1.6°) could then exist, leading to smaller satellite antenna size and thus allowing a lower frequency of operation in the case of a mixed BS/BSS implementation.

At the upper end, the limit is defined by the increase in satellite transmit power to compensate for the reduction in effective aperture of the receiving antenna (6 dB/octave). Unlike in the case of other satellite services, this cannot be compensated for by an increase in antenna gain since the receiving antenna needs to be near omni-directional as in the case of terrestrial broadcasting. Furthermore, propagation effects, such as obstruction by buildings and attenuation through tree foliage, are found to increase with frequency by a factor of about 1.5 dB/octave[8]. These losses are likely to be less severe than in the case of terrestrial broadcasting because of the higher elevation angle at reception which results in smaller attenuation and blockage. Nevertheless, since the range of possible transmission power from terrestrial stations is larger than in the case of satellites, where the cost of the spacecraft is very dependent on the required transmission power and where there is an actual limit dictated by the prime power available on current and near future satellite busses, both services are considered to constrain the practical upper frequency limit (see Table 1).

5.3 Gap-fillers and coverage extenders' aspects

In the case of the advanced digital modulation techniques described in section 3, the effectiveness in the use of gap-fillers and coverage extenders is indirectly constrained by Doppler shift produced on-board a moving vehicle. In the case of the COFDM system, the symbol duration has to be reduced with an increase in frequency because of the Doppler shift. This, in turn, reduces the guard interval and therefore reduces the allowable delay between the main signal and the active echo produced by a repeater, hence reducing the corresponding physical distance between the two signal paths[4]. For a given speed limit beyond which proper reception of digital sound broadcasting is not to be secured, the effective size of the area covered by a gap-filler reduces linearly with an

	Frequency (GHz)	0.1	0.5	0.8	1.0	1.5	2.5
BSS	BSS gap-filler coverage radius (C/I=15.5 dB) (km)[B]	50	10	6.3	5	3.3	2
BSS	Eff. receiving antenna aperture (dB)	-20	-6	-2	0	+3.5	+8
BSS	BSS propagation margin (dB)	-5	-1.5	-0.5	0	+0.9	+2
BSS	Thermal noise (NF=4 dB) and man-made noise at receiver input (dB)	+19.4	+2.9	+0.4	0	-0.3	-0.2
BSS	Beam= 1° sat. power[A] (Watts) antenna diameter (m)	13 210	16 42	28 26	46 21	118 14	439 8.4
BSS	Beam=1.6° sat. power[A] (Watts) antenna diameter (m)	32 130	40 26	72 16	116 13	298 9	1108 5
BSS	Beam=3.5° sat. power[A] (Watts) antenna diameter (m)	160 60	202 12	359 7.5	582 6	1496 4	5558 2.4
BS	Distance between omni-directional coverage extenders (km)[B]	150	30	12	15	10	6
BS	Effective antenna aperture (dB)	-18	-4	0	+2	+5.5	+10
BS	Thermal noise (NF=4 dB) and man-made noise at receiver input (dB)	+19	+2.5	0	-0.4	-0.7	-0.6
BS	ERP[A](kW) for: main transmitter cov.=33 km;E[C]=100 m cov.=50 km;E[C]=150 m cov.=64 km;E[C]=150 m	0.69 2 10	0.42 2.8 35	0.6 4 50	0.87 5.8 72	3.6 30 594	13 137 3420

[A] Power for a 12 stereophonic program multiplex, terrestrial station ERP corresponds to 46 dBµV/m at the receiving antenna at 800 MHz.

[B] For 1 dB equivalent loss caused by Doppler shift in a vehicle moving at 100 km/h.

[C] E= Effective Height Above Average Terrain of the transmit antenna.These ERP values are based on F(50,50) propagation curves for a field strength measured at 10 m above ground level. The applicability of these curves for lower receiving antenna heights and mobile reception requires further study. VHF and UHF propagation curves were used except for 1.5 GHz and 2.5 GHz where some extrapolations had to be made.

Table 1: **Variation of system parameters as a function of frequency for the COFDM system**

increase in frequency. The size of this area is related to the power of the gap-filler which is adjusted, such that when the active echo becomes destructive, its level is 15.5 dB below the power of the direct signal to avoid degradation of the received signal. The use of gap-fillers is therefore more effective, or in other words, a given shadowed area would require less gap-fillers if the operating frequency is in the lower part of the frequency range as indicated in Table 1.

Similarly, the distance between a main terrestrial broadcast station and a coverage extender is inversely proportional to the operating frequency for a given directivity pattern away from the main station. The use of coverage extenders becomes more effective at lower frequencies since the distance between these repeaters is larger and therefore a given area could be covered by less of these repeaters.

5.4 Overall best frequency range

Table 1 gives a summary of the overall situation as far as the choice of the best operating frequency, assuming the COFDM system for a mixed BS/BSS digital sound broadcasting service using gap-fillers and coverage extenders. All key parameters are included such as satellite power and antenna size, gap-filler coverage radius, terrestrial station ERP and minimum distance to the omni-directional coverage extender. Besides the well known tendency of higher frequencies to necessitate higher transmission power from both terrestrial and satellite transmitters and thus the preference for lower operating frequencies, low frequencies also have constraints such as increasing satellite antenna size and stagnation of the terrestrial transmit power because of man-made noise.

It can be found from Table 1 that the range of interest for terrestrial broadcasting is between 100 MHz and about 1.5 GHz with a relatively constant power requirement over the VHF and UHF ranges and an increase at 1.5 GHz and beyond. As could be expected, the distance between coverage extenders reduces towards the upper end of the range. For the satellite service, the range depends on the size of the area to be covered. For small beam size (e.g. 1°), the range of interest is between 1 GHz and 1.5 GHz whereas for larger beam sizes, the range drifts towards lower frequencies. In terms of a range that would be common to both services to allow a mixed operation, it is felt that the choice of a band in the range 0.8 GHz to 1.5 GHz would be appropriate.

6. SPECTRUM REQUIREMENT

The WARC-92 is to allocate a frequency band for the sound BSS and its agenda also includes considerations for complementary terrestrial services. This item of the agenda allows for a frequency allocation that can accommodate, in addition to the sound BSS, the new systems concepts described above, in particular the mixed terrestrial/satellite broadcasting service operating in the same band with the use of on-channel gap-fillers and coverage extenders.

Some preliminary studies were conducted in Canada and resulted in an estimate of the spectrum requirements for Canada and North-America[9]. All three system approaches, i.e. terrestrial, satellite and the mixed BS/BSS were analyzed. The studies were done for a carrier frequency of 1 GHz and on the basis of 12 stereo channels multiplexed in an RF channel and a required channel separation of 4 MHz.

6.1 Terrestrial Broadcasting Spectrum Requirement

An initial study was done assuming a theoretical regular coverage lattice made of circular coverage areas of equal dimension, a frequency re-use distance based on free space loss propagation and 11 dB protection ratio resulting in 17 dB carrier-to-noise ratio for a total C/(N+I) of 10 dB. This resulted in a channel re-use factor of 4. In the case of a city such as Toronto, 36 stereo channels are expected to be needed to replace all current AM and FM broadcast stations and also leave some room for growth, thus requiring three RF channels each carrying 12 stereo pairs and each with a channel separation of 4 MHz for a total of (4*3*4)= 48 MHz total spectrum requirement for terrestrial broadcasting.

A more practical exercise concentrated on southern Ontario, following the same pattern as for current television and sound broadcasting, on the assumption that the amount of spectrum required to supply this area will be adequate to supply all of Canada. The desired signal service range was estimated on the basis of the F(50,50) propagation curves for UHF-TV. The distance for the undesired signal was determined with the F(50,10) propagation curves for UHF-TV. Based on this exercise, it is estimated that Canadian terrestrial digital sound broadcasting needs alone could be met with 48 MHz of spectrum. It is estimated that with this amount of spectrum it would be possible to provide each existing AM and FM broadcasting station with one stereophonic program channel and also include an allowance for future growth.

As no data were available on U.S. requirements, the assumption was made that the U.S. terrestrial requirements in the cities close to the border would be similar to Canadian requirements. On this basis, it was estimated that an additional 25% of spectrum would be required. This would correspond to 48 MHz + 12 MHz= 60 MHz of terrestrial spectrum.

6.2 Satellite broadcasting spectrum requirement

A theoretical exercise was also made assuming a coverage of a large area based on a regular lattice of circular beams of equal size. The satellite antenna sidelobe pattern used at the RARC-83 (BS) was assumed. It was found that a frequency re-use factor of 4 is achievable with a worst case aggregate interference of 15.5 dB. This results in a C/N requirement of 12 dB for an overall C/(N+I) of 10.5 dB. The feeder-link degradation is assumed to be 0.5 dB.

A more practical exercise was made for the whole of North America. Starting from an eight beam coverage arrangement for Canada based on time zones and regional sound broadcasting needs, 4 beams were added for the continental U.S. covering the 4 time zones as well as one beam for Alaska, two beams were also included to cover the north and south parts of Mexico, and east and west regional beams were added to cover Cuba and the Caribbean. The minimum beam size was set at 1°. The frequency re-use factor for Canada only was found to be 6. It increased to 9 when the whole of North-America was included. Indications are that it is unlikely that adding the rest of Region 2 to the exercise would change the latter finding.

Assuming the provision of two RF channels, i.e. 24 stereophonic programs to each of the seventeen service areas, a minimum of 9*2*4= 72 MHz was found to be required.

6.3 Terrestrial and satellite broadcasting spectrum requirement

Assuming that the band allocated could be used by either terrestrial or satellite broadcasting, the channels unused by the satellite inside a country could all be used for terrestrial broadcasting except at the edge of the country where the terrestrial use of the same channel would create interference to the adjacent country's satellite reception.The compounded spectrum requirement for both satellite and terrestrial broadcasting can be estimated with the following rationale: setting the requirement for terrestrial broadcasting as the basis, an additional 8 MHz is needed for the satellite service in the same country, and in the special case where the most demanding coverage area is on the country's border, such as in the case of Toronto, another 8 MHz is needed since the satellite channels of the adjacent country cannot be used terrestrially at that location. The spectrum requirement for the three coverage scenarios discussed above is summarized in Table 2.

It is interesting to note from Table 2 that the spectrum requirement for the mixed terrestrial/satellite service is just slightly larger than the requirement for terrestrial broadcasting alone, a result that highlights the spectrum efficiency achievable by using the mixed terrestrial/satellite implementation concept. This approach not only results in high spectrum utilization efficiency but will also prove extremely convenient for the consumer/listener to have access to both satellite and terrestrial services with the same receiver.

Coverage scenario	Uniform circular beams	Canada	North America
Spectrum for sound BSS (MHz)	32	48	72
Spectrum for sound BS (MHz)	48	48	60
BSS channel for same country (MHz)	8	8	8
BSS channel for adjacent country (MHz)	--	--	8
Total spectrum requirement (MHz)	56	56	76

Table 2: **Spectrum requirement for the satellite/terrestrial mixed digital sound broadcasting service**

7. SUMMARY

This paper has examined the merits of a number of systems concepts for the digital sound broadcasting service. Advanced digital modulation can be used to provide a reliable service to the portable and vehicular receivers in addition to the conventional fixed receiving installations for which sound broadcasting has been planned up to now. This extension of the service to a fast growing population of vehicular receivers requires the use of modulation schemes that make constructive use of echoes produced in a dynamically changing multipath environment. The use of such schemes also allows proper operation in presence of active echoes produced by on-channel repeaters used for improving or extending the coverage.

With advanced digital technology, it is now possible to provide an integrated digital sound broadcasting service from both terrestrial and satellite transmitters. Since the receiving antenna is near omni-directional, different program transmissions from either terrestrial or satellite stations could be captured by the same receiver if they both operate in the same frequency band. This is called a mixed terrestrial/satellite type of service and allows provision of both local and regional/national service to the same receiver.

The frequency range 100 MHz to 2.5 GHz was examined for this service. It was found that in the case of terrestrial broadcasting, the required transmitter power increases slowly up to 1 GHz and more rapidly beyond 1.5 GHz to quickly become impractical. For satellite broadcasting, there is a narrow frequency range over which the systems are practical. This range is constrained at the low end by the increasing size of the transmit antenna and at the high end by the increasing satellite power. It is found that a mixed terrestrial/satellite service would best operate in the range 0.8 GHz to 1.5 GHz. At the limit, such mixed service could operate in the upper UHF-TV band if larger satellite beams could be used but this would result in a significant impact on the implementation of advanced television services and would limit the growth of NTSC services.

Digital sound broadcasting is seen by many as the next generation of radio services. Some also feel that this new service should, in time, completely replace the current AM and FM services. With this in mind, some preliminary exercises were made to evaluate the spectrum requirement for such a new service. It was found that 60 MHz would be required to accommodate all current AM and FM stations in the area of Toronto including provision for normal growth and accommodation of U.S. stations on the border. This value would seem to be representative of a maximum spectrum requirement for terrestrial digital sound broadcasting in North-America. Preliminary exercises were also done for satellite broadcasting and it was found that 72 MHz is needed to provide two RF channels (24 stereo channels) to all service areas in North-America. It was also found that in order to fully cover the terrestrial as well as the satellite service needs through a mixed terrestrial/satellite approach, the total spectrum requirement would be 76 MHz which is practically the same as the requirement for satellite service only or terrestrial service only. This confirms the high spectrum efficiency achieved by such mixed type of operation.

Presently, there is no frequency band for broadcasting digital sound to the consumer and it will be difficult to find one, as the frequency range considered for this service is extensively used. Considerable effort from the broadcasting community will therefore be needed to obtain such a frequency allocation in order to be able to compete with other means of delivering sound programming to the public and to cover the needs of the broadcasting service for the coming decades. The upcoming WARC-92 provides an opportunity to resolve this dilemma. The agenda of this Conference addresses the allocation of spectrum for a broadcasting satellite service and a complementary terrestrial broadcasting service in the same band. A WARC-92 frequency allocation could therefore accommodate the implementation of digital sound broadcasting according to the systems concepts described in this paper, in particular, the mixed satellite/terrestrial broadcasting approach using the same frequency band augmented by gap-fillers and coverage extenders.

Proper service definition and quality criteria which correspond to the public expectation and low cost receivers achieved through volume production resulting from a broadly based standardization are critical to the success of such new service. If the proper context is set through appropriate spectrum allocation and adoption of emission standards, as well as through suitable policies to protect existing broadcasting services, digital sound broadcasting could be a very successful service and provide for a flexible implementation through the various systems concepts described in this paper.

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COMMUNICATIONS SYSTEMS ENGINEERING FOR DIGITAL AUDIO BROADCAST

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Abstract - Digital Audio Broadcast (DAB) systems can take advantage of many communications theories and commercially available technologies already developed and practiced over the last thirty years. Advances in modulation, coding, and compression technology, for example, will allow successful implementation of spectrally efficient DAB systems and ensure quality audio broadcast service at affordable cost.

In this paper, various aspects of digital communications systems engineering for DAB application are introduced. The modulation, coding and compression performance trade-off, link budgets, and channel characteristics of a proposed spectrum band are discussed. To help understand the systems engineering practice, an example of DAB system suitable for utilizing the existing FM radio spectrum band while offering digital quality audio comparable to that of compact disc is demonstrated.

INTRODUCTION

In this paper, an example land-based Digital Audio Broadcast (DAB) system is considered. The design describes the transmitter and receiver functional blocks and discuss channel impairments. The system is designed to operate in 200 KHz channel bandwidth for FDMA application in pursuit of possible channel compatibility with present analog FM transmission systems. The system topology is land-based and addresses the radio communication from a local DAB radio station to local home and mobile receivers. The system structure is typical to that of standard digital communications radio equipment. Some of the key components are compression, coding, modulation, demodulation, decoding, and decompression. The frequency band for this example is assumed to be the current FM radio bands, however, the example design can be easily translated to another frequency band.

Satellite-based DAB can also bring digital stereo audio service for wide area coverage. Such national coverage can create a new market for "super-station" broadcasters: advertisement and musical service in "compact disc" quality for very large audiences. Such systems are analogous to those that already exist in military and commercial data link markets. Thus this paper treats those systems design aspects unique to a land-based DAB system.

TRANSMITTER

Digital Studio, Quantization, and sampling

Figure 1 shows the functional architecture of a typical digital audio broadcast station. The audio source is converted into analog electric signal and then fed into an analog processor. Analog processor performs distortion equalization for the audio transducer which converts audio waveform into an electric signal. Anti-aliasing filtering is performed by the analog processor to remove undesired signal outside the frequency band of interests. Automatic power level adjustment (AGC) is used to bring the signal to desired level, and etc. The processed audio analog signal is then sampled by an analog-to-digital conversion device (ADC) which converts analog signal into digital representation. The number of quantization level and the step size in the ADC determines quantization noise and signal dynamic range that can be achieved. The more quantization levels in the ADC, the less the quantization noise and more dynamic range can be achieved. This can only be accomplished at high complexity of the ADC design. The design of quantizer impacts the processing of the overall audio system since it determines the complexity of the processor for handling the digital signal and the required data rate for the digital audio signal.

Nonuniform quantization can be used to achieve greater signal dynamic range by allowing quantization noise to be increased with signal amplitude. Better audio quality can usually be achieved by nonuniform quantization than with uniform quantization. However, in terms of overall system processing complexity and implementation, uniform quantization offers more advantages than the nonuniform quantization and is predominantly used in digital audio studio.

The sampling rate of the ADC determines the frequency range of interest. According to sampling theory, sampling rate which is twice the highest frequency component is sufficient to fully reproduce the original audio waveform using discrete samples of the waveform. Digital audio used in compact disc uses a 16 bit ADC and 44.1 Khz sampling rate and achieves 90 dB signal-to-noise ratio and a dynamic range of 90 dB. This covers the range of audio frequency perceivable and the highest signal-to-noise ratio which is discernable by human ear.

In the example design, the single channel is converted to digital data by sampling the analog

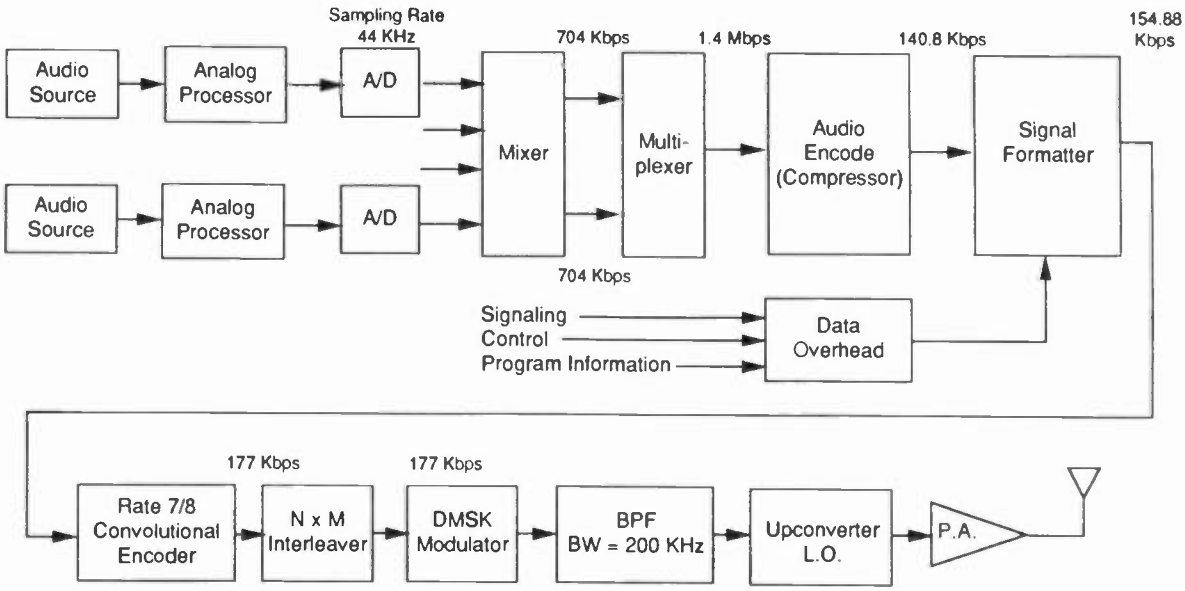


Figure 1 Functional Architecture of a Typical Digital Audio Broadcast Station

signal at 44.1 KHz using 16 bit ADC. This makes the data rate at the output of the ADC to be 705 Kbps per channel and require substantially larger bandwidth than that required by the conventional FM radio signal.

<u>Digital Multiplexer and Mixer</u>

In an analog FM radio system, audio sound track is typically produced by using several audio recording channels for a single or multiple audio sources and combined (mixing) together to produce two channel (left and right channels) stereo audio. These two channels are summed and subtracted together before it is broadcast through the air.

However, in digital radio system, the digital audio signal need not be combined. Rather, the left and right digital signals are multiplexed in time sequence, such as shown in Figure 2, such that the extraction of the left and the right signals are easily performed at the demodulator. Thus the two channel needed for stereo sound can multiplex the left and right channel data streams into a combined serial data stream.

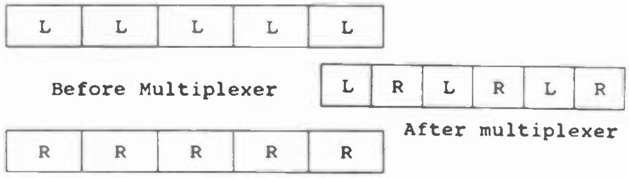


Figure 2 Multiplexing the Left and the Right Signal

<u>Audio Coding Algorithms</u>

In order to minimize the bandwidth requirements, the audio waveform encoding is often used to reduce the data rate of the digital audio signal. Audio waveform coding techniques is one of the most actively researched areas in the audio signal processing field. The major objective of audio waveform coding is to achieve high quality (less perceivable noise and distortion) audio at the lowest possible data rate. The requirements of high quality audio and a low data rate are the most important tradeoff that must be determined for DAB radio.

As an example of this tradeoff, Compact Disc-Interactive (CD-I) format developed by Phillips Corp. has introduced a multimedia extension of CD-ROM format for simultaneous combination of audio, video, graphics, text and data services that provides five different levels of audio quality. The CD quality stereo audio are offered at 704 Kbps per channel using PCM waveform coding and 16 bit quantization at 44.1 KHz sampling rate. A Hi-Fi Audio with sound quality comparable to vinyl LP records uses 8 bit ADPCM with 37.8 kHz sampling rate. Mid-Fi Audio with sound quality comparable to conventional FM broadcast uses 4 bit ADPCM with 37.8 kHz sampling rate. Speech Audio with sound quality comparable to conventional AM broadcast uses 4 bit ADPCM with 18.9 kHz sampling rate. Information Audio which provides phonetic sound information is offered at a much sampling rate than all the previous ones. The CD-I represents the audio quality and data rate tradeoff at various complexity.

Much higher performance however can be achieved with other techniques at the price of higher complexity. Examples of these are delta modulation (DM), continuous variable slope delta modulation (CVSD), transform coding and subband coding.

In addition to the data rate consideration, other system issues such as robustness of the coding algorithm to the random and bursty channel transmission errors, the ease of synchronization between encoder and decoder and complexity and the delay associated with the encoding algorithm should be taken into account in the design of the waveform coder. The Motion Picture Experts Group (MPEG) is currently evaluating the performance of 14 algorithms to adopt as the audio compression standard. The proposed algorithms fall within two broad classes: those that employ transform coding and those that employ sub-band coding. MPEG seeks to represent Hi-Fi audio signal at 128 kbps, [6]. More recent commercial development has claimed to achieved a reduction of 13 to 1 in data rate over the standard bit rate (at 2 * 55 kbps for stereo sound) while preserving nearly transparent audio quality, [7].

Data Formatter

The compressed audio data is then fed into a data formatter which combines other overhead and signaling information. These overhead data provides frames header for the purpose of synchronizing the audio data frame and signaling data for signal processing functions and information. These overhead data can provide improved service over the conventional AM and FM broadcasts such as program information, receiver control, time dissemination, copy protection, and other functions. The signal formatter can also perform encryption on the audio data for the charge service broadcast stations such as banking, brokerage firms and other information services. It is assumed that 10 % overhead is reserved for this purpose and thus brings the data rate up to 154.8 Kbps.

Error Correcting Code

In order to recover from channel induced transmission error, the data is encoded using a forward error correction code. The error correction coder inserts redundancy into data streams which are then used in the receiver decoder to correct certain amount of errors in the received data stream. Among the coding algorithms, convolutional codes with maximum likelihood decoding, Reed-Solomon codes, and Trellis codes are among the popular choices which offer good compromise between performance and complexity.

The convolutional coder with maximum likelihood soft decision decoding is one of the widely used techniques which provides strong error correcting capabilities with limited amount of implementation complexity. Commercially available low cost, compact hardware and software encoders and decoders can be used for this purpose.

Some of the coder/decoders are available in the form of Application Specific Integrated Circuit (ASIC) which are suitable for mass production and manufacturing at low cost. Since coding introduces redundancy in the data stream, the data rate is increased by a factor equal to the inverse of the code rate. A punctured convolutional code is a high rate coding technique among the class of convolutional codes and is accomplished by periodically puncturing (removing) some of the data symbols at the output of the regular convolutional encoder. A rate 7/8 punctured convolutional code is used in this design example and provide 3 to 4 dB coding gain with moderate complexity. The data rate is thus increased to 177 Kbps.

Interleaver

Since broadcast signal often suffers from fading and multipath when propagating through the channel, the received signal strength can experience deep fades over extended periods of time. This can introduce a series of burst errors and wipe out a continuous blocks of data. Interleaving is the most effective technique used to combat such bursty errors.

Basically, the interleaver reorders the data stream to be transmitted in such a way that adjacent data symbols are spread out in time. When the burst errors occur during transmission, the deinterleaver at the receiver puts the data stream into their original order which has the effect of spreading the channel errors. This will make the burst error appear to be random errors, which the error correction decoder is designed to combat.

The interleaver does not introduce any overhead into the data stream. Rather it introduces delay in the data stream to be recovered in the receiver. The interleaver size depends on the desired ability to spread out the bursty errors, the hardware complexity, and the delay constraints of the system. This depends strongly on the channel characteristics and practical hardware considerations.

Modulation Techniques

The choice of modulation technique selected depends upon a number of considerations. The ease of implementation, the spectral efficiency, and robustness against propagation impairments are the key factors in the determination of modulation schemes. Typical modulation and demodulation schemes used in a fading and multipath environment usually employs differential detection technique. This is due to the fact that the differential detection scheme is relatively insensitive to the channel induced phase changes and does not require a phase tracking loop in its implementation.

DMSK and DPSK are two bandwidth efficient schemes which offer good performance in the fading and multipath environments and can be implemented relatively easily. DMSK can be generated with the capability of about 1 bps per Hz of bandwidth of spectrum efficiency. This will allow to operate the 177 Kbps DAB signal within the 200 Khz of channel bandwidth. The modulated signal is then bandpass filtered to ensure proper spectral shape, up-converted to the proper carrier frequency, and power amplified for transmission.

TRANSMISSION CHANNEL

The transmission channel characteristics depends on the carrier frequency, terrain, weather, and time. For a given carrier frequency, the signal propagation through the urban and suburban environment suffers various degree of attenuation and multipath fading. The characteristics of the propagation media depends on a number of factors such as the center frequency foliage, building structure, humidity, air pressure and etc. It is usually determined through extensive experiments. A lot of results of the experiments are available in published material. Figure 3 shows the power fluctuation of the signal as it is transmitted through an urban mobile channel. It can be seen that the signal fades of upto 10 dB may occur as time traverses. It is also evident that the dept and duration of the fades are seemingly random. Such random fades can cause large bit errors, which will degrade the quality of sound received at the receiver unless proper design mechanisms are in place.

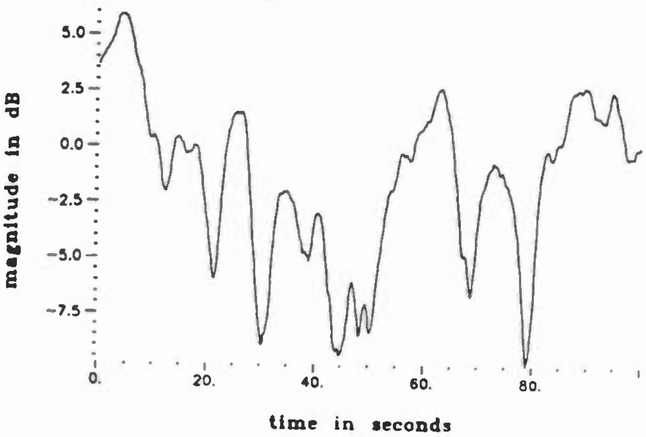


Figure 3 Channel Multipath Fading Characteristic

Channel Access Schemes

Typically DAB channels can be accessed by Frequency Division Multiple Access (FDMA), Time Division Multiple Access (TDMA), or Code Division Multiple Access (CDMA) schemes, Figure 4. FDMA is simplest of the three multiple access schemes. In FDMA system, the frequency spectrum is divided into predetermined bandwidth, and the signal is sent continuously through a specified bandwidth. Requirements for FDMA are to contain the desired signal within an assigned channel bandwidth. Any spurious signal contents spilled over by the adjacent channel may result in performance degradation for the desired channel. Such adjacent channel interferences are usually limited by channel filtering and proper choice of transmitter backoff levels and modulation formats.

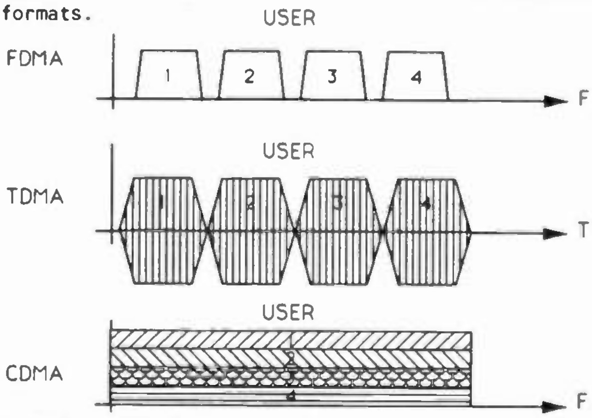


Figure 4 Channel Access Schemes

Although FDMA implementation is relatively easy, its spectrum efficiency is poorest. TDMA scheme, for example, divides the time space into tiny time segments. This allows TDMA system to allow sending small packets which consist of blocks or frames of data bits. Such time division method will allow efficient access of the channel on a time shared bases. However, implementation of TDMA system is slightly increased since frame synchronization of the data packets is required.

CDMA scheme can assign unique pseudo-noise code to each DAB radio stations. By using spread spectrum technique, CDMA can utilize the same bandwidth by all CDMA users. Thus, CDMA is most bandwidth efficient among the multiple access schemes. However, the implementation complexity is the highest due to the time and code synchronization necessary for proper operation of the CDMA scheme.

For land-based DAB system, the FDMA system is likely to be the choice since it is relatively easy to implement. Any frame and code synchronization necessary for TDMA and CDMA, respectively, are not needed for FDMA. Thus FDMA allows relatively low cost implementation with simplicity, reliability, and maintainability. However, TDMA and CDMA may still be considered later since much of the available frequency spectrum in the U.S. is very congested. Thus the complexity which is required by TDMA or CDMA may be necessary when DAB spectrum real estate is limited.

Channel Interference in FDMA System

As channel bandwidth is congested with other DAB channels, it is likely that a desired channel will experience some adjacent channel and cochannel interferences for FDMA systems. Such interferences will degrade the system bit error rate performance and may require larger transmitter power.

Adjacent channel interferences can be minimized by proper selection of transmitter power amplifier setting and signal modulation schemes. For example, Figure 5 shows a power transfer curve of a typical High Power Amplifier (HPA). The amplifier is usually power efficient at its saturation point. However, the HPA can operate quite nonlinearly at saturation and introduce performance degradation as well as adjacent channel interference through signal spectral regrowth. Figure 6 shows the spectral regrowth of DPSK and DMSK due to a saturated HPA.

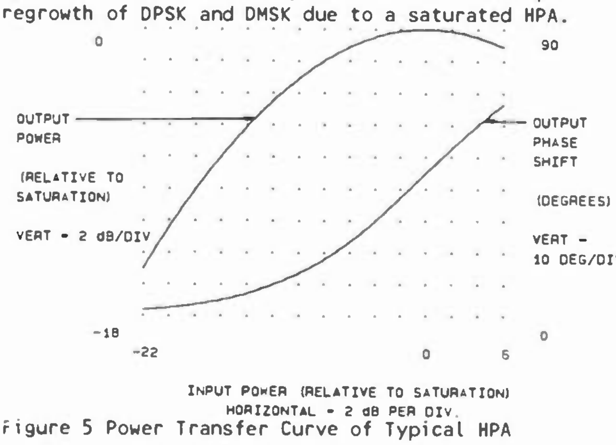
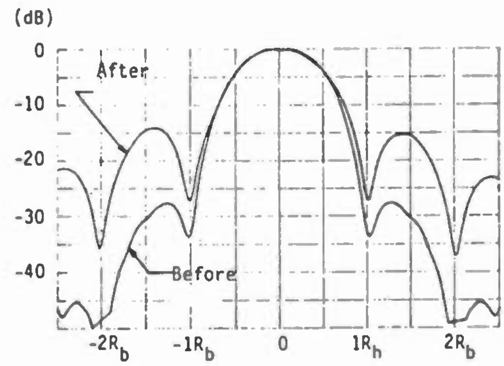
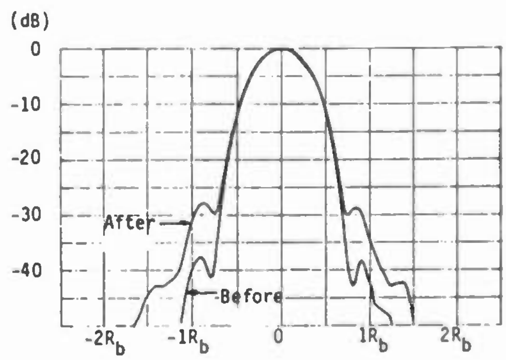


Figure 5 Power Transfer Curve of Typical HPA



a) DPSK



b) DMSK

Figure 6 Power Spectra of DPSK and DMSK through HPA

It can be seen that DMSK experiences relatively little spectral regrowth. Thus DMSK spectra can be tightly placed even while running the amplifier at saturation. DMSK will also coexist well with other constant envelope signals such as the current FM signals. Thus DMSK type constant envelope signal will not only allow the full power operation of the HPA amplifier at the transmitter, but also allow the coexistence of DAB DMSK signal with any FM signals desired for the current analog FM broadcast services.

Cochannel interference can arise due to frequency reuse exercised by a cross town radio station. Because of the ever increasing congestion in frequency spectrum, it is desirable to practice frequency reuse. Digital broadcasting system can be designed to combat such degrading channel conditions and provide a robust communications link. In order to minimize the system performance degradation due to cochannel interference, the communications systems must be selected with good frequency reuse plan, digital modulation technique, and forward error correction schemes.

RECEIVER

Receiver Front-End

Figure 7 shows the functional block of a DAB receiver. The signal is received by the antenna, filtered by a band pass filter, amplified and then multiplied by the delay version of itself and low-pass filtered. The delay and multiply operation performs the differential detection of the modulated DMSK signal. LinCom Corporation is currently developing a VLSI demodulator suitable for DAB receiver. This chip will perform the quantization, synchronization, and detection of the received signal.

The resultant baseband signal is matched filtered to increase the signal-to-noise ratio and sampled using a quantizer. The sampling clock is recovered from the received signal by a bit synchronizer. The quantized samples are fed into deinterleaver and then decoder to remove detection error due to channel noise picked up during transmission. In the case that excessive channel noise or fading causes errors after the decoder, the channel error can be concealed by using a muting technique. Muting is employed when error burst is detected and suppresses the annoying sound which would otherwise be associated with error burst.

Frame Deformatter

The detected data streams is then fed into a frame deformatter to gain access to the control information and strip-off the overhead data. The overhead data is fed into a microcontroller for display information or control receiver for audio quality enhancement and/or performs other service functions. The audio data, on the other hand, is demultiplexed into right and left channel audio data streams.

Audio Decompressor

The right and left channel data streams are fed into audio decompressor to reconstruct the original 16 bit representation of the audio signal. The audio algorithm which is to be employed in the DAB receiver has the properties that decompression algorithm is relatively simpler than the compression algorithm to make real-time and low cost decoding implementation is possible.

Audio Enhancement

Enhancement of audio is much more easily accomplished with digital processing than analog processing. Examples such as equalization can be easily done by using FFT algorithm to convert signal into frequency domain representation and adjustment portion of frequency bands according to user command. Other audio enhancements can be programmed into a software algorithm performed by the use of the digital signal processor.

Digital to Analog Waveform Conversion Processing

It is well known that direct digital-to-analog conversion (DAC) of the audio signal without distortion can only be achieved with the use of a brickwall filter. Since an ideal brickwall filter is difficult if not impossible to implement, it is a common practice to use an oversampling technique to reduce distortion associated with direct DAC.

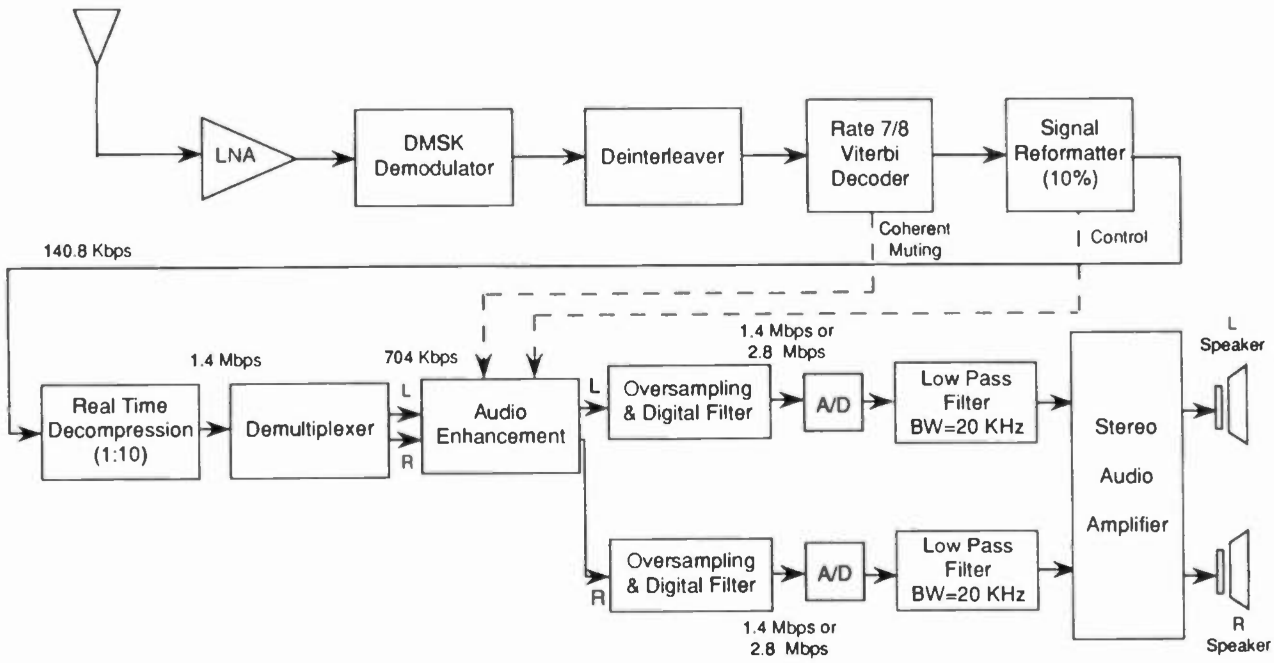


Figure 7 Functional Block Diagram of a DAB Receiver

Oversampling is performed by the zero-padding technique: first insert zeros between consecutive digital samples, and, second, low-pass filter the resultant digital signal. The higher rate digital streams is then fed into DAC and then filtered with a low-pass filter to reconstruct the analog signal. This reduces the strict performance requirement of the construction of analog low-pass filter after DAC. Other techniques such as interpolation can also be used instead of zero-padding to achieve the same or better results.

Stereo Amplifier and Speakers

The resulting output of the DAB receiver is connected with a stereo amplifier, which drives the speakers. The analog audio signal is easily integrated with existing or DAB quality audio reproduction sound system and reproduced at the receiver site with high-fidelity digital sound.

CONCLUSION

In this paper, an example design is provided for a land-based Digital Audio Broadcast (DAB) system. This system utilizes the present digital communications systems engineering practices and presents some of the insights in building a practical DAB transmitters and receivers. The system design parameters need to be optimized to meet any regulatory considerations. Given a practical frequency allocation for DAB, it is shown in this paper that proper systems engineering practice can bring a DAB system operable within the current analog FM radio station bands. Such channel bandwidth compatibility may allow graceful transition for both consumers and broadcasters toward the digital systems in the near future.

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SUBJECTIVE ASSESSMENTS ON LOW BIT-RATE AUDIO CODECS

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ABSTRACT– The Swedish Broadcasting Corporation has performed subjective assessments on low bit–rate audio codecs for ISO/MPEG/Audio. As it is likely that the same codec can be used for DAB the evaluation is of great importance for broadcasters. This paper presents the methodology, results and conclusions from the assessments performed in July 1990.

INTRODUCTION

Subjective assessments or listening tests have always been an important part in the evaluation of audio equipment. Maybe even more so today. A major part of today's, at least professional, audio equipment show electric data with no or very little degradation compared to a straight wire. Still audible differences can be detected.

For certain types of digital audio equipment there are no adequate methods of objective measurements available. This is certainly true for advanced bit-rate reduction systems. Subjective assessments will therefore play a very important role in the choice of a source coding algorithm for DAB.

MPEG/AUDIO

In 1988 the two standardization bodies ISO and IEC initiated Moving Pictures Expert Group (MPEG) with the objective to establish an international standard for encoding of moving pictures for storage and retrieval on digital storage media. Later the same year the terms of reference were extended to include audio coding and the MPEG/Audio–group was formed.

A number of sound coding algorithms were during the work reduced to four "clusters" or development groups with representatives from Europe as well as from Japan and the United States. In October 1989 MPEG appointed the Swedish Broadcasting Corporation (SR) to carry out the subjective assessments on the audio codecs.

Although this test was not done directly for DAB it was generally felt that the outcome should have a strong impact on the choice of source coding for DAB as well. A common algorithm would ensure compatibility between consumer and broadcast equipment.

Furthermore, two of the "clusters" in the MPEG competition were also candidates within the DAB–project, EUREKA 147.

METHODOLOGY

The methodology and procedures were outlined by a special Task Force Group within MPEG/Audio and were naturally to a great extent influenced by the experience and requirements of SR. Two strong requirements from our side were:

- That the test should be carried out on real–time hard ware. We regarded this as the only realistic way of controlling what was actually tested.
- That the test methodology should be "Double blind test with hidden reference". This method is sensitive and permits accurate detection even of small impairments. Results from earlier test had proven to be very reliable. The method is about to be one recommended by CCIR with the new name "Triple–stimulus, Hidden reference"

Those two points were agreed and a lot of freedom was left to SR regarding test sequences and practical circumstances for the test.

It was also agreed that a floating point (one decimal only) grading scale derived from the five–grade CCIR (Rec. 562) impairment scale should be used. The scale can be seen in figure 1.

The tests should be performed on three bit–rates, 128, 96 and 64 kbit/s per channel.

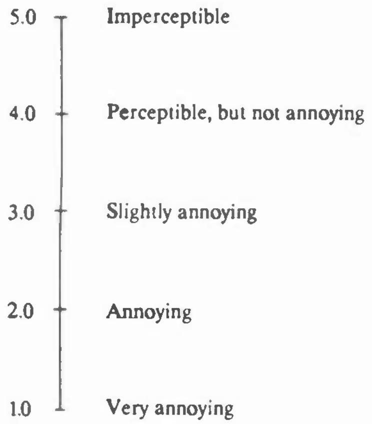

Figure 1.
CCIR Impairment Scale

The Listening Test System

In 1985 SR designed and built a system for performing listening tests where it could be guaranteed that the result was unbiased by a subject's knowledge of what piece of equipment he or she was listening to. The main objectives for the development work were:

- The system should be capable of handling a known reference signal as well as a hidden reference.
- It should be possible for the subject to switch between the objects under test and the two reference signals.
- The order of the signals connected to the select buttons should be randomized between programme items and vary between the subject control terminals, figure 2.
- All tests should be double blind tests.
- It should possible to use different grading scales.

The system consists of a system controller, a time code reader with a programmable memory for TC-related events, a section with audio switchers and 5 subject control terminals. Each terminal has its own random generator for connection to buttons B, C and D (A is always the known reference) and allows the subject to switch between the sources A-D and also to set grades. The terminal communicates with the system controller via an RS 422 interface. The grades are stored in a battery backed-up RAM in the controller.

An IBM-compatible PC is used as a host computer to create a set-up for a test and to down-load unprocessed data (grades).

GENERAL SYSTEM SET-UP

During one session all codecs (4) were tested at one bit-rate with all programme items (10). One codec was connected at a time and the subjects were to grade the buttons B and C, the object and the hidden reference without knowing which was which.

DAT

All programme items were recorded and played from a DAT with time code. The tape contained each item once for each codec in a pre-calculated random order. The order in which the objects were connected was determined by the system controller and not known in advance. The only rule was that each programme item must pass each codec once during a session.

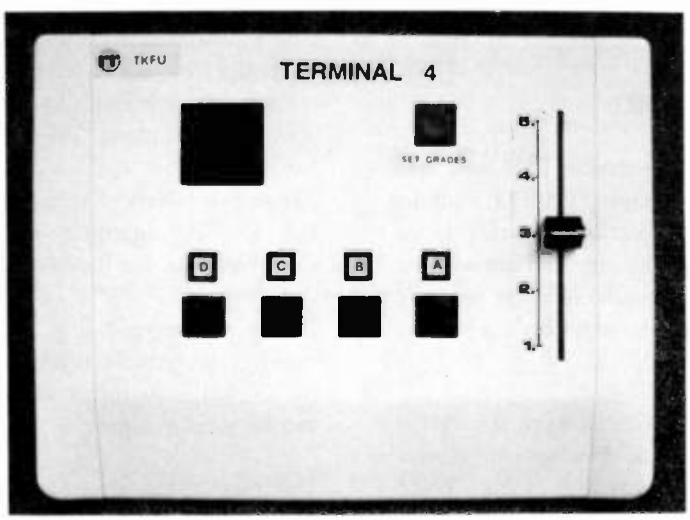

Figure 2.
Subject Control Terminal

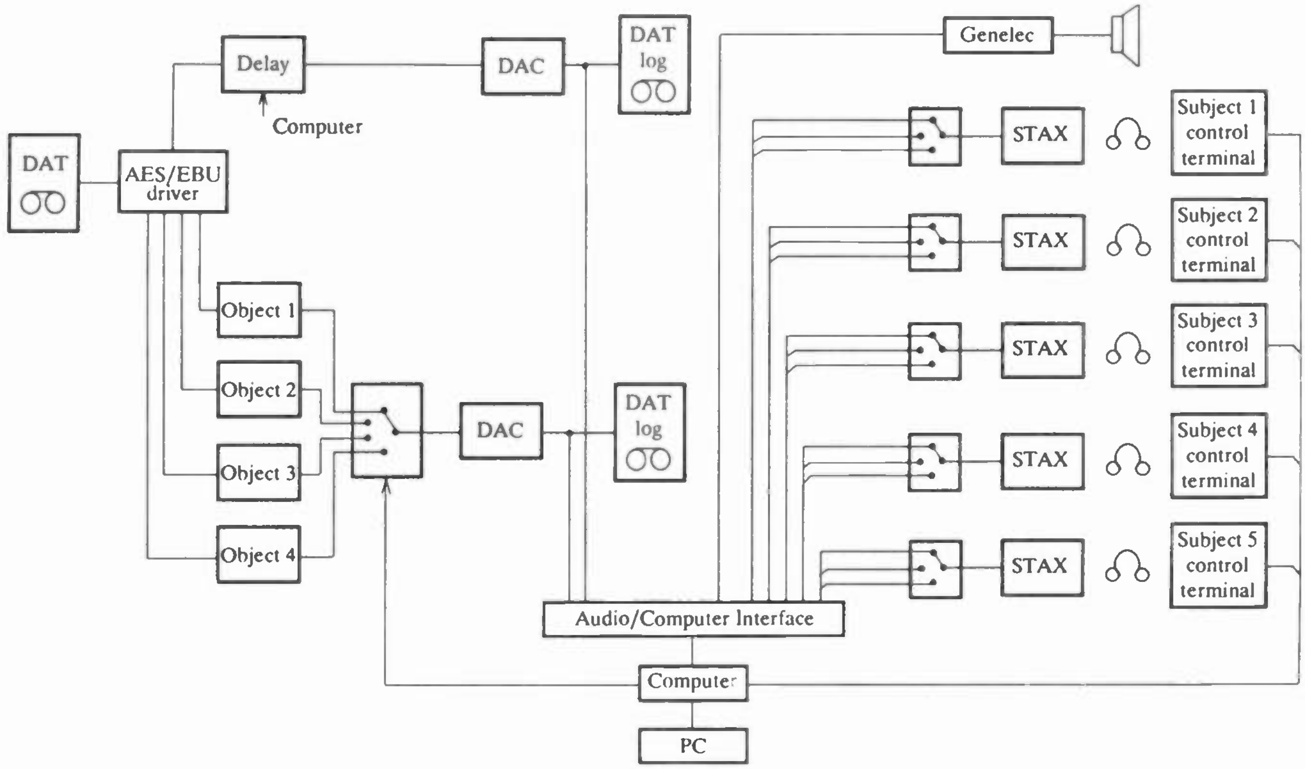

Figure 3.
General System Set-up

Objects

As mentioned before, four codecs took part in the MPEG/Audio competition.

MUSICAM	CCETT	France
(sub-band, 32 bands)	IRT	Germany
	Philips	The Netherlands
	Matsushita	Japan
SB/ADPCM	NTT	Japan
(sub-band, 8 bands)	BTRL	UK
ASPEC	AT&T	USA
(transform)	CNET	France
	Thomson/ Brandt	Germany
	Fraunhofer	Germany
ATAC	Fujitsu	Japan
(transform)	JVC	Japan
	NEC	Japan
	Sony	Japan

Unfortunately only two of the codecs (MUSICAM and ASPEC) were fully implemented at the time of the assessments (July 1990). The other two participated informally in the competition and the result from those two will not be presented here.

DAC

One DAC was used for the reference signal, direct output from the DAT, and one common was used for the four codecs. This was possible as only one codec was connected at a time. The DAC used was Philips DAC 960.

Delay unit

Different codecs create different amount of delay. In some cases the delay for a specific codec is a function of the bit-rate. In order to make it impossible for the subjects to use time difference to identify the reference and the object it was necessary to delay the reference signal with same amount as introduced by the object under test.

A digital delay with AES/EBU I/O was designed. Possible delay ranges from 0 to 255 ms, with a resolution of 1 ms. The unit is programmable in 3 groups (one for each bit-rate in the test) with 8 values in each group. When an object is connected the system controller sends a control signal to the unit and the delay is set to the correct value.

Loudspeakers

The loudspeakers used were Genelec Studio Monitor 1035A.

Headphones

Assessments on the two higher bit-rates (128 and 96 kbit/s) were performed on both loudspeakers and headphones. The agreed headphones were STAX SR Lambda Pro with diffuse field equalizer.

Listening room

A former small drama studio was used as listening room. The studio was slightly modified before the tests in order to improve the acoustics.

TEST SEQUENCES

The codecs were installed approximately two weeks before the actual test. During this period some "golden ears" from SR played possible material through the different codecs. The aim was to find two critical pieces for each codec and then add another two for test of possible changes in stereo image. As two of the codecs were not fully implemented this aim could not be perfectly achieved. The sequences should be "normal broadcast material" and not artificial signals. The list of sequences is found in table 1.

Each sequence is apr. 20 seconds long and was played 6 times to form a programme item. During loudspeaker sessions, when individual switching was not possible, forced switching, A–B–C–A–B–C was applied. B and C was randomly object or hidden reference.

SUBJECTS

CCIR Recommendation 562–2 states that a system intended for high quality broadcasting or reproduction shall be assessed by expert listeners exclusively. There is a strong argument for this in the difference of "exposure time" between a test and an introduced system. During a test a subject may listen only for a very limited time on each item. Once a system is standardized listeners will have the possibility to find artifacts during 20 – 30 years.

It was agreed between MPEG/Audio and SR that, in order to get as reliable results as possible, the target should be to have approx. 50 subjects participating in the test. It was very encouraging to find that there was no real problem to find subjects who wanted to spend 10 hours or even more on the assessments. We ended up with 60 expert listeners. 23 were appointed by SR, 24 by the four development groups and the remaining by organizations like the EBU and the AES. Half of the subjects were from outside Sweden.

RESULTS

When the assessments were completed the data base contained more than 20 000 grades, half of them on the hidden reference and the other half distributed between the objects. As 64 kbit/s was assessed with fewer subjects and only 5 programme items the major part of the grades are on 96 and 128 kbit/s. Those are also the lowest bit–rates that, for the time being, realistically can give satisfactory quality for introduction in a DAB-system.

1.	Suzanne Vega	tr 1	0.22–0.42	A&M 395 136–2
2.	Tracy Chapman	tr 6	0.36–0.57	Elektra 960 774–2
3.	Glockenspiel	tr 35/1	0.00–0.16	EBU SQUAM 422–204–2
4.	Fireworks	tr 1	0.00–0.20	Pierre Verany 788031
5.	Ornette Coleman	tr 7	--	Dreams 008
6.	Bass Synth	--	--	RR recording
7.	Castanets	tr 27	0.00–0.20	EBU SQUAM 422–204–2
8.	Male speech	tr 17/2	54.16–54.35	Japan Audio Society CD–3
9.	Bass Guitar	--	--	RR recording
10.	Trumpet (Haydn)	tr 10	5.10–5.30	Philips 420 203–2

Table 1.
Test Sequences

A usual way to present results from tests like this is to calculate mean value and standard deviation only. It is our opinion that this presentation does not give an immediate information regarding the confidence of the results. A more "honest" way is to also calculate and present the confidence interval. This also facilitates reading and understanding of the results.

Results are here presented with two different kinds of graphic representation. Only results from the two fully implemented codecs and on the two higher bit-rates are presented. Figure 4 shows the grades for MUSICAM at 128 kbit/s, loudspeaker and headphone assessment combined. Each bar has three lines at the top. The one in the middle is the mean value. The other two are the mean plus and the mean minus the confidence interval. All calculations are performed on the 95% level. An item is said to show a statistically significant impairment when there is no overlap between **object + conf.int.** and **reference - conf int.** (see item 5).

Figure 5 shows the same result as figure 4 but now calculated as difference between the reference and the object. This presentation is probably more correct from a statistical viewpoint and it also facilitates reading even more. As can be seen 7 out of 10 programme items are impaired and when the grades from all items are combined we also have a significant impairment.

Figure 6 shows the result for ASPEC on 128 kbit/s where 4 impaired items can be found. The result for item 3 (Glockenspiel) is much worse than any other result for the two codecs at this bit-rate. It was found after the test that this was caused by a hardware fault in the PLL of the encoder. This was confirmed by SR in a complementary test.

Figures 7 and 8 show the result on 96 kbit/s. Practically all items show a statistically significant impairment and the absolute grade is somewhat lower than at 128 kbit/s. **Note** the different scales in figures 5 and 6 compared to 7 and 8.

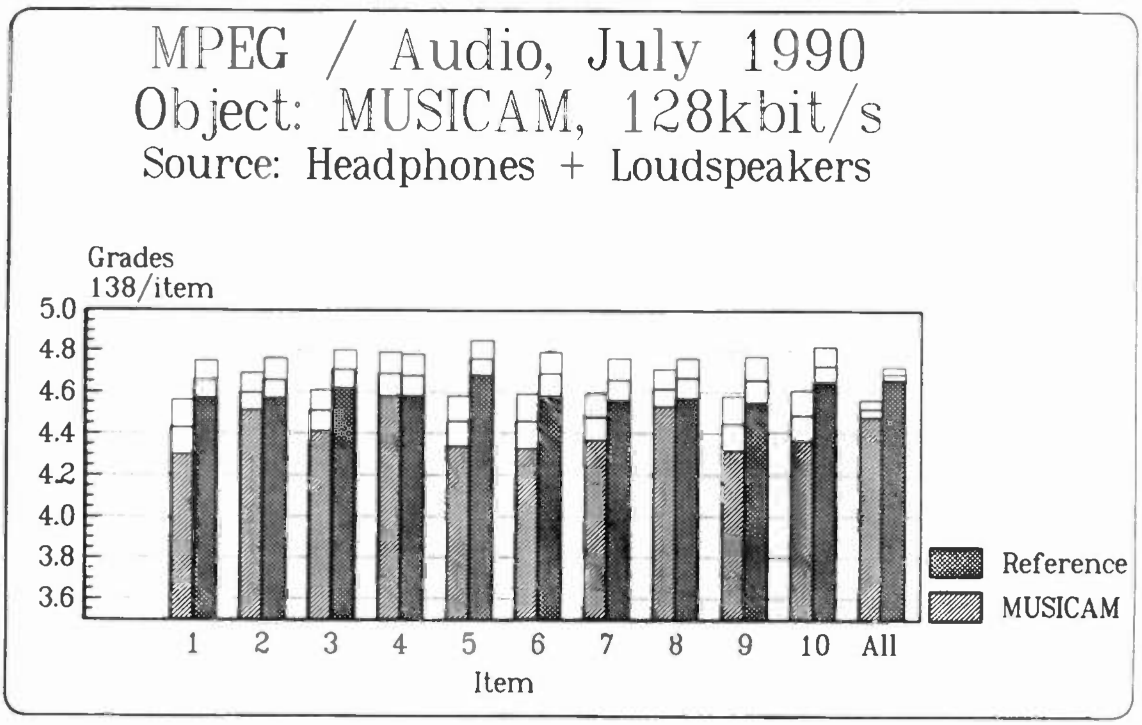

Figure 4.

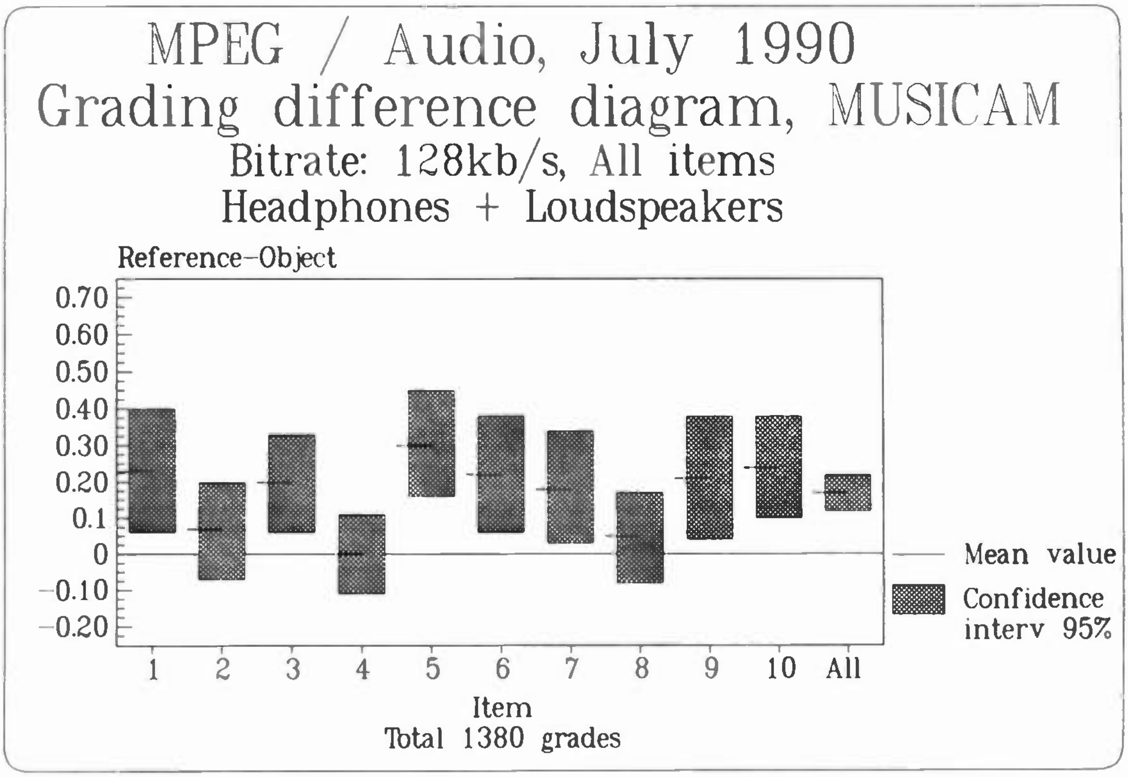

Figure 5.

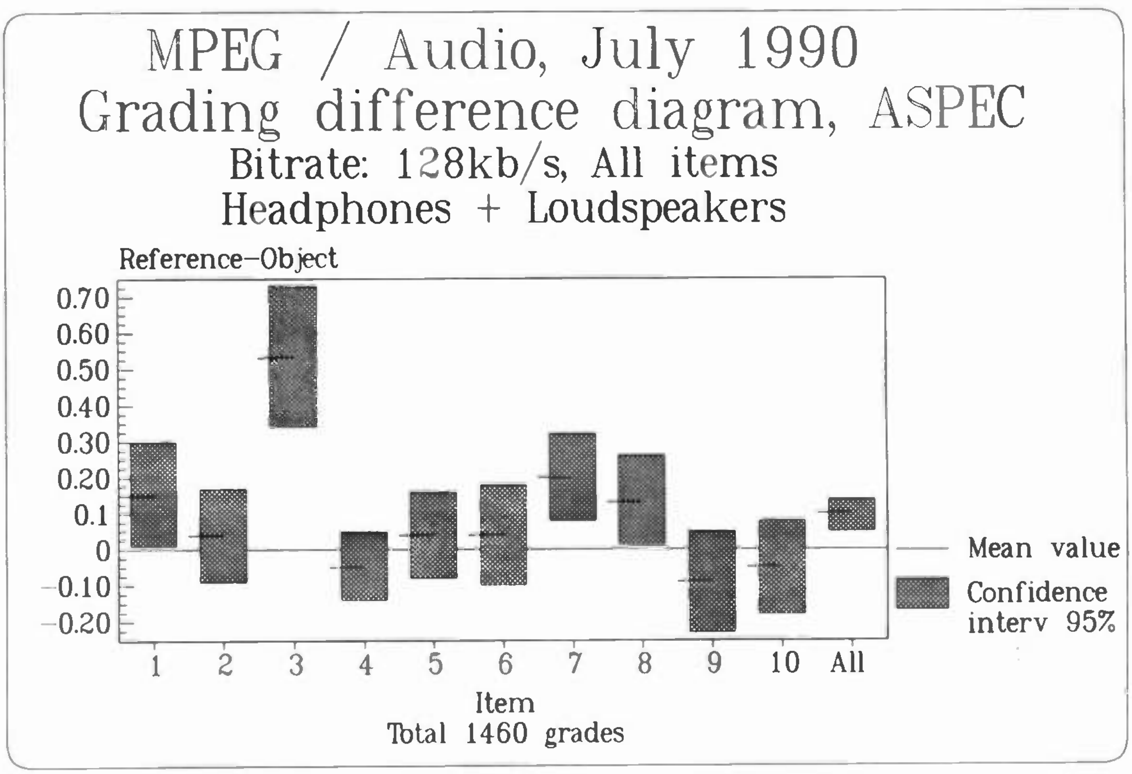

Figure 6.

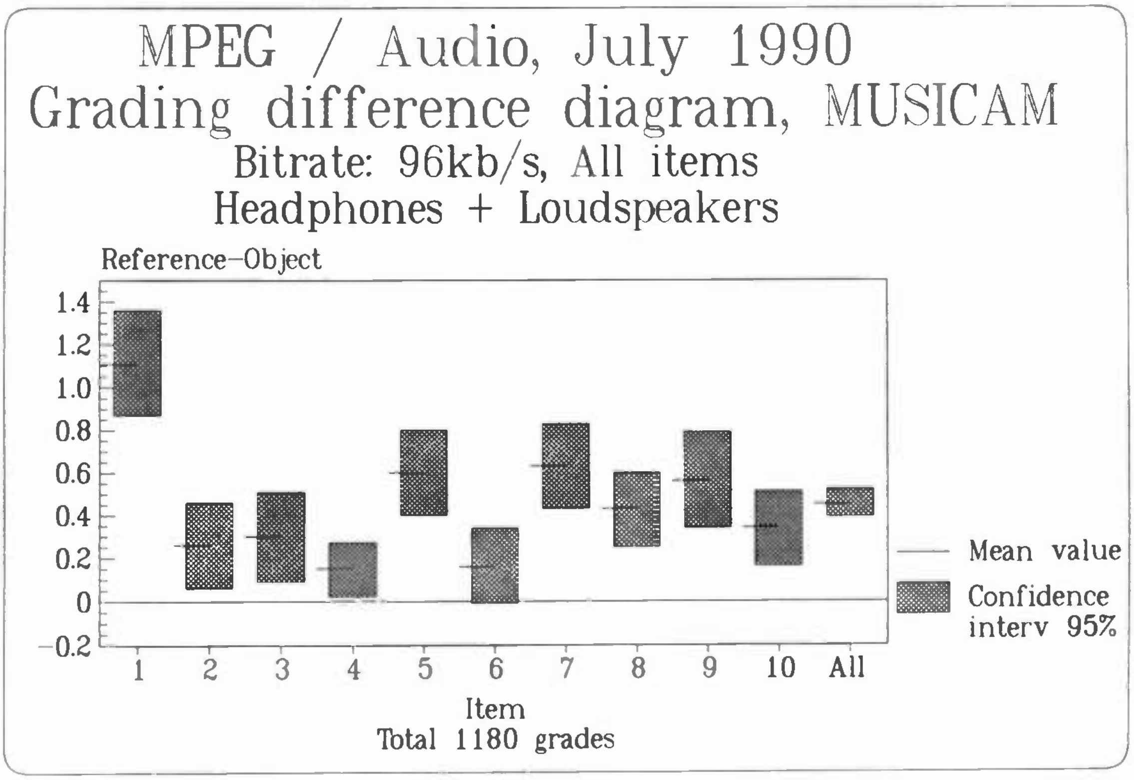

Figure 7.

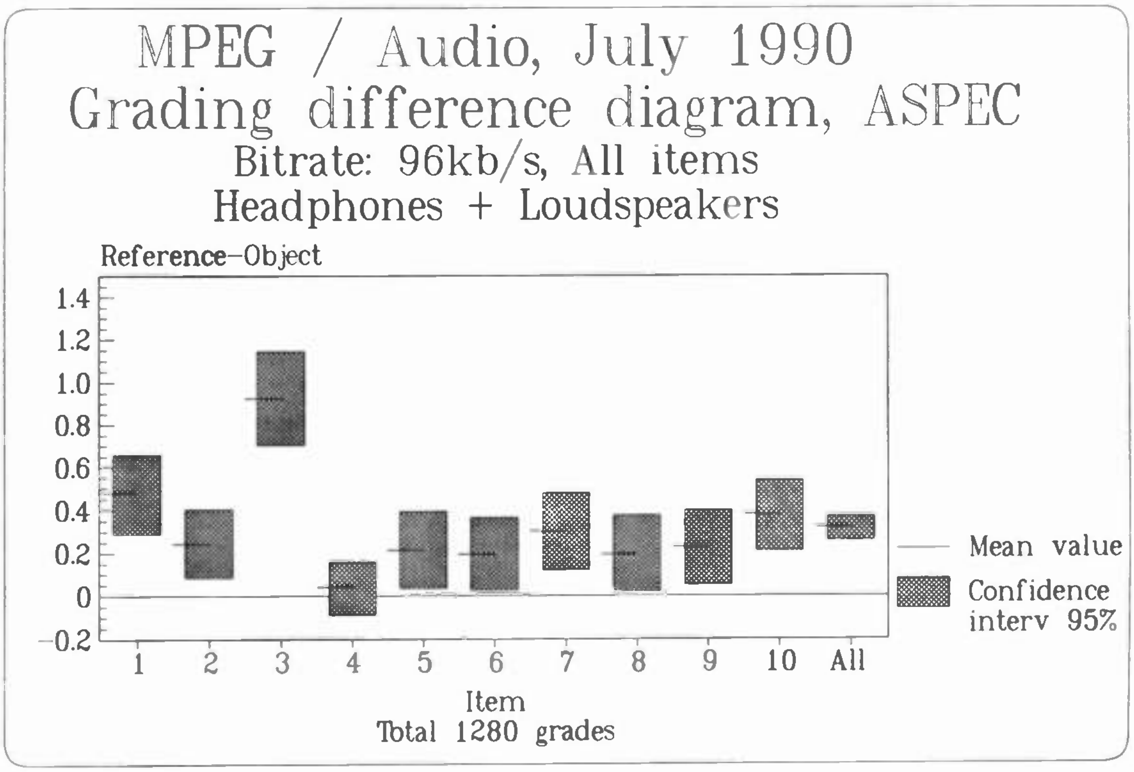

Figure 8.

CONCLUSIONS

The results show that ASPEC had a slightly better performance than MUSICAM. The difference, however, was small and it must also be mentioned that no direct comparison between the two codecs was made. The aim for the test was to find a codec that was transparent compared to a CD and not to compare the codecs in terms of absolute grades.

The sound quality of both codecs was high and the impairments are to be regarded as small. Nevertheless SR came to the conclusion that none of the codecs could be generally accepted for use as distribution codecs by the broadcasters, at the stage of development by the time of the tests in July 1990.

There were two strong reasons behind this conclusion. Firstly that it was our strong belief that there was a potential for further refinement of the algorithms and that the developers should be given more time before a standard is settled. The second reason was that it must be borne in mind that when we are talking about DAB we are talking about a system that shall live for 30 years or more. Artifacts that may be difficult to detect at a first listening will be more and more obvious as time goes by. It can be compared with somebody who moves into a new house. The first time he looks through the window he only sees the beautiful view. After a few days he detects a small damage in the glass and from that moment he cannot look through the window without seeing the damage.

At first we got a lot of criticism for our conclusion but at a later stage large groups within the EBU have supported our view.

It has also been argued that the methodology used had a tendency of enlarging differences. This well-known effect is not by any means related to the methodology but the repertoire of codecs and test sequences. A magnification may happen when the grand mean grade is considerably higher than three. However, our calculations have clearly shown that this was not the case, mainly due to the inclusion of all the four codecs and a successful selection of test sequences. In addition, we feel that the enlarging aspect has been somewhat over-emphasized.

If there is a strong interest to detect small differences it can not be wrong to put a "magnifying glass" over the objects. One absolute requirement is of course that the method itself does not introduce differences.

It must also be mentioned that as the subjects had to grade both the object and the hidden reference there was no possibility to cheat by consequently down grading the object when no difference was heard. Calculations, where subjects who had downgraded the hidden reference too often were excluded, show a result that is almost exactly the same as with all subjects included.

Our experience from the involvement in this project shows that the sound quality of the codecs is convincing. SR is confident that a refined version of the MPEG codecs will be suitable for various applications such as DAB.

THE MPEG DECISION

MPEG/Audio had drawn up very detailed rules for evaluation of both the subjective assessments and a number of objective measurements. ASPEC got a slightly higher score for the subjective part while MUSICAM was superior in the objective part (delay, complexity etc). The overall winner according to the MPEG/Audio rules was MUSICAM but the difference was very small.

At an MPEG/Audio meeting in Stockholm immediately after the test, the two development groups were requested to try to merge their two algorithms into a common one. This work was finished and presented to MPEG/Audio in December 1990.

In order to verify the expected better performance of this new algorithm a new series of subjective assessments must be performed. As the question of finding the best possible source coding algorithm for DAB (and other applications) is of greatest importance to broadcasters the Swedish Broadcasting Corporation has once again offered to perform those assessments. It is also of importance that the tests are performed under circumstances as equal as possible to the previous one.

The assessments are scheduled to take place at SR in Stockholm March 25 - April 17 1991.

INITIAL EXPERIMENTATION WITH DAB IN CANADA

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Abstract- The world's first complete digital (end-to-end) radio (sound) broadcast, which included an experimental digital production centre, digital fibre-optic studio-to-transmitter link, and digital over-the-air broadcast transmission from the CN Tower at 798 MHz (UHF band), was carried out in Toronto on June 28, 1990. This experiment was part of a much greater and audacious multi-objective project organized jointly by the CBC, CAB, CRC and DOC. The feasibility and potential of Digital Audio Broadcasting (DAB), were demonstrated across Canada, using the Eureka-147 COFDM-MUSICAM system, through an elaborate program of laboratory and field tests, as well as nation-wide fixed and mobile demonstrations. This paper describes the project and discusses the results.

INTRODUCTION

For some years, broadcasters in Canada have been aware of the need to improve radio broadcasting services from several perspectives: enhance the audio technical quality (to compete with CD and R-DAT); increase reliability of service to vehicular/portable receivers (i.e., able to cope with multipath and shadowing); eliminate inter-service interference such as FM/NAVCOM and FM/VHF-TV problems; increase spectrum and power efficiency to conserve resources and minimize the impact on the environment, e.g., non-ionizing radiation; augment spectrum capacity and availability for new services. Technological developments in Europe towards a complete digital radio broadcast system seem to respond to several of the above goals. During the first "public" demonstrations of the Eureka-147 DAB system at WARC ORB-88 in Geneva, representatives of the CRC and CBC initiated some discussions with representatives of the CCETT and IRT on the possibility of experimenting with the COFDM/MASCAM (now called MUSICAM) system in Canada. The Geneva demonstrations were very impressive, in particular the mobile one, where the signal reception was impeccable even under several multipath and severe fading conditions, such as passing through a railway underpass. The sound quality was comparable to compact disk.

In light of these promising achievements, and with the upcoming 1992 World Administrative Radio Conference (WARC-92), which is mandated to address the allocation of spectrum for Satellite Sound Broadcasting (a new service that was designed to improve upon the existing radio services and would most likely deliver digital audio), Canadian broadcast system/service planners decided that the question of DAB (or as referred to in Canada: Digital Radio or DR), should be addressed considering together the terrestrial and the satellite service, and that technology studies, system testing, and frequency planning should be undertaken as soon as possible. The first step was to examine proposed DR systems which had some hardware built and available for testing.

In summer 1989, some tentative discussions with the CCETT concerning a possible loan of COFDM/MUSICAM prototype equipment for engineering tests were undertaken by the CBC and the CRC. At that time, there was also an increasing desire from the Canadian broadcasters to evaluate and demonstrate the European DAB system in Canada. In view of their common interests, the CBC, CRC, DOC, and CAB joined together to plan and organize a program of tests and demonstrations in Canada which would attempt to respond to the needs, questions, and expectations of the national public broadcaster, the commercial broadcasters, the broadcast system researchers, and Regulators.

An ad hoc group was formed with representatives of the four above organizations, originally, under the auspices of the Canadian Advanced Broadcast Systems Committee (CABSC) - a joint Government-Industry Committee with the mandate to investigate new broadcast systems technology. The ad hoc group began negotiations with the CCETT and IRT. Agreement in principle was obtained between the CCETT, IRT and CABSC for an equipment loan on the condition that the project be a co-ordinated effort with all interested Canadian groups and that an agreement would be reached on the project work plan. The general plans for the project were finalized in March 1990, and included some laboratory tests, objective and "subjective-listening" tests of the audio source coding (i.e., bit-rate reduction codecs), and nation-wide field tests and demonstrations, in Ottawa, Toronto, Montreal and Vancouver, from June to September 1990.

In the following sections, the project's objectives, logistics, equipment configuration, tests and demonstration results will be discussed. Considering the limited length of this paper and the fact that the analysis of the data collected during the tests was not completed at the time of writing this paper, only preliminary summaries of the test results are given. For more detailed information, the DAB Technology Evaluation Commitee of the CBC/CAB/CRC/DOC ad hoc group will be issuing, in April 1991, a complete report of all the laboratory, field and listening tests. [1]

PROJECT OBJECTIVES AND WORK PLAN

The objectives of the project were the following:

- to raise the awareness level of the broadcasting industry, the government and the public to Digital Radio;
- to gather the technical and strategic information required to enable Canada to arrive at satisfactory decisions regarding spectrum requirements (including preparation for WARC 92), standards for source coding and transmission systems, and planning parameters for a new Digital Radio service in Canada;
- to evaluate the COFDM/MUSICAM technology and its current hardware in terms of technical development and applicability in Canada.

The main goal of the project was to evaluate DAB as a new service/system concept, i.e, a complete digital broadcasting system with digital audio source and digital transmission.

The work plan was composed of four parts:

- Laboratory tests, including system performance and interference measurements;
- Field tests, including area coverage assessments, channel characterization measurements, fixed and vehicular reception tests, satellite simulation measurements, and gap-filling experiments;
- Listening tests;
- Demonstrations.

The CRC was responsible for planning and undertaking the laboratory tests, the channel characterization measurements and the listening tests. The CRC, DOC, and CBC developed a field test measurement program and the measurements themselves were performed by the DOC and CRC, with the exception of Toronto, where the CBC participated extensively. The demonstrations were planned and organized by a National Co-ordinating Committee formed mainly from representatives of the CAB and CBC, with the participation of the DOC. The CAB was the major player in the promotion, administration, and funding of the demos, with the CBC providing also logistic, technical personnel, as well as the transmitter site and facilities in Toronto and Montreal. The DOC and CRC provided technical resources and expertise. Through special technology development programs, the Government of Canada contributed to the funding of the project.

As was the case for financial and human resources, each organization provided its share of the required equipment: the CRC and DOC were responsible for the test and measurement equipment and the test vehicle; the CAB was assigned to the transmitter, Tx line and antennas, the demo vehicle and its preparation, the sound system, including the audio generating equipment; the CBC was responsible for the negotiation of the DAB prototype equipment and provided special digital and RF equipment. A lot of the equipment, such as the transmitter, transmission line and antennas (including installations) were contributed to the project by major private radio groups, equipment and engineering service suppliers.

A small delegation of the ad hoc group met with the CCETT and IRT in Rennes (France) in February 1990 to discuss the work plan, the equipment configuration to be used in Canada, and finalize the joint-project arrangements. In addition, during a visit to the factory, some discussions were undertaken with Thompson-LGT (Paris) concerning the modifications required to the transmitter to interface with the CCETT equipment.

The loan of highly advanced digital prototype hardware and its import in foreign countries involved some legal and technical challenges. The CRC was the key player in the transportation, brokerage, liability and insurance arrangements associated with the loan of the DAB equipment, as well as the technical interface with the CCETT and IRT.

The DAB equipment arrived in Ottawa, at the CRC, in the first week of May 1990 with three CCETT and IRT engineers. The first installations and operational tests were performed by the French and German engineers. The CRC personnel was trained to connect and operate the COFDM and MUSICAM equipment. The first North American over-the-air DAB transmission took place during that week.

SYSTEM AND EQUIPMENT DESCRIPTION

DAB Prototype equipment

The DAB hardware used in the Canadian trials was developed mainly by the CCETT (France) and the IRT (Germany) within the EUREKA-147 project, with the involvement of the European Broadcasting Union. The DAB prototype equipment consisted of audio source coding/decoding hardware and channel coding/decoding hardware, as well as modulators and demodulators.

The source codec is required to reduce the audio bit-rate from the compact disc (CD) standard data rate to a much lower bit rate for spectrum-efficient transmission in the channel. The MUSICAM source coding hardware was provided by the CCETT and IRT, and reduced the bit rate of an AES/EBU Digital audio signal (i.e., 20 kHz audio bandwidth; 48 kHz sampling rate * 16 bit sample = 768 kbit/sec) to 128 kbit/sec, for one monophonic channel.

The channel coding system is required to process the digital audio compressed bit-stream to an appropriate form for transmission in the radio broadcasting channel, characterized in the worst cases by heavily shadowed areas and Rayleigh fading caused by the varying multipath due to mobile reception. The COFDM system, which is based on frequency and time diversity, is well described in [2]. The COFDM prototype units were provided by the CCETT and had the following parameters: 7 MHz RF bandwidth; 448 QPSK narrowband subcarriers with 15.625 kHz spacing; 80 microsecond symbol length with 16 microsecond guard interval; capacity to transmit, in multiplex, 33 digital audio monophonic channels (128 kbit/sec per channel) or 16 stereophonic pair and one data channel. The units used in Canada allowed, however, only two monophonic channels to be fed into the COFDM transmission system. The other 31 inputs were not accessible, but some sinesweep or similar test signals, generated inside the units, were modulating the multiple carriers during the time windows allotted to the 31 channels. This resulted in the transmission of a signal which is a real COFDM signal only approximately 1/16 of the time. The 7 MHz bandwidth was only a prototype implementation choice. The COFDM system can transmit at least 25 (128 kbit/sec) channels, or 12 stereophonic pairs in 3.5 MHz, and it is expected that the next generation of equipment will be able to do this.

Transmission System

A common transmitting frequency had to be found for all four field testing and demo locations, since the COFDM associated demodulators had to be preset and certain expensive and custom-made RF filters were required to prevent interference to COFDM reception, but also not to cause some to other RF services. After some frequency studies, the DOC authorized the use of UHF-TV channel 68 and 69

(794 - 806 MHz) for experimental broadcast in the four locations. The COFDM 7 MHz wide IF signal once up-converted was centered at 798 MHz thereby occupying the band 794.5 - 801.5 MHz.

A brand new Thompson-LGT 1 kW UHF-TV transmitter was modified (to serve mainly as a RF wideband amplifier) and tested at the factory in Paris to interface with the COFDM equipment. It was concluded from these tests that the transmitter could operate up to about 500 watts in a continuous mode without introducing some non-linearities which could affect the COFDM signal. Because the transmitter was damaged during transportation between Paris and Ottawa, it was operated in most locations at powers of around 150 watts. The transmitting antenna configuration was different in each location and consisted of anywhere from one to four vertically-polarized Kathrein UHF-TV panels, each one containing eight dipoles and providing a gain of about 13 dB. Each panel had a half-power beamwidth of about 62 degrees in the horizontal plane and about 28 degrees in the vertical plane. The vertical polarization was required since a vertical whip antenna was used for mobile reception.

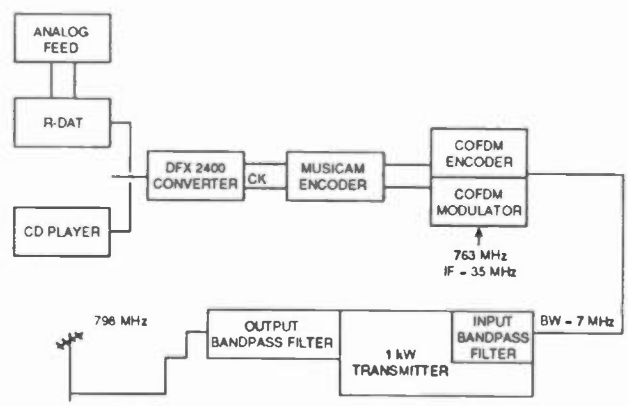

Figure 1: **Equipment Block Diagram at Transmission site**

Starting from the front end of the system, the digital audio source was either an R-DAT machine or a CD Player. The digital outputs of these units were connected to a DFX 2400 Sampling Rate convertor for two reasons: (1) to synchronize the source equipment with the MUSICAM and COFDM encoders (the master clock had to be provided by the COFDM encoder, and because the CD and R-DAT machines did not have a clock input, a DFX 2400 which had one, was required); (2) to convert the sampling rate of the CD Player from 44.1 to 48 kHz/sec. When analog sources were transmitted, they were first fed into the R-DAT machine, which then served as an analog-to-digital convertor. The output of the DFX 2400 was fed into the MUSICAM encoder which, in turn, was connected to the COFDM encoder/modulator. The COFDM 7 MHz IF signal was upconverted to 798 MHz with the help of a stable local oscillator and fed into the input bandpass filter of the RF transmitter. The transmitter was equipped with a custom-made output cavity bandpass filter (798 MHz + or - 4 MHz) to prevent out-of-channel interference. With the exception of Toronto and the demonstration launch in Montreal, all the above equipment, including the CD and R-DAT machines, was located at the transmitter site.

Receiving System

The receiving antenna was a 5/8 wave vertical whip sitting on the flat metal roof of the test vehicle or demo vehicle, at a height of about 3 metres. An 8 MHz wide bandpass filter was required at the input of the COFDM demodulator/decoder to eliminate interference from nearby mobile telephones operating in the 806-866 MHz band. Once demodulated and decoded, the received signal was fed into the MUSICAM decoder, and from there into the sound system, which included a digital to analog converter.

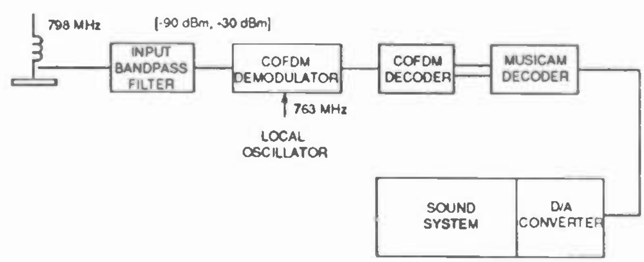

Figure 2: **Equipment Block Diagram at Receiving Site**

LABORATORY TESTS

The Laboratory tests were performed at the beginning, before the field trials. The main objectives of these tests were: (1) to verify the correct operation of the COFDM/MUSICAM prototype equipment and verify whether it met the system specifications such as required signal-to-noise ratio (SNR) for no audible impairments; (2) to evaluate the performance of COFDM in terms of interference (co-existence with other systems).

System Tests

Some Gaussian noise was introduced between the transmitting end and the receiving end of the COFDM/MUSICAM system. Degradation of the audio was just barely noticeable when the SNR reached 8 dB, at which point the bit error rate (BER) was approximately 2×10^{-4}. The CCETT/IRT specified requirement of 8.5 dB Eb/No for no noticeable effect on the audio was thus confirmed.

The required SNR (for no noticeable degradation) remained essentially constant for received input signals in the range -32 dBm to -90 dBm. Front-end overload occurred with signal levels higher than -26 dBm, indicating a receiver dynamic range of about 64 dB.

The results of the BER versus SNR measurements show that the implementation loss of the COFDM prototype system is very low since its measured performance was very close to the theoretical performance.

Interference Tests

Several interference tests were conducted. Some of them are reported here. The COFDM signal generated by the prototype equipment was not really representative of the what it would be when the final system design is ready for implementation. The results of the tests involving the COFDM signal as interferer need, therefore, to be interpreted carefully. Notwithstanding the above, some preliminary results indicate that the desired-to-undesired ratios (D/U), required to protect the NTSC-TV reception from COFDM interference, are comparable to those presently used for NTSC-TV to NTSC-TV interference.

In the NTSC TV to COFDM interference tests, performed with all the front-end filters removed, it was found that an NTSC TV co-channel interfering signal could be as much as 4 dB higher than the COFDM desired signal before the audio is affected by the interference, and this is when the input desired signal is very low, -80 dBm. The allowable U/D ratio increases to 6 dB at -60 dBm and 9 dB at -40 dBm. In the case of adjacent (1st, 2nd) channel interference, the allowable U/D ratio varies from 14 to 43 dB.

The COFDM receiver was subjected to additive narrowband interfering signals at various frequencies and power levels. For in-band CW interference, the C/I ratio (ratio of desired signal total power to interference power) for just-noticeable degradation of the received audio quality was about 0 dB, almost independent of frequency and desired signal level (in the range of -80 to -40 dBm). When FM modulation was added (5 kHz deviation), the tolerance for interference decreased by from 0 to 8 dB, and became time-varying.

FIELD TESTS

Again, only a brief summary of the tests is given here (see [1] for more information). The field tests were designed to evaluate and quantify the capability of the MUSICAM/COFDM combined system to deliver high-quality stereo digital audio to vehicular receivers equipped with simple omnidirectional antenna, within various types of North American broadcast service areas, including highly-built dense urban zones, urban, suburban, rural, and mountainous areas. The terrestrial transmissions were all carried out at 798 MHz. The tests were grouped in two categories. The coverage tests were aimed at delineating the effective coverage area obtained, including the identification of service gaps, and to relate these to the transmitting parameters. That information could serve to create propagation models to predict coverage. The second category of tests, called the channel characterization tests, were aimed to evaluate certain channel characteristics such as the type and distribution of the multipath-caused echoes (short and long ones), that can have a direct influence on the system performance. Statistics on these characteristics were derived in an attempt to calculate the coherence bandwith and optimize certain system parameters namely the guard interval.

Field Test Methodology

The CRC test van was specially equipped for the tests and one complete DAB receiver was installed. In each location, the field tests were done before the demonstrations started and other time was also provided for testing, in between and/or after the demos. Planning was important because only two complete receivers were available.

Coverage Tests. The coverage provided by the COFDM/MUSICAM system was evaluated by driving the CRC test van throughout the target area and locating spots where the reception was inadequate. The delineation between adequate and inadequate reception was facilitated by the characteristic of the COFDM/MUSICAM system which is to perform perfectly until the bit-error-rate has reached a certain threshold value that triggers an abrupt failure of the system. This sharp transition was made even more obvious by the muting feature of the MUSICAM decoder and by the absence of a gap concealment technique in the specific version used for the coverage tests. Gaps in the coverage were noted on maps and on log sheets with, for a limited number of locations, additional data on the environment, the received RF signal power and the multipath conditions observed (see below).

Channel Characterization Tests. Attempts were made to quantify the multipath environment of locations tested to determine the optimum value of guard interval for Canada. At intervals throughout the coverage area and with the test vehicle stopped, the impulse response generated by the COFDM receiver was digitized and stored in a computer for later analysis. This impulse was derived from the test signals transmitted within the COFDM signal. It depicts the channel response of the system over the 7 MHz bandwidth. From it are derived the delay interval and the delay spread. The delay interval dictates the length of the guard interval needed to overcome intersymbol interference due to multipath.

Field Tests Results

Transmitting installations varied from city to city depending on the size and shape of the area to be covered and the characteristics of the transmitter site (e.g., location's geography, tower height, etc.) of each city. Table 1 summarizes the transmission parameters for each site. The COFDM 7 MHz signal transmitted was carrying the equivalent of 16 stereo signals.

Location	OBE Tower Ottawa	CN Tower Toronto	CBC Tower Montreal	Rogers Tower Vancouver
Antenna Height AGL	76 m	364 m	30 m	38 m
EHAAT	67 m	295 m	230 m	667 m
Transmitter Power	180 Watts	160 Watts	360 Watts	150 Watts
Feedline Loss	2.1 dB	1.2 dB	1.2 dB	1.2 dB
Antenna Gain	13.2 dBi max.	13.2 dBi max.	13.2 dBi max.	10.2 dBi max.
EIRP	2300 Watts max. (33.6 dBW)	2535 Watts max. (34 dBW)	5700 Watts max. (37.5 dBW)	1200 Watts max. (30.7 dBW)
Antenna Type	2 x 2 Panels (90 deg.)	1 Panel	1 Panel	2 Panels (90 deg.)

Table 1: Transmitter Site Parameters

Ottawa Results. In Ottawa, the transmitter site was located in urban south-central Ottawa approximately 2 km away from downtown. Excellent coverage was obtained in Ottawa with uniform reception quality in the downtown core area, and up to 15 km away from the transmitter site. Some of the minor gaps found were caused by shadowing from hills and others were explained by the presence of a null in the transmit antenna pattern. It is believed that most of the gaps observed could be eliminated with simple measures such as a more uniform antenna pattern, a higher antenna, or a slight increase of the transmitted power (both of them being relatively low for propagation at 800 MHz). Few "gap-fillers" (very low power on channel re-transmitters) could also solve the problems. Large reflecting structures in the northerly direction of the radiation pattern created long echoes, often exceeding 20 usec, particularly obvious in the area west of the transmitter. The presence of multipath with excessive delay did not cause system failures but it was observed that in areas where large amounts of multipath were present, the system occasionally failed. As indicated in Table 2, the median value of the measured delay interval is 7.2 usec. The ninetieth percentile value is 17.9 usec. This means that ten percent of the locations had a delay interval greater than 17.9 microseconds. The median value of delay spread is 1.3 usec and the ninetieth percentile value is 3.7 usec.

Toronto Results. In Toronto, the transmitting antenna was installed on the CN Tower, approximately 360 metres above ground, right in downtown. The coverage extended to about 50 km north of the transmitter in the direction of maximum radiation. Only a few small gaps were found in the targetted coverage area and all of them were in parkland valley areas characterized by both blockage of direct path to the transmitting antenna and a lack of good reflecting objects to provide multipath signals.

The multipath environment in Toronto was characterized by densely built-up areas in which echoes from nearby objects predominate. This is illustrated by the small delay intervals observed (median = 4.7 usec). However, receiving sites along the river valleys exhibited long-delayed echoes (90th percentile = 20.0 usec) which, in one instance, caused audio dropouts in spite of a signal level well above the required threshold.

More detailed measurements and analysis were performed in Toronto, in an attempt to correlate the coverage area (i.e., where the signal reception is good) obtained in the field with signal level measurements, and propagation models such as F(50,50) curves. Some preliminary analysis tend to indicate that the DAB coverage area, obtained with the COFDM 7 MHz system, could be predicted by calculating the 46 dBuV/m field strength contour with the conventional UHF-TV F(50,50) propagation curves.

Simulation of Satellite Reception Test Results. With the highly elevated transmitting point (from the CN Tower) and the densily built-up and high-rise downtown core of Toronto nearby, the geometry was appropriate to a simulation of satellite reception in dense urban area, with look angles from 15 to 40 degrees (the range for Canadian metropolitan cities is from 20 to 35 degrees). CBC Engineering carried out signal level measurements along the downtown streets of Toronto at points with different elevation angles. The majority of locations did not have line-of-sight with the transmitting antenna, but the DAB reception was always excellent (even directly under the CN Tower). The signal level measurements were compared to free space calculations in order to derive some values to account for building obstruction-shadow loss. Preliminary results show that, for angles from 15 to 40 degrees, the measured received signal level in a 7 MHz bandwidth, is in average 15 dB lower than the free space calculated values.

Montreal Results. The Mount Royal site is located right in the city to the northwest of the downtown core. It is the main transmitter site for local FM and TV stations. Once again, in Montreal the coverage was excellent, reaching as far as 45 km away from the transmitter. In comparison to the current signal reception situation of local FM radio stations in downtown Montreal (characterized by heavy "picket-fencing" type of fading), the DAB reception was impeccable. The few gaps observed were due to the shadowing from a series of small mountains to the east of Montreal. The nearest gap area was at a distance of 16 km from the transmitter.

The delay intervals observed were short (median = 3.0 usec, 90th percentile = 8.8 usec) and well within the guard interval. The few and well-delimited gaps could be solved with a small number of strategically-placed gap-fillers.

Vancouver Results. The Vancouver area constituted the most severe test for this new broadcasting technology. Due to its more varied topography this city could not be covered as well as the eastern cities. Several gap areas were found in the urban area situated between the slopes of a mountain range and the very flat river delta land. Mountain shadowing, the low power transmitter, the non-uniform antenna radiation pattern and the lack of constructive multipath signals all contributed to the frequent reception dropouts. In the flatter areas to the south and east, coverage was generally good out to a distance of 40 km from the transmitter. In spite of all this, the resulting coverage was superior to that provided by the local FM stations.

The multipath environment was characterized by the predominance of short delay intervals (median = 2.2) punctuated by a significant amount of large intervals (90th percentile = 18.1 usec, 16% of points > 16 usec). As was found in Ottawa and Toronto this highlights the need for a longer guard interval to improve the reception at about 10 percent of the locations.

City	Number of Test Sites	Delay Interval (usec)			Delay Spread (usec)	
		Median	90%	>16us	Median	90%
Ottawa	440	7.2	17.9	15%	1.3	3.7
Toronto	210	4.7	20.0	12%	0.8	4.2
Montreal	230	3.0	8.8	1%	0.5	2.1
Vancouver	121	2.2	18.1	18%	--	--

Table 2: Summary of Delay Statistics

Gap-Filler Experiments. The COFDM system can cope with multipath created echoes and even uses echoes constructively. It can also cope with active echoes generated by on-channel re-transmitters. These re-transmitters can be used to fill coverage gaps ("gap-fillers") or extend coverage boundaries. In each city visited, at least one attempt was made to install a gap-filler and to analyse the impact on the coverage. The usual scenario was that after having collected some data on the coverage, a gap area was selected for the gap filler experiment. A large structure such as a hotel or a high-rise apartment building, overlooking the gap area, was chosen to provide a platform for a gap-filler installation.

The most successful experiment was in Montreal where the long curving freeway Ville-Marie tunnel completely isolates the receiver, of cars passing through the tunnel, from the transmitted signal, for a distance of about 1.7 km. A gap-filler was installed and consisted of a receive antenna on the roof of a test vehicle parked on the street above the tunnel with a good received signal from the main transmitter, and a transmitting antenna mounted on the ceiling of the tunnel, near its eastern end, pointing inside the tunnel. The vehicle also housed the amplification system. Coverage in the tunnel was practically flawless, and the transition from the main signal to the gap-filler one, and vice-versa, not noticeable.

LISTENING TESTS

The objective of the listening tests was to evaluate the capability of the MUSICAM source coding system to provide at low bit-rate a digital audio signal quality equivalent to compact disc. Following recognized scientific practices in the field of audio subjective evaluation, the CRC has compared the audio quality of MUSICAM to that delivered by high-quality FM radio and to that of CD.

These listening tests conducted at the CRC audio lab differed from the ISO-IEC/MPEG tests carried out in Sweden in summer 1990. At the time of writing this paper, the data collected during the listening tests was in the process of being analysed. The results will be presented in a specific report to be issued in March 1991. This report will also cover in greater detail all aspects of the MUSICAM listening tests.

DEMONSTRATIONS

Aptly entitled "Digital Radio: The Sound of the Future", the four-city project combined fixed reception presentation and mobile reception demonstrations. The centre of attraction everywhere was a specially-converted school bus, complete with individual headphone sets (and volume control), that enabled from 12 to 14 people at a time to hear for themselves the digital difference, during short tours of each of the four cities. Switching back and forth (under the control of the demos operator) between regular AM radio and FM radio and DAB transmissions, all of which the bus was equipped to receive, the riders were able to make instantaneous comparisons in typical urban receiving conditions. Invariably they agreed that Digital Radio is far superior to AM and FM.

Overall more than 3000 people were exposed to Digital Radio. The verdict was that Digital Radio delivers what it claims: CD quality sound, consistent throughout the area covered, and virtually no reception impairments. In each city, local demo committees were set up and were expected to raise funds to pay for a large portion of the local expenditures. Different types of demonstrations were organized in each location as well as original launches.

The National Launch of the Canadian DAB demonstrations took place on June 11, 1990 in Ottawa. More than 200 broadcasters, government officials, business people, journalists, and others attended this North American first. In addition, to the four-cities trials, special demonstrations were organized for the NAB in Montreal and for CITEL in Ottawa.

Production Facilities

For the Ottawa and Vancouver field trials, production facilities were limited to a CD player and RDAT tape player located at the transmitter site. In some cases, analog feeds of local stations were fed into the A/D input of the R-DAT. This was a direct emulation of all European field-trials, as undertaken to date. However, for Toronto, the CBC decided to implement an all-digital continuity suite and studio-to-transmitter link for the duration of the field-trials.

CBC's Studio H, the Advanced Audio Production Facilities (AAPF), in Toronto, was selected as the production and broadcast facility for the Toronto trials. Into this studio were routed contribution feeds from 10 Toronto FM stereo stations and 5 Toronto AM stations. In addition, a stereo digital audio contribution circuit was provided, in the form of a PCM F1 signal which transmits the stereo digital audio in the active picture portion of a baseband video signal.

All analog contribution circuits from local radio stations were converted to digital directly upon entering Studio H, mixed and transmitted in the digital domain to the CN Tower, where the MUSICAM encoding equipment was located.

The PCM video feed was used to broadcast "all-digital Blue-Jays" baseball live during the Toronto Demos launch (another first) by use of a video line from the Toronto Skydome via Jarvis TV Master Control. Other program sources included presentations pre-packaged on R-DAT, with CDs filling the blanks to ensure 18 hours of uninterrupted programming daily over the duration of the Toronto demonstration.

The Montreal field trials also featured limited broadcast from a CBC packaging suite, specifically for an interview with the Chairman of the project, broadcasted live during the Montreal demos launch at the Planetarium.

Studio-to-Transmitter Link

The all-digital transmission link set up by the CBC in the Toronto field trials facilitated the historic precedent of the first complete digital Studio-Transmitter-Receiver broadcast. No analog links existed in the 5 km route through the 7 separate technical facilities that processed the signal from source to receiver. The path was fully backed up throughout with metering, monitoring, and patching facilities provided at all critical interfaces.

The PCM video technology was not used in this application, as it was felt that the STL should use a more bandwidth-efficient channel code that better investigates future telcom/broadcaster interaction. For this reason, a prototype Swedish National Radio CODEC was borrowed that converts a stereo AES/EBU bitstream at 3.072 MHz to a G.704 (European DS-1 telcom channel) at 2.048 MHz. This signal was multiplexed onto a fibre-optic network belonging to Rogers Communications which carried the digital signal to the CN tower for broadcast. Although future data compression systems will allow much greater savings in bit-rates, this system could be considered a model of a high-quality audio-only contribution circuit, allowing 20 bit audio resolution.

The signal was passed on twisted pair, video coaxial and fibre-optic between Studio H, Jarvis Radio Master, Jarvis TV master, Jarvis Satellite room, Rogers fibre distribution network and the CN tower before AES/EBU to MUSICAM encoding was undertaken. The signal passed approximately 80 metres on a twisted pair, 750 metres on video coaxial (through 3 video distribution amplifiers) and 3-4 km on fibre-optic. Although the CODEC provides error detection and correction capabilities, no significant bit errors were recorded. The signal arrived in the same condition as it left, an important consideration in the context of these trials.

A further consideration was time delay, as a specially-installed 1 kW reference FM transmitter, which carried an analog version of the DAB signal, was being used in Toronto, and provided more controllable comparison. It was deemed important that the STL not significantly contribute to the delay of the MUSICAM/COFDM process. It was found to add 2 msec. to a 300+ msec. delay of the COFDM coding. This is another issue which must be carefully considered in the future when planning digital contribution circuits.

CONCLUSIONS

The Canadian project is considered to be a huge success by all those who were invoved in it. The objectives of the project were all met and, in some cases, surpassed. The co-operation between the Canadian partners was excellent and worth highlighting. This joint Industry-Government experience has considerably strengthened the Canadian Radio Broadcasting environment. The trials have shown that a Digital Radio is practical, that the technology proposed works, and most importantly, there is a public and industry demand for this new service.

The media, industry and public response was great and the reaction very positive to the new radio service concept and the quality of the product delivered by the prototype system..

All the participants to the test program were highly impressed by the excellent performance of the MUSICAM/COFDM sound broadcasting system in the laboratory and in the field. After a first analysis of the collected data, it was observed that the performance of the system could generally be predicted from the received power level alone. In spite of the relatively low transmitting powers used (considering that the equivalent of 16 stereophonic signals were being transmitted in an 800 MHz channel), the actual coverage achieved with the COFDM/MUSICAM system was amazingly large and relatively free of gaps, confirming the ability of the system to cope with multipath fading and its power efficiency. In general, it is believed that a close-to-perfect coverage could be obtained with minor adjustments at the transmitter end and the addition of a few gap-filler systems. Statistics on the multipath environment of the cities visited indicate that a guard interval in the order of 24 microseconds would be preferable to the 16 microseconds implemented in the system tested.

The effectiveness and practicality of the on-channel gap-filler concept was confirmed.

Acknowledgements

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UHF TRANSMISSION

Tuesday, April 16, 1991

Moderator:
Robert Ogren, LIN Television Corporation, Providence, Rhode Island

ALL BAND VHF AND UHF ANTENNAS

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Abstract - Antenna systems outside the Unites States require four or more FM or TV channels to radiate from the same antenna. This paper will discuss the theory of broadband antenna matching. Very sophisticated impedance matching techniques are required to obtain a VSWR less than 1.1 over the full UHF band (470 to 800/even 860 MHz). New computer software programs have been developed and are used to pre-design and select conjugate impedance matching networks to cancel reflections necessary to obtain this broadband performance. Antenna systems using this design philosophy are now in operation in Australia/New Zealand/Canada/China/England/Kuwait/South Africa/Austria/Turkey/Italy of course and many other countries throughout the world.

INTRODUCTION AND GENERALITIES

The advantage of common sites for multiple channels broadcasting are easily recognized and have led to more and more intensive adoption of broadband antennas capable of radiating simultaneansly multiple programs with the same efficiency in terms of patterns, matching and power handling capability. All around European countries this last is already a well established practice, meanwhile the actual real trend all worldwide shows a generalized running towards these techniques. The reasons why this philosophy is becoming more and more adopted are numerous, both technical as well as economic ones: avoiding proliferation of towers, optimized use of available and always limited aperture on the towers, drastic reduction of the costs per radiated channel, toghether with the current use of standard components for building up the antenna systems.

Any antenna system is or may be a tailored radiating device, but the tailoring techniques involve only the assembling of standard components, no device is tailored to the channel. In order to evaluate the problems that the broadband antenna systems have put, let us consider which are the performances that these systems are required to exhibit for TV broadcasting field as well as for FM-stereo radio broadcasting field. Essentially the main features to consider deeply are the matching at the input of the system, the radiation patterns (both horizontal and vertical, togheter with optional or requested tilting and nullfilling) and the power handling capability. As far as power capacity and radiating pattern are concerned, we will pay our attention later, when we will enter some more details on the system building up by means of radiating elements, power splitting and feeding arrangements.

Referring now to the matching performance, let us consider the standard specifications in the TV bands and FM radio band, such as we know from the most qualified Broadcasting Corporations all worldwide.

In the FM radio frequency band 87,5/108 MHz, specially for stereo trasmission, the specifications ask for a return loss up to -32 dBs on the involved channels (1,05 VSWR) while -26 dBs (1,1 VSWR) in the overall band.

In the TV bands (both VHF as well as UHF) the specifications rise up to -32 dBs return loss as well within the overall channel, or alternatively up to -36/38 dBs on the visual carrier only and relaxing till -26 dBs in the channel at the aural carrier. We can see two typical alternatives of specifications as the BBC and IBA of England are imposing for obtaining the contracts (fig. 1). These figures are looking quite rigid and difficult to achieve, but effectively the demand

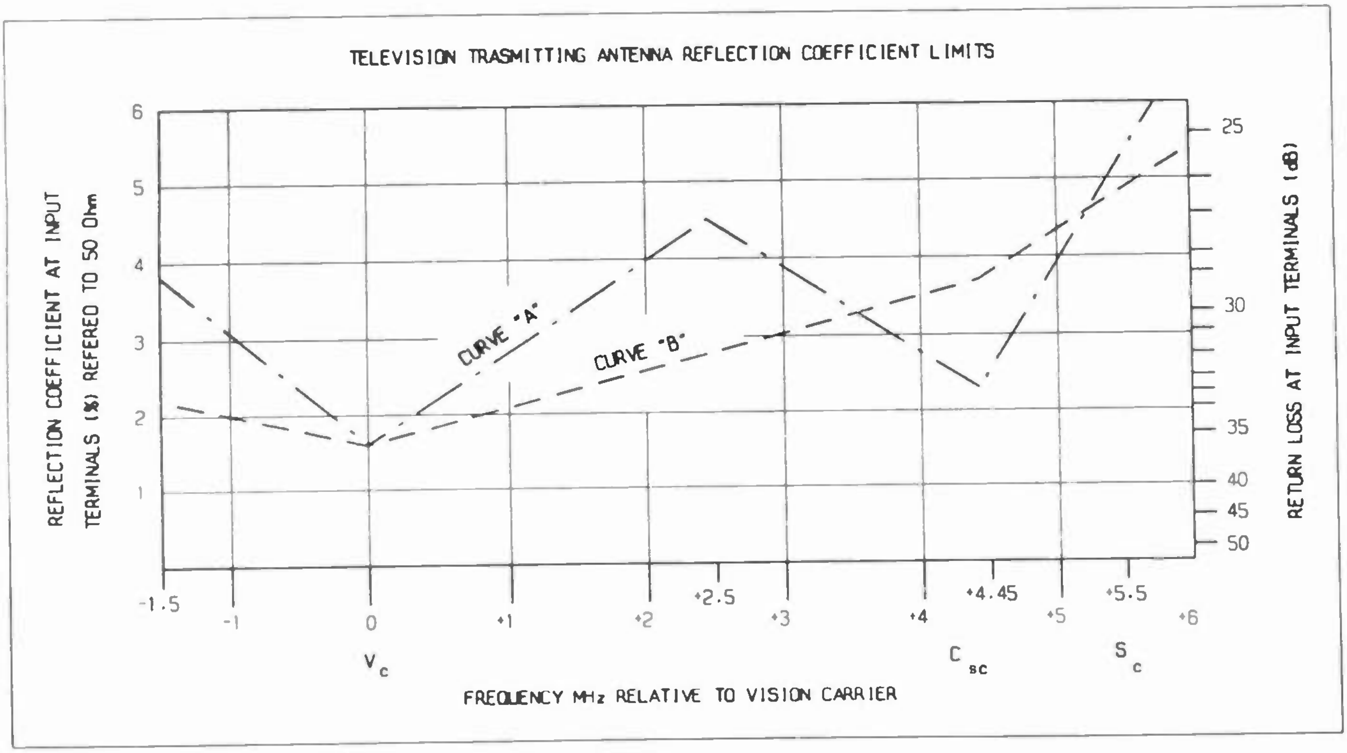


FIG. 1 - TV TRASMITTING ANTENNA SPECIFICATIONS

is rigth and justified if we consider the so called "ghost effect" on the received TV pictures.
It is well known that the "ghost effect" is due to the delayed re-radiated signals which reach the antenna after double running the feeder length, originated from the first reflection at the antenna input and reflected again at the trasmitter hall lay-out. Fortunately the feeder insertion loss, computed twice, plus some amount of absorbed signal at the transmitter, evaluated approximately in 3 dBs, put some relaxation on the figure of -36/38 dBs: the final result is that the final figure of return loss at the antenna input must be not less than -30/32 dBs. The only problem may be the final customer's choosen alternative between the -30/32 dBs figure on the whole channel or visual carrier only. The correct approach would be applying the relaxation due to feeder insertion loss mechanism +3 dBs to the original specifications, but sometimes the end user imposes -32 dBs on the overall channel and the antenna manufacturer must comply with, in spite of the correctness of the approach. One concept I would here point-out, because too many times we had to face unforeseen problems arisen from some amount of confusion contained in the specifications.

It can happen that the antenna input sometimes is interpreted at the feeder input in the station: it is too abvious that the feeder, even excellent by coax or waveguire, never is "transparent", so it introduces some degradation in the overall VSWR of the system feeder/antenna. The figure of -32 / even -38 dBs at the antenna input has its own very precise meaning (echoes reradiation), this same figure in the station has absolutely no meaning, no advantage, no benefit, has only the meaning of an academic request that is equivalent only to high costs and completely useless design efforts.

BROADBANDING ON ANTENNA SYSTEMS

The first step to be established in the broadband antenna systems is the basilar concept of how to power feeding the radiating elements.
The only system is the branch or parallel feed arrangement, never will be used the series feed arrangment, due to its too high frequency - sensitiveness.
After this first step is fixed, the building-up of the antenna system is essentially the assembling process of wide-band basic components, that means

wide-band radiators (or radiating arrays), wide-band power dividers, and coax cables or rigid lines.

We will spend no words on the cables, not so many words on the power splitters only saying that these late devices are essentially impedance transformers whose broadbanding tecniques are well known and practiced, selecting a suitable numbers of impedance steps, or directional couplers (specially 3 dBs couplers) whose broadband characteristics are instrinsic to the design.

Some more time we dedicate to the radiators, because the broadbanding tecniques here involve the radiation performances, the impedance matching at their input, and finally the mounting geometry around the supporting tower.

Let us now consider the radiators, or more commonly a radiators array, that is the main building block of the antenna system and that we currently call the "panel". It consists essentially of a certain number of dipoles in front of a reflector, whose electrical lenght is a whole wavelength shortened by parasitics. A certain amount of technical and scientific background together with well proved experience in our company have led to intensively adopt the flat dipoles instead of tubular ones (low Q/low impedance dipoles) due to better frequency response in terms of impedance. So our Company have designed and currently produce under this philosophy all the series of basic panels covering respectively the UHF European band (470/860 MHz), the high VHF band (174/230 extendable up to 254 MHz) and the two halves of the low VHF band (44/68 and 68/88 MHz). The fig. 2 shows the typical radiation patterns (horizontal and vertical) of the forementioned panels on the whole relative frequency bands.

The fig. 3 presents pictures of the pratical realization of the various panel versions. The fig. 4 describes the overall band return loss characteristics of each panel.

It is immediately evident that the return loss figures of the panels generally exhibit from -23 tc -26 dBs (VSWR from 1,15 to 1,1). Now the problem is, how to get the very high degree of input matching at the antenna system input, starting from so moderate, even if quite good, matching figures of the basic panels themselves? The principle is quite simple, nevertheless the actual application require some sophistications. Due to the parallel branching feed of the panels, the problem of enhance the total matching figure becomes a problem of impedance compensation between all the loads insisting on the power dividers involved in the system.

It is well known that by quadrature phasing identical loads it is possible to cancel completely the reflections originated from the same loads (see fig. 5).

The quadrature phasing can be realized by quarter wavelength difference in the connecting cables or by means of other devices, for instance 3 dBs couplers. The difference in frequency sensitiveness between the two solutions becomes evident. In other words, both techniques can be adopted (it is a question of cost effectiveness) according to the real bandwidth of the system.

If the broadbanding is limited, to say about 1/4, 1/3 up to 1/2 of the overall band, the cable quadrature can satisfactorily solve the problem, while for a more extended band, up to the whole band, the 3 dBs coupler quadrature phasing is mostly recommended. The sketch in fig. 6 represents the impedance compensation principle just described.

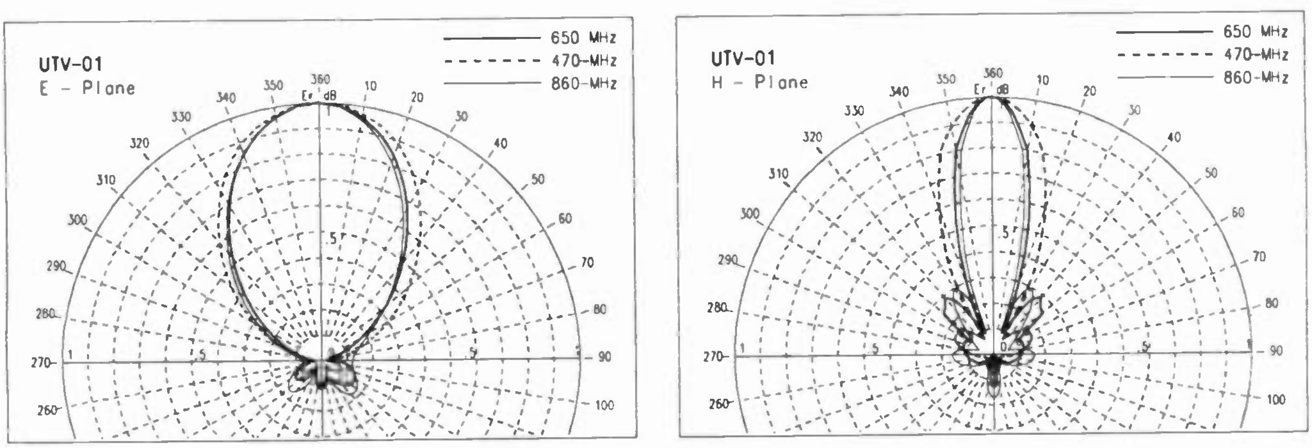


FIG. 2 - UHF TV PANEL RADIATION PATTERNS

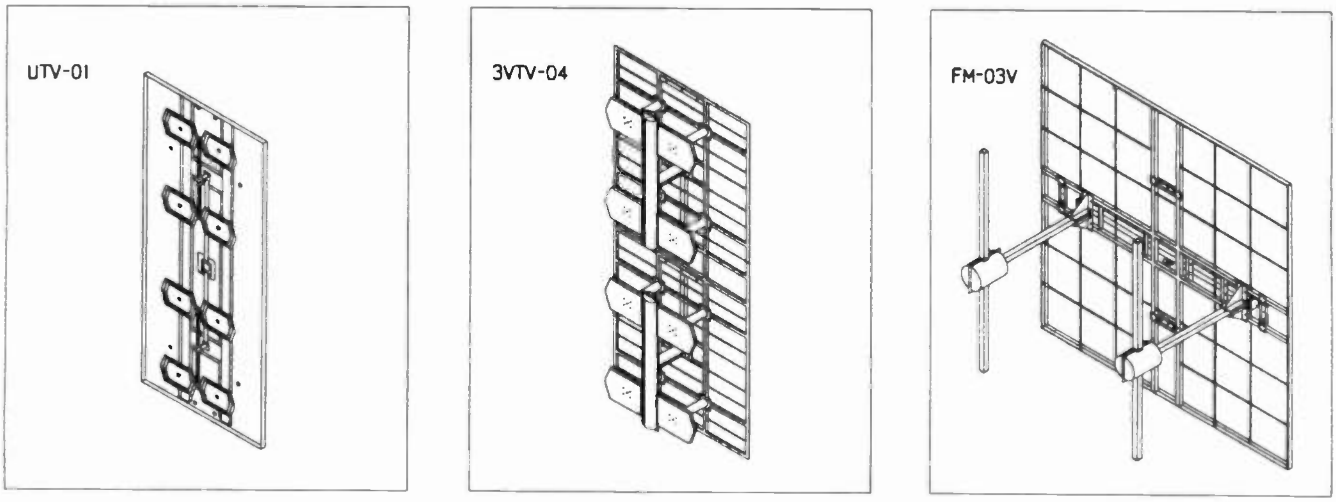


FIG. 3 - UHF - VHF TV AND FM PANEL PICTURES

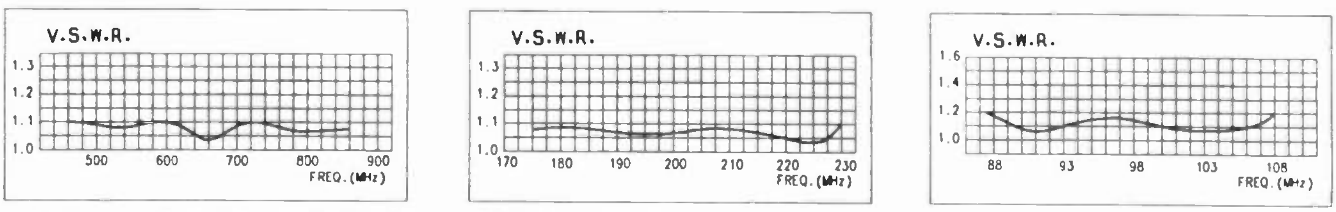


FIG. 4 - V.S.W.R. PERFORMANCE OF PANELS SHOWN IN FIG. 3

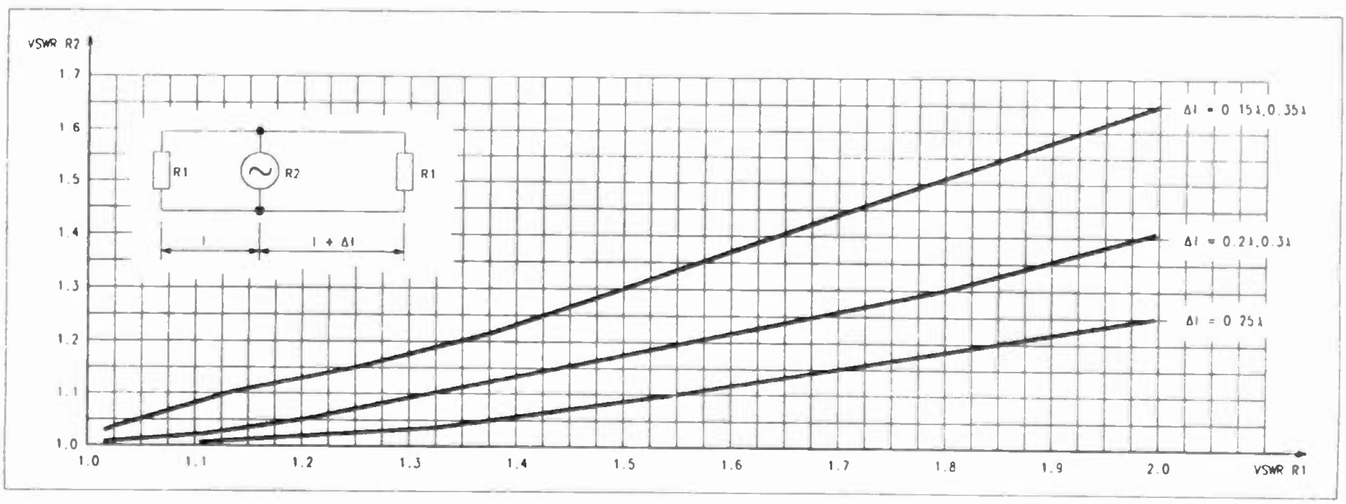


FIG. 5 - REFL. COEFF. R2 OF TWO LOADS WITH R1 REFL. COEFF. (Δ l is the parameter)

The principle is based on the phase-rotation concept that guarantees the complete reflection cancellation if all the loads, in terms of complex impedance figures (i.e. amplitude and phase) are identical. So if you feed four identical dipole array panels, mounted for example around a square tower, in order to achieve a perfect impedance matching that means total reflection cancellation, you must enter four current vectors of same amplitude but quadrature phased.

In other words the signals feeding the four panels are to be phased at 0°/90°/180°/270° respectively. Due to the frequency sensitivity of feeding cables of different lengths, a better practice consists

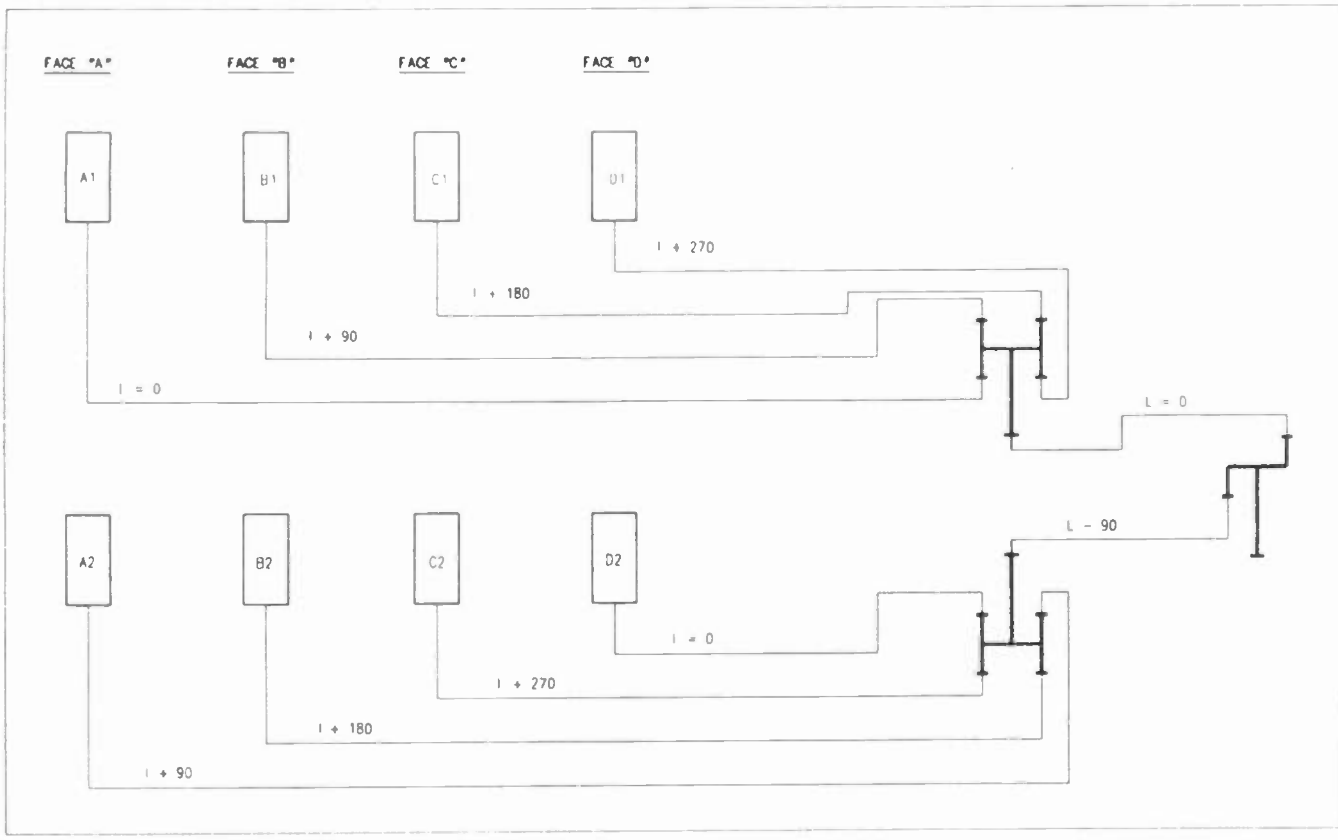


FIG. 6 - FEEDING ARRANGEMENT OF TWO BAYS/FOUR PANELS EACH

of mechanical phase inversion on the panels 180°/270° (exchanging the panels top to bottom) and eventually achieve the quadrature phasing by means of 3 dBs directional couplers.

Unfortunately it is not always possible to use the 3 dBs couplers, especially on non square geometries, as happens with directional antenna systems or with triangular or poligonal geometries (3 sided, 5/6/8 sided tower supports) but it is always possible to adopt the phase-rotation principle for the reflection cancellation, even if it is more frequency dependent due to the cable lengths.

At this point a further amount of additional phase compensation of impedance can be achieved by the application of the so-called "double step phase compensation".

This consists of an additional quadrature phasing between two vertically adjacent bays of panels which are already compensated: it is sufficient to maintain a 90° phasing relationship between the two bays involved and to recover this quadrature by complementary 90° displacement on the power splitter feeding the two bays.

In practical field application, nevertheless, this additional matching technique has not been so intensively adopted, especially because of further frequency sensitivity and because the tilting and null-filling requirements of the antenna system do not always allow for quadrature phasing between two vertically stacked bays.

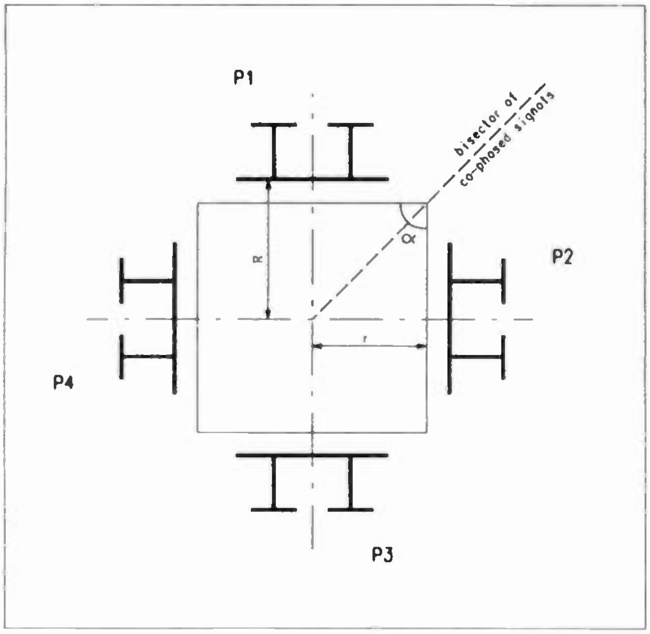
Now, after having emphasized the mechanism of phase compensation within the feeding arrangement of the radiating panels that compose the overall antenna system, we must consider how to recover all the phase displacement introduced for matching, in order to achieve the antenna performance in terms of radiation patterns.

The problem consists essentially in recovering in the space, by mechanical repositioning, what we have electrically altered, in order to obtain the correct vectorial summation in terms of electromagnetic radiated fieldstrength.

Let us refer to the sketches of fig. 7 :

<u>(A) Square tower - 4 panels</u>

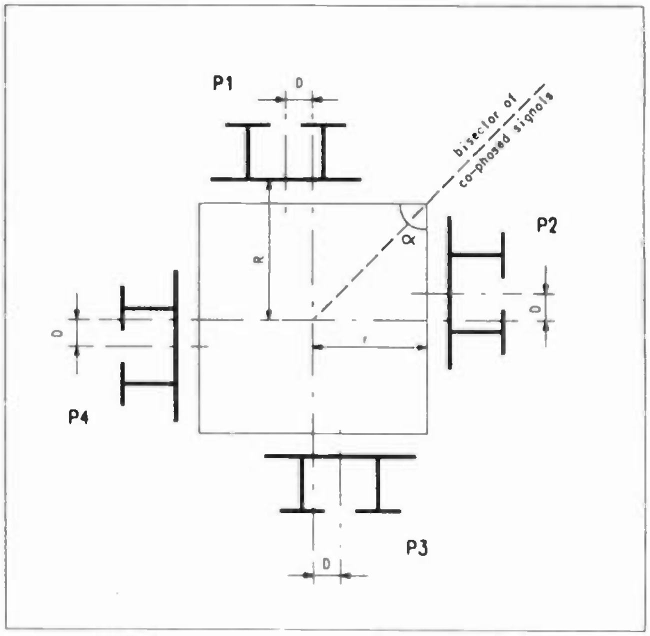
- Mutual quadrature feeding phase allows reflection cancellation on the power splitter



r = tower radius

R = hardware radius

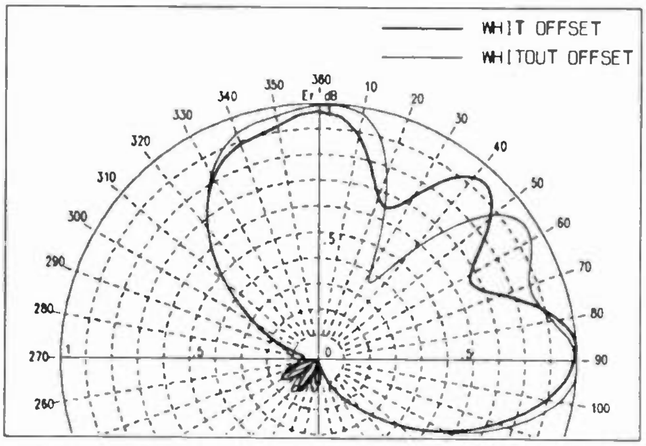
UNCOMPENSATED LAY-OUT



Feeding phase:

P1=0° P2= -90° P3=180° P4= -270° (+90°)

COMPENSATED LAY-OUT



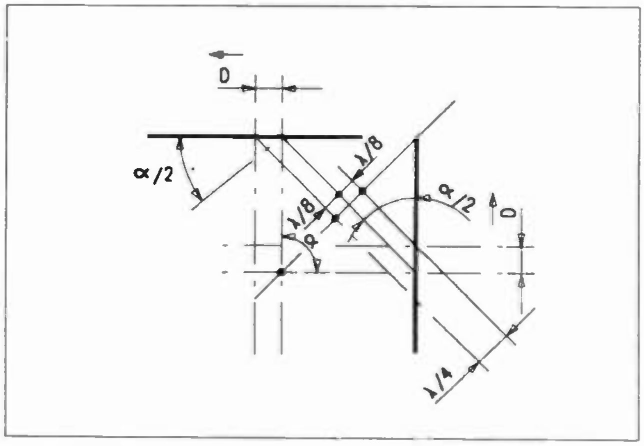


FIG. 7 - OFF-SET COMPENSATION TECHNIQUES ON FOUR-PANELS GEOMETRY

- The off-set necessary to recover the correct phase of the radiated field is achieved by introducing a mechanical displacement of $\lambda/4$ (90°) in order to lead the phase of the panel fed from 90° delayed signal. The opposite situation applies of course if the feeding phase is leading 90°. In other words D is clockwise or counter-clockwise depending on the sign of feeding phases on the panels.
- The amount of off-set D is calculated simply from the geometry of the layout:

$$D = \frac{\lambda/8}{\sin\alpha/2}$$

<u>(B) Square tower - 3 panels or triangular tower - 3 panels</u>

Due to the fact that we can achieve reflection compensation with a feeding arrangement that provides three 60° out of phase signals (120° out of phase reflections), we can calculate in this case too the off-set to be applied:

$$D = \frac{\lambda/12}{\sin\alpha/2}$$

What it is important to emphasize is the following concept: whatever the angle α between the adjacent panels may be, the off-set (space-phase) is to be related only to the electrical feeding phase of the panels. So it will be $\lambda/4$ ($2 \times \lambda/8$) for 90° out of phase, it will be $\lambda/6$ ($2 \times \lambda/12$) for 60° out of phase ...... and so on.

Even in the antenna systems built with one face only (many panels stacked in one direction) it is always possible to compensate the radiation pattern in presence of suitable out of phasing of the feeding signals entering the panels: in this case the off-set consists of staggering the position of the panels in the direction of the radiated field.

A final remark must be made regarding the impedance and radiation pattern of the antenna systems designed with the fore-mentioned criteria. In case of bad atmospheric conditions, as under heavy snow and/or icing, the alteration of the VSWR of the panels has negligible influence on the antenna performance, due to the automatic compensation of the new reflections introduced on the radiating panel (even if not yet protected by suitable radomes).

This insensitivity to the environment is one of the greatest advantages of these broadband off-set compensated antenna systems.

STATE OF THE ART IN VSWR SOPHISTICATIONS

If we refer to the specific argument of matching described in the introduction and generalities, sometimes it looks not so easy to achieve the performances requested by the contractual specifications, especially when more than -32 dBs in the overall channel are involved and/or up to -36/-38 dBs in the visual carrier on multiple channels are rigidly mandatory.

Some more detailed discussion would be necessary in this case in terms of measurement precision and test equipment, but let us consider in this very moment that this problem would not exist or would be satisfactorily resolved. What should be done in order to reach the forementioned high matching figures, even with many channels, when you have already adopted all the compensation tecniques within the antenna feeding arrangement ?

The tecniques, quite recently introduced and studied (1), (2), consider the possibility of adjusting the impedance at some specific frequencies, independently of each other, by means of introducing properly calculated discontinuities (shunt reactances) within the lines of the feeding system, at calculated positions.

These reactances could be either inductive or capacitive, but practically the lumped capacitance looks more convenient and is easily realized by soldering a metallic sleeve onto the inner conductor of a coaxial line.

The lump inductance would require to undercut the inner conductor, so a finer final adjustment would become more difficult. The limits put to the calculation of these discontinuites are essentially that the mismatches to be corrected and consequently the correctors must be relatively small: under these conditions the reflection coefficients may be vectorially combined and are linear versus frequency, so that even the relationship between them and corresponding impedance or admittance trasformation is linear.

In pratice, good results are really achieved if we can work inside a circle (on Smith Chart) of 0,11 (11 percent of refl. coeff.) corresponding to max VSWR of 1,25. It is sufficient to observe the grid of the Chart in this area for considering it approximately linear.

The mathematical explanation of the whole matter would be quite long and too specific to be entirely described here: it may be sufficient to consider that we must resolve a matrix of simultaneous equations where the reflection coefficients of the discontinuities introduced for corrections are to be equal and opposite to the reflection coefficients of the load offered by the antenna. These equations must be resolved for any discountinuity (and its position) and for any frequency involved in the correction procedure.

In short, if we measure the reflection coefficient presented by the antenna system (at input flange) at any frequency which we want to compensate and call these figures $(Ar + jAi)n = 1,2,3, ... N$ where N is the number of correction points, and call the refl. coeff. of the correcting discountinuities $(CDr + jCDi).e^{j\varphi}$ taking into account the electrical position $\varphi/2$ related always to the antenna input flange, the system of simultaneous equations to be resolved assumes the following form:

$$\left[\sum_{h=1}^{N} Kn\,(CDr_h + jCDi_h).e^{jkn\varphi h} = -\,(Arn + jAin)\right]_n$$

with $n = 1,2,3 \;........N$

At this point, where obviously a suitable computer program has been specifically developed, it is necessary to compute, by iterative calculations whose convergence is quite rapid, the positions of dicontinuites where the shunt impedances are pure capacitive reactances, to be realized by means of sleeves on the central conductor of the coax line.
This method of very low VSWR refining on multichannel antenna systems looks quite sophisticated and time consuming: it has, however, proved very satisfactory and not so difficult, especially after some experience in the specific field.
I would now like to spend a few words on a similar, more simple, patching technology that can be considered a compromise between the n frequencies/n correctors and the minimum of correctors for the maximum of frequencies. In both cases (the former and this last one), we must have a suitable rigid coax line in which to place the correcting discountinuites.
Sometimes this line already exists in the antenna system itself, otherwise we have to provide a suitable additional line that normally can be 4 meters long in the UHF band and approx 6/8 meters long in the VHF/TV or FM bands. If we take the measurement of the reflection coefficient (or impedance) at the main antenna input, (exactly as in the former method) we can resort to another computer software, by which we ask to find a position in the line (inside the antenna if this is the case of electrical distance from the antenna flange towards the load) or in the additional line (electrical distance from antenna flange towards the generator in this second case) where all the impedances to be corrected assume the value of a pure inductive reactance (or a minimum of discrepancy from this condition as an acceptable compromise). Just at this point we put a calculated capacitive shunt reactance (sleeve on the inner conductor) that cancels or minimizes the existing inductive components. This method also gives very good results, is intensively applied, is quicker and can be repeated in subsequent steps if the line length is sufficient. Obviously this method has its own limits due to the said compromise of correcting many frequencies with one patch only. In practice, because a well designed antenna system exhibits itself figures of VSWR better than 1,1, with this simple method it is not so difficult to optimize different individual channels (or at least different visual carriers) up to 1,05/1,04 that means -32/-34 dBs in terms of return loss.

SUMMARY

The Sira/MCI panel antenna solves many of the technical problems unique to the American market.

HDTV Capabilities

During the Simulcast Period, a second channel will be assigned for the HDTV signal. With an antenna that covers either the full High VHF (Channel 7-13) or the UHF (Channel 14-69) band, it will be possible to radiate both the NTSC and HDTV signal from the same antenna.

FAA Restrictions

Because of FAA Restrictions, it is increasingly difficult to find tower locations. Many new antennas will have to be mounted on existing towers that already have a top mounted antenna. The Sira/MCI panel antenna is a wrap around antenna; e.g. it can be mounted below the top and it can use the tower as its mount. THe slight increase in ripple content of the azimuth pattern, due to the larger panel spacing, will be much less that the increase in pattern distortion due to a side mounted antenna.

EPA Consideration

Unlike most current TV antennas that use one wavelenght spacing, the basic element spacing of the Sira/MCI panel is a half wavelenght. This will eliminate "all" downward radiation, permitting compliance with the FCC Non-Ionized Radiation Specifications.

Environmental

Most antennas in the American market are series fed with a 1% bandwidth and are therefore, frequency sensitive, high Q radiatiors. The Sira/MCI panel antennas, on the other hand, have a 50% bandwidth and are low Q radiators. The corporate or branch fed harness is not frequency sensitive and eliminates any beam steering effects. The combinations of the corporate harness and the low Q element makes the panel antenna insensitive to elemet impedance changes due to ice build up, eliminating the need for deicers.

EXHIBITIONS AND REFERENCES

Finally I would like to show some pictures, not for commercial or promotional reasons, of practical realizations of all the techniques I have just described (fig. 8).

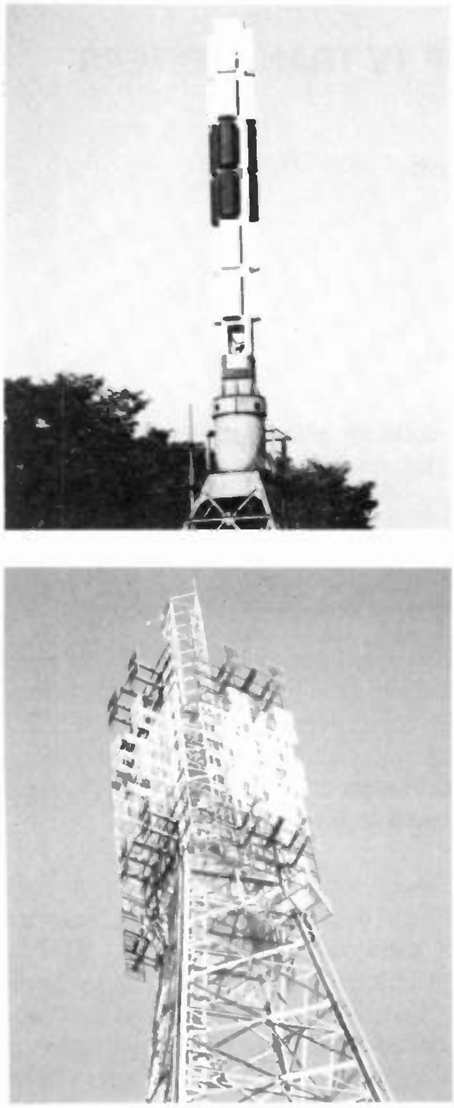


FIG. 8 - VHF NEW ZEALAND TV ANTENNA
UHF TV OMNI ANTENNA ON ROTATING TOWER

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IBA Technical Review N. 23/March 1986

(2) M.B. ANDERS - "Impedance correction of multichannel UHF Aerials"
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STATUS REPORT ON HIGH EFFICIENCY UHF TV TRANSMITTERS

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INTRODUCTION

During the 1990 NAB Conference, initial results from WNVT, Falls Church, VA's experience with the first Multiple-Stage Depressed Collector (MSDC) klystron transmitter were presented. At that time, it was projected that the station would save up to $56,000 per year as a result of the higher efficiency. Since that time, a great deal more knowledge has been obtained concerning the installation and operation of an MSDC klystron and transmitter. The objective of this paper is to share that information with the broadcast industry.

Field data has shown that transmitters with the MSDC klystron have met their performance objectives and that using the MSDC klystron is basically the same as using any other ordinary klystron. Twelve transmitters have been installed all over the country, in large and small markets. Power consumption cost data is presented along with other appropriate facts.

MSDC MADE SIMPLE

As you will remember, MSDC klystrons are identical to other broadcast klystrons except in one area, the collector. The MSDC klystron collector is a series of collectors operating at different fixed potentials. As the electron beam travels down the klystron, electron bunching occurs as a result of the drive signal and electric and magnetic fields in the tuned cavities. This means that some electrons may have as much as twice the original velocity while others may have a very small velocity. Therefore, the energy of the electrons varies dramatically. A single voltage collector cannot take advantage of the wide range of energies of the electrons. Deceleration of the electron beam takes place in the collector. The MSDC collector design is such that all electrons, will impinge upon one of the five collectors without reflecting off it and returning to the electron beam. Each collector gathers the beam electrons at different energies, resulting in less dissipation.

KLYSTRON TECHNOLOGY

Depressed collector technology is not new to the world. Depressed collectors have been widely used in military applications for about twenty years. The high cost of electricity, longer programming days, and belt-tightening to stay competitive has driven the domestic TV market to adopt this technology as a common tool.

To date, the chief supplier of MSDC klystrons is Varian. Last year both EEV and Philips announced their plans for supplying similar klystrons. EEV has named its high efficiency klystron the Energy Saving Collector (ESC) while Philips has called its the Philips Depressed Collector (PDC) klystron. By any name, the technology is identical. Significant progress in the development of each of these tubes has been made. And while these tubes may have minor differences on the inside, from an exterior and interface perspective, they are expected to be interchangeable with only minor physical differences. Also, each is expected to provide the same reliability and visual performance as ordinary wideband klystrons. Generally, it is expected that performance will be similar with the ESC, PDC, or MSDC klystron. Each of these suppliers can provide details of their klystrons.

THE BIG PICTURE

Because each chief engineer wants to make sure his equipment is optimized from a reliability, performance, and serviceability aspect, this paper will address each one of those items.

RELIABILITY

Depressed collector klystrons have had the usual start up difficulties that can be expected with any new product. As of the writing of this paper Harris has 33 tubes operating in 12 transmitters. Over 100,000 device hours have now been accumulated which makes Harris feel very comfortable with the design concept and its implementation. The longest tube life to date is about 6500 hours and it is still operating in the original socket. The experience shows pulsing this tube has the same or better reliability than the standard wideband tubes. There have been four actual tube failures. Unfortunately, man has not made a klystron that is not subject to human error. The cause of these failures would have resulted in the premature failure of any klystron -- namely, RF coupling loops installed upside down and reversed coils in the main magnetic field. None of the failures were related to a collector problem in any way. Two of the failures did cause a loss of air time at the station while the other two happened before the transmitters were officially broadcasting. The collector ceramic seals have performed as expected. The water circuits for the MSDC, although initially a small concern, have also proven their reliability. Our experience indicates that MSDC klystrons suffer failures from human installation errors and are no more prone to failure than previous generations of klystrons.

PERFORMANCE

Table 1 identifies typical operation of a MSDC klystron in a Harris TV-60UM transmitter. This is an average of the data collected.

TABLE 1

RF Drive Power	20-75 watts (channel dependent)
Body Current	45 milliamps
Pulser Voltage	1300 volts
Magnet Current	11.5 amps
Cathode Voltage	26.5 kilovolts
Filament Voltage	7 volts
Figure of Merit	1.30

Measured audible noise on a 120 kW transmitter is only 68 dBa at the front of the transmitter, while figures of merit of 1.40 have been measured.

SERVICEABILITY

Figure 1 is a photograph of an MSDC klystron installed in a Harris TV-60UM MSDC TV transmitter. One MSDC klystron is mounted in each amplifier cabinet and each klystron has its own beam supply. This makes it possible to service one klystron while the other is still operational. For this picture, the side panels have been removed. Excluding the previously mentioned failures, the MSDC klystron has enjoyed a relatively maintenance-free year of operation. When the failed tubes have had to be removed and replaced, the process has been straightforward. In one instance, a pre-tuned tube was installed and absolutely no touch-up to the cavity tuning, RF drive power, or pre-correction circuits was required. Ceramic cleaning has not been required at all. The high purity water system has performed as it should with no reports of having to change filters or flush out the system. The water system has had one improvement. The pumps for the external heat exchanger and the high purity water pumps have been combined in one cabinet, simplifying the overall installation. No evidence of scaling or other impurities has been found on the collectors of the klystron.

On one of the initial installations, video currents on one of the beam supply leads were sufficient to trip a spark gap, but filtering has resolved that problem. Also, care must be used when routing the HV wire to ensure that none of the shield braid penetrates the insulation. This will prevent an high voltage arc, as in any other klystron installation. However, there are more high voltage leads with the MSDC klystron.

POWER CONSUMPTION

The MSDC klystron was developed to reduce a/c power consumption. With that in mind, data on several stations with operating MSDC klystrons can now be presented.

Table 2 presents some results in operational savings from five stations who were on the air as of this writing.

Table 3 presents data for three new stations for which no operating cost comparisons could be made but does indicate monthly power costs.

Figure 1

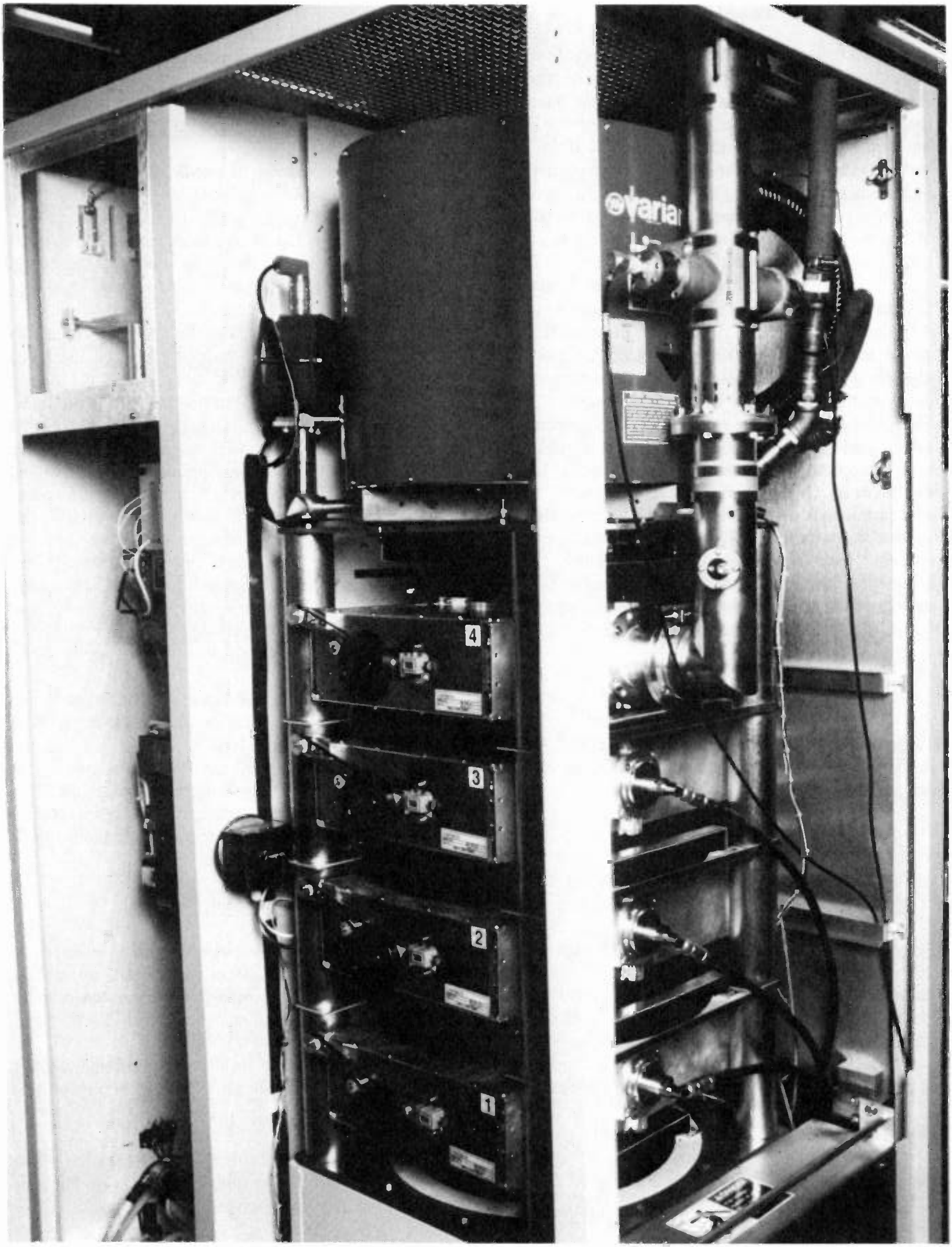

TABLE 2

STATION	PREVIOUS TRANSMITTER POWER	NEW TRANSMITTER POWER	OPERATIONAL SAVINGS PER MONTH
WNVT	55 kW	120 kW	$4,600
WFLD	120 kW	120 kW	$7,500
KITN	51 kW	120 kW	$0[@]
WPTF	220 kW 240 kW	240 kW 240 kW	$11,500[#] $3,000[#]
WEYI	55 kW	60 kW	$2000[a]

TABLE 3

STATION	PREVIOUS TRANSMITTER POWER	NEW TRANSMITTER POWER	AVERAGE MONTHLY POWER COST
WBSG	N/A	60 kW	$4,049
WKOP	N/A	60 kW	$2,800
WLAJ	N/A	60 kW	$6,054[b]

[@] Calculated based on station power consumption.
[#] Station estimate.
[a] Most of savings is offset by additional heating load of co-located transmitter and studio.
[b] Includes approximately 10 kW of other load than transmitter and .094/kwh rate.

ACCEPTANCE

Since the introduction of the MSDC, some people may have been concerned that ordinary wideband tube transmitters would remain the dominant product. It was not certain how long it would take until the MSDC demonstrated its performance, established credibility, and gained acceptance. What we have found is that this process took place very rapidly and acceptance of the MSDC has been quickly achieved as evidenced by field installations. After examining the events that have taken place, it is not so surprising. This response is a result of the evolutionary nature of technology, the savings potential, and the field results.

FUTURE INSTALLATIONS

Another major step was taken towards the next generation of high power UHF MSDC television transmitters recently when station WUNG in Charlotte, North Carolina purchased a 280 kW TVD-280UM transmitter. This equipment will use four visual klystrons operating at 70 kW each and two aural klystrons. Rather than using four-cavity tubes, this transmitter will use five-cavity tubes. The MSDC klystrons themselves are being built by English Electric Valve (EEV). The EEV klystron, although using five cavities, will still use the same number of collector stages as the Varian klystron. Therefore, the same beam power supply can be used on either tube.

CONCLUSION

The performance of the MSDC klystrons after over one year of field experience is very encouraging. No collector failures have occurred. The Figure of Merit is averaging 1.30 and the expected a/c power reductions have been achieved. A station can now update its facility with a new UHF transmitter that consumes half as much power for equivalent power output levels or, double its power output for the same energy as the present plant uses.

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3) Don Kirby, Station KITN, Minneapolis, Minnesota

4) Mike Knight, Station WKOP, Knoxville, Tennessee

5) Chuck Britt, Station WPTF, Raleigh, N. C.

6) Merlin Miller, Station WBSG, St. Simons, Georgia

7) Larry Estlack, Station WLAJ, Lansing, Michigan

8) James Barnes, Station WEYI, Clio, Michigan

AIR-COOLED COMMON AMPLIFICATION TV TRANSMITTER AT 120 KW: UHF BREAKTHROUGH

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This paper will describe the theory of operation and design as well as the physical implementation of a 120kW air-cooled, UHF television transmitter operating in common amplification. This breakthrough technology represents the first time that a UHF transmitter using common amplification on a full time basis has produced 120kW. It is particularly interesting that the transmitter is also air-cooled.

I. INTRODUCTION

The amplification at output power levels of both the vision and sound carriers in a single device is very intriguing to the transmitter designer and user. Simplified transmitter and RF systems, eliminating large and costly diplexers and switchers, and parallel high power output stages for redundancy are among the attractive features offered by the common amplification of both vision and sound carriers. The disadvantage usually associated with this technique are higher levels of distortion and low energy efficiency. Comark has perfected the linearization of the klystrode[R](IOT) device to a level that is equivalent to a diplexed transmitting system.

The Class B operation of the class of devices known as IOTs or klystrodes provides energy efficiencies that rival the most advanced klystron transmitter. With this advanced efficiency, it is possible to combine linear common amplification technology with air-cooling since no pulsing is required to obtain the efficiency.

This paper will describe a 120kW air-cooled common amplification based transmitter, manufactured by Comark and installed at WSNS-TV Channel 44 in Chicago's John Hancock Building during 1990. The impact of this transmitting system on operational reliability through parallel redundancy will be described. In addition, the FCC requirement to protect the BTSC pilot carrier frequency (73.682(c)(3)) and its impact on a common amplification transmitter will be discussed. Comark's patent pending solution to this requirement will be described.

Finally, cooling system design, noise level control and operating test data will be presented.

II. TRANSMITTER SYSTEM - RF

The design of a common amplification based transmitter allows for high levels of parallel redundancy since both visual and aural signals are always present together in each high power amplifier. Comark has chosen to use the term common amplification to describe this technology rather than the term multiplex operation. The reason for this is that common amplification implies full specification compliance while multiplex implies emergency operation outside of specifications resulting in substandard broadcast performance.

The requirements at WSNS-TV 44 were for a 120kW TPO using air-cooling to reduce dependance on the building's internal facilities. The basic building block for the transmitter would be the 30kW air-cooled Comark amplifier using the Eimac 2KDX40LF klystrode tube. Performance of this amplifier has been documented in previous papers[1,2]. With advanced linearity correction, the intermodulation of the amplifier is held to minus 60db or below. This is a level below expected video signal to noise levels and provides large margins for full broadcast quality.

A second linearity concern with any common amplification TV transmitter, regardless of amplifier technology, is the contamination of the aural signal with H sync phase modulation. This problem has been resolved by Comark and will be addressed later in this paper.

The 120kW requirement for WSNS-TV 44 in the John Hancock Building in Chicago was satisfied with four Comark 30kW amplifiers in parallel pairs.

Each parallel pair of amplifiers is driven by its own driver and exciter including its own modulator and correction systems. Thus, the 120kW transmitter is actually a pair of independent 60kW transmitters each consisting of a pair of parallel 30kW amplifiers.

Figure 1 is a block diagram of the 120kW transmitter. The independent modulators are operating in coherence. Each modulator's linearity correction circuits are adjusted for its specific pair of tubes.

The result of this configuration is deep redundancy with extraordinary linearity correction capability.

Each klystrode amplifier is combined in a 3db hybrid and then the hybrid pairs are combined in a Magic Tee. The Magic Tee can switch the output of either pair of amplifiers into the antenna, thus permitting 50% power output if either side should be shut down. The Magic Tee does not use moving contacts that are under RF power to effectuate the switching function. This enhances overall system reliability.

The complete transmitter uses two output feedback loops (Figure 1) to stabilize output power of each independent output pair. This ensures constant linear operation of each pair of amplifiers independent of the other pair. Constant output level ensures that linearity pre-correction requirements remain stable and unchanged on a long term basis.

After the Magic Tee combiner is a series of notch filters which reduce the out of band intermodulation products to levels well below FCC requirements. The specific frequency of these products is covered in an earlier paper[1].

Finally, the output switch is the only mechanical switch in the system. It is used to route the full filtered output power to either the antenna or the station's dummy load.

III. TRANSMITTER SYSTEM - COOLING

Figure 2 shows the cooling arrangement for the 120kW transmitter. Each amplifier uses its own designated blower. A significant concern during the cooling system design was ambient noise levels. This concern dictated that the blowers should be located in an air handling vault with sound isolation from the transmitting room and in line silencers on the air feed ducts to the tubes.

Additional design concerns were input air cleanliness and regulation of the input air temperature. Finally, it was necessary to limit the heat exchanged into the transmitter room from the warm exhaust ducts.

Figure 3 is a simplified site layout for this transmitter. The air handling vault is located on an outside wall of the building. Outside air is brought into the otherwise sealed vault through an elaborate multistage window filter. This air is mixed with the vault air, some of which can be supplied from the transmitter exhaust ducts to modify the ambient temperatures in winter. The four pressure blowers, with their in line noise silencers, draw air from the vault and provide flow to the amplifier tubes.

A common exhaust plenum routes the heated air to the outside of the building. Thermostatically controlled louvers permit some exhaust air to be bled to the air handling vault for temperature modification of the input air.

Figure 4 is a view of a pair of blowers in the air vault. The inline silencers are clearly visible on the output of each blower. Noise levels need to be suppressed from the air ducting. Sound proofing material was installed on surfaces of the air ducting. This metal covered fiberglass material both quieted the duct noise and insulated the ducts to reduce heat radiation.

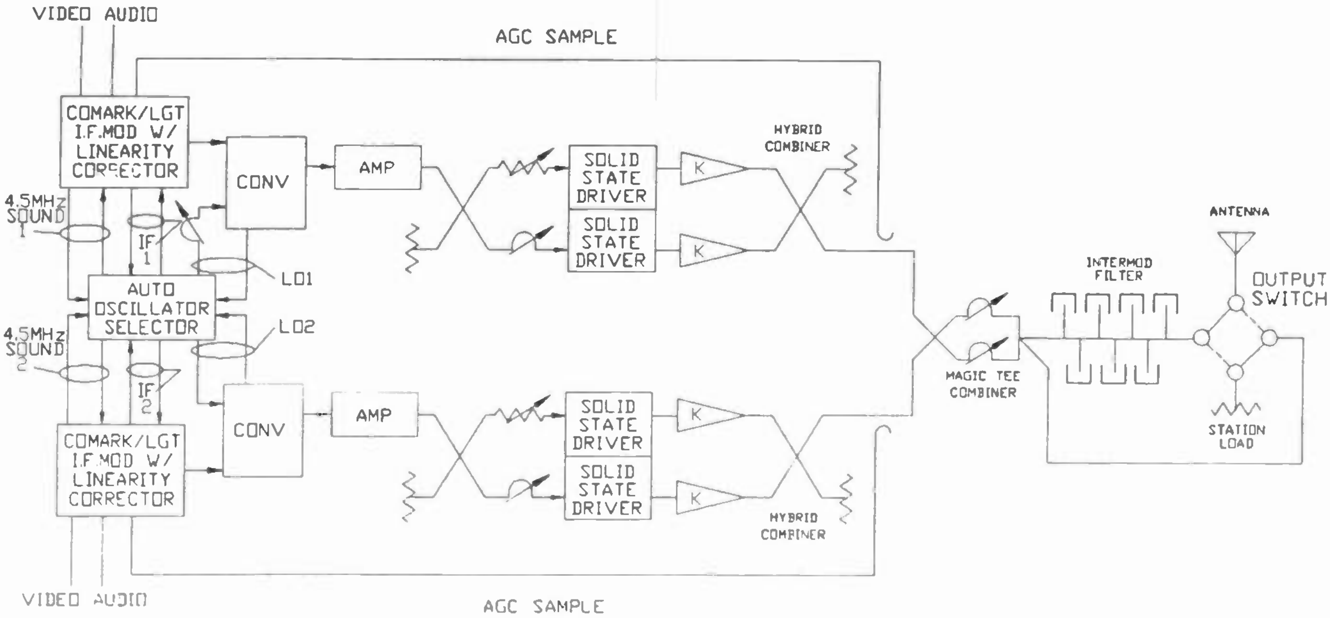

Figure 1 - 120kW Simplified Block Diagram

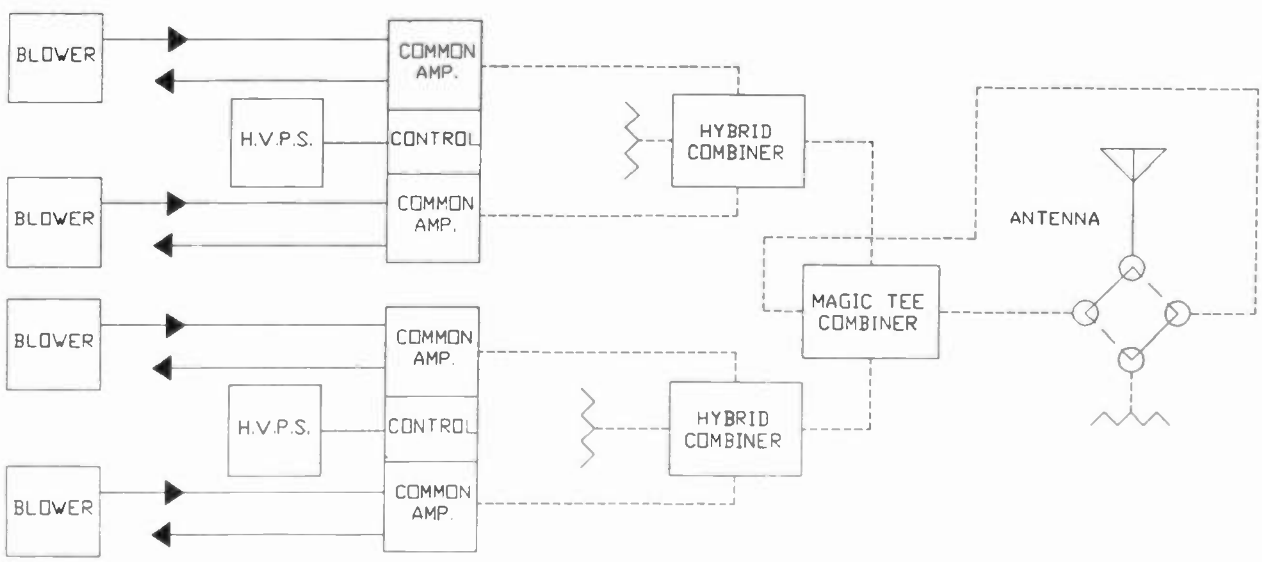

Figure 2 - 120kW Air Cooling Schematic

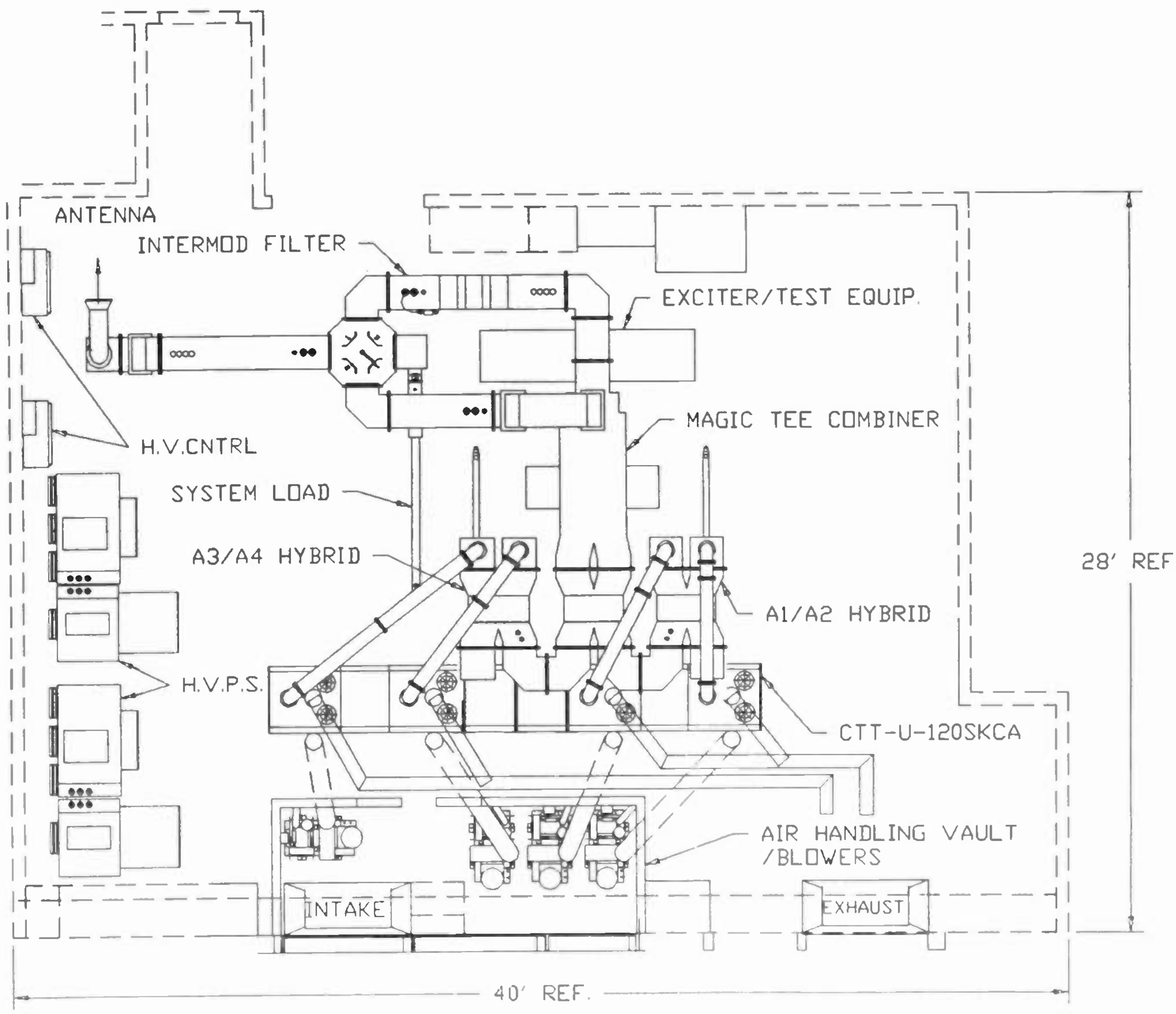

Figure 3 - Simplified Layout WSNS-TV 44
John Hancock Building, 97th Floor

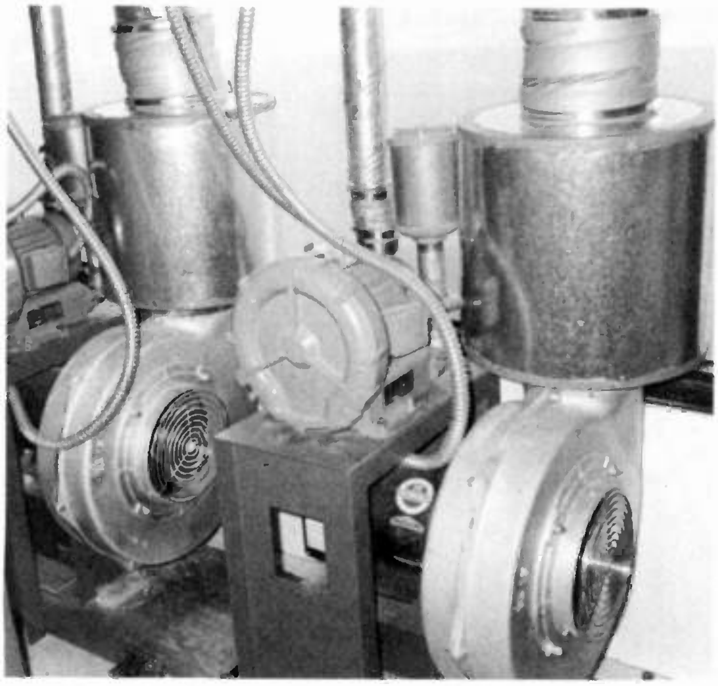

Figure 4 - Pressure Blowers in Air Vault
WSNS-TV 44

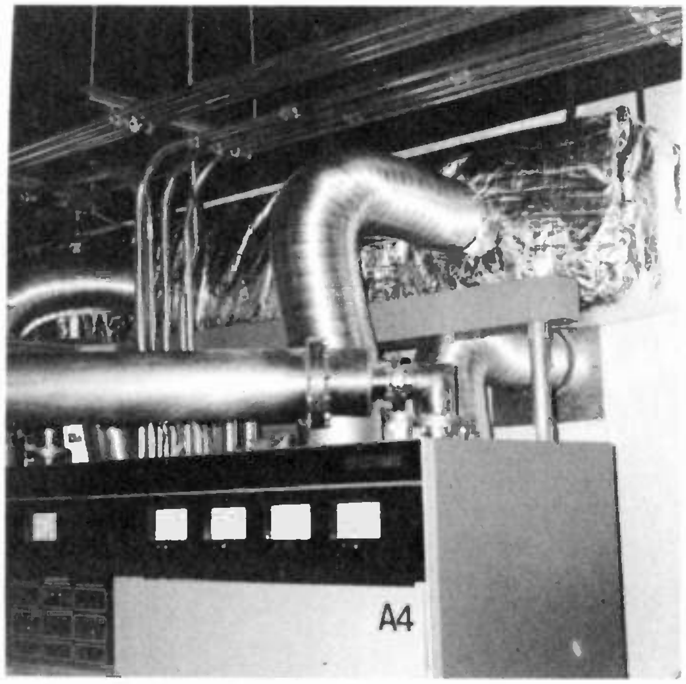

Figure 5 - Input Ducts & Output Air Plenums
WSNS-TV 44

Figure 5 is a view of the input and exhaust lines including the exhaust plenum.

The use of air-cooling always requires that the altitude of operation be considered. Increased altitude requires both higher delivered back pressure and volume, referenced to sea level, to achieve the same cooling results as would be obtained at sea level. The sea level performance of a blower is the reference point upon which pressure and flow are based. Since this transmitter was on the 97th floor of the building, altitude was a consideration.

The sea level blower performance required for each tube to operate at altitude was calculated to be 1,460CFM at 16.75 inches of water. The maximum output temperature rise for each tube was measured at 25°C. The ambient noise at the front of the transmitter was measured at less than 80dba. Noise measurements made during the same period of time on a new water-cooled UHF transmitter in the Hancock Building showed comparable noise levels.

Figure 6 is a view of the air-cooled klystrode and its hardware installed in one of the amplifier cabinets.

IV. LINEARITY CORRECTION AND BTSC PILOT FREQUENCY PROTECTION

Any high power amplifying device must establish a trade off between output power linearity and operating efficiency.

For example, the closer to saturation a Class A amplifier operates, the higher its efficiency. However, at high saturation levels, linearity is sacrificed. In a Class B amplifier, like the klystrode (IOT) or tetrode, the beam or plate current follows the drive power level giving better average efficiency. However, saturation effects still exist and the varying beam or plate current introduces additional dynamic changes in the device's input and output impedance and transfer characteristic.

Figure 6 - 30kW Air-Cooled Klystrode Tube and Hardware installed in one of WSNS-TV 44 Amplifier Cabinets

The non-linear effects of a high power amplifier create intermodulation when both visual and aural signals are present simultaneously. The standard solution to this problem is to precorrect the drive signal with a non-linear transfer characteristic that cancels the distortion of the output device. This technique works well and has been shown to reduce intermodulation by as much as 20db.[1,2]

With the introduction of BTSC stereo requirements by the FCC, specifically FCC Specification 73.682(c)(3), it is necessary to provide protection to the 15.734kHz stereo pilot frequency in the broadband demodulated aural carrier spectrum. Comark believes that this is true even if the transmitter is used in monoaural service. This requirement places new and severe demands on a common amplification based transmitter.

The non-linear effects of any HPA will cause both phase and amplitude modulation of the aural carrier by the video and sync modulation on the visual carrier. Even if the precorrection is optimized for intermodulation reduction, the aural carrier will not necessarily be properly corrected. This is evident by viewing the aural carrier from a multiplexed transmitter. AM video modulation will be detected. If the aural carrier is demodulated in a broadband FM demodulator and the resulting baseband audio is viewed on an audio spectrum analyzer, 15.734kHz components and harmonics will be seen.

Figure 7 is the broadband audio spectrum of the 120kW transmitter without aural carrier correction. This is what can be expected from a typical multiplexed HPA, whether klystron, klystrode or tetrode. The top graticule line represents 25kHz of deviation. The level of 15.734kHz is measured at -30db or about 850Hz of deviation. The picket fence spectrum of 15.734kHz harmonics, while not individually violating FCC specs, will collectively contribute to a large peak over modulation of the aural carrier by at least 6,000Hz when modulated at normal level with audio. This will cause problems with FCC 73.1570(a) and 73.1570(b)(3). The FCC Specification 73.682(c)(3) states that the BTSC pilot frequency must not contain energy at a greater level than that represented by 125Hz of deviation in a 20Hz bandwidth. Thus, any common amplification transmitter would be illegal for broadcast service in the U.S. if this problem is not solved.

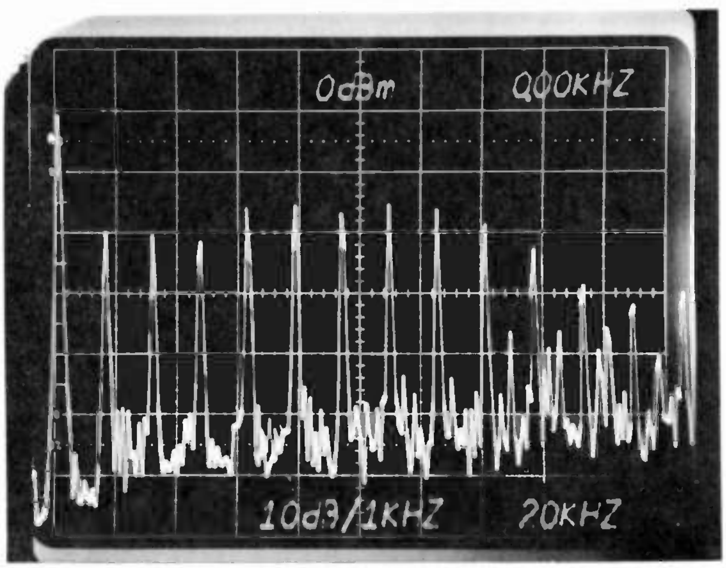

Figure 7 - Uncorrected Broadband Audio Spectrum showing 15.734kHz Components

Recent measurements have been made on uncorrected klystron and tetrode transmitters and the problem of BTSC pilot carrier protection has been found to exist. This problem is insidious because it is not apparent during normal monoaural operation and may not show itself if a normal monoaural aural proof is run on the transmitter. The problem only becomes apparent when the audio baseband of interest is expanded to BTSC widths and peak deviation measurements are made with no input modulation.

If this spec is not achieved, the performance of monoaural transmitters when received by a stereo receiver will be degraded and the quality of BTSC reception of SAP and PRO channels can be damaged. Unwanted stereo pilot light operation and activation of stereo decoder circuits will take place in the MTS TV receiver.

Obviously, FCC 73.682(c)(3) is a very important spec to a common amplification transmitter designer and user. Comark has developed an aural carrier corrector that operates independently of the visual corrector and reduces all levels of 15.734kHz and its harmonics to well below FCC requirements. An Application for a U.S. Patent has been submitted to protect this technology.

Figure 8 shows the same spectrum as Figure 7 except the patent applied for aural carrier corrector is operating. Note that the 15.734kHz energy is now 58dB below the 25kHz deviation reference and all harmonics have been similarly surpressed. This represents an equivalent deviation of approximately 30Hz at 15.734kHz, a 28db improvement, and is well within the FCC limits of 125Hz. Tests at WSNS-TV have shown that the Comark transmitter correction is stable over both time and output power level. The reduced 15.734kHz harmonics also permit meeting the requirements of FCC 73.1570(a) and 73.1570(b)(3).

Figure 9 is a generalized block diagram of the aural carrier corrector.

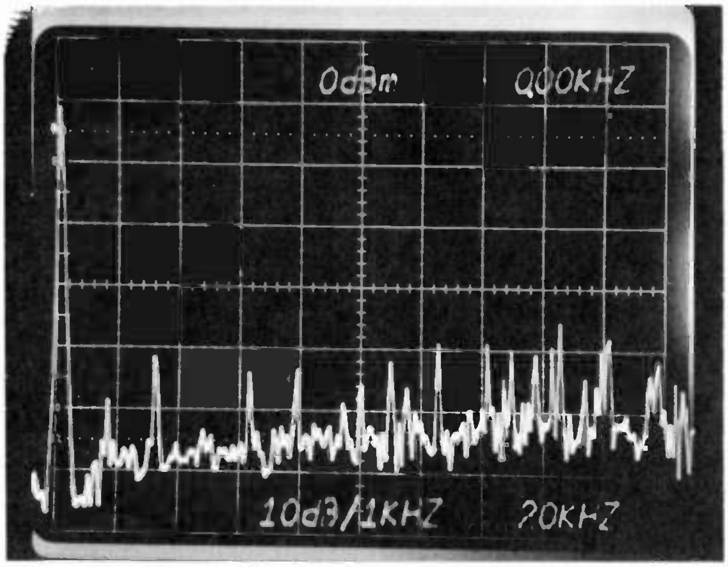

Figure 8 - Corrected Broadband Audio Spectrum showing effects of Comark Aural Carrier Corrector

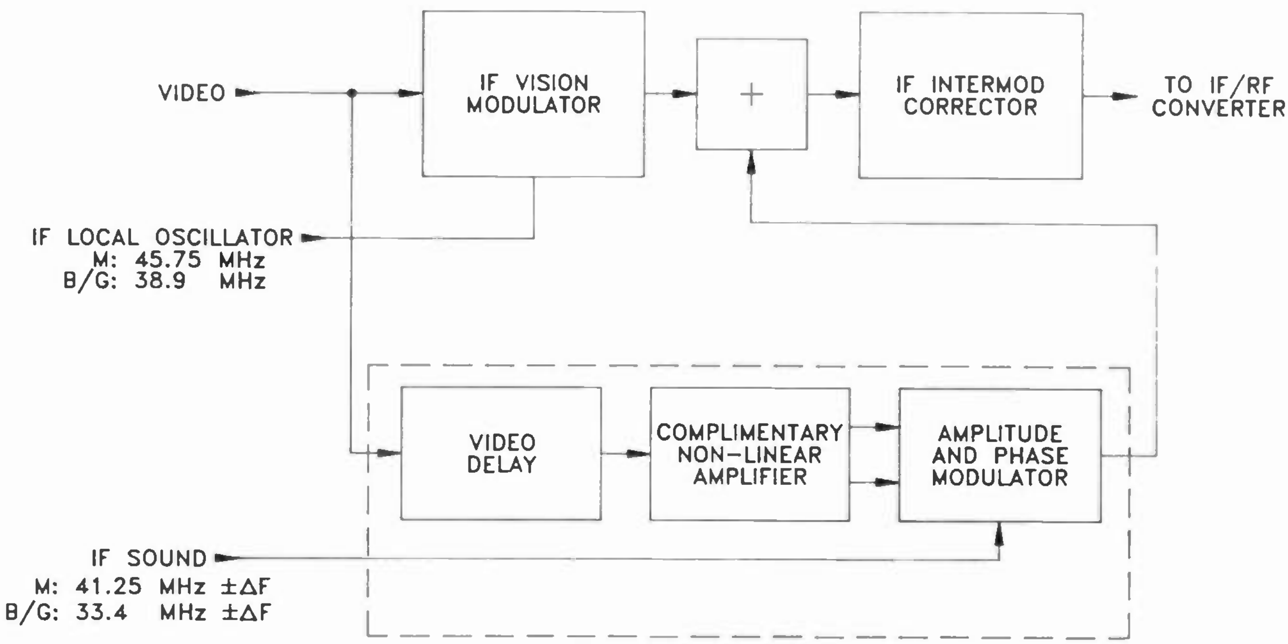

Figure 9 - Simplified Aural Carrier Corrector Block Diagram

V. TRANSMITTER PERFORMANCE DATA

In general, the 120kW transmitter at WSNS-TV Channel 44 exceeded all specifications with comfortable margins. The linearity requirements for common amplification, when met, produced extremely clean video waveforms. The precorrection systems for visual and aural carriers performed as designed and have proven to be stable over a wide range of operating conditions.

The average Figure of Merit for the four common amplification amplifiers was 90%. The definition of FOM for common amplification is:

$$\text{FOM} = \frac{\text{Peak Visual + Average Aural Power}}{\text{Average D.C. Input}}$$

The following are a representative sample of performance data.

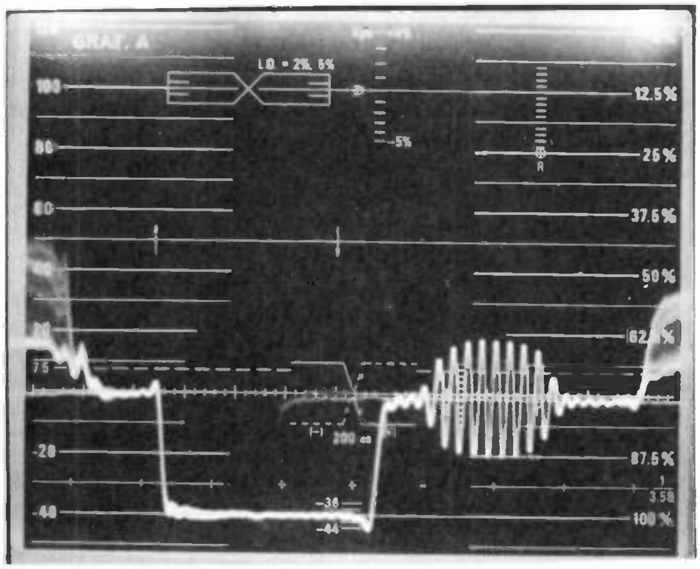

Figure 11 - Horizontal Sync

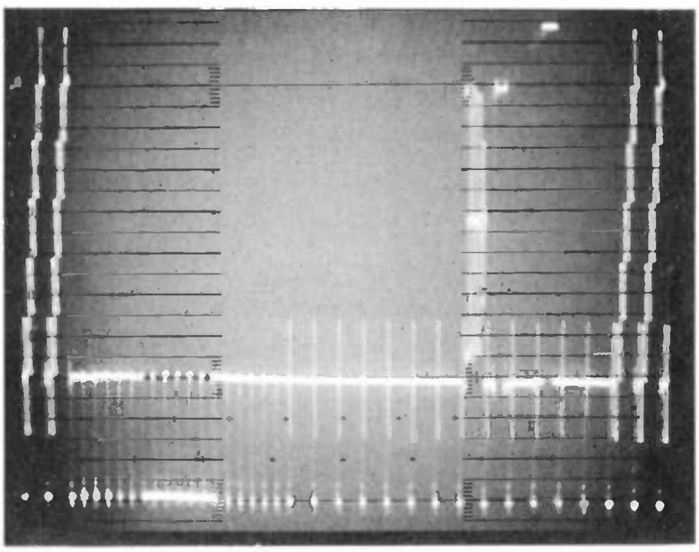

Figure 10 - Expanded Vertical Interval

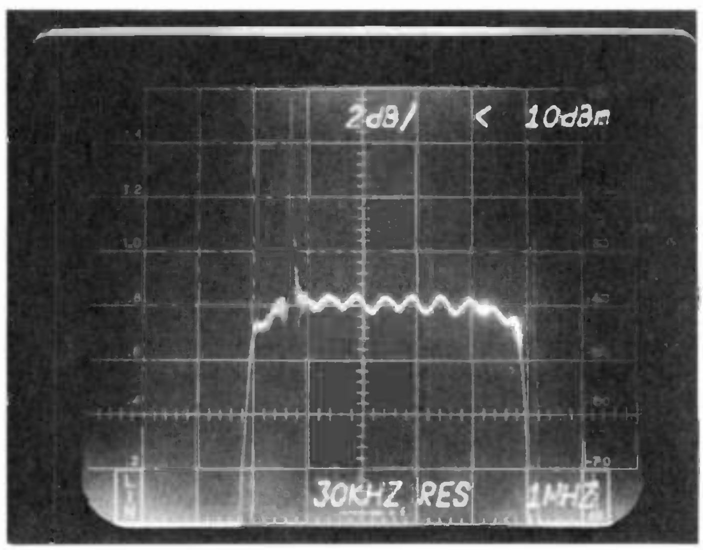

Figure 12 - In-Band Amplitude Response 2db/cm

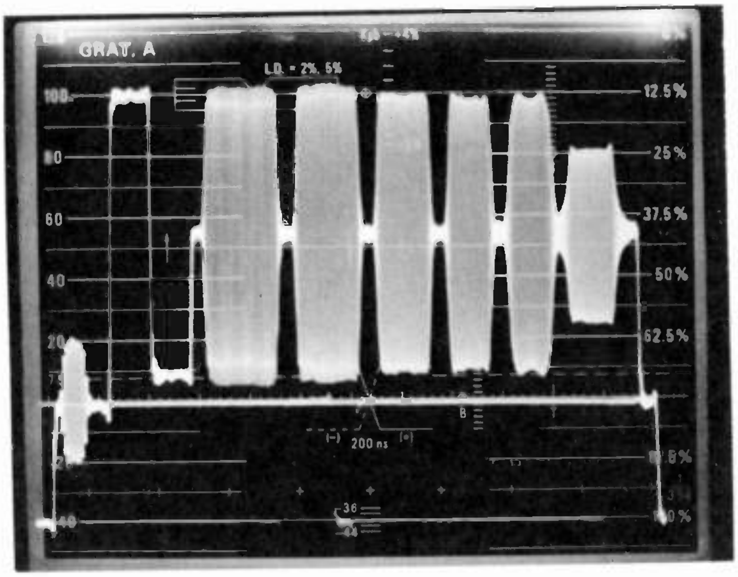

Figure 13 - Multiburst

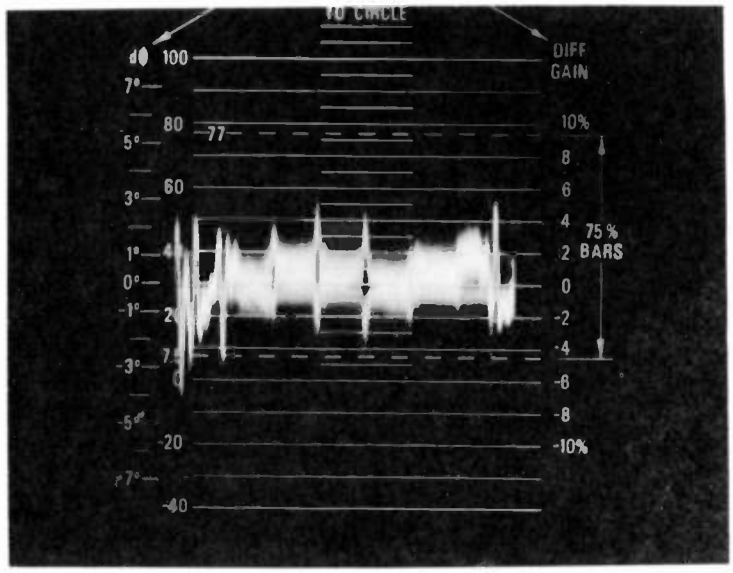

Figure 15 - Differential Phase showing less than ± 1°

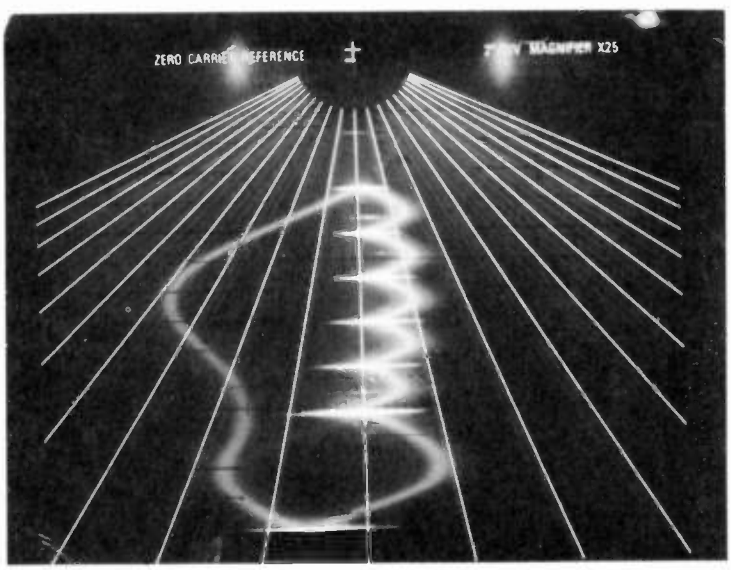

Figure 14 - ICPM showing less than 1°

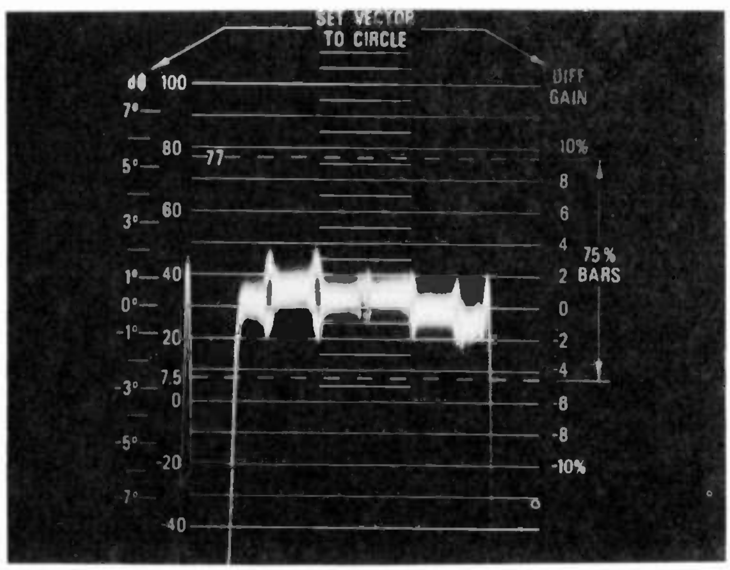

Figure 16 - Differential Gain showing less than 2%

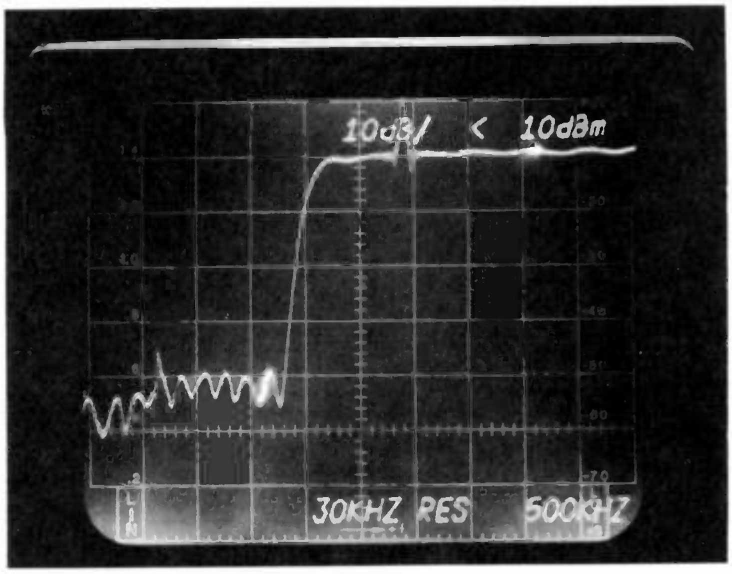

Figure 17 - Lower Sideband Rejection shown at better than 40db. A good measure of overall linearity.

Figures 18 and 19 show the output spectrum of this common amplification transmitter. The normal intermodulation frequencies of visual carrier + 9MHz and visual carrier -4.5MHz show intermodulation levels to be below 60db. In band levels are equally suppressed.

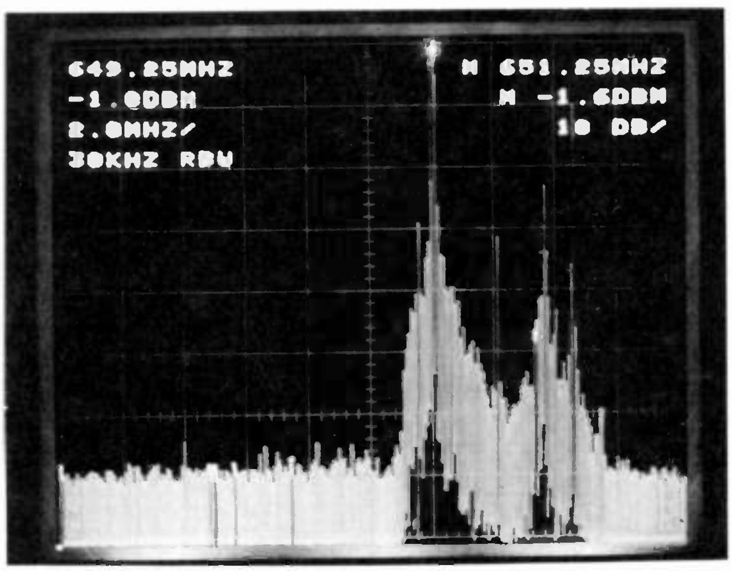

Figure 18 - Output Spectrum with emphasis on the lower frequency region

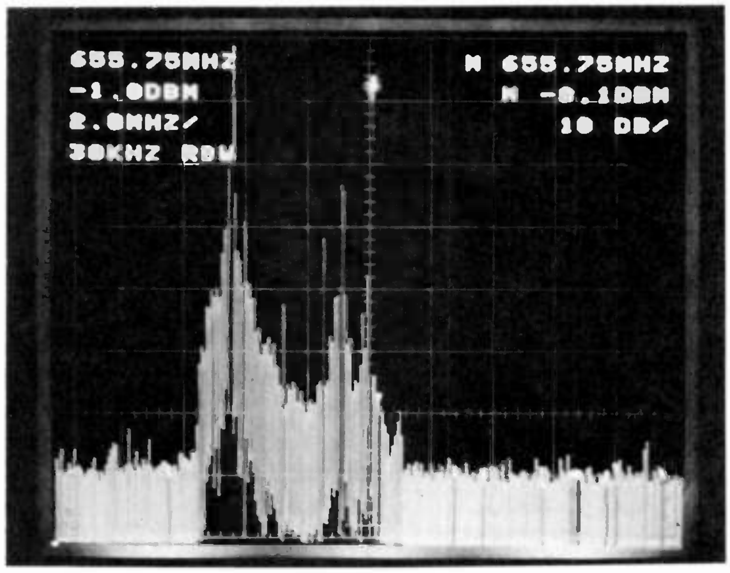

Figure 19 - Output Spectrum with emphasis on upper frequency region

The levels of H sync contamination of the aural carrier were previously shown in Figure 8.

VI. CONCLUSIONS

The state of the art of common amplification transmitters has advanced to the level where full high power broadcast service can be provided with performance levels meeting or exceeding more complex advanced technology diplexed systems.

The significance of the WSNS-TV installation is that it demonstrates conclusively the effectiveness of this technology in an operating field environment.

The impact of the aural carrier correction is that now common amplification technology is fully equivalent to diplexed technology for broadcast quality service.

Air cooling combined with common amplification permits the creation of a simple, reliable, highly redundant and efficient transmitter.

Whether air or water-cooled, common amplification will play a significant role in the future of high power UHF broadcasting.

Acknowledgement

The authors wish to express their appreciation for the support and guidance provided by Mr. Dane E. Ericksen of the firm of Hammett & Edison, Inc. and for the use of some of his data.

The authors also wish to thank Mr. Jim Church, Chief Engineer of WSNS-TV for his cooperation.

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[1]"Recent Advances in Klystrode[R] Equipped Transmitters", Nat S. Ostroff. Presented at the National Association of Broadcasters 68th Annual Convention & International Exposition, April 1, 1990

[2]"Klystrode/IOT Technology Update", Nat S. Ostroff. Presented at the 1990 SBE Convention.

[R]Klystrode is a registered trademark of Varian Associates, Inc.

UPDATE: DIGITAL TELEVISION BROADCAST TRANSMITTERS

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INTRODUCTION

In the summer of 1987 an idea was conceived whereby the digital synthesis of an RF television transmission was mathematically possible. In fact, not just television, but any arbitrary bandlimited amplitude modulated transmission was determined to be possible, theoretically. The general derivation of the mathematics was developed while always looking for the fatal flaw and lack of generality and none could be found. In the fall of 1987, a patent search was conducted looking for all previous work relating to this concept. Only dimly related patents could be found, but nothing quite as revolutionary as this concept. So, a patent application was filed with the U.S. Patent Office in December of 1987, oddly enough, one day before the December issue of Broadcast Engineering was made public. This is important because the entire contents of the patent application appeared in this issue as a technical article which was originally set aside to be written for an entirely different purpose.[1]

The magazine article was to be an overview of the various types of RF power combiners, but it wasn't long after beginning the work on this manuscript that the specific use of power combiners connected in a particular fashion looked like something new, unique and potentially very important. Needless to say, the direction and content of that manuscript changed instantly and the paper delivered to Broadcast Engineering was the result.

Some letters were received from the readership of that article. Considerable interest was given to application other than television broadcasting and no one who reviewed it was able to find any flaws or reason why it shouldn't work. This was encouraging.

The complete concept was presented at the 1988 NAB Convention[2] and appears in the proceedings, but nothing was done further during all of 1988 while awaiting the decision of the Patent Office. After all, what would be the point if it were really old news buried in some obscure corner of some obscure lab?

The patent was granted on February 14, 1989 with number 4,804,931 with myself as the inventor and Acrodyne Industries as the asignee.

From when the patent was granted until about mid summer of 1989, a period of about five months, a casual "component search" took place identifying available electronic parts that would be needed to actually construct a real working model of a real digital TV transmitter at some low level of power. Surprisingly, everything existed - even the most crucial item, a ten bit analog to digital converter capable of sampling well past the Nyquist rate required for the 4.1 MHz video bandwidth. The frosting on the A/D converter cake was that it was available on a demonstration printed circuit board complete with a digital to analog converter so that verification that the A/D was working was available to be viewed as reconstructed digital video on an oscilloscope. One was purchased and set up in the Acrodyne lab. The performance of this A/D demonstration board was most impressive and it is the one used in the development of the transmitter.

It was learned that summer (1989) that the state of Pennsylvania offers technical development grants to certain businesses within the state that qualify. These grants are for the purpose of fostering new technology developments which have the potential for increasing employment, increasing tax revenues, etc. for the general welfare of the Commonwealth. The proposal package was put together in the fall of 1989 and it was learned that Acrodyne would receive a Ben Franklin Seed Grant, as it is known, to support an effort to prove feasibility of the concept of a Digital Television Broadcast Transmitter during 1990. We accepted the grant and the challenge to make something work. Today I am here to report to the Broadcasting community that it does work!

A Digital Transmitter

Well I've said that it works, but why is that so important? We already have good TV transmitters and many of them, especially those using new amplifying devices such as the high power UHF tetrodes are very efficient and get the electric bill down to half of what it used to be with klystrons. The answer is actually quite simple: the linearity required of the amplifying devices in a TV transmitter simply cannot operate at their highest power efficiency without sacrificing linearity. Performance first, power efficiency second. This requires that money be spent on electricity to make heat that we really don't want to make in the first place. If these amplifiers were only called upon to make RF power as CW sources, the power efficiency upper limit changes from that possible with class AB operation to that of class C or even switchmode class D. However, classes C and D are so hopelessly non-linear that they are useless as amplifiers of RF carrying amplitude modulated sideband information unless that information is simple, ie ON or OFF such as Morse Code. Sufficient linearity and optimum efficiency are simply opposing physical requirements such that a good portion of the ac line power must be converted to wasted heat for the thing to work right. How can this be improved upon? Is it possible?

The Digital Approach

If the job of amplifying a small signal to make a big one is all that is required, a class C nonlinear, but highly efficient, amplifier is the right choice. The ten bit digital concept uses ten class C signal sources properly power scaled each in relation to all of the others so that some of them, all of them or none of them are turned on to produce the required summed amplitude dictated by the output of the A/D converter sampling the video. These amplifiers or CW sources are drive gated to switch on and off according to the digital expression of the analog video signal. The amplifiers themselves are simply amplifying or not amplifying the visual carrier from an oscillator which has constant output. This is CW.

The A/D converter used is a ten bit device which has shown that ten bits of video information are more than enough to meet television broadcast standards as set forth by the FCC and be indistinguisable from ordinary analog video. For a constantly changing parallel ten bit word causing the RF drive switches to change state at the sampling rate, two very important things must happen for this approach to work successfully. First, the switches must be fast enough so that they are fully on when told to be on and fully off when told to be off. The transition times cannot exceed an arbitrarily chosen length of say 10% of the total on/off cycle. For a video bandwidth of 4.1 MHz, the commands to turn on, then off could come as close together as 243 nanoseconds. And second, the class C amplifiers themselves must respond to drive or no drive commands and not be sluggish when suddenly presented with drive or have drive taken away. At high power, many amperes of current are moving very quickly and at the outset, it is not obvious that class C amplifiers are fast enough.

A block diagram of an n-bit system is shown in Figure 1.

The complete background on the topology and power levels is contained in the references, but it is necessary to reiterate here that The RF power sources with the coupler chain make up the transmitter/modulator and they are inseparable. This approach departs radically from todays transmitter where the transmitter can be broken down into its constituent pieces. Also, there is no "RF Converter" in the conventional sense of the word. Direct modulation results since the carrier is modulated with video. Of course double sideband full amplitude modulation is the result and vestigial sideband can only be created by proper high level filtering of most of the lower sideband that results. This is a known science and the way that we used to do things before IF modulation and low level filtering. For the purposes of this report, TV is generated even though it is double sideband.

The results of the digital synthesis method predicted by 1 and 2 was proved to work in the lab by means of a bench

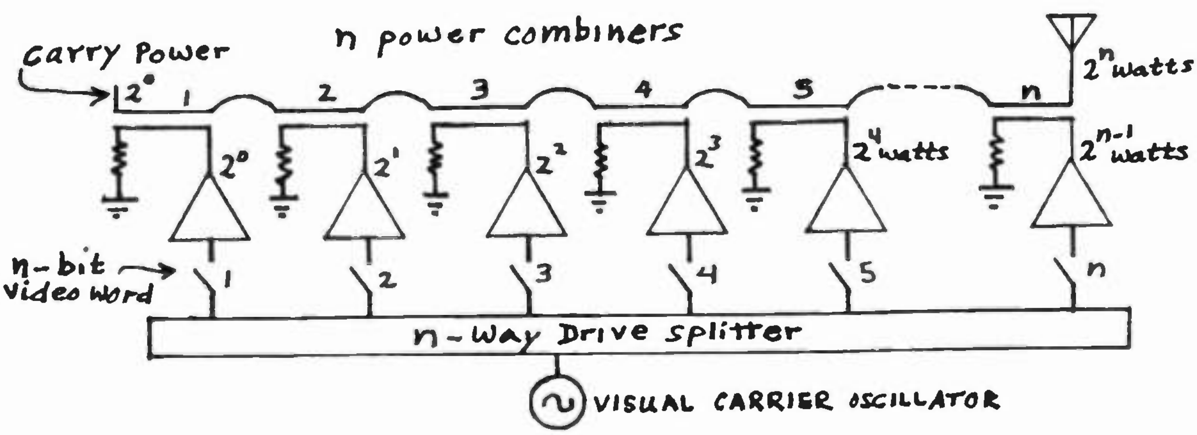

Fig. 1 An n-bit digital TV transmitter using n-3dB couplers fed by Class C power sources gated by the digital video word from an analog-to-digital converter. Peak Power output is 2^n watts scaled to any power level.

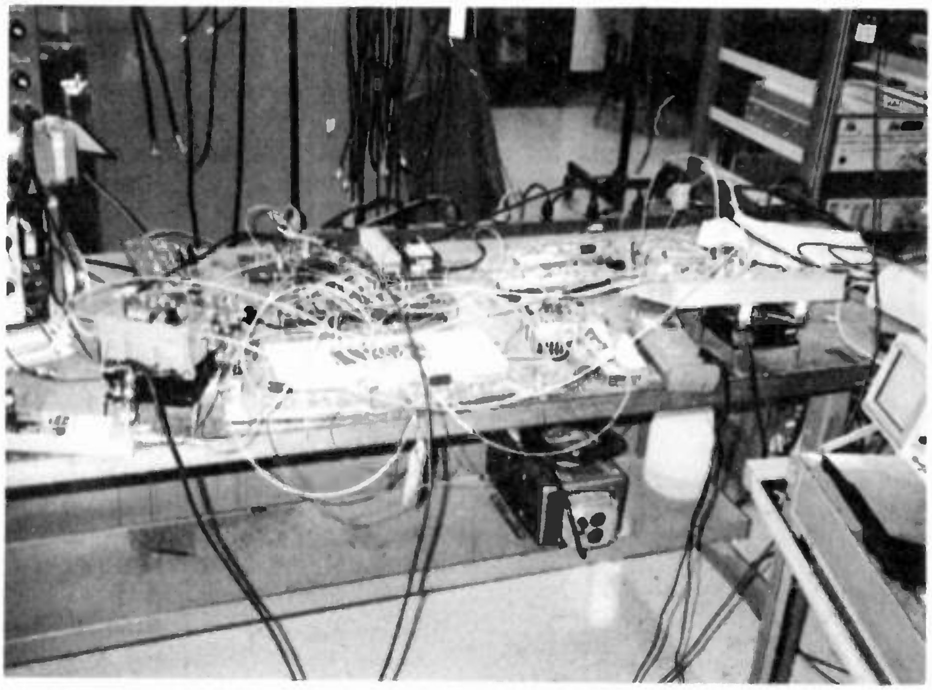

Figure 2 A distant view of the bench set-up in the Acrodyne lab. 25 watts of peak of sync power was generated on channel 43 representing the world's first demonstration of a digital TV modulator/transmitter.

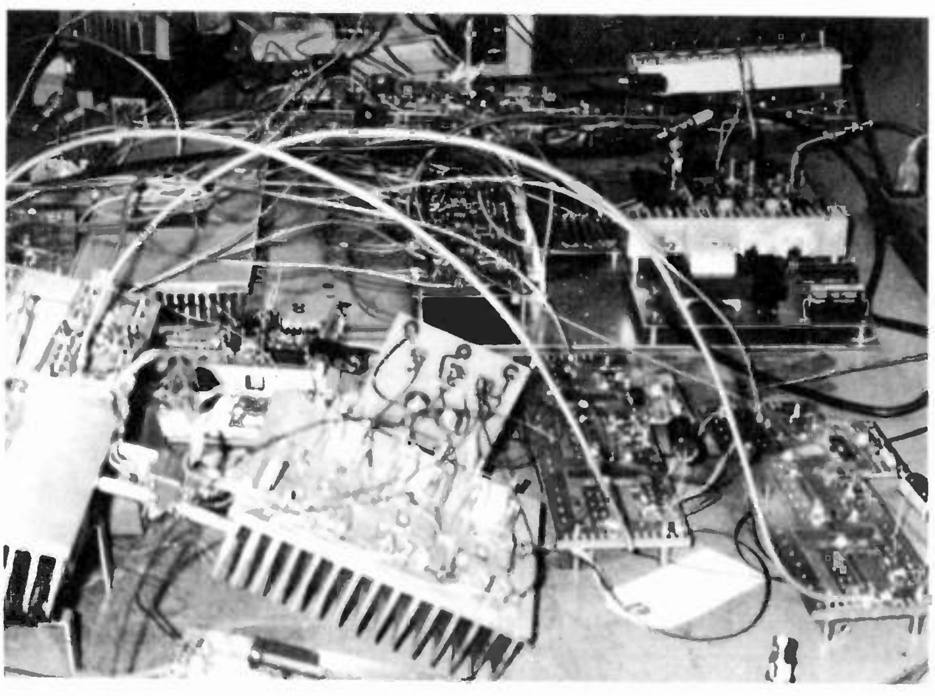

Figure 3 A clear view of the Acrodyne digital TV transmitter bench set-up. Class C amplifiers of various power levels are scattered about all feeding the chain of 3dB hybrid power combiners.

set-up that uses properly power scaled class C amplifiers connected to ten cascaded 3dB hybrid couplers made of wire-line cut for equal (3dB) coupling on channel 43. This set up is shown in Figures 2 and 3. No attempt is made to point out individual items in these figures. Suffice it to say, all things are interconnected to reflect the hook-up of the block diagram in Fig. 1 for ten bits. Fig. 4 shows ten PIN diode gate drive circuits in the upper printed circuit board and ten wire-line combiners with reject loads in the lower printed circuit board.

A close-up of the PIN diode switch circuits is shown in Fig. 5

The Pin diode circuits respond to the A/D converter ten bit TTL outputs to switch drive on and off to each of the ten amplifiers in the chain. Earlier it was stated that the on/off commands could come as close together as 243 nanoseconds. Figure 6 shows that for a 243 nanosecond wide ON command (lower trace) the PIN diode is slow to turn on. In fact it is not really fully on until the OFF command comes along (Note that only an attempt to measure ON speed is measured; that is why the upper curve continues to grow even after the OFF command is given).

The diode type used is the rather old Hewlett-Packard 5082-3042. Since then 2 nanosecond switches with built in drivers have been purchased, but not incorporated in time for this report. However, with these rather slow diodes in place, "fantastic results", could not be anticipated - and this was known. By far the weakest element in the initial lab demonstration was the PIN diode type available. Class C amplifiers were shown to respond to gated drive in less than 10 nanoseconds so they were never anticipated to be a problem nor have they been. This was

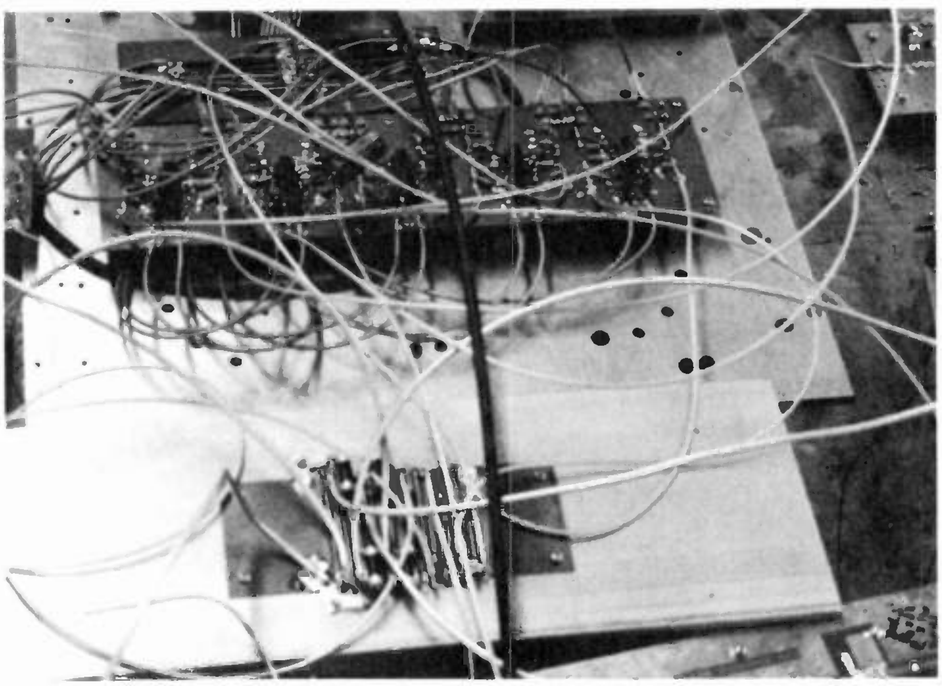

Figure 4 Five hexinverter ICs of the fast (74F) TTL family make up discrete PIN diode driver circuits which gate the amplifier chains on and off. The ten section wire-line power combiner is shown in the foreground.

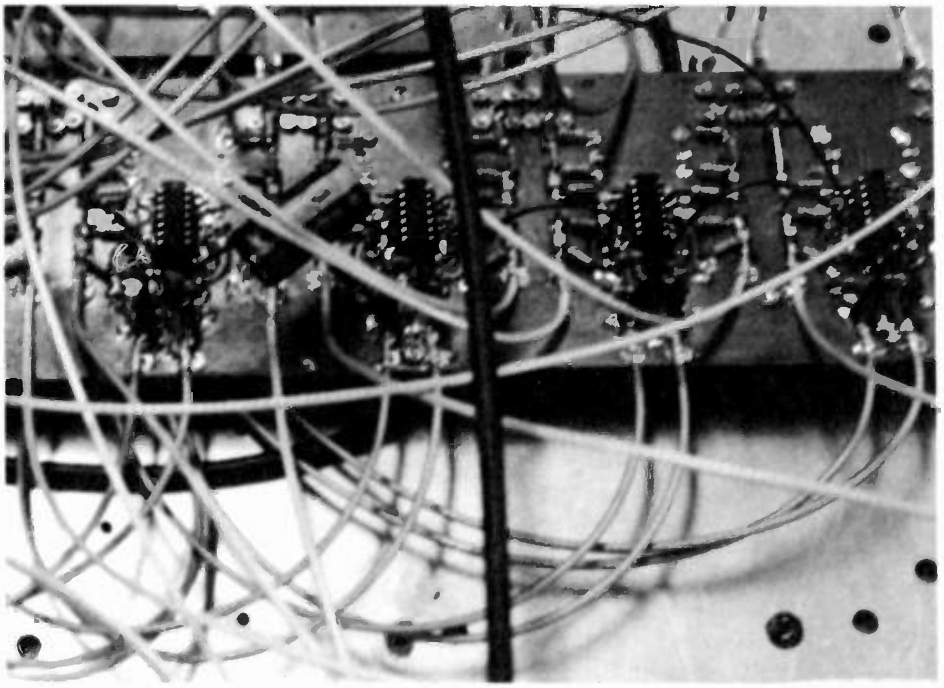

Fig. 5 A close-up of the PIN diode driver circuits is shown.

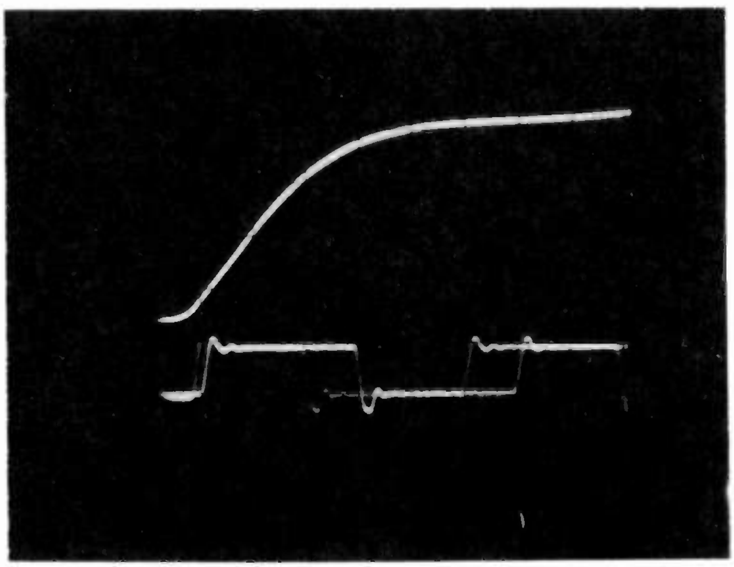

Figure 6 Turn-on time of the PIN diode switch used is shown in the upper trace responding to the TTL command to turn on in the lower trace. The response to the command to turn off is disabled to allow full assessment of the turn on time.

good news, because class C amplifiers must be used for this technique to have a significant efficiency edge over all other TV generation methods.

Test Results

After statically phasing the line lengths of each amplifier chain and matching power levels so that each hybrid input had the right power level into it, six stair-step video was applied to the A/D video input. With no further adjustment, the result is shown in Fig. 7.

In Fig. 7, the lower trace is the reconstructed video from the D/A converter. It must be the bench mark to compare with since this synthesized video is what the modulator/transmitter is responding to. The upper trace shows the detected output of the 3dB combiner chain. Ideally, they should be the same, but at least for a first turn on, the upper trace was TV. What appeared to be a large switching glitch is shown in the third step from the white level. After some refinement in phasing and amplitude matching, the result is shown in Fig. 8.

Much of the fuzziness is gone and the color signal is in better proportion with the original in the lower trace. Further refinement yields that shown in Fig. 9 where the input trace (lower) and detected RF output (upper) shows only a very slight difference.

An examination of the detected RF of the most significant bit (Fig. 10) shows what results from a slow Pin diode switch.

Digital video going out of the A/D converter is the upper trace while the MSD detected video is in the

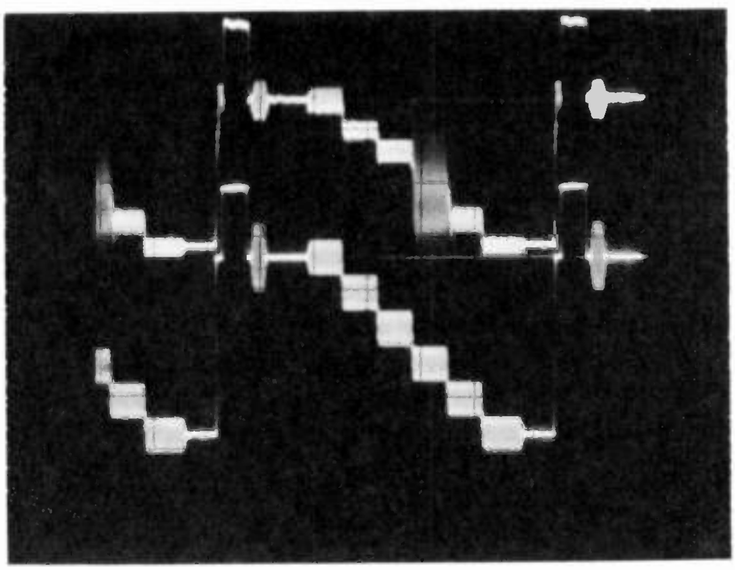

Figure 7 A six stair-step video signal is applied to the A/D converter in the lower trace. The upper trace is detected video at channel 43. Notice that "it's TV", but with severe switching glitch distortion in the third lowest step.

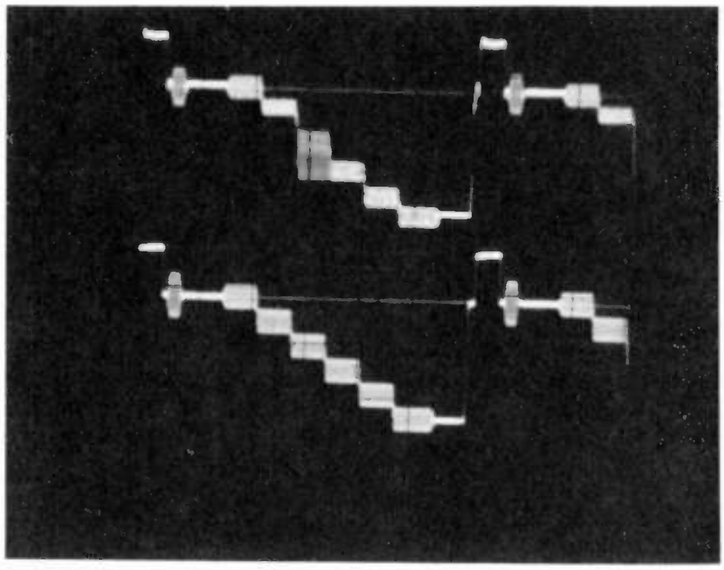

Figure 8 With minor phase and amplitude adjustment of the most significant bit, the glitch distortion is dramatically reduced and it has moved to the fourth step from white.

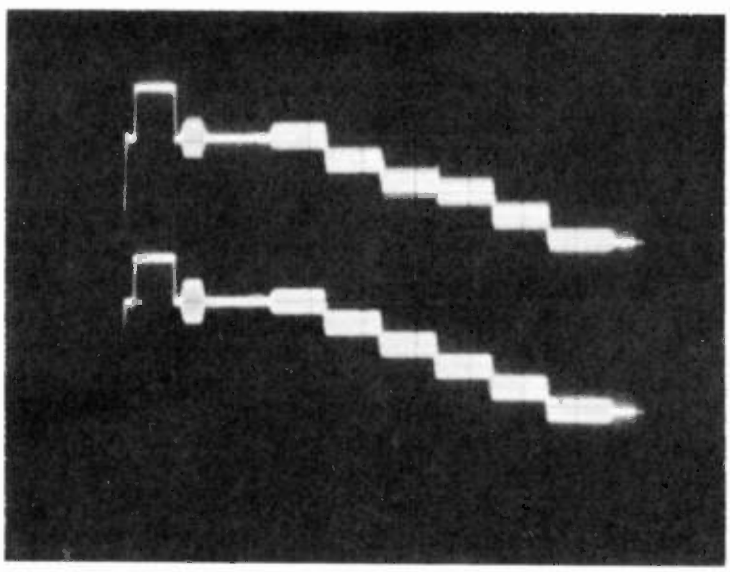

Figure 9 Video output of the A/D converter in the lower trace while channel 43 detected video in the upper trace. Almost a precise reproduction is obtained.

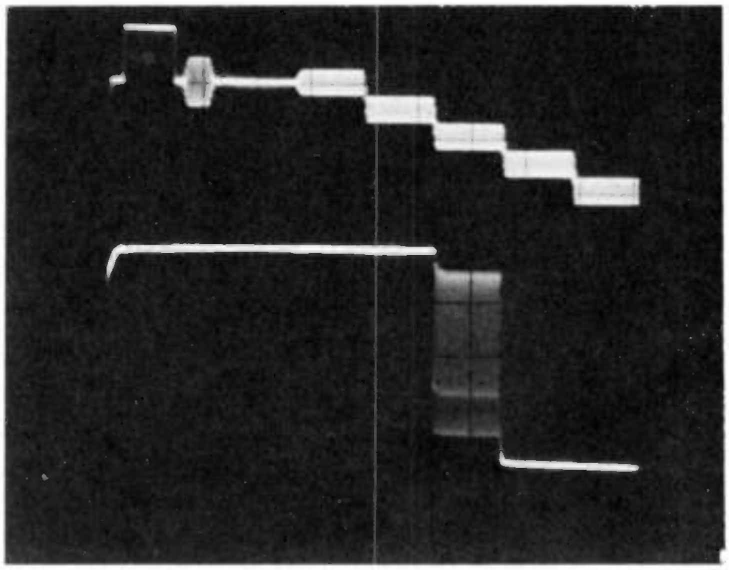

Figure 10 Looking at the detected RF of the class C amplifier making up the most significant bit. Due to the slowness of the PIN diode switch, the envelope never quite fully turns on or off as the TLL command dictates.

lower trace. The RF driver never becomes fully on and is never fully off when told to toggle at the color rate of 3.58 MHz. Incomplete modulation is the result directly contributing to the fuzziness shown in the previous figure.

The good results shown in Fig. 9 were only possible after hand picking a faster diode so that the MSB detected signal actually made it to the RF rails.

Fig. 11 is an expanded view of the sync pulse and color burst. Note the near precise match between input and output.

The vertical interval is shown in Fig. 12 with the lower trace the A/D converter output. Near precise reproduction is recorded showing very good low frequency response.

At the beginning before any "fine tuning" took place to clear up the switching glitches evident in Figs. 7 and 8, full color bars were applied to the A/D converter video input with the result shown in Fig. 13.

The severe distortion due to improper phase and amplitude alignment along with slow PIN diodes is obvious. However, a look at Fig. 14 shows near precise reproduction after the phase and amplitude are readjusted-even with the slow PIN diodes. The video input is the lower trace while the upper trace is the detected RF signal on channel 43.

Two stair step levels are shown in Fig. 15 where a finer look is available showing the fuzziness caused by the slow PIN diodes. The upper trace is the RF detected output.

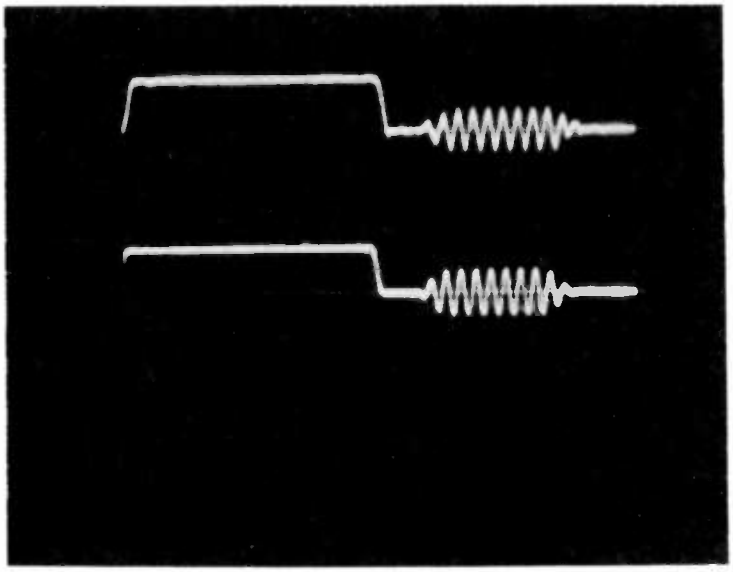

Figure 11 A close-up of the sync pulse. It is difficult to determine the difference between the digitized video and the detected channel 43 video output. (Digitized A/D output).

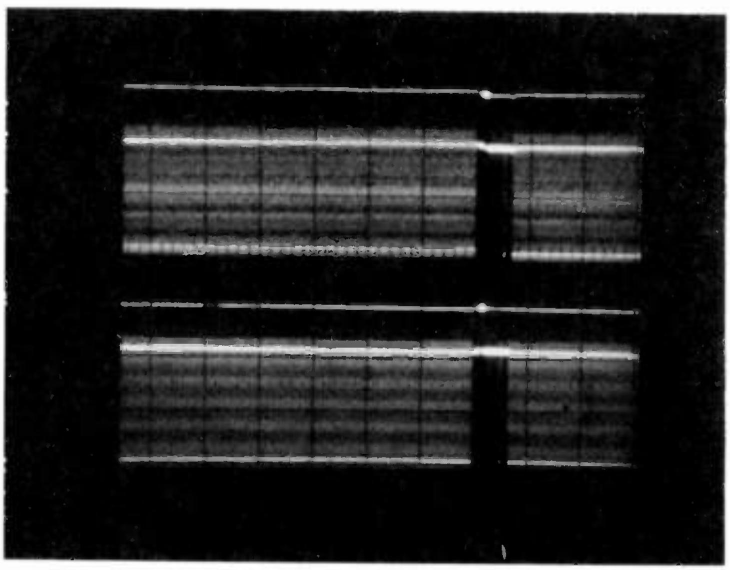

Figure 12 The vertical interval is shown with negligible difference between digitized input video and detected output video.

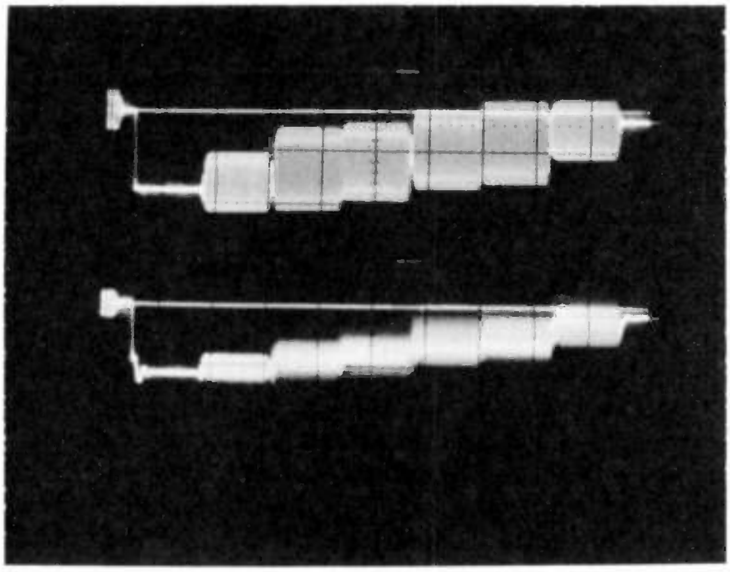

Figure 13 A first attempt at full color bars as the video input signal (output of the A/D lower trace) before any "adjusting" of amplitude and phase of the ten 3dB hybrid input signals. Detected channel 43 video is the lower trace.

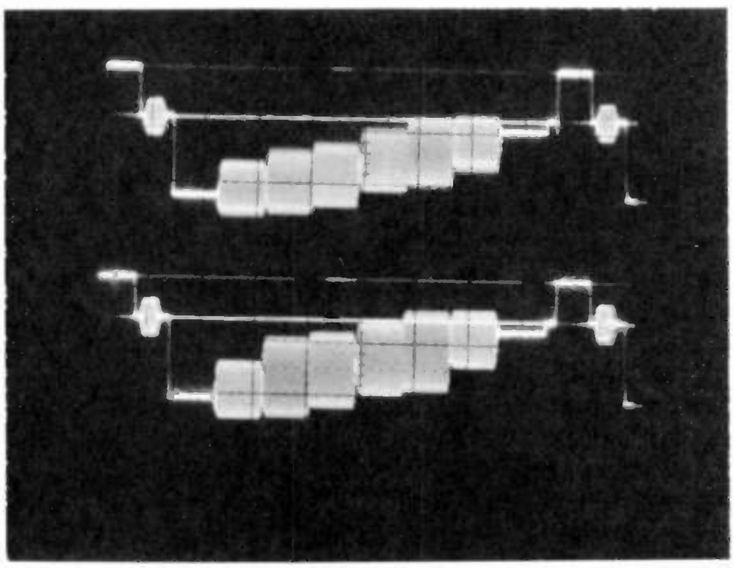

Figure 14 The same presentation as Fig. 13 after better phase and amplitude adjustment of each of the ten class C amplifier chains. Detected channel 43 video is the upper trace.

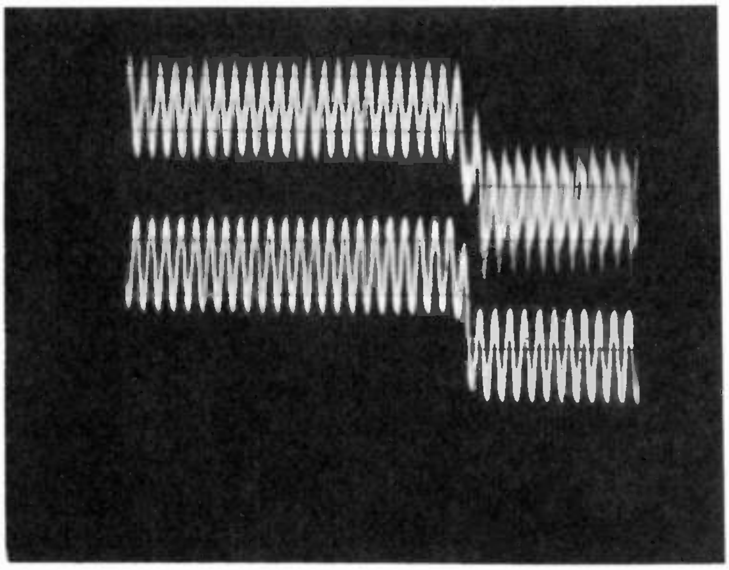

Figure 15 Two adjacent steps of six step staircase waveform showing some switching glitches in the lower step in the upper trace. A/D output video is the lower trace.

No attempt is made to show color bars on display on a video monitor because this paper is not printed in color and the information would be meaningless, but such photographs have been taken and will be shown in the oral presentation at the Convention.

How Practical Is It?

The lab transmitter produced about twenty-five watts peak of sync power. Of course, this isn't too useful on the air, but it was sufficient to demonstrate the concept. There is no theoretical power limit. Limits imposed on what is practical and what isn't will be dictated by the economics of the method. At first glance (and even at fifth glance) the digital transmitter seems to have a lot in its favor. For example, inexpensive non-linear class C RF power transistors cost somewhere around 50 cents per watt instead of the three dollars per watt for linear class AB devices. This obviously will bring the cost of a solid state transmitter way down compared to today's equivalent transmitters. Also the collector efficiency of the class C transistors is conservatively estimated at 50-60% while that of the class AB mode of operation is closer to 35 to 40%. In big transmitters, this can mean a big difference in power consumption. Also since less ac line power is needed, less heat is generated for corresponding smaller cooling machinery. Everything seems to get better! In fact, looking at some preliminary power consumption figures, plant power consumption normalized to 1kW peak of sync is about 3 1/2 to 4 1/2 times, ie for every kilowatt of peak sync power out of a UHF TV transmitter, about 3 1/2 to 4 1/2 times that in average power is consumed. With the digital class C approach, projections show that consumption will be between 2 1/2 and 3 times representing a savings of 28 to 33% - in rough numbers. These numbers represent the additional consumption saved over and beyond today's most efficient transmitters, not the pulsed klystron of ten years ago.

Is is practical? Yes, that has been shown. It's a matter of building it up. And now that Acrodyne has received its second Ben Franklin Seed Grant to continue the project toward useful product development, a real working digital transmitter of some useful power level will be developed over the next year. Watch for it!

Bibliography

1. Hulick, T.P., "Digital Amplitude Modulation.", Broadcast Engineering, Dec. 1987., pp. 68-80.
2. Hulick, T.P., "A Digital Modulator-Transmitter", NAB Proceedings 1988, April8-12, 1988., pp. 302-312.

HIGH POWER SOLID STATE AMPLIFIERS FOR A UHF TRANSMITTER

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Abstract:

The new generation of high-power solid-state TV-transmitters for B IV-V requires broadband amplifiers in modular form with high power capability.

Output power of a module is 800 to 1000W.

This paper concentrates on the design and application of the new PHILIPS bipolar transistor BLV62 for this purpose.

The transistor is able to produce at least 150W in Class-AB operation over the frequency range 470-860MHz.

Main emphasis will be made on modelling the combination of transistor and circuitry.

Lay-out and results will be presented.

INTRODUCTION

Although a klystron is able to produce sufficient peak sync power for terrestrial TV-transmitters in B IV-V, a modular solid-state solution has the advantage of better on-air availibility because of simplified maintenance, whilst a reserve transmitter becomes superfluous then.

To limit the number of active elements in these at least 800W modules, powerful transistors are necessary. Here the newly developed PHILIPS type BLV62 is advised.

Characterization of BLV62:
- push-pull; Gemini-package (SOT-262)
- $P_L \geq 150W$ (-1dB); "narrow-band" G_p up to 860MHz>9dB (typ. 10dB).
- V_S=28V; Class-AB operation.
- Input: prematched; Output: $C_L \leq 1pF/W$.

In UHF, matching already starts inside the active device. So, it has been concentrated on this aspect first and attention has been paid to geometry, bonding method, effective bandwidth and choice of V_S.

Three typical TV-amplifier designs are discussed in which the output circuitry is optimized with TOUCHSTONE and the input one with PHILIPS software.

Attention is paid to Class-AB biasing and methods to tune-up designs for lowest gain compression versus frequency.

It should be noted that an excellent compression point does not assure an optimum for essential TV parameters like differential phase and amplitude. However it is a good starting-point for design and asks only minor corrections afterwards.

A complete BLV62 application report[1] is available.

BIPOLAR CHIP FOR BLV62.

As a rule, broadest bandwidth asks for lowest possible output capacitance per W. A new 1pF/W bipolar 30W chip has been developed of which six are applied within SOT-262 Gemini-package.

This new chip has an interdigitated structure with emitter-pitch of 8 micron and width of 1,5 micron. Poly-silicon emitter ballasting resistors are used to assure excellent current distribution and uniform heat distribution. PHILIPS' transistor dies utilize T_i-P_t-A_u top metallization for long lifetime and excellent reliability. Only Gold bonding-wires are applied.

BLV62 IN GEMINI PACKAGE.

The modified Gemini-package SOT-262 contains two separate BeO substrates on one flange. Parallel connection of chips irrevocably will introduce losses in power and linearity.
This is reduced by applying equal and direct emitter grounding to the flange by means of via-holes through BeO substrates and double wire-bonding each emitter pad. Paralleling chips causes rather low transistor input- and output impedances.

PREMATCHING.

These low impedances cause large losses in matching networks outside the devices. They have been increased by inserting single prematching inside of the package.

Besides taking care of a higher, more ohmic complex impedance, it corrects G_p in a positive way, has some negative influence on stability and decreases collector efficiency somewhat.

Correlating design results have been obtained with the aid of the Giacoletto simplified large signal model for fundamental frequency calculations.

OUTPUT MATCHING; EFFECTIVE BANDWIDTH.

The input matching is responsible for achieving level gain over the operating bandwidth, whilst the collector loading has to be designed independently to satisfy P_{out} and $eff._c$ considerations over a large bandwidth.

A pragmatic approach to the design of an overall stage is to determine a suitable output matching first, then design the input-matching section to satisfy one's gain requirements.

It can be proved that V_S=28V is an optimal choice for the BLV62 if one combines: 1pF/W, BW>400MHz and P_L>150W. Techniques to extend the intrinsical bandwidth have been considered. A possible method is INSHIN (INtegrated SHunt INductor).

Ultimate intrinsical output circuitry, which can be matched to 50 Ohm, is shown in Figs. 1 and 2 (left of the dashed line).

BROADBAND OUTPUT MATCHING.

BLV62 is a push-pull type, so it is clear a balun is needed in the input- and output chain just before the 50 Ohm connectors. To limit the impedance ratio in the balanced parts, an already transforming balun version is applied here.

Figs. 1 and 2 respectively show the output circuits for B IV-V and B IV as results of TOUCHSTONE optimization. The available application report much better explains the procedure we followed here.

BROADBAND INPUT MATCHING.

Both Z_L and G_p exhibit variations versus f and P_L. The main function of the network between balun and transistor input is flattening G_p over the desired operating frequency range by accepting mismatch against a fixed source impedance (here 25 Ohm via balun). The 50 Ohm source can be arranged by means of a circulator (during experiments) or 3dB quadrature hybrids for combining in practice.

Topology of input- and output matching circuitry sometimes can be the same. For the input-side a special PHILIPS' optimization program is used.

OPTIMIZATION, TUNING, POWER SWEEPING.

A useful method to verify how far a wide-band transformed 50 Ohm indeed reaches the expected value inside of intrinsical dies can be done with a passive dummy and a Network Analyzer. This is an ideal system for the output-side.

For optimization of the input matching, an amplifier is inserted in a power-sweep able to stabilize output levels over a given frequency range.

Both methods are described in the application report.[1] As a final check G_p and $eff._c$ are measured at a number of spot frequencies under c.w. conditions. Figs. 3, 4, 5 and 6 show results.

BIASING.

For Class-AB operation the knee and cross-over distortion are reduced by some quiescent current. A good choice for designs being described here is 400mA per BLV62 side, what may be re-adjusted under TV-conditions.

Fig. 7 shows a possible bias circuit. It contains balancing and leveling adjustments,has a temperature sensor and is failsafe.

SUMMARY.

BLV62 is applied in two typical TV-amplifier designs being checked in c.w.practice. A third design is only theoretical.

Fig. 8 shows the B V design of which the output circuit covers B IV-V completely, whilst B V limitation is caused by gain egalisation on the input-side. The TOUCHSTONE block diagram is depicted in Fig. 1. C.w. results are shown in Figs. 3 and 5.

The second design, for B IV is given in Fig. 9. The TOUCHSTONE block diagram is depicted in Fig. 2. C.w. results are shown in Figs. 4 and 6.

A theoretical design contains B V input- and output circuit. (See Application Report[1])

Figs. 10 and 11 give PCB lay-outs. They are double Cu-clad with microfibre reinforced PTFE dielectric (ε_r=2,2); thickness 1/32 inch (0,794mm) and copper sheets 2 x 35 micron.(for example RT/duroid type 5870/5880 or other same quality material)

One may expect reasonably higher peak-sync power levels under Class-AB TV-conditions, because average (black) power only reaches approx. 60% of c.w.values.

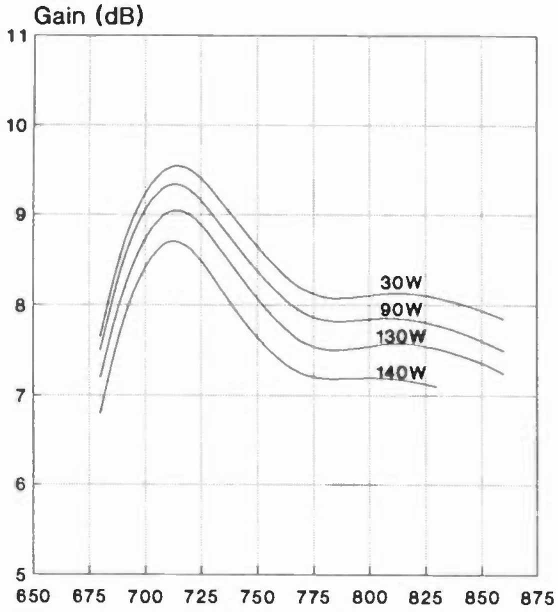


Figure 3

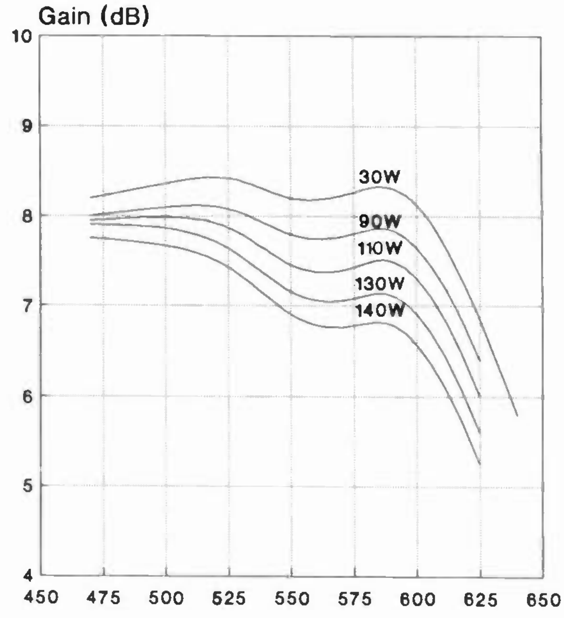


Figure 4

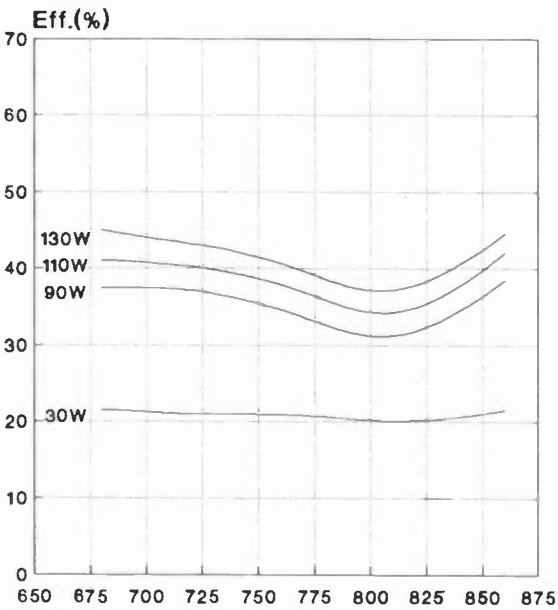


Figure 5

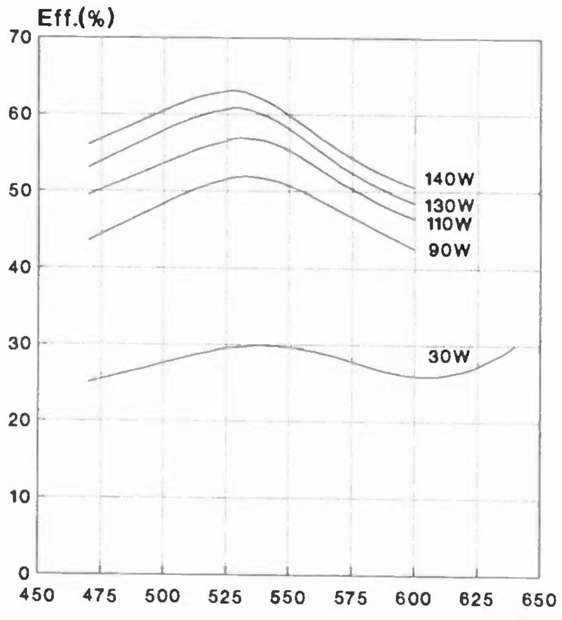


Figure 6

REFERENCES.

[1] PHILIPS Application Report NCO-9101.

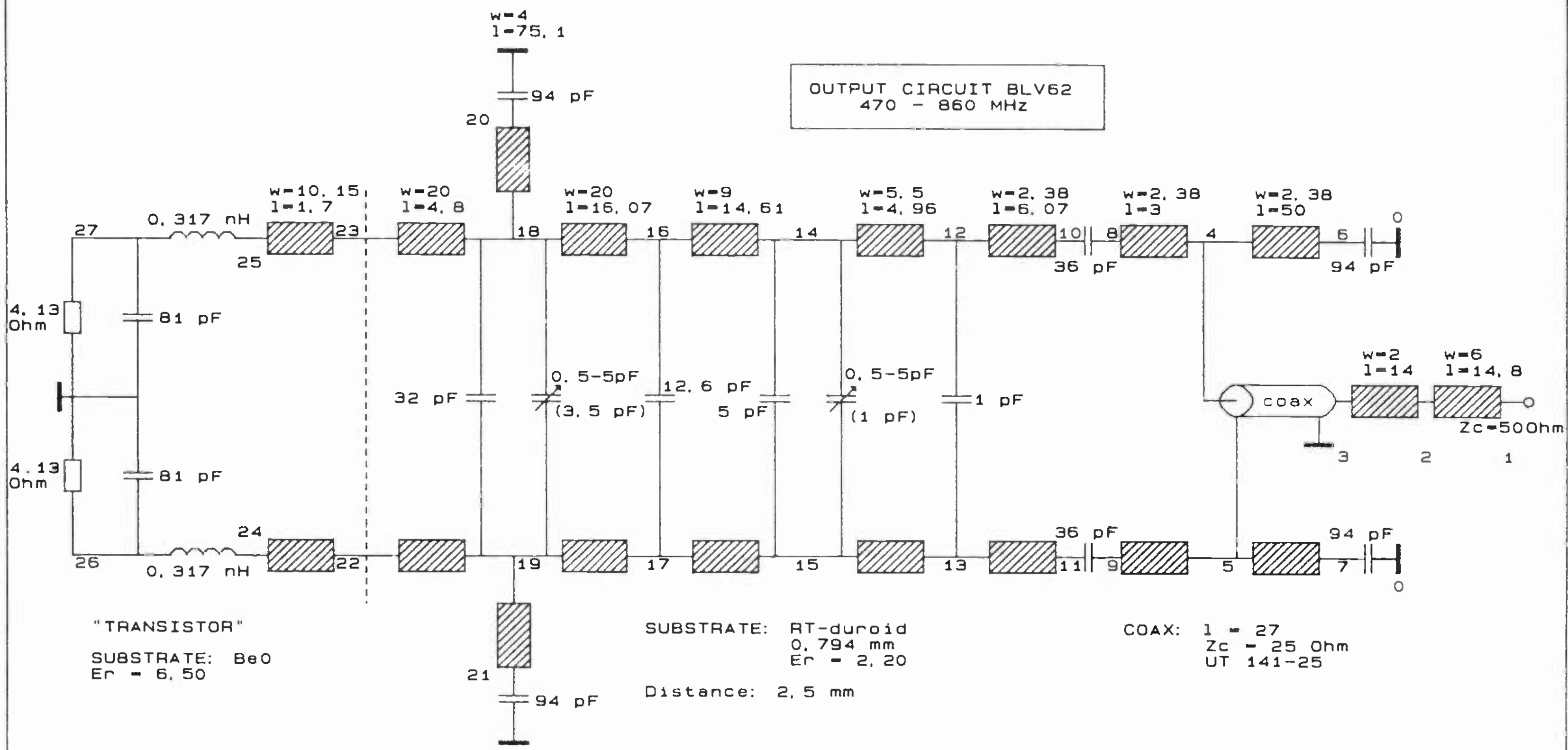

OUTPUT CIRCUIT BLV62
470 - 860 MHz
w=4
l=75, 1
94 pF
20
27
0, 317 nH
25
w=10, 15
l=1, 7
23
w=20
l=4, 8
18
w=20
l=16, 07
16
w=9
l=14, 61
14
w=5, 5
l=4, 96
12
w=2, 38
l=6, 07
10
8
36 pF
w=2, 38
l=3
4
w=2, 38
l=50
6
94 pF
4, 13
Ohm
81 pF
32 pF
0, 5-5pF
(3, 5 pF)
12, 6 pF
5 pF
0, 5-5pF
(1 pF)
1 pF
coax
w=2
l=14
w=6
l=14, 8
Zc=50Ohm
3
2
1
4, 13
Ohm
81 pF
24
26
0, 317 nH
22
19
17
15
13
36 pF
11
9
5
94 pF
7
21
94 pF
"TRANSISTOR"
SUBSTRATE: BeO
Er = 6, 50
SUBSTRATE: RT-duroid
0, 794 mm
Er = 2, 20
Distance: 2, 5 mm
COAX: l = 27
Zc = 25 Ohm
UT 141-25

Figure 1

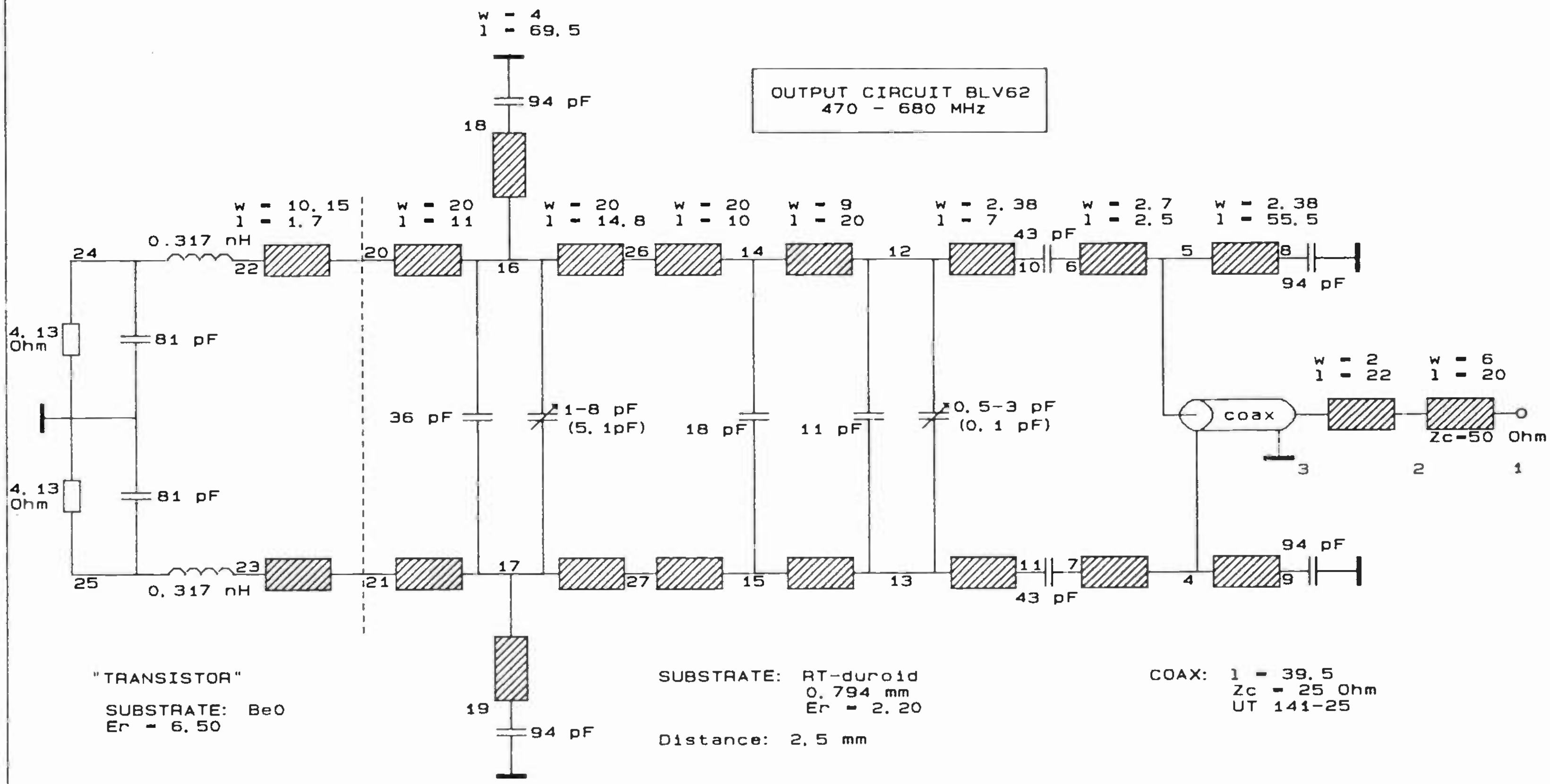
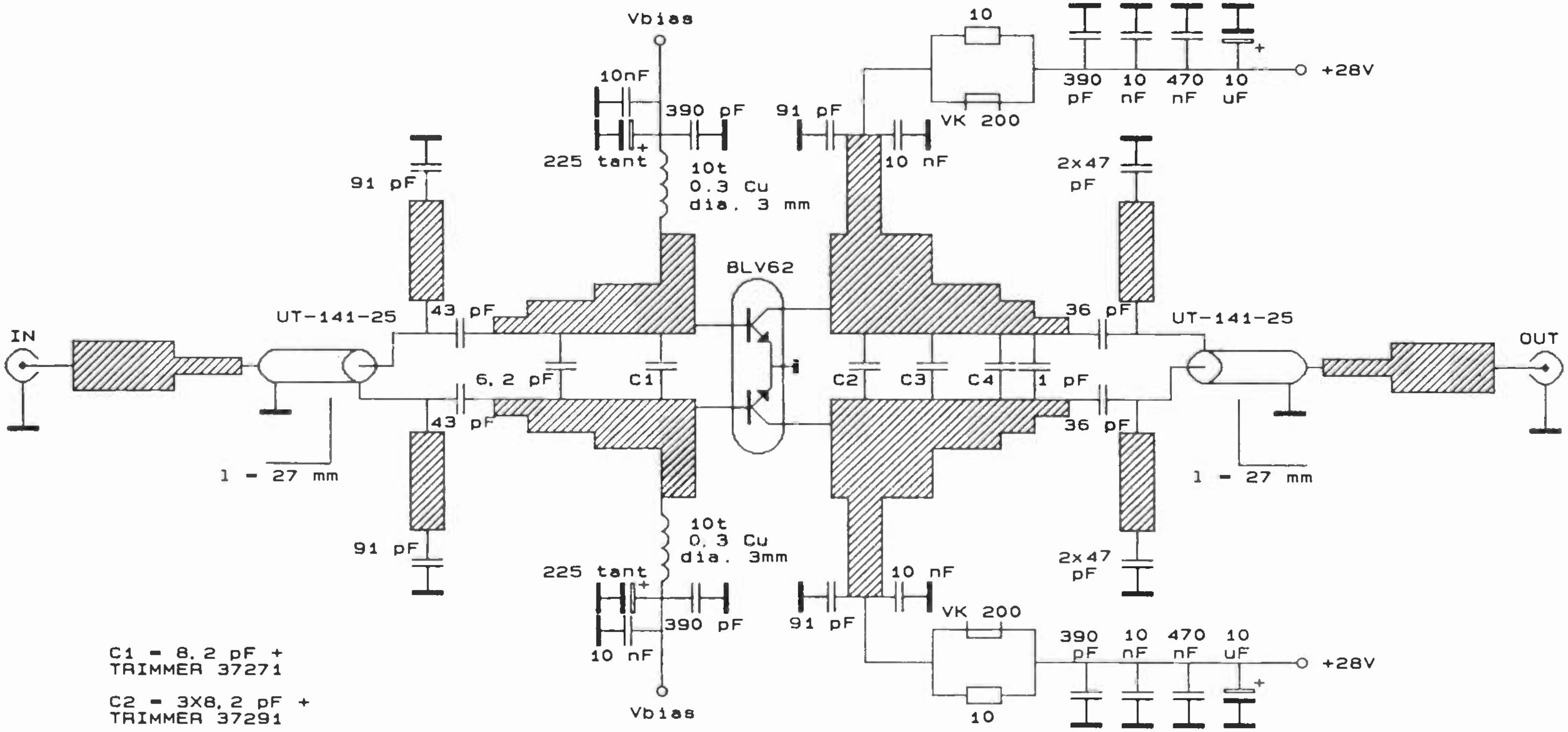

OUTPUT CIRCUIT BLV62
470 - 680 MHz
w = 4
l = 69.5
94 pF
18
w = 10.15
l = 1.7
w = 20
l = 11
w = 20
l = 14.8
w = 20
l = 10
w = 9
l = 20
w = 2.38
l = 7
w = 2.7
l = 2.5
w = 2.38
l = 55.5
24
0.317 nH
22
20
16
26
14
12
43 pF
10
6
5
8
94 pF
4.13
Ohm
81 pF
36 pF
1-8 pF
(5.1pF)
18 pF
11 pF
0.5-3 pF
(0.1 pF)
coax
w = 2
l = 22
w = 6
l = 20
Zc=50 Ohm
3
2
1
4.13
Ohm
81 pF
94 pF
25
0.317 nH
23
21
17
27
15
13
11
7
43 pF
4
9
19
94 pF
"TRANSISTOR"
SUBSTRATE: BeO
Er = 6.50
SUBSTRATE: RT-duroid
0.794 mm
Er = 2.20
Distance: 2.5 mm
COAX: l = 39.5
Zc = 25 Ohm
UT 141-25

Figure 2



C1 = 8.2 pF +
TRIMMER 37271

C2 = 3X8.2 pF +
TRIMMER 37291

C3 = 2X3.3 pF

C4 = 2.2 pF + 2.7 pF +
TRIMMER 37271

37271/91 Tekelec Giga-Trim

ML chips : B-type

PCB : Er=2.2 D=0.794 2x35u

Figure 8 TV-BAND 5 design

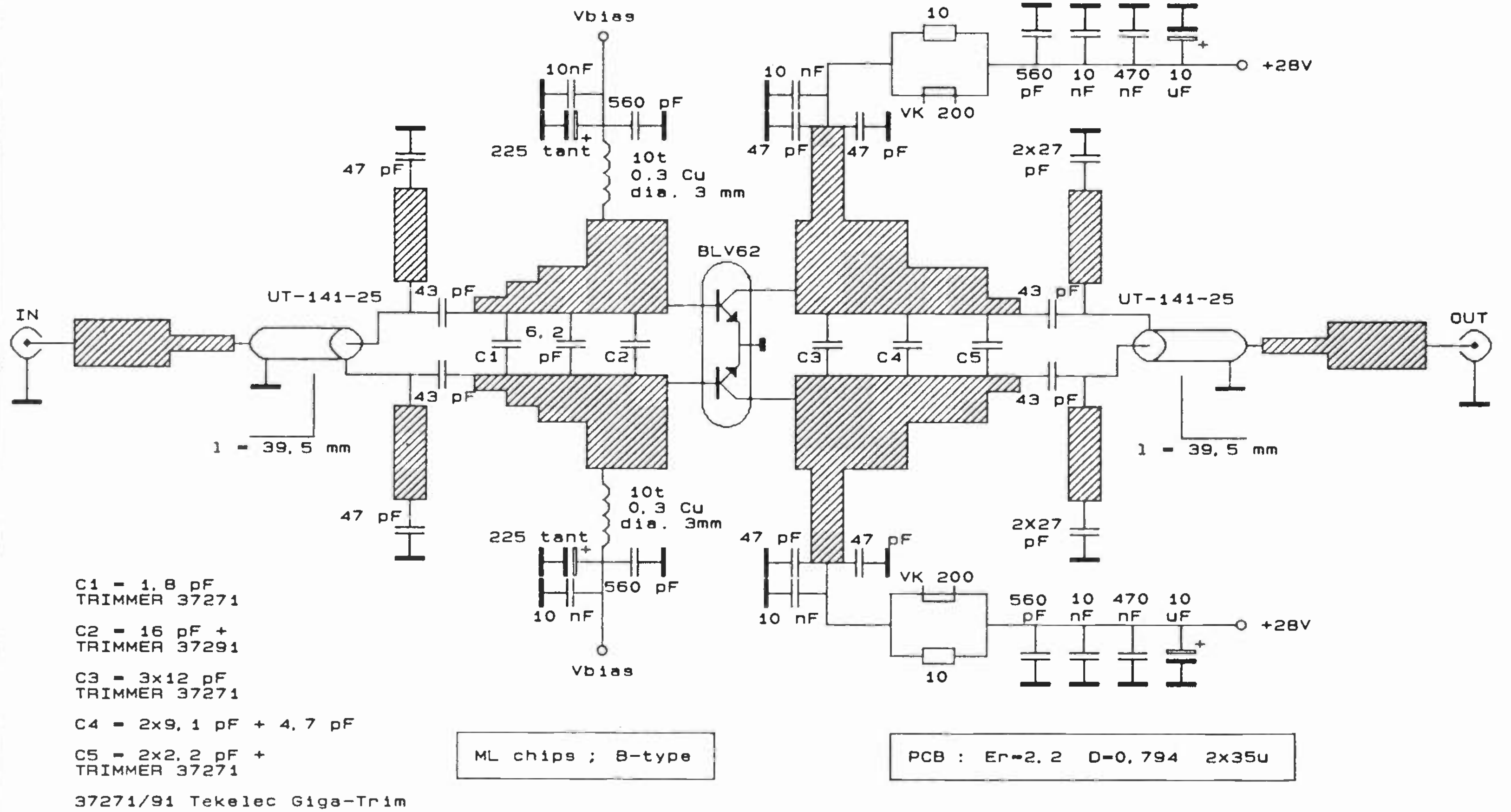


Figure 9 TV-BAND 4 design

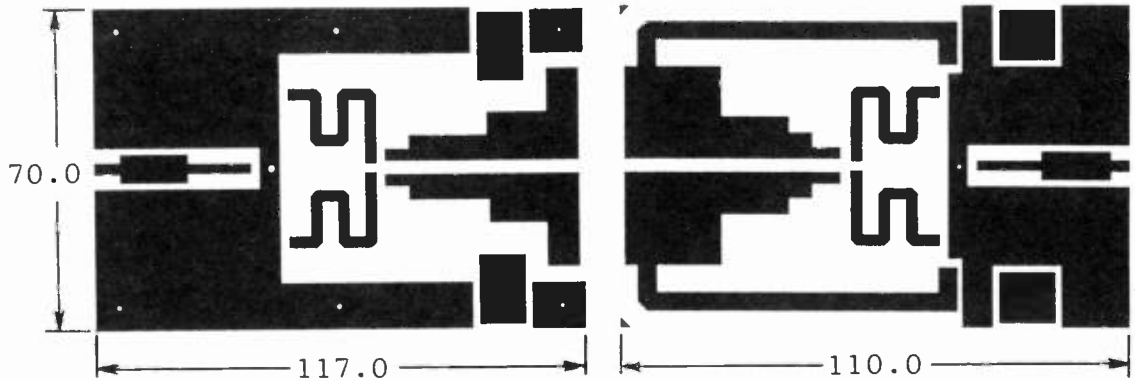


Figure 10
PCB B V

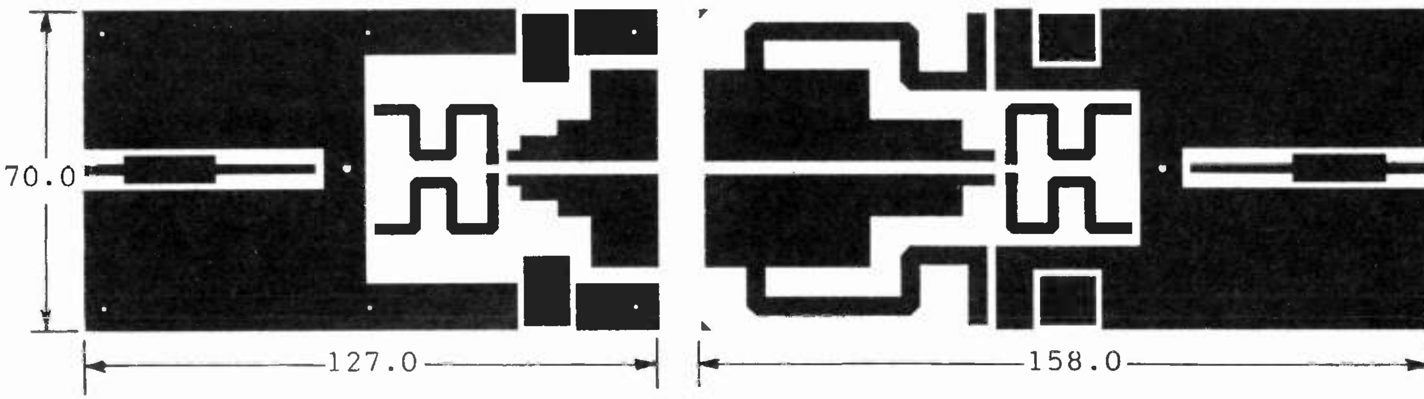


Figure 11
PCB B IV

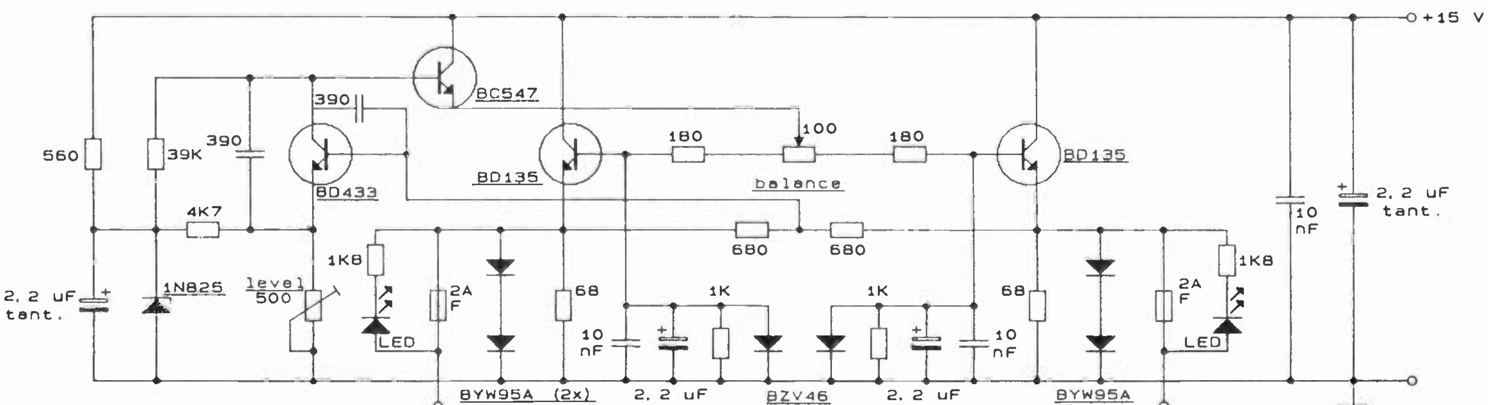


Figure 7. Biascircuit

DIGITAL VIDEO AND TRANSMISSION SYSTEMS

Wednesday, April 17, 1991

Moderator:
Richard Streeter, CBS, New York, New York

THE MIGRATION PATH FROM ANALOG TO DIGITAL*
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Utah Scientific, Inc.
Salt Lake City, Utah

A PRAGMATIC APPROACH FOR DIGITAL PANDEMONIUM
Thomas R. Goldberg
Ampex Corporation
Golden, Colorado

A ½" COMPOSITE DIGITAL VTR FORMAT
Y. Oba, T. Uehara, T. Nakayama
NHK (Japan Broadcasting Corporation)
Tokyo, Japan

K. Suesada, P. Livingston
Matsushita Electric
Industrial Company, Ltd.
Osaka, Japan

SPECTRE—DIGITAL TELEVISION TO UK HOMES IN THE EXISTING UHF TV BAND
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National Transcommunications Limited
Winchester, Hampshire, United Kingdom

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*Paper not available at the time of publication.

A PRAGMATIC APPROACH FOR DIGITAL PANDEMONIUM

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Abstract - Proliferation of video recording and processing formats has created a confusing situation for purchasers of equipment. Suggestions for appropriate application of technology and minimizing investment dollars are made in this paper.

INTRODUCTION

During the transition from the 1970's to the 80's the video industry saw a revolution in recording formats - moving from quad 2 inch to helical 1 inch. This change was relatively painless because the format for signal distribution remained in the familiar PAL or NTSC encoded form. The arrival of digital video has created yet another recording format revolution that has left users in a quandary. Not only has the distribution and processing format changed, it has blossomed into 4 different distribution alternatives (component or composite, both in parallel or serial), each with its own advantages and drawbacks. Were the digital situation not confusing enough, a growing range of component analog production equipment vying for today's video dollar further muddy the situation. In this proliferation of formats, users face an inordinately expensive array of supporting production and facility equipment to take best advantage of the additional quality digital can provide.

Unquestionably, component digital provides the highest quality approach available today. It is the native format of almost every digital source now commonly in use such as character generators and graphics devices and, therefore, can reproduce images with virtually no degradation at all. It also can retain more information from a component analog source than any other recording media. Digital effects devices which process video in this format also can be recorded with no losses. The drawback of component digital, as anyone who has looked knows, is that between acquiring recorders and associated facility changes, it is likely to continue to be too expensive for most users for some years. With the inevitable arrival of some form of HDTV on the horizon, many users will never be able to invest economically in the current generation of component digital equipment.

While D2 composite digital was conceived as an analog replacement format, it has the potential to provide some of the same benefits as the more expensive D1 precursor. Though equipment to support production in the digital composite domain is now available from some manufacturers, it can be almost as expensive as investing in top end component digital. Because D2 is composite and requires less bandwidth, recorders are less expensive. A complete studio may cost somewhat less than digital component but still enough that purchasers question the economics of creating a full-blown composite digital suite. It is a format with many advantages over traditional analog, but one that will ultimately give way to the admittedly superior component digital format as technological development drives costs down.

Where does this leave most users? With one alternative out of sight from a cost standpoint and the other created as an analog replacement format, many producers of video will continue to process in analog. This is especially true as component analog continues to grow with benefits of its own. Sensible choices must be made weighing many factors including client demands, the technical demands for equipment and when it can or must be purchased in a time of fast moving change.

Though Ampex is in the business of manufacturing composite digital recorders, we do not recommend that our customers invest in extensive composite digital routing, distribution and production equipment. There certainly are users who will find good justification for doing so, but it will be uneconomical, inadequate, or wasteful overkill for many. There are many reasons to purchase D2 recording equipment that do not extend to D2 production and distribution. For many users of D2 tape machines, it is our belief that small digital layering devices working with existing analog equipment would yield a cost effective alternate approach. This approach can provide the additional quality of digital production without most of the incremental costs. With thoughtful blending of present equipment, good control interfaces, and some new and moderately priced digital devices, users can create hybrid suites that can provide all the benefits of digital production at a fraction of the cost for a "full up" digital facility.

THE FORMAT IS MORE THAN JUST WHAT GOES ONTO TAPE

The recording format proliferation is clearly driving the changes in facilities to incorporate those formats more fully into their plants. But, in fact, the electronic distribution and production formats are only loosely coupled to the format in which the signal is recorded. Except D1, all recorders have composite analog I/O available without regard to the native recording format of the individual machine. With growing availability of decoders, encoders, and transcoders, any format can be changed to any other. Now, even Betacam machines can be outfitted directly with 601 outputs. The choice of interconnection standard must be based upon the range of sources to be used as well as recorder format and ultimate production needs.

One universal generalization can be made that, if origination, distribution, processing, and recording are all done in the same format, the highest possible quality from the system can be achieved. This is especially true in productions involving more than a single generation.

This is a truth, however, that is easier to state than practice. A single production today could well have to deal with a camera whose native format is component analog, source footage in composite analog, a C.G. or graphics computer whose native format is 4:4:4 and recorders that are D2. Picking the best format in which to process and distribute can't be based on sources alone in such a scenario. Glamour, such as that represented by the raw appeal of "digital" or as embodied in large impressive looking switchers, is a consideration in making the choice. Yet these cosmetics are becoming ever less important in our increasingly cost conscious industry.

Half the decision making criteria really rests with the goals for the production - how many passes will be required, how much and what kind of keying and special effects will be required, what level of quality is required - the technical issues. The other half of the decision is economic - what available equipment meets these needs most inexpensively, for how long a period can it be used considering rapid product evolution and HDTV, and what is the ultimate price for quality that the client is willing to pay.

RELATIVE MERITS OF THE AVAILABLE FORMATS

Each of the 4 standards, component digital, component analog, composite digital and composite analog, has its own advantages. The only intelligent choice between them must be made with the specific application in mind.

Composite Analog

Everyone knows that composite analog is on it's way out - right?

Surely the day will come when only digital formats are in use, but a central question is "When?" How long can video facilities continue to use the substantial investment they have made in analog equipment? The answer to this question depends on the two factors outlined above - how long existing equipment meets the user's technical requirements and how long it remains cost effective. It's a very safe bet that at some point in the future, analog will be phased out failing to meet both these criteria. It's equally safe to say that there will still be good uses for it for many more years in various areas of video production and distribution.

As long as the final link in the chain is the composite home receiver we all recognize, any functioning type C will still have more than enough quality for source playback. If video is destined for broadcast, using composite analog is justified. If, for whatever reason, a source is best available in composite analog, it should be played in that format. The economics of maintaining aging analog recorders particularly when offset by the conveniences of newer cassette based formats will eventually force users to replace them. Still, it will take more than that to kill off composite analog. Even when traditional analog recorders are replaced with D2, many purchasers will be happy to continue to use it as an analog device, feeding analog production equipment. With a huge installed base of routing, distribution, and processing equipment, this format will continue to be appropriate for broadcasters, cable operators and the like until a major change in transmission of signals occurs. It would be wasteful to abandon the core format upon which all video is based until there is adequate motivation to re-tool for some new form of video.

Source playback is only one piece of the production puzzle. The three other alternatives clearly win out on quality when layering starts, and each has its own advantages.

Component Analog

Component Analog has existed since the first color cameras but, until recently, has only been used for production in the creation of RGB chromakeys. When Beta and M recording formats arrived providing color difference outputs, people started to think seriously about doing more in component because it had one really great advantage - the ability to layer without edge crawl. As anyone who has had the pleasure of combining pictures in the component domain knows, certain things will always look better. Key edges in particular and all images in general can be superior in component whether done in the digital domain or in analog component. The PAL video system created better quality inserts because it does lessen these problems. Still, despite that advantage, Europeans have led the move to component layering to yield even better results. NTSC users stand to gain even more improvements in quality when the layering is done in component.

The reason is simple. In terms non-technical people can appreciate, processing video is cleaner when the color information is not all mixed up with luminance. Whether slicing and dicing pictures (switching) or fold, spindle and mutilating pictures (digital effects), when each component is handled separately, the relationships between color and brightness at layer edges are preserved. This creates more natural appearing composite images.

Component analog has the additional advantage that it will more faithfully capture the output of component digital devices than either composite analog or D2. A 4:2:2 (or 4:4:4) character generator outfitted with component analog outputs will provide a better representation of what's inside its electronic brain than composite outputs can provide. That allows a Betacam or similar color difference recorder to save a better quality copy than any composite machine. The same thing holds true for digital effects; the benefit is doubled because, in component production, both the inputs and the outputs suffer less translation losses.

So why shouldn't everyone run right out and invest in Beta machines and component production and distribution? Many production facilities have done so and have been very successful. There are also important drawbacks to this approach - costs for distribution and processing, uneditable audio AFM track limitations, a limitation to number of passes that can be used in a production, and the technical expertise available to the facility.

The cost to distribute component analog is three times as much as composite because the raw number of coax connections in and out of each piece of equipment has tripled. This can be more expensive than distributing digital in a large facility and is the reason why component production has typically been done within component only "islands." Component switchers also tend to cost about one and a half what equivalent composite units sell for. So, while there are clear cost savings in recorders, the signal distribution equipment can easily send the price tag above that for analog composite. These costs are justified if most of the acquisition is in Beta and the format is appropriate for the application.

One of the most important considerations here is how many passes are expected for the most sophisticated work to be done on this equipment. When one goes to this expense and trouble to gain quality, it makes little sense to go down many generations. All that layering quality could be lost to generational softness, chroma infidelity, or a host of other video sins in the final product. There does not seem to be a consensus on how many high quality generations should be expected in a component analog environment, largely because such systems vary greatly in how well they are "tweeked." Triple analog signal paths mean three times the chance for equalization, timing, noise and reliability problems both between and within the boxes. Component analog production requires a better planned and executed suite design plus careful alignment and maintenance to produce good results. These are problems that greatly diminish with any digital format.

Component Digital

Anyone paying attention knows that component digital video is the processing format of choice. It's used inside virtually all high quality digital effects. It is the electronic domain used today by almost every character generator, art system, and graphics computer. It is unarguable that, when color images need to be processed or created digitally, it is much easier to deal with video components. Even so-called D2 digital effects use video components internally at a different sample rate from D1.

Most of the additional cost associated with component digital, however, is due to the cost of recording it. This cost is strictly related to the number of bits of information that must be written on the tape. One frame of CCIR 601 component digital video at 8 bits requires about 700 kilobytes of data versus composite digital's 390 k (in NTSC), about 1.8 times more information. Although this ratio drops to about 1.5 in PAL it clearly requires a more sophisticated and expensive recorder in either standard. The challenge of building high performance recorders to meet this criteria is no small task. Few manufacturers have attempted it and common features on less sophisticated recorders like slow motion don't yet exist on these machines.

Routing and distribution are also expensive in this domain. Parallel signals require expensive cables that can't run more than about 30 feet. Serial converters are still expensive with standards still being determined and the component digital data rate converts to a demand for almost a 500 MHz bandwidth distribution system for serial 601 signals. Because of these problems, even those high end houses who have made substantial investment in component digital are often keeping signals within digital islands with limited parallel distribution.

For manufacturers, these data rates also demand expensive high bit rate parts to build the equipment. These parts continue to get cheaper but still have several years of volume and technology growth to reach real cost effectiveness in many applications. Manufacturers also must begin tooling for HDTV and that requires even faster, more expensive technology. When the time to move to HDTV production arrives, almost all production will undoubtedly be done in digital component. This is becoming ever more apparent as more players jump into the HDTV melee with digital approaches. That is fortunately seeming to coincide with the ability of technology to meet this demand. Until then, costs for component digital productions are likely to continue to be too high for all but the most state-of-the-art houses.

Composite Digital

Composite digital recorders already make sense to thousands of users who are not currently doing digital production. These machines were initially conceived to serve as type C replacement devices and are extremely well suited for that capacity. The advantages they offer over their type C forerunners are myriad. Like their more expensive component digital predecessor, D2 is a cassette based format with improved signal quality and 4 channels of digital audio. As today's cost of D2 recorders is only a little more than type C machines, the added advantages in faster transport speeds, improved editing performance, read before write, and multi-generation performance, the benefits are sufficient to merit purchase of D2 machines even while doing production in the composite analog domain. These costs will continue to fall, making D2 even more attractive in the future.

As long as production remains in analog, there is one significant digital benefit users will never get. D2 can provide the ability to make flawless generational copies with no degradation in background layers. This fact has driven investors in D2 recorders to seek out the associated equipment to do production in composite digital and utilize this multi-generation advantage of equipment they've already purchased. Manufacturers are offering an increasingly broad range of video switchers to enable production in this format. If a cost effective studio for D2 can be assembled, especially with D2 recorders that have already been purchased, many facilities will have justification for investing in that equipment.

Unfortunately, most of the devices available for this purpose are, as with component digital, inordinately expensive. Further, as with component digital, routing and distribution is still a problem that can't be handled by the equipment in place today. Although bandwidth requirements for D2 serial digital signals drop to the range of 250 MHz (in NTSC), this is still well beyond the capacity of existing routers and D/As. The problem is compounded when one considers that, aside from recorders (and disks), there are virtually no other composite digital sources being manufactured . Even with rapidly dropping prices for transcoders, the process of converting enough sources to composite digital to feed a switcher in that domain becomes an expensive proposition indeed. Assuming such investments are made, quality trade offs for transcoding everything, processing in composite, and both starting and finishing with analog sources may create a situation where a small advantage is attained at a high price.

SORTING IT ALL OUT

Facility decision makers are wondering whether to invest in any costly digital production equipment at all given the current doubts over a new HDTV television standard. As our industry becomes ever more competitive both for manufacturers and for equipment purchasers, every investment will be closely scrutinized and thoroughly thought out. As pointed out in the preceding sections, no format should be selected without considering the application requirements and associated costs to get the desired results.

Financial analysis will have to asses not only how much equipment must be purchased, but also how much existing plant can still be used. Such an analysis will differ significantly for each individual case. However, it is not evident that clients will pay more for a D2 suite leaving equipment amortization to take place at analog rates. Buyers will need to polish up their crystal balls to get a vision of the future to anticipate the useful life of equipment purchases. Errors in those predictions will have dire consequences for financial success of a digital endeavor. We can point out the relative merits of the various formats, but will not predict at what point in the future HDTV or other technological breakthroughs may happen.

Except for those few who have well thought out justifications and commitments to either analog component or digital component suites, most facilities will continue to use some of their existing composite analog equipment. This makes sense from both an investment viewpoint and a technical one. At the same time, everyone currently using composite analog will find adequate justification to install some D2 recorders for the reasons pointed out above - both to replace old VTR equipment and to fill new VTR requirements.

A PRAGMATIC APPROACH

Buying a few D2 recorders, however, is not sufficient cause to go whole hog in setting up a composite digital facility. The limited number of 4FSc sources available and the relatively high costs likely to be associated with doing so will limit such investments. Granting that it is dubious to jump deeply into 4FSc production, existing D2 users would still like to take advantage of the benefits. Ampex agrees with the legitimate value of doing so but we question whether high priced multi-layer switchers are even warranted when additional generations do not yield any losses. To provide a more reasonably priced alternative to meet this need, Ampex has created an inexpensive D2 layering device dubbed ADAPT™ (Ampex Digital / Analog Production Tool).

This new ADAPT layering device does a single high quality key per pass in the composite digital domain. There are other devices similar to this already on the market, and because of its suitability for this phase in the video industry, we can be sure there will be more such devices appearing in the future.

We have taken the approach that existing analog equipment is far from dead. In all likelihood, with the integration of small hybrid keying devices into analog suites, facilities can realize a hybrid application where composite analog and composite digital equipment function seamlessly together providing the best benefits of both domains. Even component analog sources used for chromakeying can be utilized in this hybrid marriage. Large existing analog switchers can be used to layer together any combination of sources that can then be incorporated into a digital background in the digital domain. Further passes can be used to continue layering, adding either digital or analog layers to a D2 tape with no degradation in the background material.

To achieve such a hybrid possibility, we have provided both digital and analog inputs to ADAPT in a combined analog and digital switching bus. We have also given it a variety of interfaces and control methods to simplify incorporation into any existing analog suite. An example of such an integrated system is shown in the following diagram (figure 1).

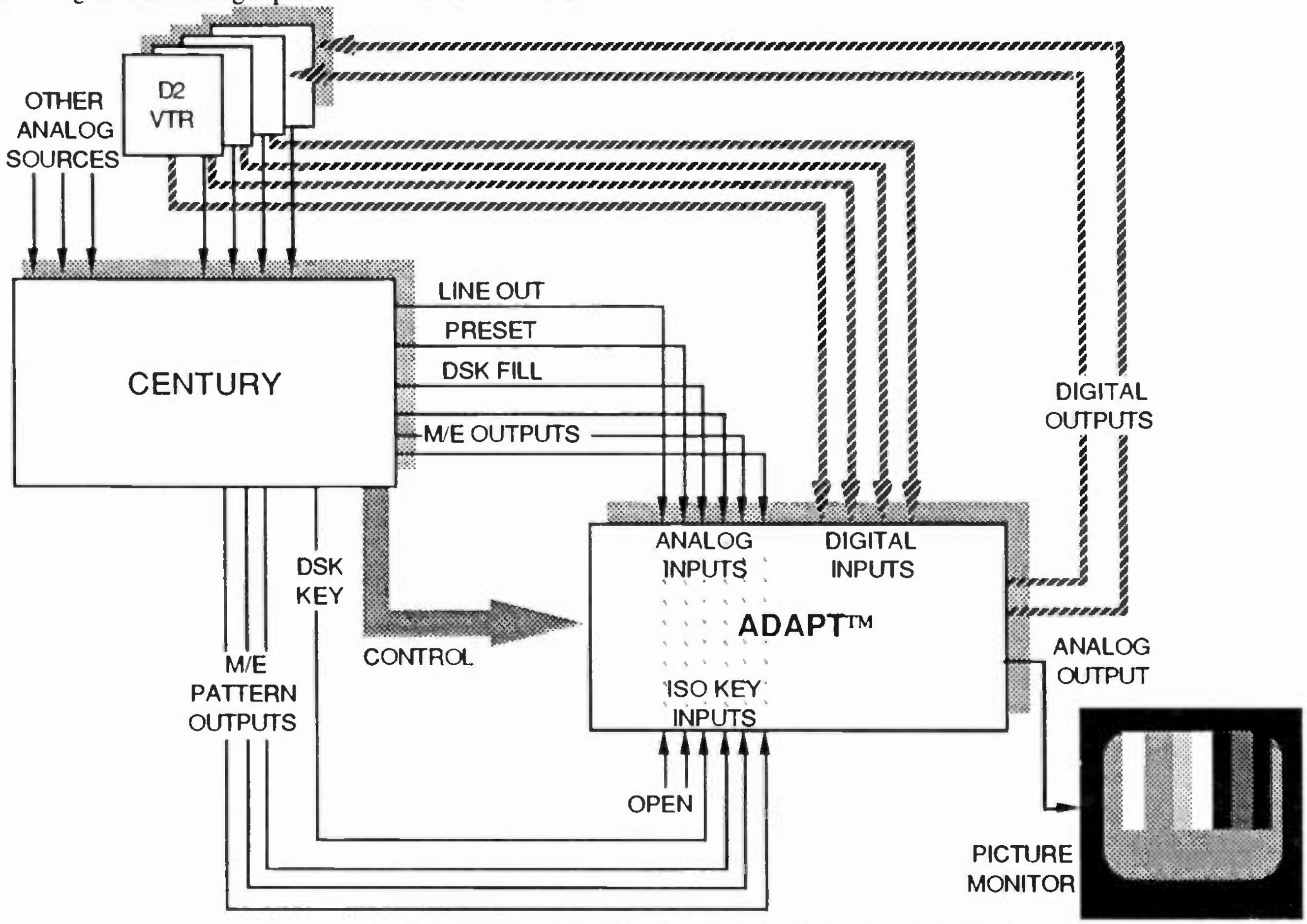

Figure 1. Typical Integrated Composite Digital and Analog Hybrid Suite

When incorporated into a studio in this manner, digital layering can be virtually transparent to the user. Using intelligent control systems between analog and digital devices, the digital layering device can be called upon when appropriate sources are available. When effects requiring analog capabilities are called for, the control will automatically employ the analog portion of the system. In complex layering situations, any number of analog images can be combined in the traditional manner, up to the capacity of the analog switcher, and then incorporated into a digital background with the digital processing electronics.

Because most of the sources in a typical application will be analog, there is no penalty for combining them in the analog domain. Many users may think that, in this type of system, quality will be lost when using the analog output of component digital devices. In most cases, there will be no appreciable difference between transcoding the 4:2:2 signal directly to 4FSc composite digital, or taking the composite analog output of the device and using the ADAPT A/D to do the conversion. Either way it will be combined digitally. While it is true that slightly better signals may be obtained with very high quality devices and very expensive D1/D2 transcoders, the difference is not likely to be visible. More importantly, once the video has been converted by either means, it will never suffer from any additional degradation.

Such a device is not limited to being tied to a big switcher. Endowed with an editor control port of its own as well as programmable GPI's, and a simple dedicated control panel, ADAPT can serve in stand alone layering applications tied only to D2 recorders as shown in the next diagram (figure 2).

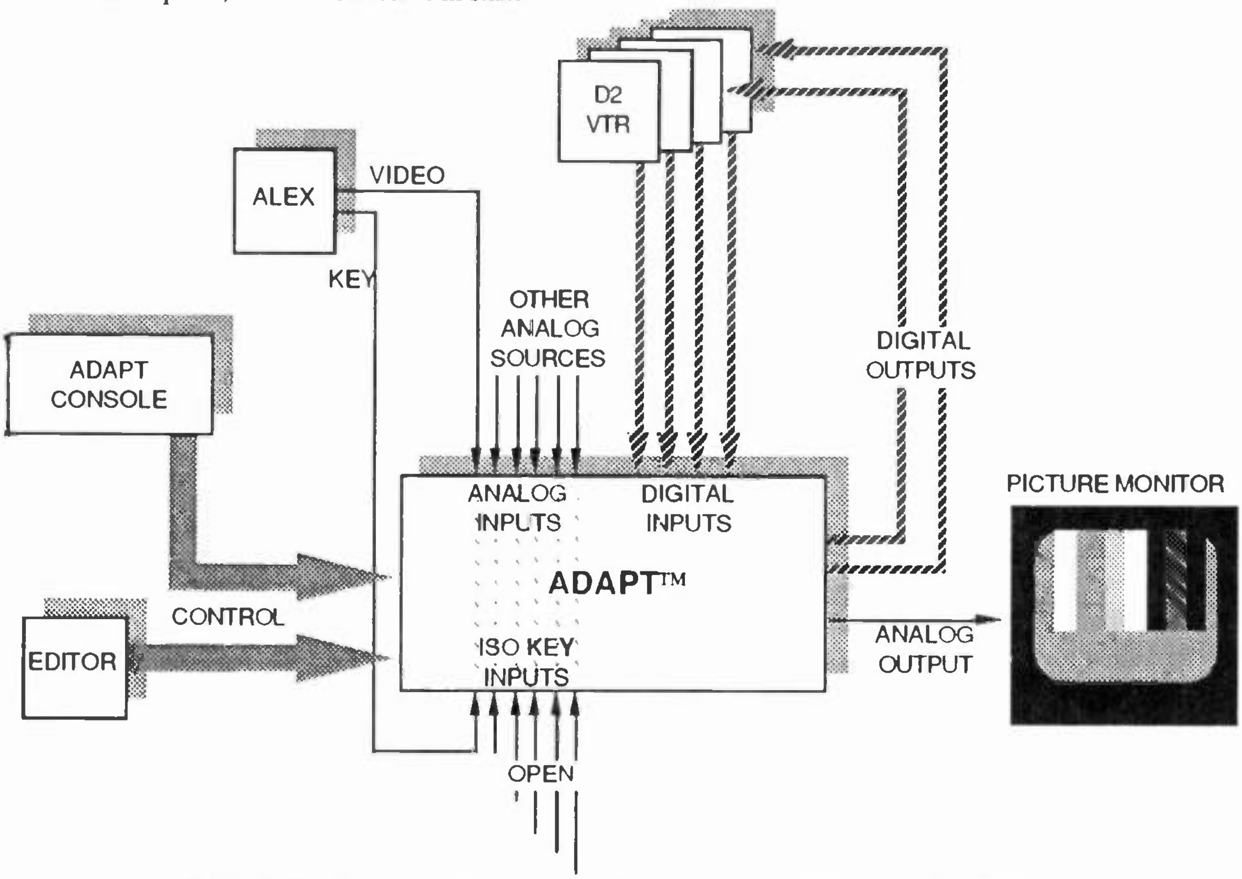

Figure 1. Typical Stand Alone Application for a Simple Digital Layering Device

Using the "read-before-write" feature available on composite digital recorders, digital layering can be accomplished with even just one D2 machine and a digital keyer. To further facilitate compatibility for a wide variety of applications, we have provided our layering device with the ability to create chroma nulled chromakeys using Ampex's Spectrakey™ process. Layering can be done with traditional keys or using true additive mattes. It can provide a digital key output to enable recording a matte reel in the D2 domain, and do background mixes and fades or even create monochromes or sepias. Working internally in parallel composite digital, we also offer individual I/O serial options to promote incorporation into any facility structure.

CONCLUSION

This paper has attempted to show that there are benefits to be gained from each of the four video formats in wide use today. Decisions being made over formats to use for the future should be made in light of individual application demands and individual views of how soon future change will be upon us.

For most users, D2 recorders will make sense in some part if not all their operations over the next few years. It is not necessary to make large investments in basic plant changes or in big, expensive composite digital switcher to take advantage of the superior performance available from these D2 recorders. In many applications, simple, low cost layering devices used alone or in conjunction with new or existing analog switchers. This approach can provide all the advantages that many equipment buyers would have expected to require much larger investments and still take advantage of their existing equipment.

A ½" COMPOSITE DIGITAL VTR FORMAT

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INTRODUCTION

Present broadcast analog VTR formats are the composite 1" format and the component 1/2" MII and Betacam formats. First generation picture quality is satisfactory for broadcast use. However, repeated dubbing severely degrades picture and sound quality. Digital VTR's were developed to solve this problem. The first digital VTR standard developed was D-1 component, and this was followed by D-2 composite. Both of these formats utilize cassettes and offer greater convenience than the 1" format. However the tape is 19 mm wide, so the cassettes are large and do not compare well with the compact 1/2" analog MII and Betacam that have become so popular all over the world. Especially noteworthy is the fact that these formats cannot be easily adapted for ENG/EFP applications, where the 1/2" analog formats are widely used. It is possible to use D-1 or D-2 in the studio and 1/2" analog VTR for ENG/EFP, but this method invites several problems. One is that dubbing is often necessary. Another is that provisions for using and storing two physically different styles of cassettes must be made. Both entail considerable complexity and operational inconvenience in addition to increased tape and storage costs. Also, using an analog source tape for a digital VTR does not allow the high quality of the digital video and audio recording to be fully exploited.

The above factors have led us to develop a single tape format with 1/2" cassettes that can be used for all applications from studio VTR's to camera/recorders and cart machines. However, to achieve more than twice the recording capacity of the D-2 format the following two problems had to be overcome. They are:

1) The achievement a low error rate in spite of high-density recording.
2) The achieving tape interchange compatibility in spite of a narrow track pitch.

We strove to develop new techniques that would allow us to meet the above challenges. These are:

1) A new 8-14 Code modulation scheme, allowing for 14% greater recording density when compared to Miller squared,
2) An error correction strategy and an LSI error correction processor allowing random and burst error correction capabilities

several times better than D-2 without increasing redundancy,
3) Provisions for a format structure which improves digital audio, since audio has greater susceptibility to degradation than video,
4) A recording method with guardband only at the editing points, which are especially susceptible to tracking errors, thus minimizing the error rate at editing points and throughout the edit,
5) A low tension tape transport allowing gentle head-to-tape contact,
6) An optimized drum diameter, which lengthens head life by reduction of relative head-to-tape speed, decreases power consumption by lighter drive load, and gives advantages in smaller size, and lighter weight, and
7) A superstructured nitride film head which improves the recording capability over the entire frequency range.

These new techniques have led to the goal of the digital VTR using a 1/2" cassette only slightly larger than that of MII with a recording time of two hours. In addition to reducing tape operating costs, reliability has been increased.

For increased convenience we also strove to develop other new technology that would allow:

8) A format to accommodate the recording of auxiliary data of 768 bytes x 1 lines without increasing redundancy, and
9) A format to provide superior color pictures during shuttle playback.

The 1/2 inch digital format is summarized below, and followed by a more detailed explanation. Finally, some of the more novel features of the first 1/2" digital studio VTR are outlined.

ADVANCED TECHNIQUES OF 1/2" DIGITAL FORMAT

Figure 1 shows the location of recorded tracks of the 1/2" digital format and Table 1 shows some of the main items of the 1/2" digital format. Figure 2 shows a size comparison between 1/2" L cassette with maximum 125 minute play time and D-2 M cassette with maximum 93 minute play time.

After the video signal is sampled at 4 times the color subcarrier frequency and quantized at 8 bits, it is divided into two channels. Both channels are slant azimuth recorded by head pairs at a rate of 6 helical tracks per field. The audio signal is sampled at 48 kHz for each of the 4 audio channels, quantized at 16 to 20 bits and recorded at both edges of the video helical track, two sectors per edge. In addition to the helical tracks, there are also 3 longitudinal tracks for cue, control, and Time Code.

New 8-14 Code

Various channel codes have been tried for every digital recorder over the years but yet no one method has been able to receive general recognition as the best. In the digital VTR field different channel codes are also used, as seen by the use Miller squared for D-2 and randomized NRZ for D-1. Randomized NRZ has the advantage of relatively low high-frequency characteristics. However, because of its DC content it is easy for intersymbol interference to be generated.In addition, the error rate is easily influenced by the content of the input signal. Miller squared is DC free and its low-frequency characteristics are excellent but it has the disadvantage of having relatively high high-frequency characteristics.

The new 8-14 Code is also DC free like Miller squared. And by using azimuth recording, cross-talk from adjacent tracks can be kept to a minimum. Intersymbol interference can be more readily suppressed as well. Moreover, it has high frequency characteristics of only 14/16 (87.5 %) of those of Miller squared. In other words, higher density recording is possible. The merits of high-frequency characteristics equivalent to those of randomized NRZ are obtained without any of the disadvantages.

Figure 3 shows the change in the error rates of the various channel codes when tracking is misaligned. As can be seen, the new 8-14 Code offers a core technology that combines the strong points of both randomized NRZ and Miller squared allowing more than twice the capacity of D-2 for high-density recording.

Error Correction Code

One of the fundamental differences between a digital and an analog VTR is that the digital VTR has an error correction capability that enables complete recovery from deterioration in the S/N ratio of the RF channel as long as that deterioration does not exceed certain limits. This capability also greatly improves the reliability of digital VTRS. However, if a digital VTR does not have good error correction capabilities then it is not much different from an analog VTR, especially in regard to dubbing performance. Therefore, this error correction capability should be maximized without exceeding reasonable hardware and software limits.

Figure 4 shows the video and audio block structure of the 1/2" digital format. Both use the Reed-Solomon product code block for their basic structure. The inner code is the same for both video and audio with 8 check bytes, the same as D-2. However, the error correction processor is more powerful and is capable of correcting 4 error bytes. As a result, random error correction capability is greatly improved. Also, the check bytes for the outer code, both video and audio, are 8 bytes, twice that of the D-2 method and allowing for correction of 8 error bytes. Thus, error correction capability from a random error to an error which is one several hundreds of bytes long is vastly improved. The audio outer code is 16 bytes long and is quite short in comparison to that of the video code (136 bytes). Therefore, audio correction capacity has been further increased in comparison to that for video. In addition, the product code block for both audio and video covers 1 field whereas that of D-2 covers only 1/3 field. This makes the maximum correctable burst error length about 3 times longer (6984 bytes) than that of D-2.

Figure 5 shows a comparison of random error correction capabilities in video. As can be seen, error correction capacity has been improved across the board from random errors to burst errors several thousands of bytes in length. The above mentioned improvements in error correction capabilities result in a 1/2" digital format which is very robust and resilient to degradation, even in non-ideal conditions.

Digital Audio

There are two possible reasons why digital audio signals have greater susceptibility to degradation than video. One is that the human sense of hearing is more attuned to audio errors than the eye for visual errors. The other reason arises from problems related to audio channel editing. When an audio channel is edited, it is not only the audio sectors at the edit in/out points which become adjacent to the pre-recorded old sectors,but all the new audio sectors between as well. This results in both the new and old sectors being subject to degradation from tracking error. Digital audio therefore requires an even more robust recording than video.

To solve this problem for the 1/2" digital format, the following steps have been taken:

1) The error correction power of the outer code structure is increased by making the check block 8 bytes long, the same as the data bytes.
2) The burst error correction power is improved by increasing the dimension of the Reed-Solomon product code block to cover one field.
3)The sample shuffling range is increased to cover one field. This distributes contiguous. Samples of the audio signal onto different sectors so that a burst error one sector long is converted to single occasional errors in continuous samples of the audio signal. Thus, highly precise interpolation can be provided from the adjacent samples of an error sample.
4) The adjacent audio sectors to be recorded by head pairs are distributed in the direction of the track pitch (Figure 1). In addition, samples of the same audio channel are distributed to the adjacent audio sectors (Figure 1). As a result, during editing of a single audio channel, one sector cannot be degraded by the simultaneous editing of both adjacent sectors. In other words, degradation is reduced by half.
5) By sector shuffling (Figure 1), which distributes sectors of an audio channel to the opposite edges of the tape, more effective compensation can be made for longitudinal tape dropouts.
6) Interference factors arising from single audio channel editing are reduced by a recording method with guardband only at the editing point. (This is explained in the next section.)
The above mentioned improvements in error correction and format structure results in a digital audio recording that is very robust and powerful.

Recording Method with Guardband at Editing Point

In order to maintain tape compatibility for the 1/2" tape format with a track pitch about half that of D-2, several items had to be improved:
1) The reduction of inter-channel cross talk through the use of the new DC-free 8-14 Modulation Code and azimuth recording,
2) The improvement of error correction capabilities across the board, from random to burst errors, and

3) Improving the handling of digital audio. These 3 items have been discussed above. The fourth, improving the recording footprint, will now be discussed.

Tracking errors can be attributed to two factors. One factor is the linearity errors caused by drum machining tolerances or tape weave. The other being track shifting errors caused by head height or tracking adjustment errors. Using two VTR's with opposing tracking error, the worst case error rate is at the in/out insert points when observing a tape which was recorded on one machine, insert-edited on a second machine, and then played back on the original machine. In this case the degradation in the recording footprint can be attributed to two factors. One is that the track being recorded over by a track of the same azimuth is not fully erased and to some extent still remains intact. This is the so called "same azimuth incomplete erasure" problem. The other factor is that adjacent tracks are also partially erased by the new information being recorded. This is the " track shaving" problem. Of the two, the incomplete erasure problem is the more serious, requiring a tracking error of only half that required by the track shaving problem to the produce a given increase in error rate.

The 1/2" digital format recording footprint has azimuth and guardband-less recording as its base but in addition, at edit points only, a guardband is adopted. This results in a greatly reduced amount of errors due to the same azimuth incomplete erasure problem. Also, for areas other than the edit points, since the entire track pitch width is used for recording, margin dropout is reduced. Therefore, a significant error rate reduction has been achieved at edit points and a safe margin at areas other than edit points has also been obtained.

These new techniques overcome the traditional limitations of narrow track recording and result in a broadcast recording system ideally suited to repeated editing and robust tape interchange.

<u>Video Sample Recording Limits</u>
Color shuttle picture in playback is an important aspect in the design of the recording format for composite digital VTR's. The samples in a shuttle playback which comprise a field are simultaneously gathered from several fields on the recorded tape. However, if there is no fixed color sub-carrier phase relationship among the samples, the color of shuttle pictures will become chaotic. Therefore, the system prescribes that recorded samples of a field are taken so that the color subcarrier phase is the same at the same address within the limits regardless of the recorded field. The D-2 method uses 2-line vertical shift recording to achieve this however, as a result vertical resolution is considerably degraded during shuttle playback. The 1/2" digital format uses 1-line vertical shift and 2-sample horizontal shift recording. As a result, the shuttle playback picture quality resolution is improved. Additionally, it is possible to assign one line to the auxiliary data area, which is not possible with D-2.

In the future, the serial digital composite/component interface will be increasingly used in broadcasting studios. By making use of video blanking and this serial interface, it will be possible to transfer auxiliary data related to the video information. Therefore, a digital VTR should also be capable of recording this auxiliary data as well as the video information. Traditionally, user bits in Time Code was one method of storing additional data on tape. This however is likely to be insufficient for automation systems which are advancing in sophistication. For example, one line will provide 768 bytes of user data which, when addressed by the serial digital interface, will provide valuable extra information for such systems.

Low Tension Tape Transport

The 1/2" digital VTR tape transport system has only one stationary post with a small friction angle. Increases in load due to tape-post friction are kept to a minimum. Stationary posts easily gather dust that can scratch the base or magnetic surfaces of the tape, increasing dropouts and demagnetization. Even though the base material is the contact surface, the same problems occur with thin tapes.

The low friction design also prevents increases in tape tension (approximately 30 g). Head-to-tape contact pressure greatly influences head wear. The higher the tension, the faster the head wear and, conversely, a the lower the tension, the lower the wear. Therefore, a low tension tape transport ensures longer head life. As tapes become thinner, allowable tape tension is also lowered. A low tension tape transport makes it possible to use thinner tape and also ensures longer tape life. When head-to-tape contact pressure is optimum, the lubricant contained in the tape works more effectively. This effectiveness is lost when tape tension is too high, causing head clogging and abrasion of the tape, thus tape life is shortened. In addition, when tape tension is high, the contact pressure between the lead surface of the lower drum and tape edge is also high, wearing the lead portion of the drum, distorting the lead portion of the tape, and adversely affecting tape interchange. If tape tension is high, the acoustic noise of the tape slapping against the rotating head also increases. The faster the rotation, the louder the sound becomes. The smaller 1/2" digital drum size combined with low tape tension provides for quite operation with excellent interchange.

SPECIFICATIONS AND FEATURES OF STUDIO VTR

Figure 6 shows the studio VTR with the camera/recorder. A field recorder-player is also planned. Table 2 shows the preliminary specifications of the studio VTR. The height of the studio VTR is 6 rack units and the input/output specifications for video and audio are compatible with both D-2 and analog composite VTRs. The VTR also has operational features not normally found on existing broadcast digital VTRs.

The audio system is of particular interest. The audio analog to digital converters use the new floating conversion technique (3), and the digital to analog converters use the new Multi-stage Noise Shaping Principle. These converters have a capability of 20 bits per sample. The frequency response of the audio system is from 20 Hz to 20 kHz (+/-0.5 dB), the dynamic range is over 100 dB, and the distortion is less than 0.01 %. Thus, the studio VTR is the product which makes the best use of the 20 bit format.

Using the Automatic Tracking (AT) track following heads and some novel digital signal processing circuitry (DSP) the 1/2" digital VTR mimics the feel and sound of conventional analog audio recorders being "rock and rolled". This digital audio in jog feature increases the ease with which audio edit points can be located and reduces operator fatigue.

The studio VTR is fully equipped with assemble, insert, and audio split editing functions, as well as spot erase and variable memory editing capabilities. The read-modify-write video and audio editing function is also implemented. Audio edits are true cross fade with variable duration. There is no output timing delay between the digital/analog video, the 4 channels of digital audio, and the cue channel in any generation.

CONCLUSION

As discussed in this paper, the 1/2" digital format includes a significant number of new developments in recording technology. They provide the format with input/output signal specifications equivalent to D-2 but a cassette size only slightly larger than MII with two hour play time. Thus, tape running costs and tape storage costs are reduced. This is particularly important for producing a compact digital cart machine with a large library. All applications, from ENG/EFP to studio recording, post production, and on-air playback via cart machine, can be achieved in a single format, contributing greatly to studio automation and reducing labor costs whilst maintaining picture and sound quality of the highest standards.

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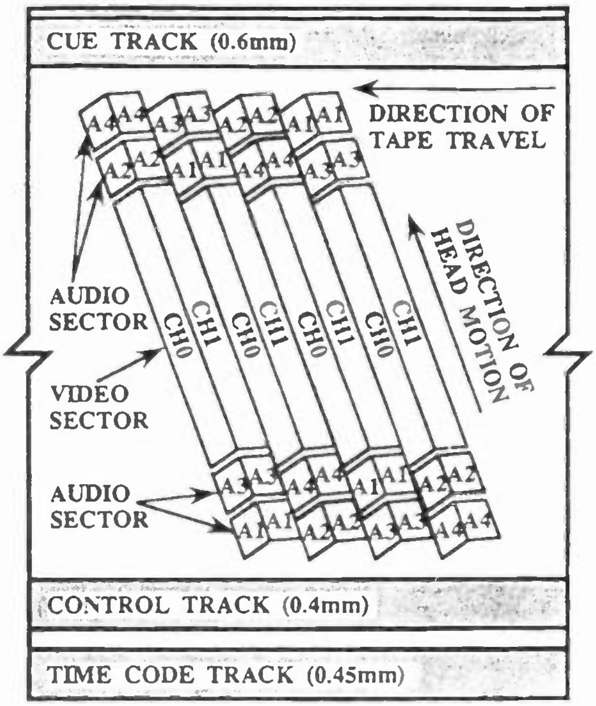

Figure 1. Location of recorded tracks

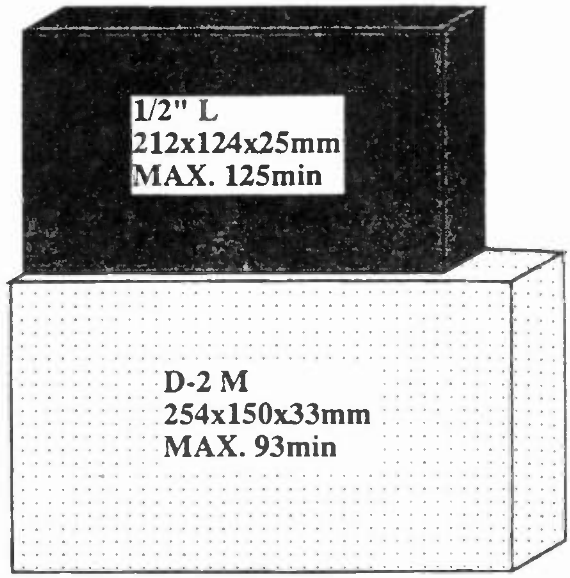

Figure 2. Cassette size comparison

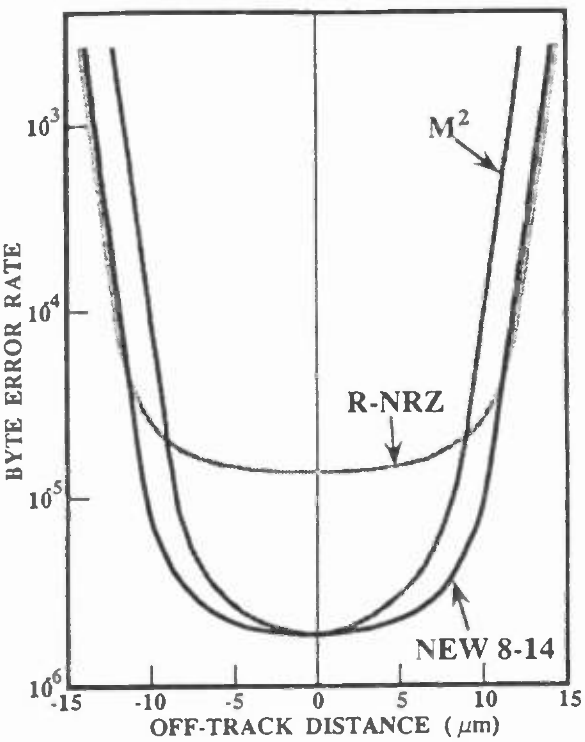

Figure 3. Off-track error rate

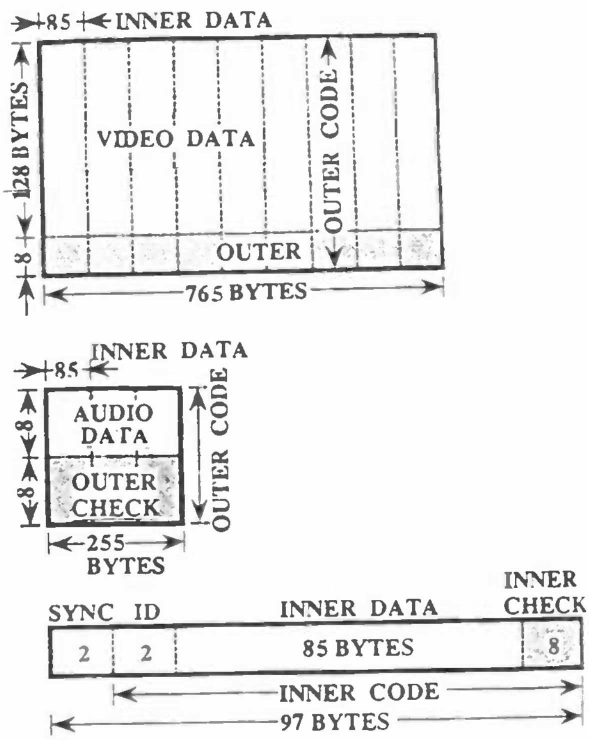

Figure 4. Block structure

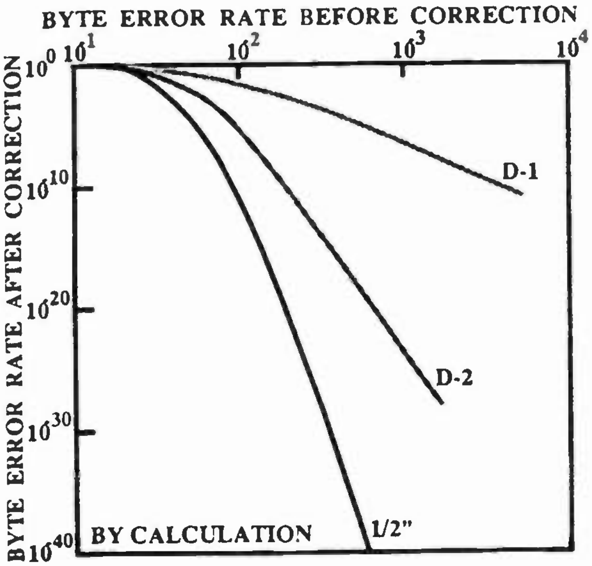

Figure 5. Error correction capability

Figure 6. 1/2" digital VTR

1/2" DIGITAL FORMAT

Tape width/material	12.65 mm/metal particle
Cassette Size/Max. Play Time	S: 161 x 98 x 25 mm/64 min. L: 212 x 124 x 25 mm/125 min.
Video	
Input signal	NTSC composite signal
Sampling frequency	4fsc=14.31818 MHz
Sampling Phase	123°, 33°, -57°, -147° (I/Q)
Bits per sample	8 bits
Recorded samples	768 bits/line
Recorded Lines	255 lines/field
Audio	
Sampling frequency	48 kHz
Bits per sample	16-20 bits
Number of channels	4 channels
Channel code	New 8-14
Error correction code	
Type	Reed-Solomon product code
Check bytes	Inner: 8 bytes, Outer: 8 bytes
Correctable burst	6984 bytes (Maximum)
Shuffling range	
Video	1 field
Audio	1 field
Drum diameter	76 mm
Effective wrap angle	178.1°
Scanner rotation speed	5400 rpm
Herical tracks per field	6 tracks/field
Tape speed	83.88 mm/sec
Track pitch	20.0 um
Minimum wavelength	0.77 um
Tape consumption	42.4% (D-2: 100%)

Table 1.

Specifications (Preliminary)

General	
Power consumption	Max. 400W
Weight	Approx. 48 kg
Dimension	424(W) x 265(H) x 667(D) mm
Video	
Video bandwidth	0 to 5.5 MHz ±0.5 dB 6.0 MHz -3 dB
S/N ratio	54 dB
Differential gain	Less than 2%
Differential phase	Less than 1°
Linearity (quantization noise)	Less than 2%
K factor	Less than 1%
Input level ajustable range	±3 dB (analog input only)
Output adustable range	
Video gain	-00 to +3 dB
Chroma gain	-00 to +3 dB
Hue	±15°
Setup	±15 IRE
Video phase	±1800 nsec
Sync phase	±1800 nsec
SC phase	More than 360°
Digital audio	
Frequency response	20 Hz to 20 kHz ±0.5 dB
Dynamic range	More than 100 dB (at 1 kHz, NAB unweighted)
Distortion	Less than 0.01%
Operating level	+8, +4, o, -20 dBm adjustable (LINE IN/OUT)
IN/OUT gain range	-00 to +12 dB

Table 2.

SPECTRE—DIGITAL TELEVISION TO UK HOMES IN THE EXISTING UHF TV BAND

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ABSTRACT

The paper describes low bit rate digital video coding, OFDM digital modulation (which will work in hostile interference and multipath conditions) and frequency planning methods in order that digital television may be transmitted to homes in the United Kingdom. The service would share the existing UHF TV band with the present analogue transmissions. The work described in this paper is being carried out under contract, for the Independent Television Commission, UK.

1. INTRODUCTION

The existing United Kingdom Terrestrial television service consists of 51 main stations and about 900 low power relay stations which deliver four programmes per transmitter to some 99.5% of the population. This service uses 44 8MHz channels in the UHF band. Over recent years there has been interest in increasing the number of programmes broadcast, and a fifth terrestrial service is planned which uses two 8MHz channels that have been released from the aeronautical radar service which also operates in the UHF band. Studies were made of a sixth service using precision offset within the present 44 channels, but this was found to yield a service for only a small population and it would have been uneconomical to run.

From this background, the work described in this paper began some two years ago [4]. A project named SPECTRE (Special Purpose Extra Channels for Terrestrial Radiocommunications Enhancements) was started, and looked at whether the existing 44 channels could be used for new services if modern more efficient modulation methods were used. Studies were made with three inherently different types of modulation: the first was an improved analogue AM/VSB scheme, the second was an all digital scheme and the third was a hybrid of the analogue and digital methods - it used some of the ideas put forward in the United States for power reduction by sending the low frequency picture information in the digital part of the signal.

The improved AM/VSB method used

suppressed carrier, suppressed synchronisation pulses and precision offset working. Although some channels were found in certain parts of the country it was thought unlikely to yield a useful service and had the disadvantage of requiring the existing 950 transmitters to be locked to precision offset. Moreover, all the continental transmitters in neighbouring countries rapidly began to require adjustment to precision offset as well! Even with this disadvantage this scheme was given due consideration because of its simplicity of receiver implementation.

Both the hybrid and the digital schemes did appear to find useful numbers of channels for a new service, and neither of these schemes used precision offset. The digital scheme produced a more elegant technical solution and under certain conditions was capable of creating more channels than the hybrid method.

The remainder of this paper describes the digital approach used as this has been the main emphasis of our work to date. However, the reader should not gain the impression that we have totally ruled out the hybrid approach just because we have only mentioned it briefly.

The digital scheme involves three major areas: digital video coding, digital modulation and frequency planning. The video coding scheme we have studied bit rate reduces CCIR Rec. 601 pictures from 216 Mbits/s to 12 Mbits/s, and gives about MAC type picture quality. The digital modulation scheme used has special anti-interference and anti-multipath properties and provides a 13.5 Mbit/s data capacity in an 8 MHz TV channel. The remaining 1.5 MBit/s of data capacity is used for sound and error protection. The frequency planning makes use of redundancy which is inherent in the existing frequency plan and predicts some four extra new channels at many of the main UK transmitter sites.

2. LOW BIT-RATE VIDEO CODING AND SERVICE QUALITY

The last few years have witnessed significant strides in the development of rugged and practical methods for redundancy reduction coding of television pictures from a source rate of over 200Mbit/s to very low bit-rates. System specifications are now emerging from the combined efforts of several international collaborative projects and World standardisation bodies, covering visual communication over a wide variety of source standards and transmission bit-rates. Perhaps the most striking outcome from all this work is not the picture quality achievable, remarkable though it is, but the fact that there has been almost total convergence on the same basic redundancy reduction algorithm - that of motion-compensated hybrid discrete cosine transform (DCT) coding.

The motion compensated hybrid DCT algorithm is shown in figure 1. The word 'hybrid' is used because a number of different techniques are employed to reduce the bit rate. One technique used to reduce the bit rate is transform coding and the method chosen is the discrete cosine transform (DCT). The DCT is used because it has properties which result in data organisation sympathetic with picture statistics and in a form eminently suitable for bit rate reduction. The DCT is a way of describing a picture in terms of spatial frequencies and the quantiser shown in figure 1 as the Q function is a method of providing a different number of bits for the different spatial frequencies found.

A second technique used to reduce the bit-rate is motion compensated prediction. The feedback loop in figure 1 which encompasses the DCT, Q, DCT^{-1}, Motion Estimator, Predictor and subtraction functions comprises the basic predictive loop. The predictive loop subtracts samples, which have been offset by the motion compensated predictor, in a previous frame from samples in a current frame and outputs the difference. The motion estimator finds where movement has occurred between a current frame and a previous frame by comparing groups of samples and generating a displacement (motion) vector which is transmitted in the data channel. The use of this vector in calculating differences aids the prediction process thereby reducing the differences and minimising the transmitted information. Further refinements of this technique are possible, such as interpolative coding in the temporal direction, and this enhancement is in fact being used in the SPECTRE system.

A third technique for bit-rate reduction is variable length coding. This is a technique where sequences of bits which occur more often are coded using short word lengths, and infrequently occurring sequences are coded with longer word lengths. This function is shown in figure 1 as VLC.

All these techniques when taken together produce a bit rate which varies with picture content. A buffer is used at the output to smooth this irregular data rate, and match it to a continuous data channel of 12Mbit/s.

The buffer level in figure 1 must be constantly monitored and feedback action taken in the video coding algorithm to prevent buffer overflow. This is achieved by scaling the DCT coefficients before quantisation in order to reduce the data input to the buffer. The level of scaling required varies with the level of information in the scene being coded. Scenes with lots of high detail and large areas of motion are much more critical than 'average' television scenes. Hence, an increase in level of feedback from the buffer is required to reduce the data rate produced by the algorithm which can only be accomplished by introducing some distortion into the critical areas of the picture.

Given this nature of low bit-rate coding, and accepting that there will always be some types of picture sequence which will result in their own perceptible, and possibly even objectionable distortion during coding, how is it possible to make a judgement on the adequacy of service quality? A practical approach is to determine that perceptible distortion should be present for less than some very small specified fraction of typical broadcast television material and objectionable distortion for significantly less than this. To enable such a quality assessment to be made for a low bit-rate coding method, it is helpful to have an analytical measure of picture sequence "criticality" which is related to the statistical redundancy (or lack of it) as experienced by a typical coder employing the motion-compensated DCT method. This measure can then be applied to thousands of short picture sequences taken from typical television, to construct a distribution similar to that shown in figure 2. Indicated on the graph is the condition corresponding to the most critical picture sequence which can be conveyed without perceptible distortion. Recordings have been made of many thousands of typical broadcast pictures. We are in the process of analysing these for the purpose of calibrating video codecs in terms of percentage of real broadcast pictures successfully coded.

3. DIGITAL MODULATION METHODS

The transmission of digital information in a band shared with existing broadcast television is not simple. The first criterion to be met is that the transmission of digital information is done in such a way that it does not interfere with the existing services. Studies to date have indicated that the only way to achieve sharing with the existing service is to radiate the digital signal at a very low level. More will be said on the methods of sharing in the section on frequency planning. It is quite possible to cover the same service area with a low power digital service because typical carrier/noise ratios (C/N) for digital signals are about 15dB whereas the analogue television service requires a C/N of greater than 40dB for a grade 4 picture. Moreover, it is quite possible using 'today's' technology to improve the noise figures of receivers so that transmitting the digital signal at 30dB less power than the conventional analogue television signal can provide the same coverage.

Since the digital signal is at a very much lower power than the existing television service it has to operate in a very hostile interference environment. Hence a suitable choice of modulation scheme is necessary to overcome this problem. A further difficulty arises from multipath propagation where reflected signals can impair the received data 'eye-height' due to intersymbol interference. The problem with intersymbol interference occurs when the delay spread of the multipath propagation is in the same order of magnitude as the symbol period of the digital transmission. There are basically three modulation methods which can be used to reduce the effects of interference and multipath. They are as follows:

i) adaptive equalisation - eg Kalman, gradient algorithm

ii) spread spectrum transmission

iii) parallel data transmission - orthogonal frequency division multiplexing

The technique of adaptive equalisation will help with the multipath problem but it will cause useful information to be filtered out when removing the interference. Spread spectrum will cope with both multipath and interference but requires a large RF to modulating signal bandwidth ratio in order to be effective. Since bandwidth is limited, spread spectrum is not really a solution. The third option of using orthogonal frequency division multiplexing (OFDM) is particularly attractive because of the way information is sent with this modulation scheme and the nature of the existing analogue television signal as an interferer. The OFDM signal also behaves well in the presence of multipath for reasons which will be described in the next section.

3.1 THE ORTHOGONAL FREQUENCY DIVISION MULTIPLEX MODULATION SCHEME

Orthogonal Frequency Division Multiplexing (OFDM) consists of a large number of carriers equally spaced in frequency, with each carrier modulated by some digital modulation method, eg QPSK. The spectrum of each modulated carrier is arranged to overlap the spectrum of its neighbouring carrier in such a way that the information content of each carrier is mutually orthogonal. In this way the OFDM modulation scheme becomes spectrally very efficient.

The orthogonality criterion can be achieved if the spectrum of each modulated carrier has a sinx/x shape as shown in figure 3. When all the carriers are combined the overall spectrum is a good approximation to a rectangular spectrum which just fits in the transmission channel. Since the OFDM signal has a flat spectrum that just fills the channel it closely approaches the 2bits/s/Hz theoretical limit of QPSK. The modulation scheme is simply a parallel data modem where there are n parallel data paths corresponding to the n carriers. Since the information on each carrier is independent and uncorrelated each individual carrier adds like noise according to a power law, and the overall signal in the time domain is in fact an excellent approximation to noise. The OFDM technique has been used in HF data links where Rayleigh fading is a problem [1, 2] and more recently in the EBU satellite sound broadcasting studies [3]. More detail on the OFDM technique and means of error coding are given in the above references.

The OFDM signal gives excellent performance in the presence of multipath because the bit period of each carrier is much longer than the delay spread of typical multipath reflections. Hence delayed versions of the OFDM signal do not significantly alter the eye-height. For fixed reception conditions there is only a 1-2dB degradation in the OFDM performance in the presence of very large multipath echoes.

The OFDM spectrum is ideally suited for use in a hostile interference environment where the interference can be approximated to single tones of continuous waves (CW) at known positions in the spectrum. This property of OFDM arises because information need not be sent on the carriers affected by the CW interferers. Hence, provided the interference can be well defined, portions in the transmitted spectrum can be essentially cut out leaving a spectral template where bona-fide information is transmitted. The receiver would naturally only look for information within the given template - hence it would ignore the interference.

4. THE EXISTING ANALOGUE TV SIGNAL AS AN INTERFERER

The spectrum of a television signal has two main components that represent continuous high power interfering elements, these are the vision carrier and the sound carrier. Although the colour sub-carrier and the digital sound sub-carrier are also present these are reduced in level by the dispersal effect of their modulating signals. Hence, these sub-carriers have a similar energy level to the vision modulation which has much less peak power than the vision and sound carrier levels. It then follows that a television signal may be approximated to a spectrum consisting of two continuous wave tones (CW), the vision carrier at 0MHz and the sound carrier at 6MHz.

Since the UHF television channels are spaced 8MHz apart, transmissions from a given mast will resemble four pairs of CW signals as shown in figure 4. The pairs of CW signals will always be spaced apart by an integer multiple of 8MHz. Given this property of the interferer it is possible to show that the wanted digital channel, which is the interference victim, is able to resist interference from the analogue TV signal. This makes it possible to transmit the digital signal at a level of 30dB less than the existing TV service - which is necessary in order to prevent the digital service causing interference to the analogue service.

Principal modes of interference into the digital service are as follows:

i) co-channel interference
ii) adjacent channel interference
iii) image channel interference
iv) third order intermodulation products

Co-channel interference from a distant analogue television transmitter on the same frequency as the digital transmitter may be taken into account by not transmitting information in the OFDM spectrum at the vision and sound carrier spectral positions. If two slots are cut out in the OFDM spectrum, one at the vision carrier position (0MHz) and one at the sound carrier position (6MHz), the level of interfering TV signal that may be tolerated may be considerably increased, compared to the case of not having slots in the OFDM spectrum; this is shown in figure 5.

Overlapping adjacent channel interference may be overcome by removing the transmission of information at the edges of the OFDM spectrum, again figure 5 indicates a template to achieve adjacent channel protection.

The effect of the image channel can be taken into account by suitable choice of receiver intermediate frequency (IF). The correct choice causes the vision and sound carrier components of the image channel to fall into the OFDM spectral template cut outs at 0MHz and 6MHz. Since the image channel is mixed into the IF amplifier with a local oscillator that is lower than the image channel, the spectrum of the image channel is reversed compared to the wanted channel. This property of the image channel is taken into account when determining the IF frequency.

Third order Intermodulation Products (IP) may also be taken into account if this is needed in the OFDM spectral template. Given an incoming signal as shown in figure 4, the low-noise front end of the receiver may generate intermodulation products from the large amplitude analogue TV carriers. Given that each TV channel is spaced 8MHz apart, and the vision and sound carriers are 6MHz apart for PAL-I, it can be shown that the most troublesome third order intermodulation products can only fall in four places in the spectrum 0MHz, 2MHz, 4MHz and 6MHz. The OFDM spectrum shown in figure 5 indicates the positions where additional cut-out portions would be placed if intermodulation products are a problem in the receiver. Studies to date indicate that intermodulation products are not likely to be a problem and these additional spectral holes are unlikely to be needed. However, an additional hole at the subcarrier position might be necessary to resist interference from the PAL-I signal when highly saturated signals such as colour bars are being transmitted.

5. EXPLOITING THE REDUNDANCY IN THE UHF TELEVISION FREQUENCY PLAN

The method of determining the sites for the main transmitter stations of the existing analogue UHF TV frequency plan was based on the use of nine frequencies A-I as shown in figure 6. A nine group lattice was used because the analogue television system is very sensitive to interference and requires a large frequency re-use distance. The large frequency re-use distance can be seen in figure 7, and is equal to three transmitter separations. The nine frequencies represent a redundant element in the plan because, if the analogue system were not so interference sensitive, a lattice comprising three frequency groups could be used. The following paragraphs explain how this piece of redundancy can be exploited.

Consider now adding to the mast at site B in figure 7 a new channel 'A' at much lower power so that it does not interfere with the existing analogue service on channel A. Since the digital service is able to accept very high levels of interference from the analogue service, because of the spectral conditioning described in the previous section, the frequency re-use distance may be drastically reduced between the analogue and digital services. The process of adding extra digital channels from existing masts may be continued as shown in figure 8 and will provide two new digital channel groups from each mast. In this way the nine frequency lattice becomes a lattice of three groups of three frequencies (ABC), (DEF), (GHI) and the present number of channels per transmitter can in theory be trebled! This would result in four analogue channels and eight digital channels from each main station mast in the UK!

Unfortunately, the real world is not as simple as the above analysis. The above would only be true providing the transmitters were sited on the ideal lattice as shown in figure 8 and there were also no relays present in the network. In practice the transmitter coverage areas form a more random distribution and the number of frequency sets in the lattice increases from three to four as the following will explain. In figure 8 the three frequency sets in the lattice were (ABC), (DEF) and (GHI). Where the transmitter coverage areas form a more random pattern, a four frequency set must be used; this arises because of the theory of the four colour map. Four colours are needed to colour a two-dimensional map without the same two colours touching. Without going into the analysis, the use of a four frequency set rather than a three frequency set would lead to about four analogue and four digital channels per transmitter mast.

A practical restriction to the above frequency plan concept is the fact that the UK transmission consists not only of a main station network but also of a low power relay network which uses the opposite hand of polarisation. Normally the relays would occupy the frequencies required for the digital transmissions. Hence it is necessary to look for redundancy in the use of channels used for relays in a given area in order to exploit the above ideas.

5.1 RELAY REDUNDANCY - THE 'TABOOS'

Channel redundancy occurs in a given area because relay stations in that area cannot use certain channels because of frequency planning restrictions. These are known as the 'taboo' channels. Further redundancy may be exploited in some transmitter areas where few relays are needed because the terrain is flat. In this case extra channels, in addition to the 'taboo' channels may be exploited for use.

The 'taboo' channels are as follows:

(i) The four channels that are co-channel with the main station's frequencies.

(ii) The four upper adjacent and four lower adjacent channels to the main station's frequencies.

(iii) The four channels that are N+5 and four channels that are N-5 to the main station's frequencies. These need to be excluded to prevent local oscillator interference.

(iv) The four channels that are N+9 and the four channels that are N-9 to the main station's frequencies. These need to be excluded to prevent image interference.

Combining all these restrictions together reveals that there are about 21 channels sterilised in a given locality and cannot be used by relays. Hence, in theory there is considerable redundancy in the relay network. Although in practice there are exceptions to these rules, it is likely to be possible to find channels available to broadcast a new digital service from many of the masts in the UK. It should be noted that coastal areas with a short sea path to neighbouring countries are more difficult due to different propagation conditions over water.

6. DISCUSSION

Studies to date indicate that digital television technology is likely to be able to make more efficient use of the UHF TV spectrum and possibly offer significant extra channels which can share the band with the existing service in the United Kingdom. To date we have only looked at QPSK modulation of the OFDM carriers. There is scope for considering higher order modulation schemes with OFDM, either to further improve the error performance or to provide a means for upgrading to higher bit rates in order to carry higher definition television.

The error performance may be improved by using a scheme such as 8PSK with trellis coding. The 8PSK scheme provides 3 bits/symbol; 2 bits/symbol are used to carry information giving the same capacity as QPSK, and the remaining 1 bit/symbol is used for error correction purposes in a trellis decoder.

At the expense of error performance, a higher bit rate can be transmitted in an 8MHz channel by using a 16 level modulation scheme such as 16QAM. 16QAM provides 4 bits/symbol and has a better performance than its 16PSK counterpart because it has more efficient modulation characteristics. However, there are some disadvantages of QAM systems over PSK systems and these are increased susceptibility to non-linear distortions and more complex carrier recovery requirements.

There is likely to be growing interest in 16 level modulation schemes because they enable the 12Mbit/s data rate to be increased to 24Mbit/s. This might make HDTV a possibility in an 8MHz TV channel. Our experience is that 12Mbit/s provides about MAC type quality and some 24Mbit/s is needed for higher definition pictures which work with larger screens. Clearly, there are a variety of options within these two extremes, which trade between error performance and picture quality.

7. ACKNOWLEDGEMENTS

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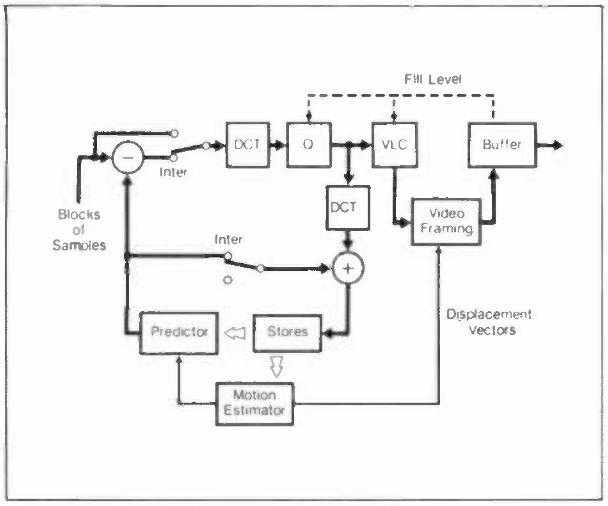

Fig. 1 *Motion compensated hybrid DCT algorithm*

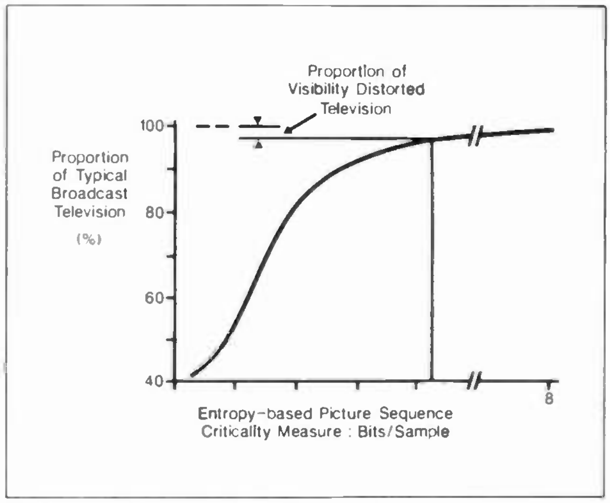

Fig. 2 *Calibration of picture sequence criticality*

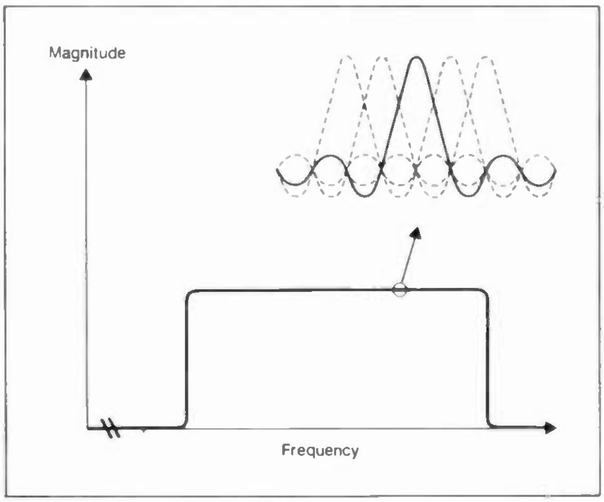

Fig. 3 *The O.F.D.M. spectrum*

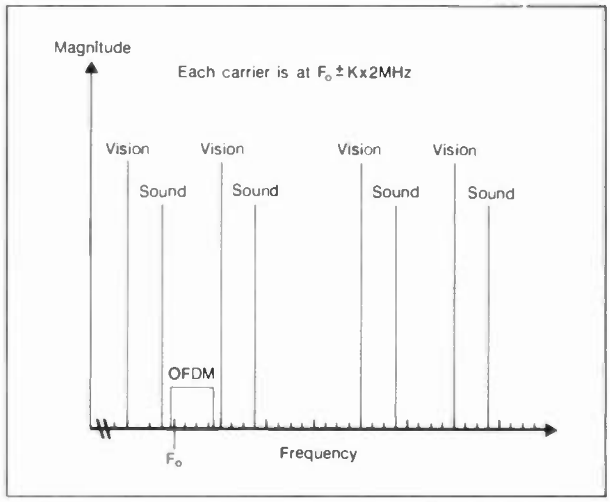

Fig. 4 *A typical spectrum of incoming signals to an O.F.D.M. receiver front end*

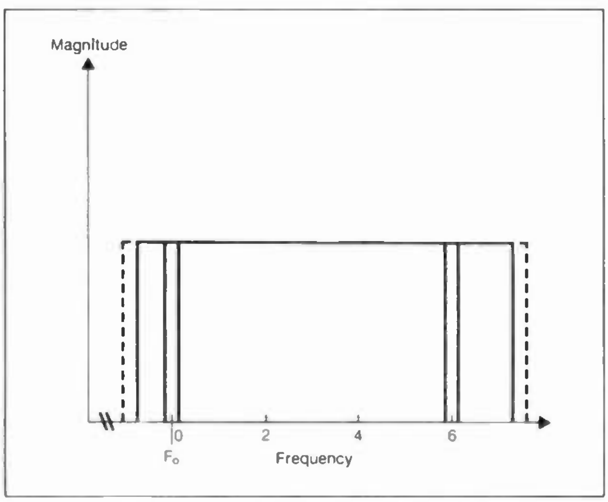

Fig. 5 *The conditioned, interference rejecting, O.F.D.M. spectrum*

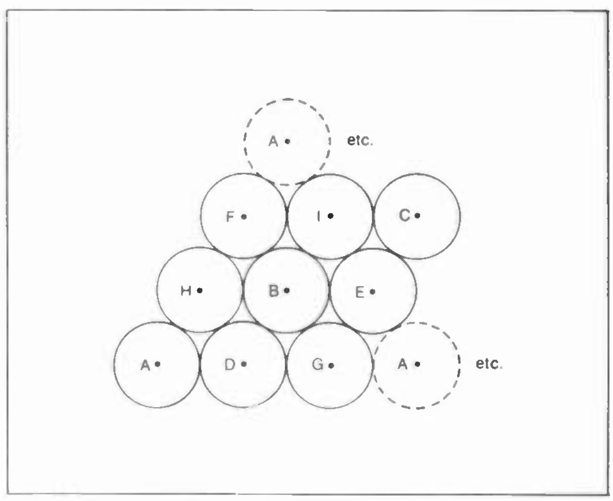

Fig. 6 *The present UHF plan*

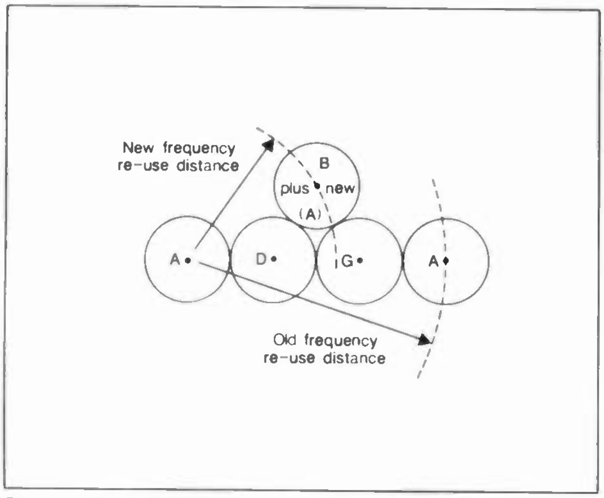

Fig. 7 The addition of a new service with reduced frequency re-use distance

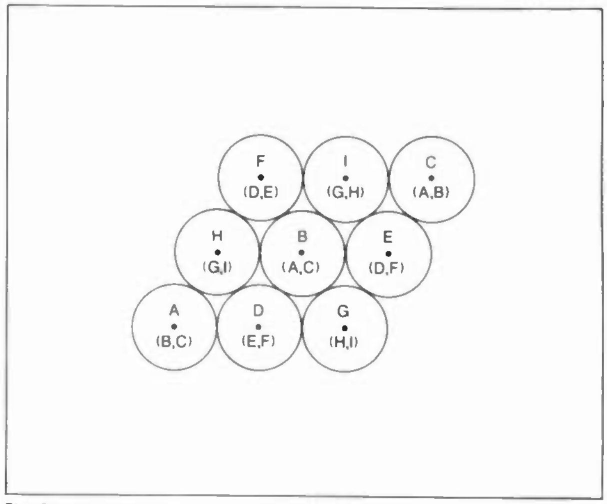

Fig. 8 A theoretical new lattice

FM SYSTEMS ENGINEERING AND IMPROVEMENT

Wednesday, April 17, 1991

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David Reaves, WHTZ-FM, Secaucus, New Jersey

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THE TECHNICAL FUTURE OF FM RADIO
Thomas B. Keller
Broadcast Technology Partners
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THE TECHNICAL FUTURE OF FM RADIO

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Bloomfield Hills, Michigan

ABSTRACT

The commercial dominance of FM radio over AM is now history, but the continuing success of FM radio is not guaranteed. The installation of high quality DAT and CD audio systems in automobiles will soon give the FM broadcaster keen competition for the driving audience. This competition will be intensified with the introduction of Digital Audio Broadcasting (DAB) by the end of this decade. The FM broadcast industry, which has not widely adopted any new transmission or audio technical improvement system for the last twenty years, should not repeat the mistakes made by AM broadcasters. If the FM broadcast industry does not keep up with technical developments, the FM industry, like AM, will have to take second place to the new technologies, and once FM's audience is lost it will never return. FM stereo quality can be as good as DAB. FM's limitations are multipath and stereo noise. To overcome these transmission problems the broadcasters will have to adopt and promote new FM technologies.

This paper proposes the adoption of four new technologies for the improvement of the FM stereo system. Operating procedures under the control of the broadcast engineer which will have a direct effect on the quality of the transmission system are also covered. Recommendations for future development in FM broadcasting are presented.

Audio processing, a subject that could stimulate several papers is not covered. Excessive audio processing will have an effect on interference.

REVIEW OF THE PROBLEM

Digital radio will not have the same dramatic quality improvement realized with the introduction of high definition television. The human eye resolves picture quality far in excess of the capabilities of the present NTSC television system. (See Table 1.)

TABLE 1. HDTV/NTSC		
	VERTICAL	HORIZONTAL
NTSC	310 LINES	330 LINES
HDTV	650 LINES	650 LINES

On the other hand, very few human ears can hear sound beyond the 15 kHz bandwidth of FM radio. The 20 kHz capacity of one of the proposed digital systems will be heard by only a few people.

Single channel stereo digital transmission is inherently sensitive to multipath distortion. To solve this problem, digital designers are spreading the data over hundreds of carriers operating over a frequency band of about 4 mHz. To make the service efficient many stereo programs share the same carriers in the 4 mHz band. Because multipath only affects a narrowband of frequencies at one time, only a few of the carriers will be lost. The effect of bit loss in this process will be corrected by the bit error correction system. This is a unique form of frequency diversity.

FM radio's major limitations are caused by transmission artifacts. The most troublesome of these artifacts as with digital is multipath distortion, especially as experienced in a moving vehicle. Stereo noise is a system problem associated with stereo transmissions which can be reduced with the introduction of a new technology

(FMX Stereo™). FM radio quality is also limited by allocations policy. When the FCC allocates stations too closely, receiver manufacturers will be forced to reduce receiver bandwidth to minimize interference [1]. Narrowband receivers have poor stereo performance, greater distortion, and poorer multipath performance. It is imperative that sufficient receiver bandwidth be maintained if the FM stereo system is to remain a quality service. Minimum receiver bandwidth for high quality performance should be established by the NRSC or other industry committees. Receiver manufacturers have already started to reduce bandwidth because of problems attributed to FCC Docket MM No. 80-90. The relationship between the FCC frequency allocations policy and the quality of an FM Stereo transmission system is critical to the future of the medium.

The future viability of FM broadcasting depends on the timely introduction of new technologies that overcome the problems of the FM stereo transmission system. Also, the FCC strict enforcement of a no-interference allocations policy must be maintained. This will allow the local terrestrial broadcaster to compete with the new audio delivery systems on a level playing field. The FM broadcaster is faced with problems of multipath, system stereo noise, receiver limitations, and adjacent channel interference. To solve these problems the receiver manufacturer and the broadcaster will have to simultaneously introduce the new technologies.

FM PROBLEMS

For the purpose of this paper, I have listed seven major FM transmission problems in Table 2. Table 3 shows the new technology that will reduce or eliminate each transmission problem.

TABLE 2. FM TRANSMISSION PROBLEMS	
1	MULTIPATH
2	STEREO NOISE
3	RECEIVER BLEND
4	RECEIVER BANDWIDTH
5	HIGH LEVEL MULTIPATH
6	FIRST ADJACENT INTERFERENCE
7	RECEIVER INTERMODULATION

TABLE 3. NEW TECHNOLOGIES	
1	DIVERSITY AUTOMOBILE ANTENNA
2	FMX Stereo™
3	WALSH FUNCTION DECODERS
4	NOISE CONTROLLED BLEND
5	SYNCHRONOUS AM CONTROL

1. MULTIPATH

With the introduction of stereo in the 1960's the problems of reception under multipath conditions were accentuated. Multipath is the propagation of the same signal over two or more different path lengths due to reflections. The reception of the two signals causes an increase or decrease in the level of the received signal. Theoretically, if a 180-degree out-of-phase exits between the direct and rejected signals, cancellation of the signals' relationship will occur. This may be thought of as a notch which occurs at the particular frequency (wavelength) where the 180° relationship exists because of the x/2 difference in path length. In fixed receiving locations this notch remains stationary. With the receiver in motion the notch sweeps through the FM signal and selectively distorts narrow bands of frequencies throughout the broadcast FM channel. When the notch distorts the 19 kHz pilot or the 38 kHz, the stereo decoder is disrupted.

To reduce these multipath problems almost all receiver manufacturers use the RF level blend system. This circuit uses the received radio frequency signal level to determine the amount of blend needed to reduce multipath distortion and stereo noise. Receiver blend circuits reduce the stereo difference signal at the expense of stereo separation. The weaker the signal, the greater the blend. This system does not have the ability to detect signal distortion resulting in the reproduction of annoying high signal level multipath noise. To reduce this problem the blend circuit of the receiver should be activated only when the stereo signal is distorted or a complete loss of pilot occurs, or a loss of signal level is experienced.

Four technologies are recommended to reduce the effects of multipath. Diversity receiving antenna systems will have the greatest effect on multipath. The control of synchronous AM at the transmitting plant [4], the implementation of the FMX Stereo™ system, and receiver noise controlled blend will further reduce the effects of multipath.

The development of the space diversity antenna system for the automotive FM receiver is one of the most important developments for reducing the effects of the multipath problem [2,3]. Diversity receivers have been available for several years, but their market penetration is low. The performance of these receivers has been good to excellent. With further research the performance of diversity systems can be improved. The receiver manufacturing industry has a halfhearted approach to the marketing of this technology. Automobile manufacturers have resisted the use of a second antenna even when this antenna is hidden in the rear window of the auto.

Synchronous Amplitude Modulation is a multipath aggravating distortion that a broadcaster has under his direct control [4]. The operation and design of the FM station antenna system, transmission line, and transmitter will significantly affect the performance of his station.

Broadcast engineers now have the tools and technical information to control source aggravated multipath. In most cases this will reduce the noise during the multipath event. Usually the solutions will not be costly. The benefits realized by reducing the multipath noise will justify the expense.

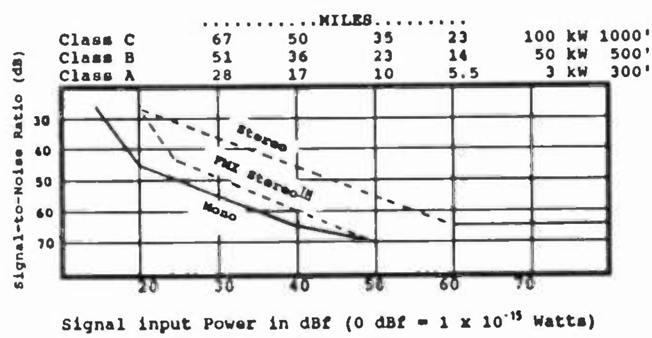
FMX Stereo™ is the system that reduces noise in stereo reception by compressing the difference signal before transmission and expanding it to its original dynamic range at the receiver [5]. With an improvement of 14dB in audio signal-to-noise ratio, some multipath effects will be reduced.

2. STEREO NOISE

All broadcasters are familiar with FM stereo noise. FM Stereo noise has been thoroughly documented. FMX Stereo™ is the technology that will significantly reduce stereo noise without reducing separation. This system will deliver an audio signal-to-noise ratio of 60 dB on all receivers using an average indoor antenna system within the 1 mV/m contour.

In field tests conducted in the Detroit area by Broadcast Technology Partners it has been found that when an automobile is located on a station's 1 mV/m contour, an average signal level of 43 dBf will be present at the FM radio input. For comparison purposes let us assume that the signal level delivered to the input terminals of a home receiver by an indoor antenna is about the same as the signal level delivered to an automobile receiver. With this assumption we can use the receiver quieting curve in Figure 1. It can be seen from the quieting curve that for signal levels of lower than 60 dBf, the received stereo noise will start increasing with distance from the station. The 60 dB S/N point is at an approximate distance of 7 miles for a Class A, 18.5 for a Class B, and 28 miles for a Class C station. Table 4 illustrates the signal-to-noise improvement zones which can be expected for the three classes of stations on home radios using FMX Stereo™. It is interesting to note that the 60 dB signal-to-noise coverage area has increased 3.8 times for a Class A station, 3 times for Class B station, and 2.5 times for a Class C station.

Combining FMX Stereo™ with a high quality tuner like the NAD Model 4300



TYPICAL TUNER WEIGHTING QUIETING CHARACTERISTICS

Figure 1.

TABLE 4. 60dB Audio S/N Contour		
	FMX Stereo™	Stereo
Class A	14 miles	7 miles
Class B	32 miles	18 miles
Class C	45 miles	28 miles
Receiver Input	43 dBf*	55 dBf*

TABLE 5A. 70dB Audio S/N Contour		
	FMX Stereo™	Stereo
Class A	12 miles	6 miles
Class B	28 miles	16 miles
Class C	41 miles	25 miles
Receiver Input	46 dBf*	58 dBf*

TABLE 5B. 75dB Audio S/N Contour		
	FMX Stereo™	Stereo
Class A	9 miles	5 miles
Class B	22 miles	12 miles
Class C	34 miles	20 miles
Receiver Input	51 dBf*	63 dBf*

* dBf is dB above one femtowatt (0 dBf = 1×10^{-15} Watts)

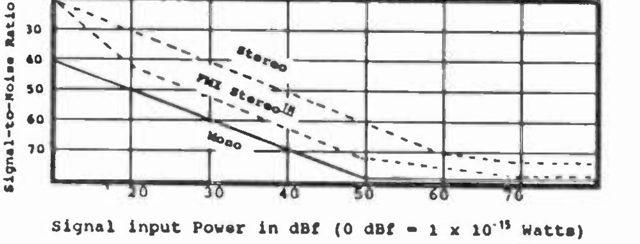


Figure 2. NAD Model 4300 Quieting

and using an average indoor antenna will give an audio signal-to-noise ratio of __70 dB__ within the 12 mile contour for a Class A station, 28 miles for a Class B, and 41 miles for a Class C station. (Class A = 3 kW at 300'; B = 50 kW at 500'; C = 100 kW at 1000') With the audio processing used by broadcasters, CD-like quality will extend to within 9% of the 1 mV/m contour for the three classes of stations.

Again for comparison purposes, assume that the signal level delivered to the input terminals of a home receiver by an indoor antenna is about the same as the signal level delivered to an automobile receiver. With this assumption and an input signal level of 46 dBf, the NAD Model 4300 receiver's quieting curve in Figure 2 is used to determine the 70 dB signal-to-noise contours for the three classes of stations. Table 5A illustrates the signal-to-noise improvement zones (70 dB) which can be expected for the three classes of stations on a home receiver of the NAD model 4300 quality with FMX Stereo™. Table 5B also show the 75 dB audio signal-to-noise ratio contour as compared to the conventional stereo contours with the NAD receiver for the three classes of stations.

3. RECEIVER BLEND

FM stereo has been accompanied by a significant increase in noise. Blend reduces stereo noise and has increased the mono coverage of FM stations at the price of stereo [6]. Blend is incorporated in all automobile radios. The problem is that a good portion of stations' automobile listeners are listening in areas where the signal level results in a blended or a partially blended receiver. Blend has resulted in maintaining station coverage with the loss of stereo, seriously limiting the ability of FM broadcasters to compete with the new technologies. The new 14 dB stereo noise reduction system allows radios to be built with full stereo separation down to input levels of 40 dBf and with usable separation to 32 dBf. Figure 3, the JVC Model KS-RX5500 and Model KS-RG9 FMX Stereo automobile radios, shows the stereo blend curve and the FMX Stereo™ blend curve. One can see that stereo coverage is greatly improved with this system.

100 kW	1000'	67.3	50.2	34.9	23.0	MILES
50 kW	500'	51.4	36.0	23.0	14.3	
3 kW	300'	27.7	17.0	9.6	5.5	

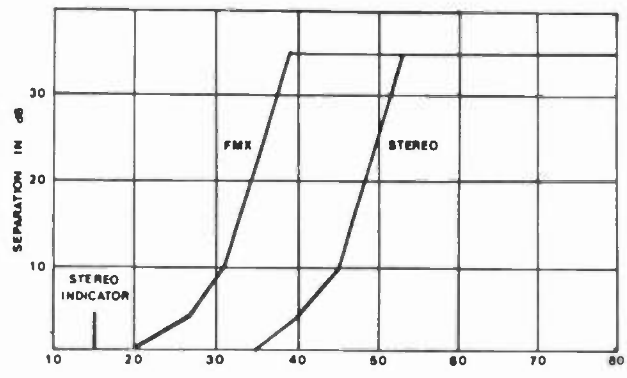


Figure 3

FMX STEREO SEPARATION vs. SIGNAL LEVEL JVC RECEIVER

4. RECEIVER BANDWIDTH

Receiver designers are concerned with an increase in FM interference caused by the use of short-spaced directional antennas, station power increases, and the increasing number of FM stations [1]. To solve the adjacent channel interference, receiver manufacturers reduce the bandwidth of FM radios. This is the same kind of allocations situation that forced AM receiver manufacturers to reduce AM bandwidth. The FM stereo broadcasters will pay the price of reduced bandwidth in exaggerated multipath noise, IM distortion, and stereo separation. European FM broadcasters are already experiencing the problems of a poor allocations policy. European FM receiver manufacturers have reduced the bandwidth to 120 kHz because of interference and an even poorer allocations policy as compared to receiver bandwidth in North America of 200 kHz. (Table 6.) Aggravated multipath, stereo separation reduced to 27 dB, IM distortion doubled, and increased distortion make up the situation in Europe. It's no wonder they are pioneering digital radio systems.

5. HIGH LEVEL MULTIPATH

A new FMX Stereo decoder IC has been developed with a noise-operated blend circuit that is an improvement over the conventional signal level system. This

TABLE 6. COMPARISON

AUTO RADIOS	EUROPE	U.S.
IF BANDWIDTH	120kHz	200kHz
STEREO SEPARATION AT 2 kHz	27 dB	35 dB
IM DISTORTION AT 5 kHz	1.7%	0.5%

circuit design assumes that the spectrum around the 19 kHz pilot is clear of program signals and is usable for measuring the received noise. The IC uses a quadrature detector to measure the noise and distortion at the pilot frequency. With the quadrature detector, the system will blend on distortion or noise. This means that the blend circuit will be actuated by multipath distorted or noisy signals at any RF level. Signal level controlled blend circuits will not respond to these distortions.

6. FIRST ADJACENT INTERFERENCE

First adjacent channel interference is a problem only too familiar to many FM broadcasters. Because of grandfathered allocation compromises, station power increases, and more stations on the air the problem will most likely get worse. In the past receiver designers have used narrowing the bandpass of the receiver to reduce this interference. Another solution is now available to the receiver designer, but first we must review the function of the stereo decoder.

The stereo decoder in the FM receiver is the device designed to convert the composite stereo signal to the left and right component signals. The PLL decoder is the only type commonly used in today's receivers. The PLL stereo decoder's switch operates in synchronism with the switch in the broadcast encoder (stereo generator). Because of its comparing square waves with sine waves, the PLL decoder has the disadvantage of being sensitive to the odd multiples of 38 kHz. This results in decoding the odd multiple signals at 114 and 190 kHz, making the receiver sensitive to the first adjacent channel interference.

With the introduction of the Walsh

function PLL stereo decoders, the PLL decoder is no longer sensitive to the odd multiples of 38 kHz. This reduces the sensitivity of the decoder to 190 kHz, resulting in an increased first adjacent channel rejection of 20 to 25 dB. Therefore, the use of PLL stereo decoders with Walsh functions reduces interference, increasing the coverage of FM stations and will make it unnecessary for receiver designers to narrow receiver bandwidth with present station separations.

7. RECEIVER INTERMODULATION

Intermodulation is a major cause of interference for FM broadcasters especially in strong signal areas where weaker signals are to be received. In the last few years FCC Docket MM 80-90 has encouraged stations to go to full facilities. Because of a lack of land to build tall towers, stations have found it necessary to unite with other stations and build community antenna systems. Because these high power facilities are often located in or near communities, these sites are ideal locations for intermodulation interference.

To overcome the stereo noise problems, receiver RF gain may be set higher than good intermodulation performance will allow. With the introduction of the FMX Stereo™ system, the need for a super-sensitive receiver to overcome FM stereo noise is reduced. This system will give future designers greater latitude in minimizing the effects of blanketing and adjacent channel interference.

RECOMMENDATIONS

The following recommendations are made to improve the quality of FM radio and to allow the service to be able to compete with the up-coming new technologies.

FCC:

* Maintain separation in allocation system

NRSC:

* Recommend automobile receiver minimum bandwidth
* Recommend an inter-modulation standard
* Review D/U ratios

Broadcasters:

* Maintain high standards of technical operating quality
* Implement new technologies
* Control synchronous AM
* Maintain high quality wide-band RF antenna system

Receiver Manufacturers:

* Implement FMX Stereo™
* Maintain sufficient bandwidth in receivers
* Implement diversity antenna systems
* Implement noise controlled blend
* Implement Walsh function decoders

CONCLUSION

Table 7 shows the relationship between seven problems and the five technologies. It can be seen that most of the problems of FM radio can be solved with the available technologies. The FM system can deliver a high quality artifacts-free signal to its audience.

FM broadcasters can be assured of a bright future for the industry if they aggressively work to implement the technologies which will improve the quality of the existing transmission system. A lesson can be learned from AM broadcasting, not to wait until the system is in trouble to implement quality improvements in the service. With the present success of the FM system, the broadcaster has a strong influence over the receiver manufacturing industry. It is the broadcasters' responsibility to implement these technologies at their own facilities and to take the lead in assuring the technical future of this industry.

FMX Stereo™ is a trademark of Broadcast Technology Partners

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[1] Richard D. Zerod, "Trends in FM Broadcasting: A Receiver Manufacturers Perspective" Presented at NRSC Meeting at the
NAB-89 Radio Conference.

[2] Emil L. Torick & Robert A. Finger., "FM Receivers in Automobiles: A Case For Diversity Reception" IEEE

	FM PROBLEMS / TECHNOLOGIES	DIVERSITY AUTO ANT.	FMX Stereo™	WALSH DECODERS	NOISE OPERATED BLEND	SYNC. AM CONTROL
1	MULTIPATH	P	S		S	P
2	STEREO NOISE		P			
3	RECEIVER BLEND		P		S	
4	RECEIVER BANDWIDTH			P		
5	HIGH LEVEL MULTIPATH		S		P	
6	FIRST ADJACENT INTERFERENCE		S	P		
7	RECEIVER INTERMODULATION		P			

P = PRIMARY S = SECONDARY

TABLE 7. FM PROBLEMS/TECHNOLOGY

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[3] William R. Rembo, "Switched Diversity for Mobile FM Broadcast Reception", Proceedings of the 38th NAB Engineering Conference Spring 1984.

[4] Mendenhall, Geoffrey N., "Techniques for Measuring Synchronous AM Noise in FM Transmitters", Proceedings of the 42nd NAB Engineering Conference, Spring 1988.

[5] E.L. Torick and T.B. Keller, "Improving the Signal-to-Noise Ratio and Coverage of FM Stereophonic Broadcasts," J. Audio Eng. Soc. Vol. Vol. 33, No. J. Audio Eng. Soc. Vol. 33, No. 12, December 1985

[6] Thomas B. Keller, "FMX Stereo Separation Versus Receiver Blend" Proceedings of the 1989 SBE & Broadcast Engineering Conference.

EFFECTS OF LIMITED BANDWIDTH TRANSMISSION PATHS IN FM ON SCA/RDS PERFORMANCE

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Abstract - This paper explores the effects of limited bandwidth transmission paths of FM broadcast stations on audio performance, and SCA/RDS performance.

Introduction

There has been much study and discussion regarding the effects of bandwidth limitation on FM signals in the past five years. Particular attention has focussed on the parameter of Synchronous AM. Indeed, this parameter is useful as an index of the total effective bandwidth, but experimentation and computer modeling has shown that Synchronous AM alone is not the most damaging effect of bandwidth limitation.

Types of Bandwidth Limitation

Bandwidth in an FM transmission system may be limited in many ways. The most often discussed limitations include the bandpass cavities which make up antenna combiner systems, and the output cavity design in the FM transmitter. Other areas which contribute include the antenna system, the grid matching circuit in the transmitter, a tuned intermediate power amplifier in the transmitter, and the front end / IF stages in the receiver. It is the cumulative effect of these filters which determines the overall performance of the system.

It is important to realize that there are at least two subcategories of these filters, those which precede a limiting amplifier, and those which do not. Traditionally, we have ignored those filters which precede limiting amplifiers, because it is presumed that the limiting amplifier will, by its nature, remove the amplitude effects on the signal and leave only the frequency modulation.

An example of a filter which precedes a limiting amplifier is the grid tuning / matching network, which resonates the substantial tube input capacitance and matches the 50 ohm input impedance from the IPA to the grid of the final amplifier. As the grid is normally driven into limiting, the amplitude effects of the input circuit Q masked, however the time delay induced damage that this filter may cause can be substantial.

Asymmetrical Group Delay is the problem

Group delay may be defined as the difference in time that is required for the various elements of a signal to propagate through the device. Group delay asymmetry is defined as the difference in group delay between the upper and lower sidebands. In the case of FM transmission, it is the difference in nanoseconds that it takes the upper vs lower sidebands to propagate through the transmitter. When the sidebands do not arrive at the output of the transmitter at the same time, distortion is created which manifests itself in crosstalk.

There is a group delay effect to any bandpass filter, and the effect of this group delay can be more damaging to the audio and subcarrier quality than the Synchronous AM effect. Worse, the group delay effect is NOT removed when the signal is passed through a limiting amplifier, nor can its effect be diagnosed with conventional Synchronous AM tests.

Computer Modeling

To study these effects, Broadcast Electronics co-operated with Quantics of Nevada City, CA to develop a software program which models the effects of these distortions on an FM signal. This program is available from Broadcast Electronics. The following series of graphs are generated by this program and show the differences in the generated composite baseband distortion components which are caused by the amplitude response effects (Synchronous AM) and asymmetrical group delay effects for two different cases. The first case [Figures 1 to 5] shows a second order synchronously tuned filter, each section 1 MHz wide, with the amplitude response centered on the passband, and the group delay of one section offset by 90 kHz. This is indicative of typical offset between amplitude and group delay found in many single tube transmitter designs.

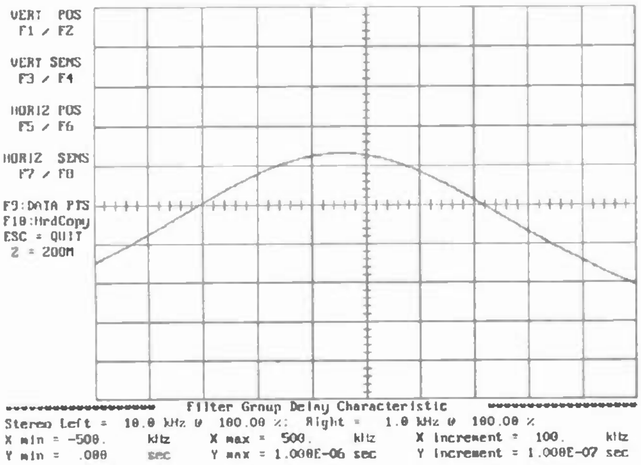

Figure 1. Group Delay Characteristic

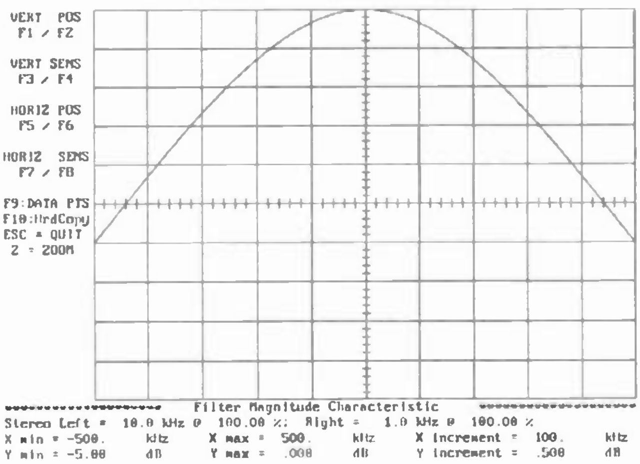

Figure 2. Magnitude Characteristic

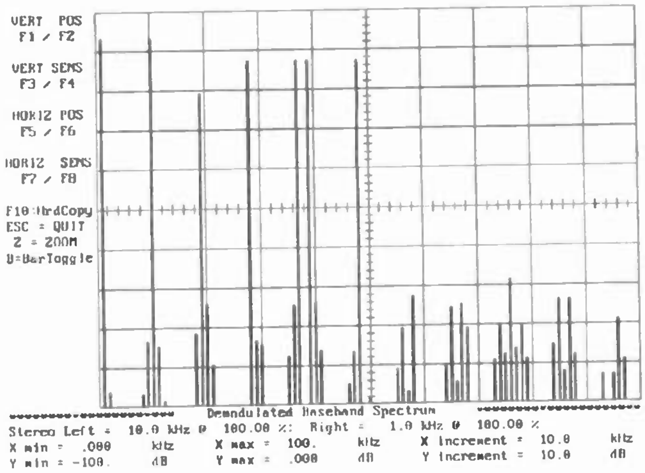

Figure 3. Total Composite Spectrum Effects

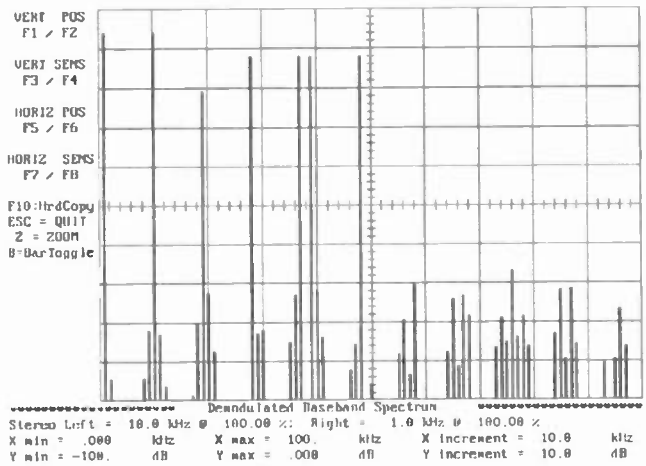

Figure 4. Group Delay Effects Only

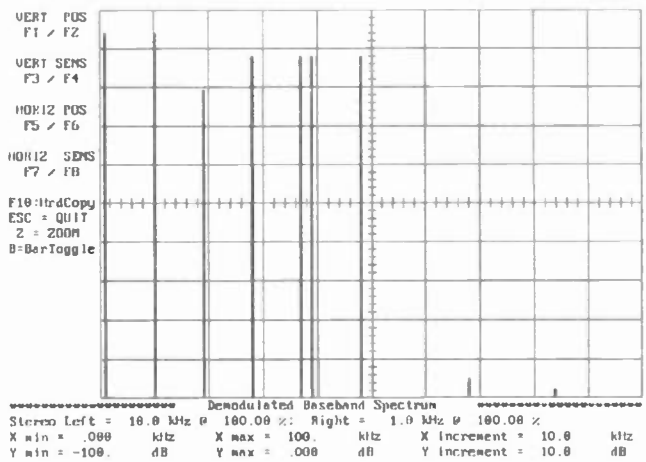

Figure 5. Magnitude Effects Only

Ignoring the six desired signals, which are at 10 and 1 kHz (L+R), 19 kHz (pilot), and 28, 37, 39 and 48 kHz (L-R), note the generation of significant distortion components, even though the Synchronous AM has been minimized by centering the amplitude response.

It is these baseband distortion products which interfere with not only the L+R and L-R signals, degrading the stereo performance of a station, but also with the SCA and RDS subcarriers which may exist above 53 kHz.

In case 2 [Figures 6 to 10], the identical modulating conditions and filter bandwidths are used. The only difference is that the group delay and amplitude response are now both centered in the passband. In both cases, the most serious effects are caused by the group delay effects, as opposed to the amplitude effects.

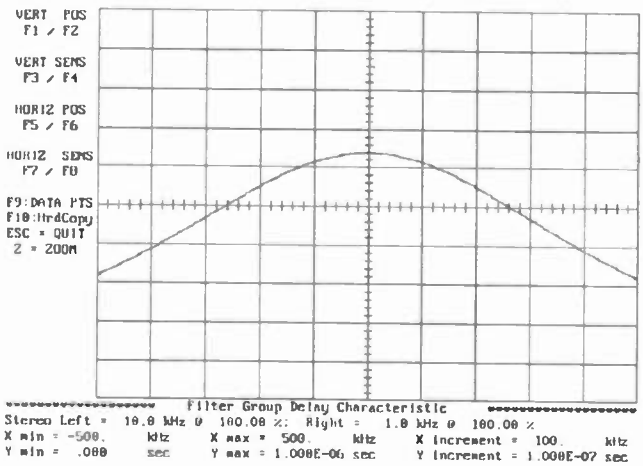

Figure 6. Group Delay Characteristic

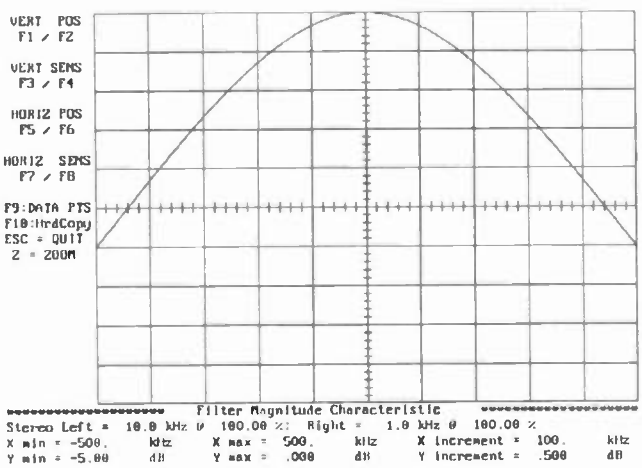

Figure 7. Magnitude Characteristic

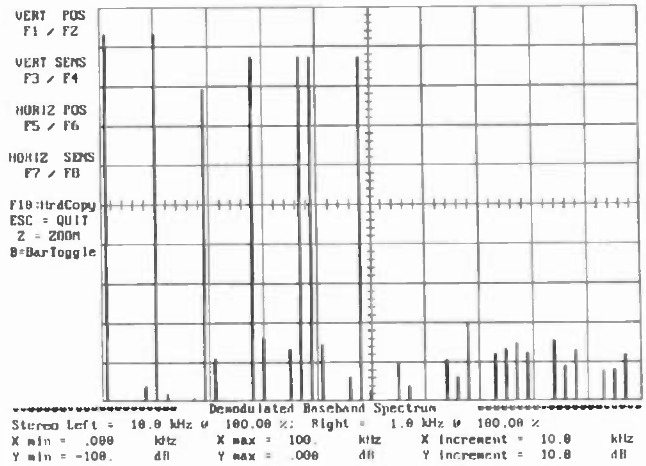

Figure 8. Total Composite Spectrum Effects

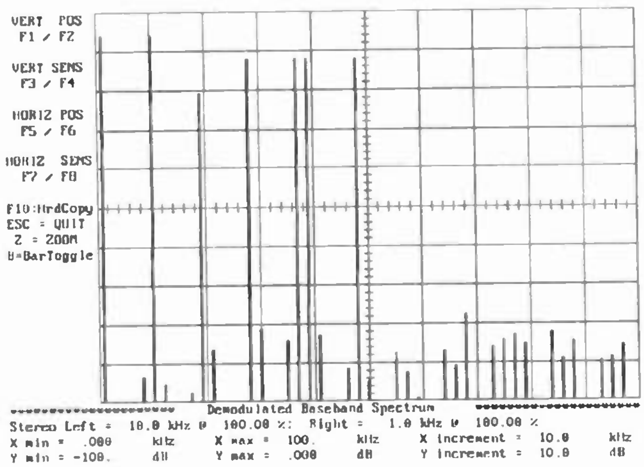

Figure 9. Group Delay Effects Only

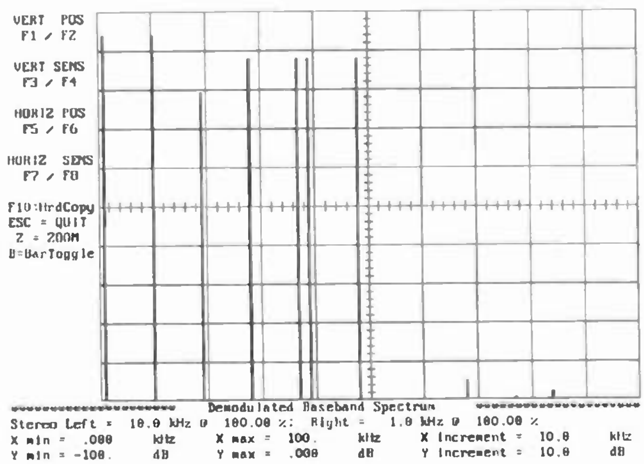

Figure 10. Magnitude Effects Only

FM Subcarriers

FM subcarriers have been growing in importance around the world in the last few years, with Radio Data service gaining in acceptance in Europe and elsewhere. These services rely on low level subcarriers being added to the FM baseband signal above 53 kHz. They are generally 20 dB lower in amplitude than the L+R signal, and thus are susceptible to crosstalk.

Typical subcarrier frequencies for SCA are 67 and 92 kHz, and for RDS, 57 kHz. It is easy to see that the distortion products generated in the graphs will cause significant interference to the subcarrier services.

Tuning for minimum crosstalk

As broadcast engineers, we have witnessed an interesting change in the way we tune FM transmitters. Most of us learned to tune for maximum power and maximum efficiency. Then, a few years ago, we learned how to tune for minimum Synchronous AM. However, now we have seen that, in order to minimize the effects of group delay on our baseband signal, we must find a way to tune for group delay symmetry.

FM broadcast transmitter RF power amplifiers are typically adjusted for minimum synchronous AM (incidental amplitude modulation) which results in symmetrical amplitude response. This will assure that the transmitter's amplitude passband is properly centered 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 will be true if the RF power amplifier circuit topology results in simultaneous symmetry of amplitude and group delay responses.

Actually, symmetry of the group delay response has a much greater effect on FM modulation distortion than the amplitude response. Tuning for symmetrical group delay will cause the phase/time delay errors to affect the upper and lower sidebands equally or symmetrically. The group delay response is constant if the phase shift versus frequency is linear. All components of the signal are delayed in time, but no phase distortion occurs.

The tuning points for symmetrical amplitude response and symmetrical group delay response usually do NOT coincide, depending on the circuit topology. Therefore, simply tuning for minimum synchronous AM (symmetrical amplitude response) does not necessarily result in best FM modulation performance.

Measurements taken on a typical FM transmitter as well as computer simulations showed that tuning the RF power amplifier for symmetrical group delay response resulted in minimum distortion and crosstalk. It confirms that group delay response asymmetry causes higher FM modulation distortion and crosstalk than amplitude response asymmetry. Therefore, RF power amplifier circuit topologies that exhibit coincidence of symmetrical amplitude and group delay responses will result in a better overall FM modulation performance. 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.

All optimization should be done with any automatic power control (APC) system disabled so that the APC will not chase the adjustment in an attempt to keep the output power constant. The transmitter should be connected to the normal antenna system rather than to a dummy load. This is because the resistance and reactance of the antenna will be different from the dummy load and the optimum tuning point of the transmitter will shift between the two different loads. The tuning sequence is:

Initial Tuning And Loading

The transmitter is first tuned for normal output power and proper efficiency according to the manufacturer's instruction manual. The meter readings should closely agree with those listed on the manufacturer's final test data sheet if the transmitter is being operated at the same frequency and power level into an acceptable load.

Input Tuning And Matching

The input tuning control should first be adjusted for maximum grid current and then fine tuned interactively with the input matching control for minimum reflected power to the driver stage. Note that the point of maximum grid current may not coincide with the minimum reflected power to a solid state driver. This is because a solid state driver may actually output more power at certain complex load impedances than into a 50 ohm resistive load. The main objective during input tuning is to obtain adequate grid current while providing a good match (minimum reflected power) to the coaxial transmission line from the driver. In the case of an older transmitter with a tube driver integrated into the grid circuit of the final amplifier, the driver plate tuning and the final grid tuning will be combined into one control which is adjusted for maximum grid current.

Output Tuning

The output tuning control adjusts the resonant frequency of the output circuit to match the carrier frequency. As resonance is reached, the plate current will drop while both the output power and screen current rise together. Under heavily loaded conditions this "dip" in plate current is not very pronounced, so tuning for a "peak" in screen current is often a more sensitive indicator of resonance.

Amplifiers utilizing a folded half wave cavity will display little interaction between output tuning and output loading because the output coupling loop is located at the RF voltage null point on the resonant line. Quarter wave cavities will require interactive adjustment of output tuning and output loading controls, since changes in loading will also affect the frequency of the resonant line.

Output Loading

There is a delicate balance between screen voltage and output loading for amplifiers utilizing a tetrode tube. Generally there is one combination of screen voltage and output loading where peak efficiency occurs. At a given screen voltage, increasing the amplifier loading will result in a decrease in screen current, while a decrease in loading will result in an increase in screen current. As the screen voltage is increased to get more output power, the loading must also be increased to prevent the screen current from reaching excessive levels. Further increases in screen voltage without increased loading will result in a screen overload without an increase in output power.

Automatic Power Control Headroom

Automatic power control (APC) feedback systems are utilized in many transmitters to regulate the power output around a predetermined set point with variations in AC line voltage or changes in other operating parameters. Most modern FM broadcast transmitters utilize a high gain tetrode as the final amplifier stage with adjustment of the screen voltage providing fine adjustment of the output power.

For each power output level there is one unique combination of screen voltage and output loading that will provide peak operating efficiency. If the screen voltage is raised above this point without a corresponding increase in loading, there will be no further increase in power output with rising screen voltage and screen current. If the screen voltage is raised without sufficient loading, a screen current overload will occur before the upward adjustment in power output is obtained. To avoid this problem, it is a good idea to tune the transmitter with slightly heavier loading than necessary to achieve the desired power output level in order to allow for about 5% headroom in adjustment range. The output loading can be adjusted for a "peak" in output power of 5% over the desired level and then the screen voltage can be reduced enough to return to the desired level. This procedure will allow headroom for an APC system controlling screen voltage and will result in about a 1% compromise in efficiency, but it will assure the ability to increase power output up to 5% without encountering a screen overload.

Centering the Passband

A simple method for centering the transmitter passband on the carrier frequency involves adjustment for minimum synchronous AM. If the bandpass is narrow or skewed, increasing synchronous amplitude modulation of the carrier will result. A typical adjustment procedure is to FM modulate 100% at 1 kHz and fine-adjust the transmitter's grid tuning and output tuning controls for minimum 1 kHz AM modulation as detected by a wideband envelope detector (diode and line probe). 1 kHz is used as the FM modulating frequency rather than 400 Hz so that the audio highpass filter in the audio analyzer can be used to eliminate the AC line frequency related asynchronous component from the synchronous AM component. It is helpful to display the demodulated output from the AM detector on an oscilloscope while making this adjustment. Note that as the minimum point of synchronous AM is reached, the demodulated output from the AM detector will double in frequency to 2 kHz, because the fall-off in output power is symmetrical about the center frequency causing the amplitude variations to go through two complete cycles for every one FM sweep cycle. It should be possible to minimize synchronous AM while maintaining output power and efficiency in a properly designed power amplifier.

Effect of Transmitter Tuning on FM Sidebands

The higher order FM sidebands will be slightly attenuated in amplitude and shifted in time (group delay) as they pass through the final amplifier stage. These alterations in the sideband structure that are introduced by the amplifier passband, result in distortion after FM demodulation at the receiver. The amount of distortion is dependent on the available bandwidth versus the modulation index being transmitted. For a given bandwidth limitation, the distortion can be minimized by centering the passband of the amplifier around the signal being transmitted. This will cause the amplitude and group delay errors to affect both the upper and lower sidebands equally or symmetrically. Tuning an amplifier for minimum plate current or for best efficiency does not necessarily result in a centered passband. One way to center the amplitude passband is to tune the amplifier for minimum synchronous AM modulation while applying FM modulation to the transmitter. Since the circuit topology of most transmitters exhibits a difference in tuning between the symmetrical amplitude response and the symmetrical group delay response, FM modulation performance can be further improved by tuning for symmetrical group delay rather than for minimum synchronous AM. The symmetrical group delay tuning point usually does not coincide exactly with the symmetrical amplitude tuning point and falls between the point of minimum synchronous AM and the point of maximum efficiency.

The transmitter may be tuned for minimum intermodulation distortion in left-only or right-only stereo transmissions. Stereo separation will also vary with tuning. For stations employing a 67 kHz SCA, transmitter tuning becomes very critical to minimizing crosstalk into the SCA. Modulate one channel only on the stereo generator to 100% with a 3.5 kHz tone. This will place the lower second harmonic (L-R) stereo sideband within 2 kHz of the 67 kHz SCA. Activate the SCA at normal injection level without modulation on the SCA. Tune the transmitter for minimum 2 kHz output from the SCA demodulator. This adjustment can also be made by listening to the residual SCA audio while normal stereo programming is being broadcast.

A more sensitive test is to tune for minimum even order harmonic distortion which will result in a symmetrical group delay response and will optimize distortion, separation, and crosstalk.

The latest generation of power amplifiers have been designed to operate without compromising subcarrier performance. By providing broadband matching circuits, adjustment of these transmitters for optimum FM modulation performance (minimum distortion, minimum crosstalk, maximum separation, etc.) is very repeatable and stable. The field adjustment techniques are listed below in ascending order of sensitivity:

1. Tune for minimum synchronous AM noise.
2. Tune for minimum IMD in the left or right channel only.
3. Tune for minimum crosstalk into the unmodulated SCA subcarrier.
4. Tune for minimum even order harmonic distortion. (symmetrical group delay)

In any of these tests, the grid tuning is frequently more critical than the plate tuning. This is because the impedance match into the input capacitance of the grid becomes the bandwidth limiting factor. Even though the amplitude response appears flattened when the grid is heavily driven, the group delay (time) response has a serious effect on the higher order FM sidebands.

Optimum tuning versus efficiency

VHF amplifiers often exhibit a somewhat unusual characteristic when tuning for maximum efficiency. The highest efficiency operating point does not exactly coincide with the lowest plate current because the power output continues to rise on the inductive side of resonance coming out of the dip in plate current. If the amplifier is tuned exactly to resonance, the plate load impedance will be purely resistive and the load line will be linear. As the output circuit is tuned to the inductive side of resonance, the plate load impedance becomes complex and the load line becomes elliptic instead of linear since the plate current and plate voltage are no longer in phase. Apparently best efficiency occurs when the phase of the instantaneous plate voltage slightly leads the plate current. This effect is believed to be caused by the non-linear gain characteristics of the power amplifier tube operating on an elliptic load line.[1]

Summary

FM broadcast signals are significantly affected by asymmetrical group delay response. This effect is most pronounced in SCA and RDS subcarriers. Careful tuning of the transmitter can minimize the distortion, and maximize the performance of these services.

Acknowledgements

The author wishes to thank the Engineering Department of Broadcast Electronics and Quantics, Inc., in particular, Geoff Mendenhall, Mukunda Shrestha, Ed Anthony, and Dave Hershberger, without their assistance, this paper would have been impossible.

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A NEW DESIGN IN MULTI-USER FM ANTENNAS

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ABSTRACT

In this paper a unique new design of FM antenna will be introduced that will give many broadcasters a new option. Each circularly polarized sidemount element is completely balanced. The bandwidth in FM applications exceed 5 Mhz with a VSWR of 1.1:1. Two stations with a frequency separation of up to 10 Mhz can combine their signal into this antenna and each have a VSWR of better than 1.1 to 1 over their channels. This paper will discuss the **JBBP** design and its many advantages for FM, high band VHF and UHF TV applications.

INTRODUCTION

FM stations historically face few options in choosing a broadcast system. Large markets can often support a station to use a multi-user master antenna, ranging from an array of flat panels up to the top mounted Spiral Antenna. The results are typically quite pleasing. This improved performance generally justifies the high cost of the project. Many stations, however, are not left with this option due to reasons such as politics, budgets, tower constraints, etc., thus being forced to use a standard sidemount antenna.

Many FM broadcasters are relegated to the use of sidemount antennas for many reasons such as lower cost and reduced tower constraints versus comparative panel systems. Until now, all sidemount antennas shared some disadvantages, such as a strong interaction with its support tower (distorting tuning and pattern characteristics), and poor bandwidth.

Balanced Feed

The key to the new effectiveness of this antenna is its balanced feed system. Until now, all current sidemount feed systems whether having an internal, external or capacitive feed lacked balance and symmetry. Capturing all the advantages of the Penetrator skewed Vee dipole antenna, baluns were introduced, creating the first completely balanced sidemount FM antenna on the market.

The balanced feed eliminates the electrical asymmetry of the antenna, i.e., all arms radiate equally. Equally important is the elimination of stray currents, which flow through the rest of the surface of the antenna and its feed line. These surface currents radiate in random directions, introducing distortion in its 3 dimensional radiation pattern.

The Balun

The dipole easily lends

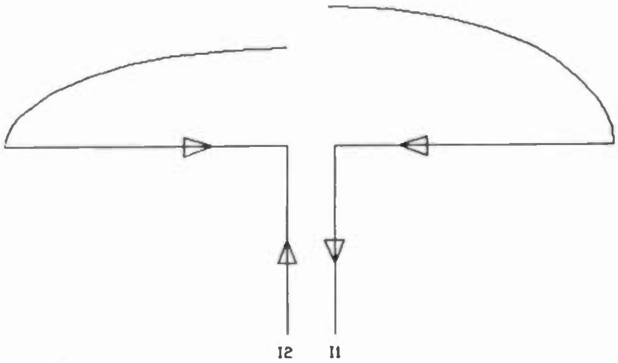

Figure 1

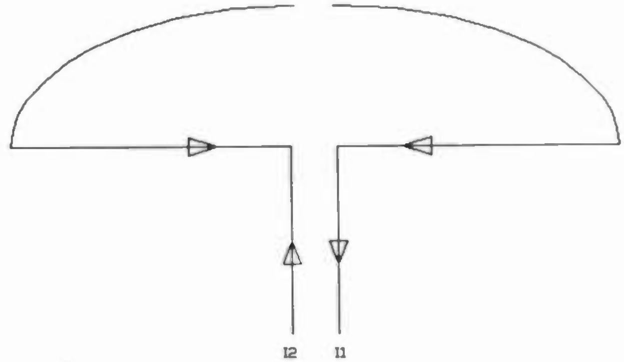

Figure 3

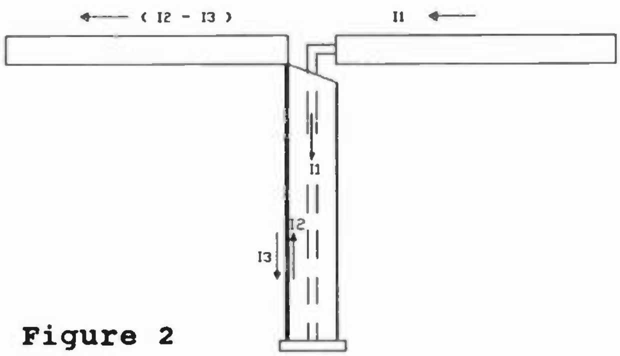

Figure 2

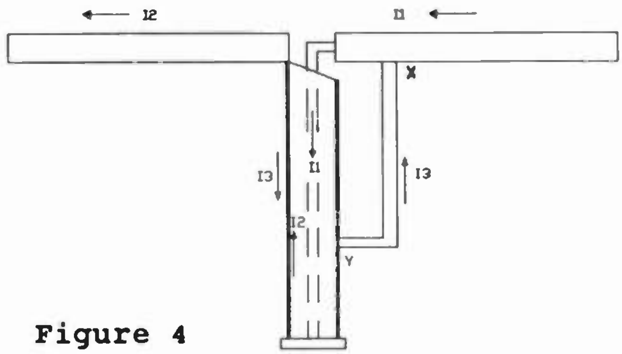

Figure 4

itself as a model to illustrate the basic principles of a balun.[1] In figure 1, a dipole has an unbalanced current flow (I1 > I2). This results in a net current (I1 - I2), required to flow on the transmission lead, causing unwanted radiation of an uncontrolled polarization. It is easy to follow that this results in distortion of the antenna's pattern.

A practical example of this case is when a dipole is fed directly by a coaxial line. Coaxial transmission line is in a balanced mode when the current on the outer surface of the inner conductor induces an equal magnitude current in the opposite direction on the inside surface of the outer conductor. This condition is upset when the coax line is directly connected to the symmetrical dipole antenna. As shown in figure 2, the inner is shorted to one (hot) arm, while the outer is shorted the other (cold) arm of the dipole. This connection of the outer to the cold arm induces a current on the outer surface of the coax feed as well as a current on the cold arm of the dipole. Since I1 must equal I2 in a coaxial line, the current on the cold arm is equal to I2 - I3. The resultant condition of unbalanced currents is much like the condition of all currently available designs of sidemount antennas. The current I3 causes the boom, transmission line and even the support structure to radiate the unbalanced current in a non controlled polarization.

Figure 4 illustrates one type of balun. This configuration has a second line which connects the hot arm to the outer of the coaxial line. In this case, the length of this section is a quarter wavelength. This apparent short at the connection to the coax is transformed (rotated 180 degrees on the Smith chart) to have an infinite impedance at the point of connection to the antenna arm. Thus, the antennas input impedance is not effected by the balun. Likewise, the high impedance to ground at point x, is transformed to a very low impedance at y. Thus as I3 flows down the outer, a current equal in magnitude flows in the opposite direction up the balun. The outer of the coax now simulates a balanced twin lead transmission line, and thus, there is no net current below the point y.

THE DESIGN

The physical configuration of the balanced antenna is illustrated in figure 5. The antenna is fed with a 50 ohm boom which connects to the baluns of the antenna. These baluns assure uniform radiation of all four arms. Further, these arms are the only source of radiation, since no exterior surface currents on the boom or feed line of the antenna exists.

The elements can be sidemounted on a pole or tower by similar methods used with other standard sidemount antennas. Depending on the application, the feed system will either be a rigid shunt line or a branch feed system. Branch systems consist of a power divider which directly feeds each element through semi-flexible cables. The advantage of the branch feed is that the bandwidth of the feed system can cover the FM band, and that the phase of the feed to each element is independent of frequency.

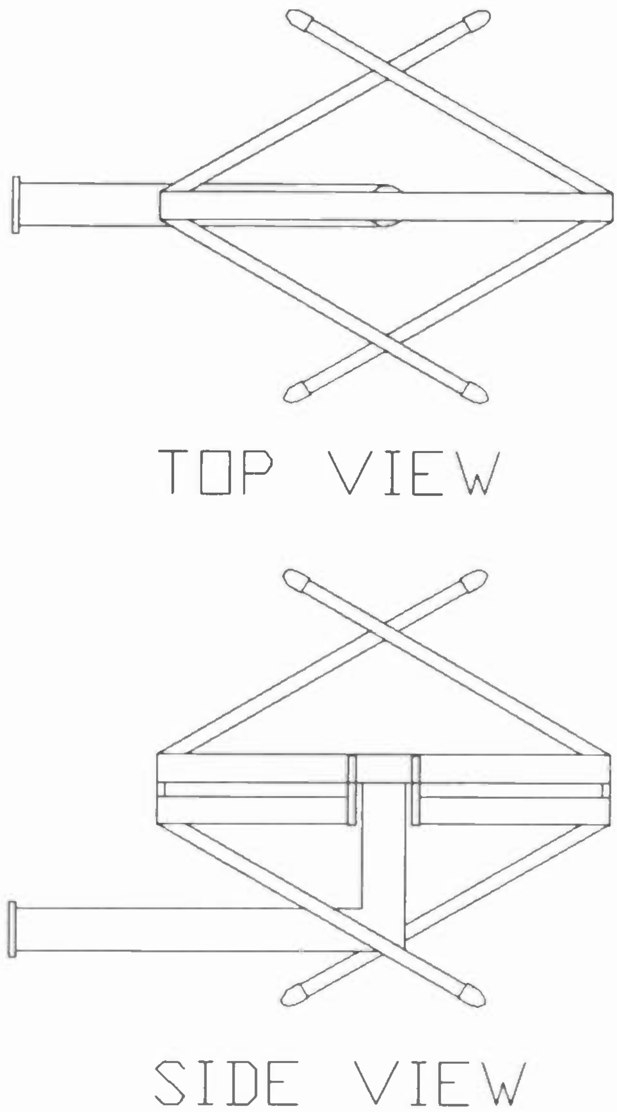

Figure 5

Pattern Performance

The new element produces patterns based on the physical configuration of its arms. Positioned as skewed Vee dipoles, the intention was to capture the advantages of the JSCP design. Figures 6 and 7 are free space measurements of the element's azimuth pattern. The patterns shown were measured in full scale at 96 Mhz. Similar patterns were recorded throughout the FM band, each with a circularity of +/- 1.5 dB or better. Equally

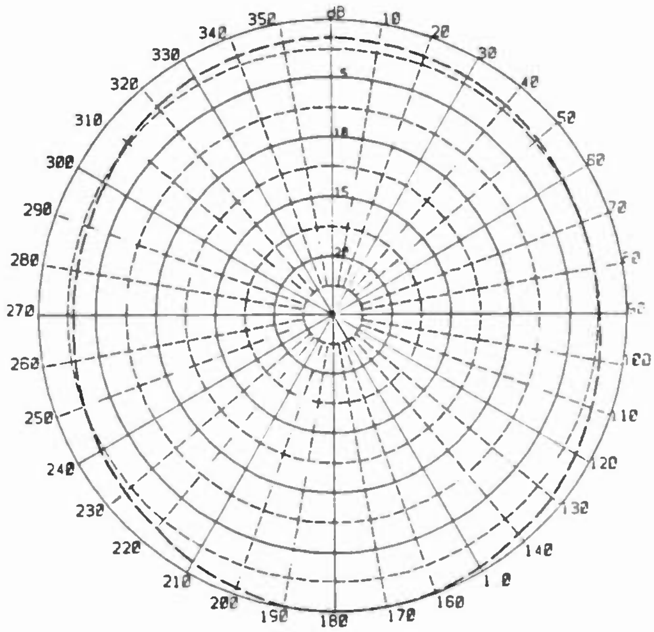

Figure 6 HPOL Freespace

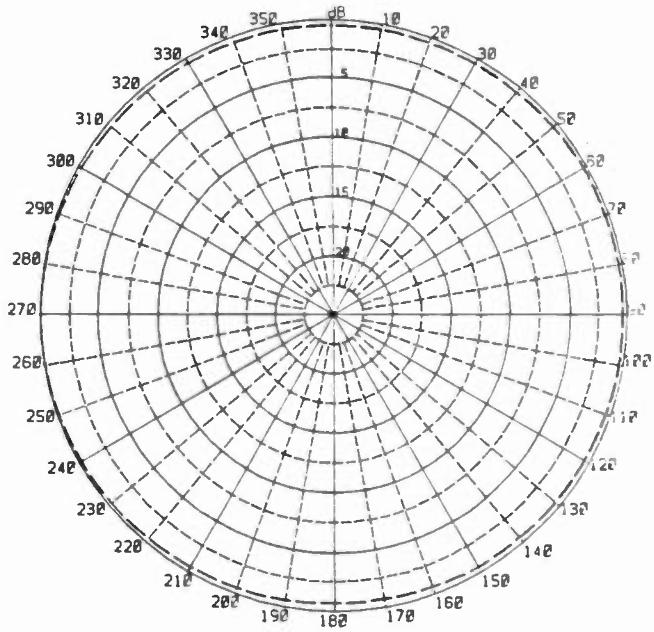

Figure 7 VPOL Freespace

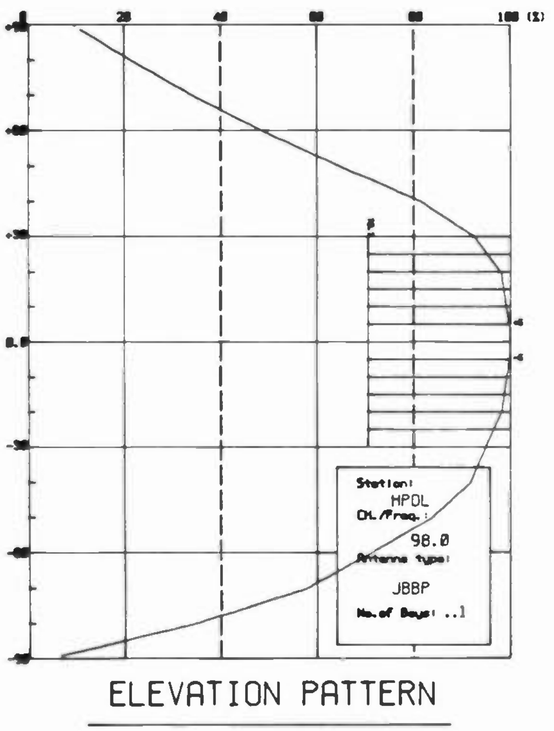

Figure 9 VPOL Elevation

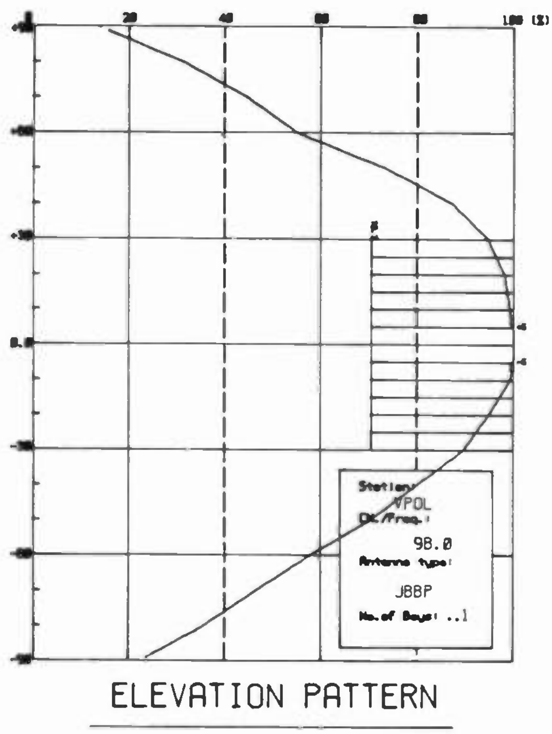

Figure 8 HPOL Elevation

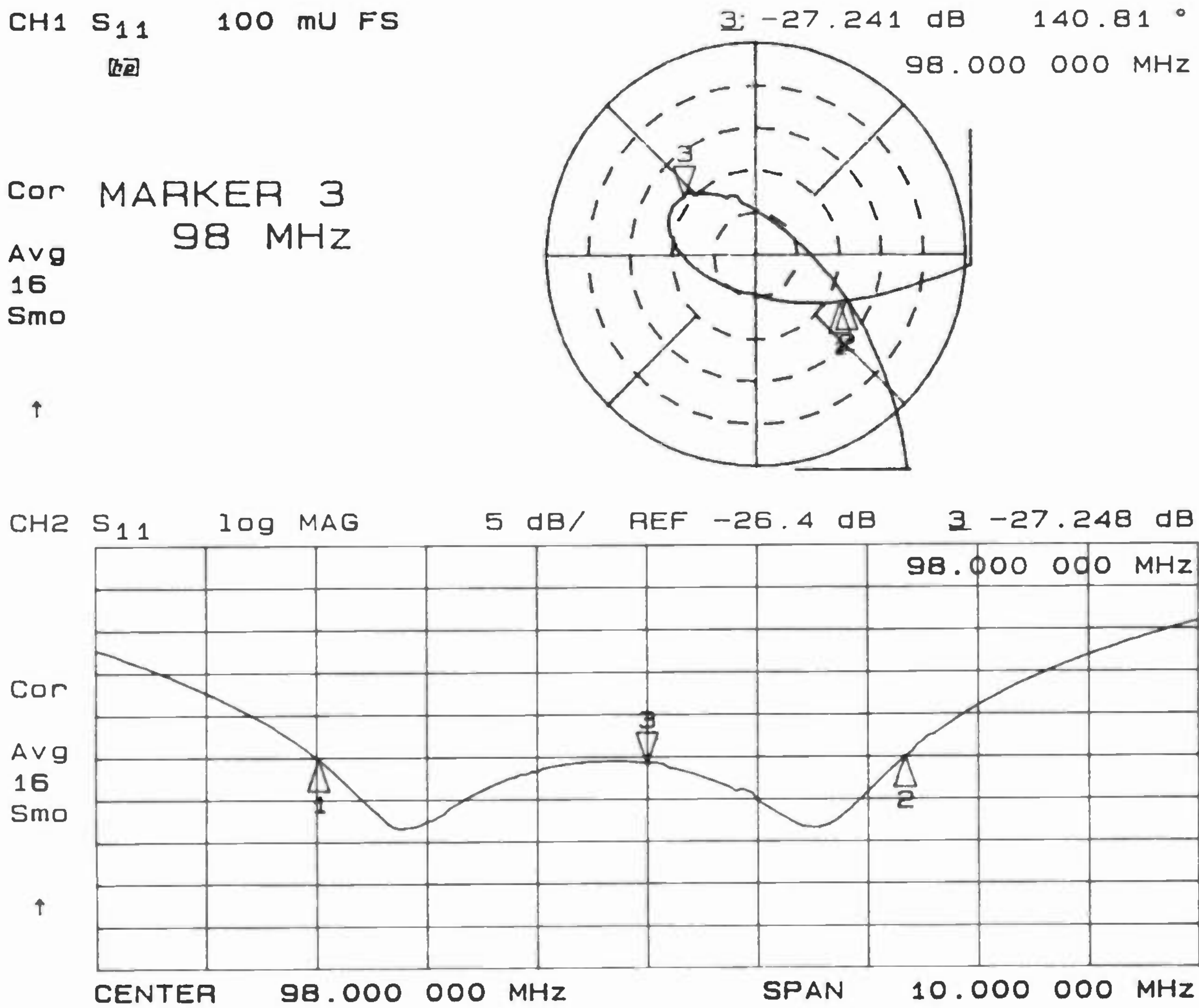

Figure 10

impressive were the measured elevation patterns, a sample of which are shown in figures 8 and 9. The anticipated patterns illustrate the lack of distortion with which standard sidemounts display.

It is acknowledged that a support structure will distort the antenna's azimuth pattern. This distortion, though, is not as drastic as exhibited by standard sidemount antennas, due to the elimination of boom currents.

Antenna Bandwidth

The bandwidth of this antenna is many times greater than currently available antennas. Figure 10 shows the tuning characteristics of a single element. The Smith chart has been expanded for purposes of clarity. As easily seen on the accompanying log magnitude chart, the bandwidth where the antenna element has a VSWR of 1.1:1 exceeds 5 Mhz. Through simple adjustments, the antenna's impedance can be shifted to the desired range.

Applications

Aside from single FM channel

use, the most obvious application of this new antenna is when 2 or more stations within a 5 Mhz spread intend to share a radiating system. In lieu of requiring a series of panels around a tower, the cost of a broadband sidemount offers an alternative that many stations can afford. Further, since the windload of a sidemont antenna is only a fraction of a comparable panel system, the tower size requirements and installation costs are also greatly reduced.

The 5 Mhz spread is not a limiting factor. By adjusting the impedance of the antenna, the loop on the Smith chart increases in size. With proper adjustments, two stations 10 Mhz apart can each have a VSWR of 1.1:1 over their channels. The use of branch feeding in this application assures each channel optimum performance.

The bandwidth of this antenna makes it an excellent choice for TV broadcasting. For upper band VHF, the antenna is able to maintain a VSWR of 1.1:1 over the entire channel. An omni-directional circularly polarized sidemount VHF antenna offers a unique option to many TV stations. Its low price and small space requirements should be welcome news to low power VHF stations. The same advantages should make it a popular choice for auxiliary use for full powered stations.

The bandwidth of this design also makes it applicable for use in the UHF band. Two or more UHF stations within a 5 channel spread can share the advantages of this antenna. More than 30 sidemount elements can be stacked vertically to achieve the high gain that UHF broadcasters require.

The future of radio may be headed for major changes. Recent developments have shown that digital broadcasting can greatly increase the quality of signal, as well as greatly reduce the effect of multi-path on signal propagation.[2] The digital systems proposed require a bandwidth on the order of 4.5 Mhz.[3] More than a dozen stations can be multiplexed in that band. With power requirements slashed by 20 to 30 dB, there will be concentrated effort to combine all the stations into a single broadcasting facility. If the current FM spectrum is used, this new design is up to the stringent design requirements.

Summary

With its new feed design, this new element has opened up a variety of new options. Its bandwidth permits multi-channel use in the FM and UHF bands. Further, high band VHF channels also can take advantage of the design improvement. The improved radiating characteristics have set a new standard for sidemount antennas. As a result, many broadcasters may choose this design for single channel applications.

Acknowledgements

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COMPUTER APPLICATIONS FOR BROADCAST ENGINEERS

Wednesday, April 17, 1991

Moderator:
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COMPUTER APPLICATIONS FOR BROADCAST ENGINEERS
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COMPUTER DOCUMENTATION: AN ENGINEER'S FRIEND OR FOE
Dr. Walter P. Black
Video Design Pro
Las Cruces, New Mexico

COMPUTER APPLICATIONS FOR BROADCAST ENGINEERS

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Brookfield, Connecticut

Abstract- The evolution of the personal computer over the past several years has brought forth an invaluable tool for the broadcast engineer. Many tasks from office management to tedious mathematical calculations can be relegated to a computer saving both time and money. The focus of this presentation will be on the software available to enhance the business of today's broadcast engineer.

INTRODUCTION

Many broadcast engineers are employed or retained by more than one station, or have the responsibility for supervising several employees. The requirements for AM/FM/TV facilities vary considerably from one to the other. The computer can be a very versatile tool, more specifically a valuable investment as an instrument for enhancing operations and insuring FCC rule compliance. There are several categories of software that are generally available on the open market:

Word Processor

Spreadsheet

Database

Calendar/Appointment

CAD/Drawing

Languages (BASIC, C, PASCAL, etc)

Software specific to the broadcast engineering community can be categorized as follows:

AM/FM/TV frequency allocation and channel search programs.

AM DA and RF network design, analysis and adjustment programs.

Terrain analysis programs.

Specific task oriented programs i.e. STL system design, FM/TV antenna pattern, RFR radiation, calculations etc.

SOFTWARE USAGE

Word processing programs can perform a variety of tasks. If you are a contract or consulting engineer, you represent yourself as the President of your own company. The paperwork that you generate in the form of letter correspondence, inspection reports, and invoices reflects the professionalism of your company. Most word processors feature spell checkers and thesaurus while add-on programs are available to enhance grammar and syntax. FCC related documents such as Equipment Performance Measurements, AM directional proof-of-performance tabulations, and frequency measurements can easily be prepared using a word processor.

Spreadsheet programs are sometimes thought to be used only by large corporations. A broadcast engineer who has several clients or a corporate D.E. can make use of a spreadsheet to keep track of invoices/collections, salary levels, utility expenses, taxes, etc. A spreadsheet can also be used as an AM DA proof analysis tool or to calculate harmonic radiation for the annual EPM's.

Databases can be used to store client information, parts and equipment inventory, and some resourceful engineers have compiled record/CD databases for their program directors. A database of spare parts can be useful to the contract engineer who maintains several facilities so that in the event failure, an immediate determination can be made if the necessary part is within local reach. A database can also be used to keep track of equipment repairs so that a handy record is immediately available for every equipment serviced. This is an invaluable time saver and provides an instant failure trend analysis. Another use of a database is for wire routing and locating. When you want to know what wire #240 is being used for and its destination, just consult your database!

Keeping track of appointments and routine events such as quarterly tower inspections, EPM's, frequency checks, tube changes, tower lamp changes, air filter changes, etc can easily be relegated to a computer. Trying to keep a mental note on each of these and other routine tasks for a number of facilities can be overwhelming. Routine maintenance can be enhanced and a greater degree of security against FCC violations is attained if the required measurements and inspections are performed on schedule.

Computer Aided Design or CAD programs have become an everyday part of our lives. From drawing relatively simple schematic diagrams, flow charts and PC board layouts to precise engineering drawings, CAD has proven to be a most valuable tool. Production personnel have found many uses for CAD type programs to design video fonts and logos. You can also use most CAD programs to draw up operating logs with neatly aligned gridwork for ease of operator use.

Up until a few years ago the most popular languages among computer users were BASIC (which is usually supplied with each computer along with DOS), FORTRAN which was a favorite for engineers who programmed on mainframes and COBOL for business applications. The evolution of computer languages has been astounding in the past five years. BASIC has evolved from its infancy in the interpreted mode to structured form in the compiler environment. C and PASCAL are now the favorites among engineering and scientific communities. I will devote some time to computer languages since many engineers prefer to "homebrew" programs specific to their needs.

BASIC BASICS

It has been said that 90% of people who learned BASIC outside of the classroom learned it wrong. This is no doubt a true statement. Let us consider the following short BASIC program:

```
10 DEFDBL A-Z
20 pi = 3.141592653589793#
20 FOR x = 1 to 10 : IF x = 5 THEN GOTO
   30 : PRINT x * pi : NEXT
30 PRINT "X Has Reached Five"
```

Some early BASIC interpreters would find no syntax problems with this program and allow it to execute. Since the FOR..NEXT loop would bail out at the value of five, 6,7,8,etc times pi would never be printed since the loop was improperly exited. This ungraceful exit could lead to improper calculations if an array was being formulated and stack problems would result as well. X would be treated as double precision because of the line 10 declaration, but X really should be an integer. Speed and memory usage will suffer. Under the interpreter rules, it was a speed advantage to cluster lines together and leave off the X after NEXT. These practices lead to "spaghetti code" which is almost impossible to decipher at a later date.

Microsoft QuickBasic and Borland Turbo Basic (now Spectra Publishing PowerBASIC) led the way into a lightning fast compiled environment. The interpreter translates each line of code immediately prior to executing. The compiler first translates the entire program resulting in much faster execution. Almost all language compilers today take advantage of a math coprocessor chip, if installed and provide excellent editing and program debugging features as well. Although modern BASIC compilers allow recursion (calling a routine from within itself) and other advanced features, a program developer must choose the language best suited for the particular program requirements.

The C and PASCAL languages are more "structured" than their BASIC counterpart. In C for example, every variable must be declared by type (integer, precision, etc) and usage (local or global). FORTRAN assigns the I through N variables as integers. The language evolution trend is towards object oriented programming. That is, creating various modules that contain code performing specific tasks. these modules can then be shared between many programs. There is no need to rewrite the same code many times for use in different programs.

SPECIALIZED PROGRAMS

Many engineers do not fancy themselves as computer programmers. They see the computer as a tool of the trade and rely on professional programmers to supply software for their needs. Consulting engineers and corporate D.E.'s frequently make use of channel search and allocation software to determine if a station can be built as new, increase power or relocate to a new frequency. Computer programs can search all available channels and query the FCC allocation tables, resulting in a tabulation of possibilities if any were found.

The design of AM directional antennas was perhaps one of the first tasks assigned to a computer. The mathematics involved are intense (hemispherical RMS determination using FCC 73.150 formulae, loss factor, standard pattern calculation and augmentation factors where appropriate) and repetitive for each iteration tried by the computer. The differences in methodology employed in the various commercial programs provide a variety of output results. Entire phasing systems can be computer modeled in order to determine what shape the radiation pattern takes at each sideband, how the feeder line VSWR's look at the sidebands and finally the curve of the common point impedance. The computer can be used to design or redesign a phasing system to obtain the best pattern and impedance bandwidth possible. AM directionals can be initially set up using computer software as well.

Analysis of AM proofs is another task best handled by a computer. The calculation of arithmetic and log ratios, the elimination of certain tainted monitoring points from the average and production of a neatly formatted output is a relatively simple task for most computers. Computers now determine conductivity zones and Inverse Distance Fields for AM original proofs. Before computers, it was necessary to overlay conductivity charts and manually "curve fit" each segment to the appropriate line. This left a wide disparity between the engineer's determination and that of the FCC at times. This task has been computerized, replacing guesswork with speed and precision.

Analyzing a directional antenna system for its stability characteristics is a repetitive task which tests for radiation values exceeding the FCC Standard Pattern while varying the field ratio and phase angle of the various elements of the array. For each permutation, the new fields are compared against the original standard pattern. Performing this function could take days if done manually while a computer program is more suited for the job.

The FCC now uses a computer routine to calculate the FM and TV F50,50 and F50,10 propagation curves. Manual interpolation is no longer necessary. Source code in the FORTRAN language for all FCC programs, including the RADIAT AM directional program, is available to the public through its contractor and other sources.

In the past several years, the inclusion of terrain effects on FM and TV signal coverage determination has become very popular. The cellular telephone industry relies heavily on terrain effects in order to provide adequate coverage to the "cells" in their systems. Too great of a coverage area in their case can be a disadvantage. Most engineers who have designed STL systems are familiar with the manual procedure of "clearing a path" for their systems using a topographic map and manually determining Fresnel zone clearance and obstruction locations. Doing this at one degree increments for 360 degrees can be tedious to say the least. By making use of terrain analysis software, one can choose the best possible transmitter site by accounting for multipath generating obstructions and adequate clearance to all pertinent markets. Terrain calculations allow more accurate coverage maps to be generated for a given FM or TV station allowing management to focus sales efforts into the areas of best signal penetration.

PUBLIC DOMAIN PROGRAMS

Many programs that perform relatively simple tasks are in the "public domain". This means the programs are freely distributed and widely available. All commercial programs are copyrighted, you cannot legally copy the programs and give them or sell illegal copies to any other person. Some commercial programs require multiple-user licensing. Public domain programs, on the other hand, may be copied and given to anyone provided there is no charge involved. Bulletin boards are an excellent source of public domain programs. If you write programs yourself, you may want to place them in the public domain. This will encourage others to do likewise and stimulate interest in computer programming while providing benefit to the broadcast engineering community as a whole. A sampling of the public domain can find programs calculating intermodulation products, transmission line efficiencies and power ratings, RF network design and analysis, RFR levels for FM stations, metric conversions, Bessel nulls for FM mod monitor calibration and more.

Some bulletin boards host demo or shareware programs. These are partially functional programs designed to familiarize the user with the product and to give an opportunity to exercise its functions, usually within a preset limited range. A fee is often required in order to obtain the fully functional program and to become a registered user who is eligible for upgrades and technical support.

ON LINE SERVICES

If you own a computer or are contemplating the purchase of one, you can take advantage of on-line services offered by a number of companies. A modem can be internally or externally connected to your computer and allow you to access via landline bulletin boards or on-line services which allow you to run specialized programs for a fee. On-line programs cannot be downloaded as their bulletin board counterparts. The program can be run using the parameters you specify with the program output results sent to a disk file on your machine. Most on-line services offer AM/FM/TV allocation, channel and unused call-letter searches, stations within and other services that may only require occasional use. This is an ideal solution for a contract or consulting engineer for whom the outright purchase of this type of software may not return profit in a reasonable period of time. Databases of this nature require constant updating whereas calculation programs are more consistent in nature.

HARDWARE

It is difficult to address the question of which machine, printer, drive type, modem, monitor, etc to buy or upgrade to. A computer system is an investment. Its price range can be comparable to those of oscilloscopes to spectrum analyzers. For serious users who make frequent use of computers for calculations, terrain extraction, etc the 386 class of computer should be considered a minimum. A number of computer trade journals feature ads from well established manufacturers to do-it-yourselfers. Rock bottom prices on name brands can be found through computer magazines. If you want local support you can visit a nearby computer retailer who will generally charge a few percent more (and local tax) for this service. A 2400 baud modem and 40 MB hard drive are recommended. The choice of dot matrix vs laser printer generally favors the latter.

CONCLUSION

Computers should not be an intimidation. They are both a tool and an investment through which you can increase your productivity and profit. The network of computer users and experts is wide, and the availability of on-line help through networks and local computer clubs is vast. Just as the pocket calculator became an everyday part of our lives in the 70's, computers now surround us in our work environment from the sales, accounting, music and traffic office to the on-air program automation and remote facilities monitoring system.

COMPUTER DOCUMENTATION: AN ENGINEER'S FRIEND OR FOE

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Abstract - A computerized cable documentation system can reduce documentation time by 50-90%. Does this mean ...

- the engineer is no longer needed?
- low level computer technicians can do engineering?

This need not be true because computers do not have the intelligence to replace engineering knowledge and abilities, unless engineers refuse to implement a comprehensive documentation design and maintenance program.

Answers to these problems will be examined in light of current computer hardware and software technology to show that the computer can indeed be the friend of the engineer—even if s/he is new to computers.

INTRODUCTION

This paper is a compilation of actual situations in the broadcast industry, compiled during seven years of consulting with engineers regarding computerized documentation. Only the names have been changed to protect the guilty.

Hypothesis: Computerized documentation systems can be an engineer's friend if s/he is friendly back, a formidable foe if feared, and a hard task master if not integrated into the work environment.

DEFINITION OF TERMS

Computer: An electronic machine that solves complex mathematical (engineering) problems in a very short time when given the right information.

Documentation: A collection of engineering facts, e.g., rack and diagram drawings, cable information, cable labels, cable schedules, runlists, jacks designation strips, interconnect drawings, equipment and cable bill-of-materials (see Figure 1).

Friend: A person who favors and supports another.

Ally: A person or group united for a specific purpose.

Task Master: A very exacting boss.

Foe: Anyone or anything that harms or is likely to injure.

OBSERVATIONS

Case Study 1

Facts DL was chief engineer in a medium market TV station. He knew he had a major rebuild project in two years, so he purchased a computer documentation system and attended a training class. He figured that it would take him six months to learn the system and he wanted to be ready for the big project in plenty of time.

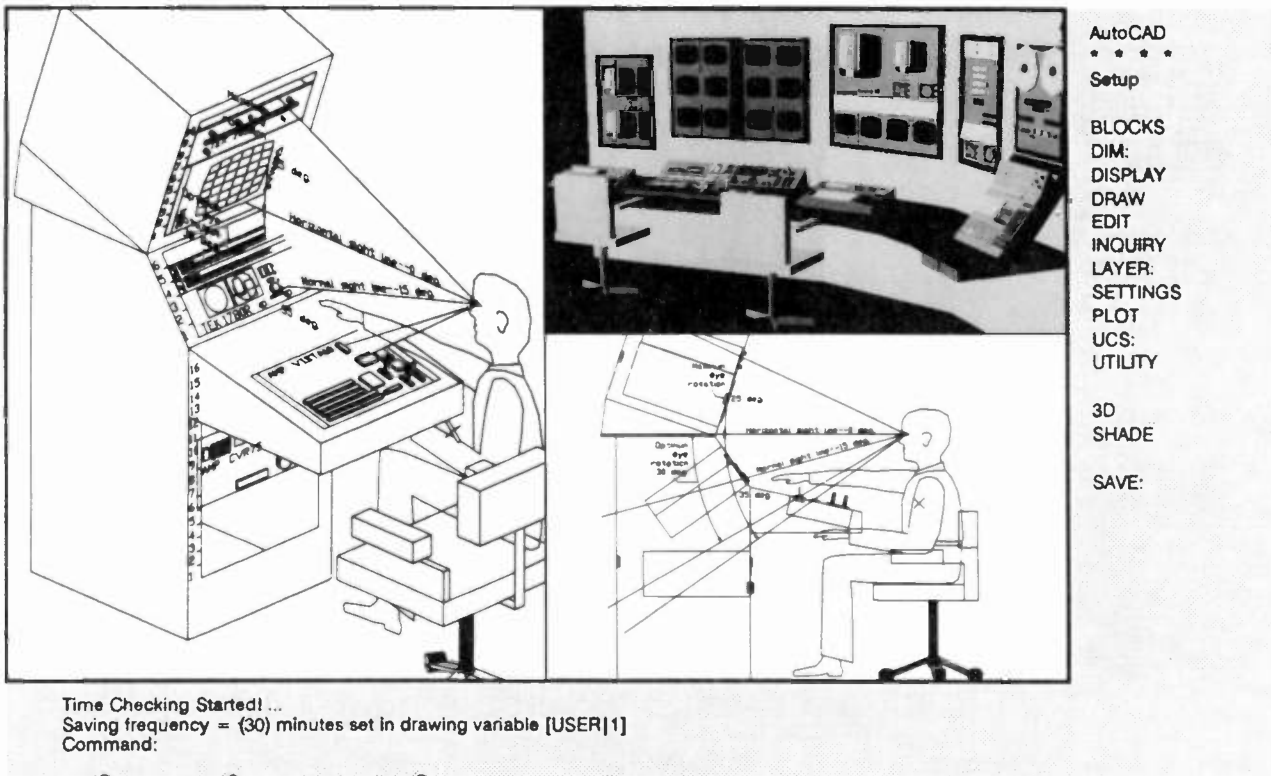
He first tried the system by rebuilding a small edit room, and found he finished the project in less than two months—with 50% less time and mistakes than he considered normal. With this success, he knew enough to tackle several larger projects. He has continued to upgrade his computer and software, each time with productivity gains. Now in the midst of the major project, he finds it going on-time and on-budget.

Conclusion Computerized documentation clearly was the friend of DL, because it supported him. But DL supported and liked the computer, received proper training on a regular basis and kept upgrading his software to better achieve his aim. The result: 50% or better gain in productivity.

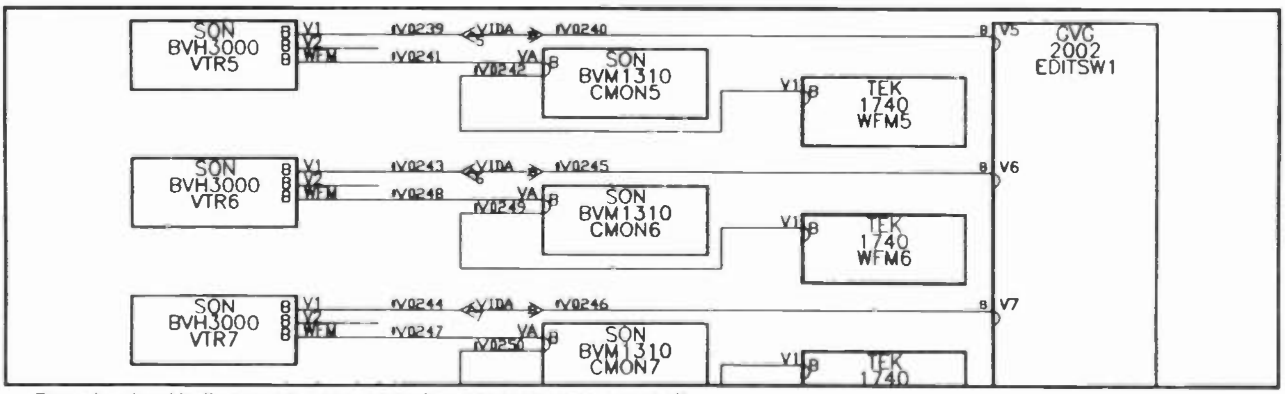
Case Study 2

Facts HJ said he did not need computerized documentation—his station was well documented in his head (he was convinced this was job security). He did have a cable log book, but it was never up-to-date. Computer documentation, he stated, was unnecessarily complex for his station's size (#1 station in a top 50 market). Besides, he could not spend time documenting with a computer because the projects had to be installed immediately. When the station was off the air for hours due to a key component which burned out and had to be worked around, and HJ was not able to remember what cables went where, the station manager found HJ was not irreplaceable. The new CE immediately purchased a computerized documentation system. HJ is on unemployment.

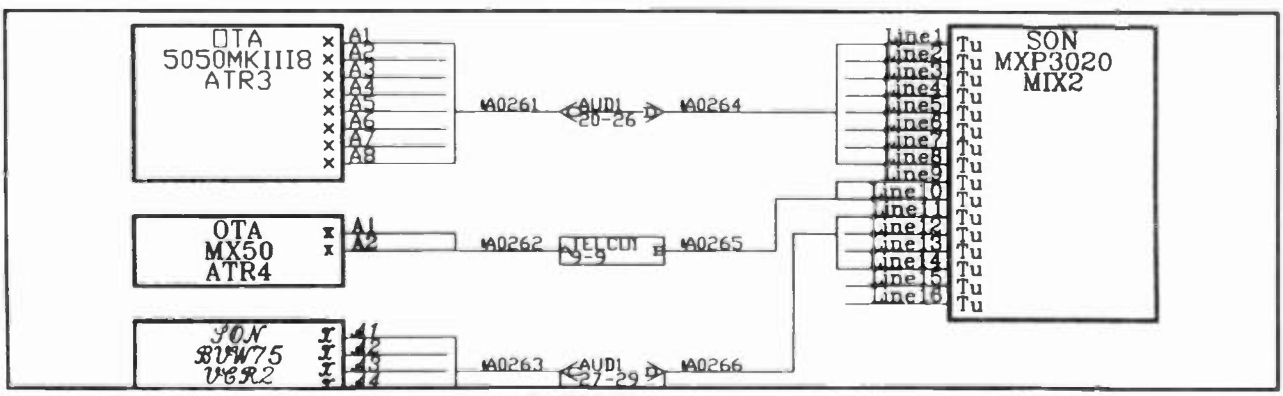
Figure 1. Samples of Rack, Ergonomic and Diagram Documentation Created With VidCAD/AudCAD 386



VidCAD© and AudCAD© drawings using EZ3D© and rendered in AutoShade.™ Rendering within AutoCAD™ available in R11.



Clutter is reduced in diagrams when unnecessary inputs and outputs are turned off.



This diagram displays some of the typestyles available through FlexiBLOCK.

Conclusion Computerized documentation was HJ's foe. He assumed it was the enemy and refused to associate with it. While believing mental documentation was adequate and led to job security, his ability to remember documentation proved faulty under fire. HJ is like countless other engineers, remembering when life was simple and decrying the computer age. But this need not happen. Advances in computerized documentation make it easy to become a documentation expert, even with no computer experience.

Case Study 3

Facts JR was hired to build a new post-production facility. During the first two weeks JR had to purchase millions of dollars worth of equipment. Convinced that documentation was essential, she also ordered a computer documentation system. By the time the equipment began to arrive, she had only made a few test drawings. When the boss said, "Get it on the air," JR gave up on documentation—figured she would do as-builts when the job was finished. A year later, JR is still trying to get the bugs worked out of the facility and has not had time to do the computer documentation. She realizes now that more time spent in design would have shortened the final job completion and resulted in fewer lost billing hours spent fixing problems.

Conclusion JR viewed computerized documentation as an ally—someone to help with a specific task. But she failed to realize how long it takes to build an alliance—almost as much time as a friendship. Consequently, in the heat of battle, the alliance faltered, this immediate battle was lost and the result of the war is in question. If only a little more pre-planning and training time had been allocated, much time and money could have been spared.

This "build-it-quick" construction is often called "Fast-Tracking," and is commonly used in major building projects. This may work well in construction, but when creating a video or audio work space, it leaves a lot to be desired, including:

- Inadequate drawings or written documentation for installers, resulting in excessive and inaccurate verbal instructions.
- Too much time spent in cable labeling or finding correct cables.
- Pulling too many or too few cables between racks or rooms.
- Poor space utilization—too much or too little space assigned for task areas or equipment in racks. (One of our customers paid for his computerized documentation system in one week by saving over 600 square feet of space improperly assigned to master control.)
- Non-existent ergonomic work space planning which can lead to operator fatigue, injuries, Workman's Compensation, lawsuits and untimely remodeling of work space.
- Little growth potential for new projects or new technology without major rebuilds.

Case Study 4

Facts Station X had a large technical staff. Design engineers designed systems, installers installed, maintenance assigned cable numbers and operations assigned jack rows. Design engineers were given computerized documentation systems, but could not assign cable numbers, which the computer system did automatically, because that was someone else's job. Cable numbers in the cable log book were not integrated into drawings ... that was maintenance's job. Since maintenance kept cable documentation on a mainframe, it was not easy to integrate into drawings except by hand. In essence, a computerized documentation system was a fast Etch-A-Sketch® for preliminary drawings.

Conclusion Computerized documentation was like a frivolous pal to the design engineer, which is fine for "fun" times but not there when real work is to be performed. For the company, computerized documentation became a cruel task master, actually taking more time for complete documentation than before. Why? Essentially, computerized documentation was "tacked on" to an existing pre-computerized scope of work. **When new tools are introduced, the work assignments need to be modified to fit the tools.** We would never think of using our old film editor to edit video tape, yet we often fail to analyze how to implement computerized documentation. In this situation, splitting up the design from cable numbering resulted in taking a design tool that could do both tasks in 30 seconds to a procedure that took five or more minutes per cable. The resulting overload in work caused the company to question their purchase of computerized systems rather than perform a systems analysis of the work at hand and making appropriate organizational changes.

Case Study 5

Facts SB was assigned the task of computerized documentation at the TV station and went to CAD school. After months of work, little had been accomplished. There were countless "hardware problems" (hard disks accidentally reformatted, files mixed up, configuration files lost), resulting in no output. Upon closer examination, it was determined that SB was assigned that job because management did not know what other work assignments to give him. Union jurisdiction precluded disciplinary action. The computer is currently used for word processing and SB has been assigned elsewhere.

Conclusion SB found computerized documentation to be an enemy, but not so cruelly as HJ did in Case Study 2. SB's boss made a poor choice by putting someone into computers who a) knew and cared nothing about them and b) had nothing better to do. Either problem will usually doom computerized documentation in that organization; both guarantee failure.

In selecting an employee to take responsibility for computerized documentation, use it as a reward for someone who wants to make computers a part of his/her career. On the other hand, do not give it to a "Techno-nut" who always wants to dink with programming or hardware—this person will also never get anything accomplished.

Case Study 6

Facts JB was VP of Engineering at a large TV station. Unlike most stations today, JB was able to hire a draftsman, DM, to begin computerized documentation. DM had technical training in AutoCAD® at the community college and loved to write programs in AutoLISP®. DM was always talking about how his drawings met ANSI standard X and Y. To help him begin, JB bought DM a powerful CAD system, a prepackaged documentation system and training on how to use it.

After three months, few drawings were completed. When JB asked about this, DM replied that the menu structure in the other program was not as automated as he preferred and he had spent a little time (one month) fixing it. The libraries were also not up to "ANSI" standard and he had carefully redrawn more than half of the libraries he needed, and would be finished by the end of the month.

After two more months with little accomplished, DM explained that he had rewritten the cable documentation database and AutoLISP routines to work better with the new standard drawings. Of course, they were no longer interactive between this database and drawing. It took half an hour per drawing to extract the data and put it into the database, but the new database was "user friendly" because it did the kind of reports he liked. After several more months, the database and libraries were working, although no one but DM could ever figure out how to get a report from the database. The drawings, which slowly began to emerge, were beautiful. However, on new projects, JB or another engineer had to draw the first sketch of every drawing on paper because DM did not know how to design systems. After the drawings were put on the computer, the engineers then had to proof several versions of prints for accuracy.

Conclusion The computer and AutoCAD were friends of DM, but he considered someone else's programs and libraries inferior to his. By the way, there are no true ANSI standards for video and audio diagrams, it was just the way his instructor had taught him.

For JB, computerized documentation became a hard task master, requiring him to spend a lot of time doing the designs on paper for DM. This is a common problem for companies with draftsmen. **Computerized documentation is not a drafting tool but rather is an expert system for engineers to speed their designs.** You must understand design before you can use documentation systems. Because draftsmen are interested in their particular (or peculiar) style and do not feel comfortable with engineering, they often spend considerable effort "customizing" or "reprogramming" a packaged system. In these cases, it is better to give the draftsman AutoCAD or some other CAD program rather than purchasing a computerized documentation system. Then, when the draftsman has learned enough about design to be independent and has completed months of library, menu and programming customization, he can save you a lot of your time—except by then there will be a new version of the CAD or database program which will not run with his customized programs unless they are rewritten.

Lest you get the impression that I am anti-draftsmen, I must assure you, I am not. If fact, many engineers do the same type of customizing. Also, draftsmen have a place in mechanical or architectural drawings. It would be inappropriate for an engineer to spend his/her time dimensioning a console. My point is this: low level computer technicians cannot replace knowledgeable design engineers—unless said designers ignore and oppose their potential, good friend, computerized documentation.

Summary

Computerized documentation can be beneficial and become a true friend in solving the key design and documentation problems—lack of time. However, to be a true friend, these principles must be followed:

- An engineer needs proper engineering and documentation tools so s/he does not become bogged down in programming and drafting.
- This engineer must believe that computers will make him/her more productive.
- Training, regular updates and retraining will help create and maintain productivity.
- Pilot projects should be done to develop proficiency before starting a major project. When purchasing a system, allow three to six months before beginning that major project.
- Assign draftsmen to mechanical and architectural designs until they are proficient at design engineering. It is often more practical to contract this type of service.
- Do not assume that documentation in your head will be useable when needed or that this enhances job security.
- Study work patterns before implementing computerized documentation. Change them now rather than jeopardizing the benefits of computerization.

DIGITAL AUDIO BROADCASTING—METHODS AND SYSTEMS

Wednesday, April 17, 1991

Moderator:
Donald Wilkinson, Fisher Broadcasting, Seattle, Washington

*Paper not available at the time of publication.

MODULATION AND CODING FOR DAB USING MULTI-FREQUENCY MODULATION

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Abstract - Advances in digital audio compression have demonstrated that CD quality audio music can be achieved at compression ratios of six to one. Multi-frequency modulation, with trellis coding and interleaving, provides an extremely bandwidth efficient, multi-path resistant transmission method for unequalized mobile reception of DAB. A design suitable for low power VHF terrestrial broadcast of a 256 kbps stereo pair and six 16 kbps sub-carriers in a 200 kHz channel is described. It is concluded that broadcast of DAB using tellis-coded and interleaved MFM in the first adjacent channels to existing FM transmissions is an attractive option for introduction of DAB in the USA.

DIGITAL AUDIO BROADCAST

Approach

The objective of DAB is to provide compact-disk (CD) quality voice and music to the community, both fixed and mobile, now served exclusively by frequency modulation. Two approaches are technologically feasible, one via satellite and one using the existing FM band. This paper addresses the latter possibility, which envisions that DAB would be provided as an additional service by existing FM stations using new antennas on existing towers. In order to avoid interference with currently licensed FM transmissions, DAB would be assigned to the first adjacent frequency bands, which are presently unused, and broadcast at a low enough power level so that no degradation of FM broadcasting would be subjectively noticeable.

Problems and Solution Methodology A variety of technical problems must be overcome before DAB in the FM band (88-108 MHz) becomes a reality. Here we outline the major considerations and our approach to their solution. A block diagram of the system being designed to provide the desired service is discussed in Section II.

Bandwidth Compression: The first difficulty is to fit CD-quality music into the FM broadcast bandwidth of 200 kHz. An essential part of the solution is a robust music compression scheme which compresses digital sound for a full stereo pair down from 1.536 Mbps to something under 400 kbps at the transmitter and reconstitutes it at the receiver with no loss of fidelity apparent to the listener. Several algorithims are currently capable of meeting this requirement. The MUSICAM algorithim [1], for example, produces very high quality compressed music at 128 kbps per monaural channel and is capable of being implemented in a small integrated circuit chipset.

High-Density Modulation: In order to fit 256 kbps into the FM band and still accommodate guardbands and ancillary services such as paging requires fairly high density modulation, in terms of bits per Hertz. We envision operating at two bits/Hz, using 8-PSK signals and rate 2/3 trellis coding in conjuction with multi-frequency modulation (MFM). MFM[2] is a technique for transmitting a large number of modulated tones simultaneously. The tones are chosen to be orthogonal to each other by separating them in frequency by 1/T Hz, where T is the baud duration, so that they do not interfere with each other.

Propagation: Propagation in the FM band is characterized by significant multipath effects, due to reflection and refraction of the radio waves by natural and man-made obstacles. Most of the multipath spreading is of less than 15 μs [3]

duration, however, which means that baud lengths T much longer than 15 μs will not experience a great amount of intersymbol interference. We choose T = 1 ms in order to assure this, and in addition allow 60 μs guard time between bauds in order to make certain that a full 1 ms interval with no intersymbol interference will exist.

A second propagation effect which must be overcome is fading caused by destructive interference between propagation paths of different lengths. We counter this difficulty by a combination of interleaving and trellis coding, plus vehicle motion. The interleaving disperses the block of faded signals due to slow movement through a faded region so that errors in reception fall sparsely across a large number of coding epochs where the trellis decoder can eliminate them. We anticipate being able to operate with vehicle speeds as slow as 15 mph. If a vehicle is stopped in a totally faded zone, of course, there will be no signal at all and consequently no reception. But this should be a rare occurence, since fading regions are typically only a few feet wide[3].

The third propagation effect of note is Doppler frequency shift due to high vehicle speeds. This problem is countered by a modicum of frequency tracking, in combination with differential phase shift modulation, which makes the receiver insensitive to a small amount of residual frequency offset. The design goal is operation at any vehicle speed less than 100 mph.

Signal-to-Noise Requirements: One significant advantage of DAB over FM is its lower transmitter power requirement. In order to provide a 60 db signal-to-noise ratio (SNR) at the audio output, an FM receiver requires an input SNR of 33 db. On the other hand, recent work[4] on a satellite mobile voice communications system design using much the same techniques described here indicates that a DAB receiver should operate at a 10^{-4} bit error rate and produce CD quality music at a receiver input SNR of only 17 db. Thus for equal coverage a DAB transmitter can operate at a power level 16 db (or a factor of 1/40) lower than an FM transmitter. It is this fact which makes it reasonable to operate both services simultaneously on adjacent frequencies.

Cost: The key to successful development of DAB service in the FM band is to produce at moderate cost a robust system that operates compatibly with existing FM broadcasting. Care is being taken in system development to make certain that the technology is amenable to implementation using current high-quality FM receiver front-ends, plus digital processing reducible to a small integrated circuit chip set.

THE MDC DAB SYSTEM

Transmitter

The block diagram of the DAB transmitter being developed by Mercury Digital Communications (MDC) is shown in Figure 1. The system is fed by a stream of digits output from a music processor which represents stereo CD quality sound in a compressed format, nominally at 256 kbps for a stereo pair. This, or any equivalent bit stream with an aggregate data rate of 352 kbps or less, is an appropriate source for the transmitter described below.

Trellis Coder: Trellis coding is a technique for transmitting digital information at relatively high information densities, measured in bits per Hertz. It uses a convolutional encoder to generate (k + 1)

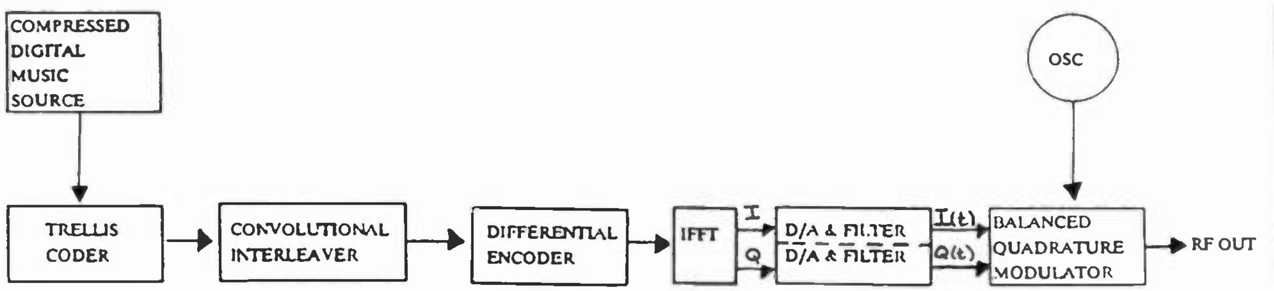

Fig. 1 - Transmitter

bits at the encoder output for every k bits of encoder input. These (k+1) bits are then used to choose one of 2^{k+1} waveforms to be transmitted during each corresponding time interval. For our case k is taken as two, and we have elected the set of 8 phase shift keyed sinusoids (8-PSK) as the modulation set. Although the information rate is therefore the same as if we had used uncoded QPSK, it has been shown[5] that the trellis coded transmission can operate at a low bit error probability in additive white Gaussian noise with 3 db less signal-to-noise ratio than QPSK can.

Convolutional Interleaver: Conventional trellis coding is very effective against random errors but is quite sensitive to bursts of errors, such as will occur when a car drives slowly through a zone in which multipath transmission produces fading of the radio signal. The convolutional interleaver alleviates this problem by mixing up the order in which the signals are transmitted, and then de-interleaving them at the receiver so that a burst of contiguous received errors appears to the decoder as individual errors separated from each other by symbols that are received correctly. The important parameters of the interleaver are how far apart the individual errors are separated, and the length of burst that can be accommodated. We have chosen the separation distance to be ten, which assures that the effects of one error will have been flushed out of the trellis decoder before the next one enters it. The maximum design burst length has been taken to be 32 ms, which is approximately the amount of time required for a vehicle at 15 mph to traverse a faded zone one meter wide (or 1/3 wavelength at 100 MHz.) The advantage of a convolutional[6] interleaver is that this can be accomplished with only a 30 kilosample memory, whereas a block interleaver would require twice this much.

Differential Encoder: In addition to amplitude effects (fading), multipath causes slowly varying phase shifts across the propagation band. One way to protect against these effects would be to try to track them and use coherent detection. A simpler alternative, however, is to use differential modulation. The specific technique used in our system is proprietary to MDC[7] and relies upon the fact that adjacent frequencies across the modulation band (spaced 1 kHz apart) will suffer very nearly the same phase shift as their neighbors. One frequency in each baud is left unmodulated, and its phase used as the zero phase reference for its neighbor. The phase of that neighbor is then used as the zero-phase reference for the next tone, and so on across the band. So long as the relative phase shift from tone to tone due to propagation and receiver combined is substantially less than $2\pi/8$ radians, the system works very well and fits economically into the digital implementation. The only real disadvantage seems to be the well-known 3 db loss in signal-to-noise ratio inherent in all multi-phase differential modulation schemes[8].

Inverse Fast Fourier Transform: The heart of the DAB system described here is the IFFT. Multi-tone transmission has long been of theoretical interest, but not until the advent of the fast Fourier transform was it practical to consider actually generating several hundred modulated tones in parallel[9]. With the FFT, however, the procedure is straightforward: sequentially load the (unit magnitude) phasors specified by the differential encoder into points 0-93 and 162-255 of a 256-point IFFT, set the 68 points from 94-161 to zero to provide a frequency guard band, and turn on the transformer. Since these spectrum samples do not have conjugate symmetry, the IFFT converts them into a set of 256 complex time samples of the waveform corresponding to that spectrum. In order to maintain orthogonality of the tones after multipath propagation, we allow 60 μs (16 time sample points) of guard time between each 1 ms (256 time sample points), and fill this guard interval with a repetition of the first 60 μs of the preceding baud. Because of the periodicity of the DFT, any 1 ms sub-interval out of the resulting 1.06 ms period yields the same transform. To avoid cross-talk between tones, we therefore need only to avoid inclusion within the selected 1 ms interval of any part of the transient smear between bauds caused by the multipath.

Digital-to-Analog Converters: The first step in generating the transmitted waveform is to convert the output samples from the IFFT from digital-to-analog form. This is done separately for the real (I) and imaginary (Q) components of the IFFT output. A parallel to serial readout is performed before D/A conversion, so that only two D/A converters are needed.

Baseband Filters: In order to convert from time samples to modulating waveforms, it is necessary to clock the samples into low-pass filters, one for each of the I and Q streams. Since each baud lasts 1 ms, the clocking rate is 256 kilosamples per second and the low-pass signal bandwidth is 94 kHz. It is important that the filters have a sharp enough cut-off so that the spectral components around the clock frequency are attenuated -50 db. Since there is a guard band with no signal from 94 kHz to 161 kHz, this is easily accomplished by integrated circuit switched capacitor filters of moderate order.

Frequency Translation: To convert the baseband signal into a signal around an intermediate frequency (i.f.) f_o convenient for driving the transmitter, we use two balanced quadrature modulators to form

$$I(t)\cos 2\pi f_0 t - Q(t)\sin 2\pi f_0 t.$$

The resulting waveform is an SSB signal 188 kHz wide centered on f_0. A second mixer stage is used to heterodyne from i.f. to the final transmitter frequency in the FM band.

Power Amplifier: It is important to realize that the power amplifier of the transmitter must be linear. A saturating amplifier of the type used for FM broadcasting will not work because the signal is a sum of sinsusoids with arbitrary phases. The resultant is a waveform which looks much like band-limited Gaussian noise and has significant dynamic variation around its rms amplitude. On the other hand, the power level required for DAB is 16 db lower than that required for FM reception with the same coverage. Thus transmitters in the 200 w to 2 kw range should be adequate--- indeed, will be desirable in order to avoid interference with existing FM service on first-adjacent channels. We conclude that standard SSB transmitter technology should suffice, albeit operating in a somewhat higher frequency band than HF.

Receiver

As shown in Figure 2, the DAB receiver is largely the inverse of the transmitter. As in the case of the transmitter, the analog stages are relatively conventional.

Front End: The receiver front end is similar to any high-quality FM receiver with good noise figure and frequency stability. It is important, however, that the i.f. strip be linear; otherwise, limiting of the noise-like received signal will result. Experimentally, the effect of such limiting on the FFT proves to be tolerable provided that at least 10 db of linear dynamic range above the rms signal level is available.

Analog-to-Digital Conversion: The first step in A/D conversion is to convert the i.f. input back down to baseband, again using balanced quadrature modulators and 94 kHz bandwidth low-pass filters. The two quadrature components are then sampled at 256 kilosamples/sec, and the analog samples fed into two ten bit A/D converters, which in turn feed the FFT.

Fast Fourier Transform: Ideally, the output of the FFT would be the same as the input to the IFFT at the transmitter. In addition to noise, the effects of unknown absolute phase and small oscillator and Doppler errors keep this from being so. But it is important to realize that even a small frequency shift over the baud interval of 1 ms is in fact nothing but a small phase shift in the FFT output. Since this phase shift is nearly identical for all the tones in a baud, the relative phase shift from tone to tone will be nearly the same at the receiver as it was at the transmitter.

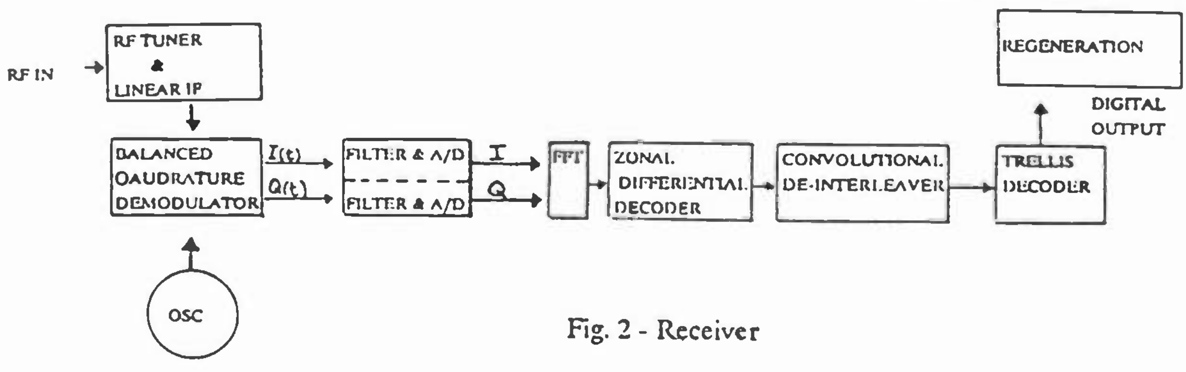

Fig. 2 - Receiver

Zonal Differential Decoder: This uniformity of unkown phase shift across the FFT output is, of course, the reason for using differential phase modulation: by subtracting the phases of adjacent tones, one regains an estimate of the original modulation[8]. The problem is somewhat complicated, however, by the need to de-interleave and to trellis decode, both of which require substantial digital processing. A straightforward solution to this problem is to quantize the received phase plane into zones, and to use the zone identification number as a discrete representation of the differential phase (and magnitude, since we must also be concerned with noise and fading.) We choose to use 16 different phase sectors and 3 different magnitude boundaries, leading to 64 different zones and a 6 bit representation of the received differential vectors.

Convolutional De-interleaver: The job of the de-interleaver is simply to reorganize the symbol stream into the same order as was generated by the trellis encoder. The only significant difference from the interleaving operation in the transmitter is that when de-interleaving each symbol is represented by 6 bits instead of 3.

Trellis Decoder: An ideal trellis decoder searches through the set of all possible transmitter sequences looking for the maximum likelihood cause of the received sequence. It does this efficiently by using the Viterbi Algorithim[10], which exploits the fact that the sequences generated by a convolutional encoder are constrained to lie on a trellis in order to discard less likely hypotheses as soon as possible.

Audio Regeneration: The details of reconstituting the audio from the 352 kbps digital output from the trellis decoder depends upon the specifics of the algorithim used to compress the music at the transmitter.

IMPLEMENTATION CONSIDERATIONS

Analog Sections

Receiver: The requirement for stability in the receiver oscillators is dominated by inherent variations in the received frequency due to Doppler shifts caused by vehicle movement. For example, at a carrier frequency of 100 MHz and relative velocity of 100 mph, the Doppler shift is approximately 16 Hz. It follows that there is little point in prescribing receiver frequency stability better the 10 Hz (or 10^{-7}). Even this, however, is probably too expensive for mass-produced receivers. It seems better to use a 10^{-5} or 10^{-6} crystal oscillator, and track the overall frequency offset (including the Doppler) to one part in 10^{-7}, which corresponds to a 3.6 degree phase error during a 1 ms baud. This can be accomplished by tracking the frequency of the unmodulated tone used as the phase reference for the differential encoding in each baud. Otherwise, as noted before, a standard, good quality FM receiver front end should suffice, provided its noise figure, skirt selectivity and i.f. linearity are state of the art.

Transmitter: The requirement for linearity in the power amplifier stages of the transmitter has already been noted. In order to make certain that degradation due to the receiver dominates, the specifications on the transmitter should be somewhat tighter (linear to 15 db above the rms amplitude, vice 10 db for the receiver.) A frequency stability of 10^{-8} in the crystal oscillators will guarantee no substantive contribution by the transmitter to the receiver frequency tracking problem.

Digital Sections

The digital sections of the transmitter and receiver are being implemented currently using Motorola DSP 56000/1 chips. It is anticipated that this implementation will be ideal for both bench and field testing due to ease of code alteration and performance monitoring. Ultimately, however, it would be desireable to design special integrated circuit chips for mass production of the receiver.

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A COMPATIBLE DIGITAL AUDIO BROADCAST SYSTEM

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Abstract - Compatibility implies existence without disrupting. The concept explored in this paper is that of a digital audio broadcast (DAB) system that can successfully coexist with contemporary FM broadcasting stations.

INTRODUCTION

Digital audio broadcast (DAB) is generally perceived as radio (RF) delivery of compact disk (CD) quality stereo audio by transmission of an appropriate digital bit stream. A compatible digital audio broadcast (CDAB) system is defined here as DAB that coexists with existing FM broadcast signals. In other words, CDAB is a terrestrial based DAB system operated by existing broadcasters, under current modes of operation, much as stereo was added to monaural.

This concept relies on the technology of Power Multiplexing™. The features of Power Multiplexing are presented. The application of Power Multiplexing to a DAB system that occupies no more than 200 kHz of RF bandwidth and that is compatible with current FM service is developed.

This paper traces the development of the technology which now makes possible such a compatible DAB system. The concept, operation, features and state of the art of power multiplexing are presented. The application of this technology to a compatible CDAB system is explained.

BACKGROUND

The designer (engineer) of the radio system has available three parameters whose values are chosen to satisfy the system requirements. These parameters are the time (duration) of the transmission, the bandwidth of the transmission and the power level of the transmission. Quite often these parameters are related. Increasing one affects (increases or decreases) the necessary value of the others.

In some applications, it is desirable to transmit many messages in the same channel (on the same carrier). This is called multiplexing. Present practice assumes two forms.

Historically, the messages shared bandwidth (frequency spectrum). Each message (customer) is assigned a unique frequency "slot" on a dedicated basis (sole use at all times). This is called frequency division multiplexing (FDM). The customers share frequency, and all messages are sent simultaneously. See Fig. 1. In the receiver, the individual messages are isolated (sorted) in frequency and then recovered.

Recently, with the advent of digital technology, the messages are made to share time. Each customer is assigned a unique time "slot" on a dedicated basis (sole use during that time interval). This is called time division multiplexing (TDM). The customers share time, and all messages occupy the same bandwidth. See Fig. 2. In the receiver, the individual messages are isolated (sorted) in time and then recovered.

With TDM and FDM, the bandwidth required to transmit N messages increases linearly with N. If five kHz of bandwidth are needed to send one message, then at least 15 kHz will be needed to send three messages, etc. (These two multiplexing methods are also used on guided communication channels - such as coaxial cable - where carriers may not be required. With TDM, processing of bit streams may occur in an attempt to maintain constant bit rate when another message is added.) Note that no reference is made to power when discussing TDM and FDM. In usual practice, all messages consume an equal amount of power in the process of transmission.

Until the present time, these two multiplexing methods are the only practical choices available to the designers of multi-message communication systems. Note that TDM and FDM connect with two of the three basic design parameters (time, bandwidth, and power). Use of power as a shared resource had been recognized as a possibility, but the concept had eluded practical embodiment.[1]

In 1989, a patented realization of Power Multiplexing (PMx) was successfully demonstrated by Dr. Glen A. Myers.[2,3] This demonstration completes the connection of multiplexing methods with the three basic design parameters of a radio system.

With PMx, each customer (message) is assigned a unique power level (relative to that of adjacent messages). See Fig. 3. All messages are sent simultaneously in the same frequency band. The only commodity that is shared is transmitter output power. No increase in bandwidth occurs.

The realization of PMx relies on the "capture effect" of FM demodulators.[4,5] This effect occurs when two FM carriers in the same band (passband of the superheterodyne receiver intermediate-frequency amplifier) are simultaneously present at the input to the hard limiter part of the demodulator. The effect is that with unequal amplitude carriers, the modulation of the weaker signal does not exist (or is greatly attenuated) at the output of the demodulator. The message of the dominant signal is recovered intact. So, the dominate carrier is said to capture the receiver (demodulator).

With the proliferation of FM broadcast stations, it is now fairly common to experience capture effect on vehicular radios. Because the required power separation of the carriers is slight and because weak signal suppression (capture) is so marked, it is possible to hear with good quality at all times the programming switch from one station to another in traveled distances as short a few hundred meters.

Recent computer simulation and experimental work has established the power level separation necessary for "good" capture occurrence. When the power levels P_d and P_w of the two carriers are equal, the output of the demodulator is unusable. Computer simulations show that capture occurs with power differences as small as 0.17 dB. That is, $P_d = 1.0197P_w$ where P_d is the power level of the dominant carrier and P_w is the power level of the weaker carrier.[6] Measurements on conventional phase locked loop (PLL) demodulators show good capture with separations slightly less than 0.4 dB ($P_d = 1.047P_w$).[7]

The principle of PMx requires that two or more messages (carriers) each be isolated solely on the basis of power. If FM carriers of unequal amplitude are used, then an ordinary demodulator can isolate the message of the stronger carrier by virtue of capture effect.

The engineering challenge has been to recover the message of the weaker carriers as well. Previously, circuits have been proposed which claim to suppress the dominant carrier so that the weaker carrier can be demodulated.[8] No practical system has been devised which successfully permits simultaneous recovery of the messages from both carriers until the introduction of the Myers' Demodulator (MD) in late 1989.

A block diagram of the MD is shown as Fig. 4a for the case of two FM carriers in the same band and as Fig. 4b for N such carriers. A feature of the MD is a phase tracking circuit (PTC) (patent pending).[9]

The operation of the PTC is as follows. With reference to Fig. 4a, a particular PLL design is used to recover $m_d(t)$, the message of the dominate carrier. An auxiliary loop has been designed to recreate, using $m_d(t)$, a replica of the dominant FM carrier. This replica is, by design, 180° out of phase with (the inverse of) the dominant FM carrier. This 180° relationship is made to occur over the entire broad lock range of the PLL. (A method has been devised to observe, using a single channel of an oscilloscope, the preservation of the 180° phase shift over the peak-to-peak frequency deviation range of an FM carrier. With an I.F. of 455 kHz, the 180° phase shift has been observed from 200 kHz to 600 kHz -- the lock range of the PLL.)

The replica of the dominant FM carrier is added to the PMx carriers to cancel the dominant FM carrier. The remaining FM carrier is then demodulated with an ordinary PLL (or other FM demodulator).

Again, because of the capture effect of FM demodulators, it is not necessary to cancel the dominant FM carrier. It is only necessary to attenuate this carrier by a few dB such that it no longer dominates at node B of Fig. 4a.

It is a feature of the MD that the PTC can be used repeatedly without modification to recover all the messages from N PMx carriers as shown in Fig. 4b.

A COMPATIBLE DAB SYSTEM

There are two main technical implications of a CDAB system.

1. The digital carrier occupies the same bandwidth as existing FM stations.

2. The digital carrier is transparent to existing FM receivers.

There are also two practical aspects of any radio broadcast system (apart from profit considerations).

A. The effects of multipath propagation on reception of the broadcast signal.

B. The coverage provided by the transmitter power.

These issues are addressed in the rest of this paper.

Use of the recently developed Power Multiplexing technology allows another (digital) carrier to occupy the same bandwidth as the analog FM stereo carrier. A variety of proposed modulation methods permit the transmission of several bits per Hz. These provide the opportunity to place the bit stream required for digital audio in the normal 200 kHz band assigned to a commercial FM station. This transmission is assigned a power level less than the existing analog stereo signal. With proper power management, the digital transmission is transparent to existing stereo FM receivers. The modulation representing the bit stream can be AM or FM or a combination thereof. In any case, the new circuitry can allow recovery of the analog stereo signal as well as the digital audio bit stream.

With Power Multiplexing, the dominant FM carrier is assigned a major portion of the available transmitted power. The subdominant carrier(s) consume the rest. The assignment of power for the case of two carriers is diagrammed in Fig. 5. There, C_r represents the power at the input to the receiver; C_s is the power of the dominant carrier (FM stereo); C_d is the power of the subdominant (weaker) carrier (digital audio); N is receiver input noise power (in the bandwidth of the IF amplifier). Decibel level separations are represented by I, x, y and z.

It is apparent from Fig. 5 that x and z can be controlled or managed by choice of y (for a given value of I). For example, increasing y places more power in the FM stereo carrier and hence less in the digital audio carrier. This will increase the range or coverage of the FM stereo signal at the expense of coverage of the digital audio signal.

Assume no increase in transmitter power with the addition of the digital signal. Then, from Fig. 5,

$$C_r = C_s + C_d.$$

Since $C_s < C_r$, then the stereophonic broadcast range is reduced relative to that occurring when $C_s = C_r$. This reduction in range is easily determined as follows. Let R_s be the maximum range of the stereophonic broadcast when $C_s = C_r$. Thus, at a distance of R_s miles from the transmitter, the broadcast is by definition of R_s at a reception "threshold." The spatial attenuation in dB of the propagating broadcast power is 20log R where R is the distance from the transmitter. So, a reduction in broadcast power by x dB is equivalent to a reduction in range of $20\log(R_s/R_{s,d})$ where $R_{s,d} < R_s$ is the stereophonic broadcast range given a CDAB signal (when $C_s < C_r$). For example, when $x = 1$ dB, then $R_{s,d} = 0.891R_s$. The range of the FM stereo broadcast with a digital audio signal is 89.1% of that of the stereo signal alone. Table 1 shows the values of interest for y in the range 10 dB to 35 dB. <u>A conclusion is that the reduction in FM stereo broadcast coverage is negligible when the power separation of the carriers exceeds 10 dB</u>.

Of interest next is the coverage of the CDAB signal. For digital transmission, z of Fig. 5 is typically in the range of 0 to 15 dB. The actual value depends on the specified error probability, carrier modulation, error control coding employed and receiver operation (processing). So, a requirement for most systems is that $z > 0$ which implies, from Fig. 5, that $z = I - (x + y) > 0$. Since $y >> x$ (Table I), then the requirement is that $(I - y) > 0$ for a usable system. Note that z is SNR at the receiver input and not at the demodulator input. The difference is the noise figure of the superheterodyne front end.

We wish to relate the parameters of Fig. 5 to CDAB coverage. Now, z is merely an indication of SNR available for the CDAB signal. Coverage depends on SNR required. So, define z_p as that value of SNR required to provide the minimum specified value of error probability. This value is know for a given CDAB signal and CDAB receiver.

It is useful to relate CDAB coverage (maximum usable range R_d) to FM stereo coverage. To this end, select a value z_e of z corresponding to some range $R_{d,e}$ where $R_d = R_{s,d}$ (equal coverage of FM stereo and CDAB). Now define a margin $M = z_e - z_p$, dB. If $M > 0$, then $z_e > z_p$ (available SNR greater than that required) and CDAB coverage exceeds that of FM stereo. Conversely, if $M < 0$, then $z_p > z_e$ and the CDAB coverage is reduced until the required SNR is achieved. As was done in the case of the FM stereo signal, we can relate R_d to $R_{d,e} = R_{s,d}$ as M varies. Values are shown in Table II and plotted in Fig. 6.

Recall that $z \approx I - y$. So, z (and M) can be increased by increasing $I = C_r/N$ or by decreasing y. The value of I is determined by stereo quality specifications and receiver performance. A way to increase I is to increase the effective radiated power (ERP). The effect is to increase both the stereo and digital audio coverage areas. This is an effective way to make $M = 0$ for systems where $M < 0$ otherwise. Recall that $M = 0$ is the condition of equal coverage areas.

Acceptable values of y depend on the performance of receivers in the field (capture characteristics), on acceptable reduction, if any, of audio quality and on the CDAB waveform broadcast. The CDAB waveform affects y in two ways: by its relation of error probability to SNR and by its effect on FM stereo audio quality. Testing is required to determine the effect on analog audio of candidate CDAB waveforms. Appropriate design of CDAB waveforms coupled with design of new receivers intended to receive both FM stereo and CDAB may substantially decrease the acceptable value of y with commensurate increase in CDAB coverage.

There are, then, at least three important considerations in the selection of a waveform to carry the digital signal in a compatible DAB system. First is the relation of probability of error (P_e) to signal-to-noise ratio (SNR). Second is the impact on existing receivers. Third is the effect of multipath propagation.

All wireless transmission systems are subject to multipath reception. Because waves reflect and refract, it is possible to receive a signal after propagation along more than a single path. The simultaneous presence at the receiver antenna of many waves of differing amplitudes and phases is called fading when the interference is destructive.

Frequency diversity and space diversity are common ways of compensating for fading transmission. Both techniques use redundancy. With frequency diversity, additional bandwidth is used. With space diversity, one or more additional receiving antennas are used.

Because of the two-state nature of digital signals, the transmission of digitized messages is more immune to the effects of fading than analog messages of specified fidelity. In addition, digital signals can be "carried" in a variety of forms. This ability to design waveforms creates choices which can be intelligently used to address peculiarities of a given communications link. To combat fading, the choices involve redundancy of some type in either the transmission or the reception of the bit stream.

There are many choices of waveforms. Because of the 200 kHz bandwidth constraint of a compatible DAB system, a waveform capable of reliably carrying more than one bit per Hertz of bandwidth is needed. Some examples are a family of pulse waveforms with matched filter reception and multiple carrier modulation (multiple frequency modulation) as in the Eureka 147 system.

Other examples which are being pursued and which are waiting test results are the following. Power Multiplexing can be extended to three carriers. Two digital carriers separated by about 1 dB in power and spaced y dB and (y + 1) dB from the analog carrier can be used to double the bandwidth available for the CDAB signal. This provides redundancy in transmission which can be used to combat multipath effects.

Also being considered is a combination of FDM and Power Multiplexing. We call this composite multiplexing. This proprietary technology uses patented circuitry to isolate in the receiver several FDM modulated carriers in the 200 kHz band. These several carriers form the CDAB signal. Features of this proposed compatible system include redundant bit transmission, relaxed synchronization requirements and expected low cost of the receiver.

CONCLUSIONS

A compatible DAB system is technically possible. Tables I and II indicate that adding a CDAB signal has negligible effect on FM stereo coverage (for expected values of y = dB carrier separation). To be determined are CDAB waveforms, CDAB power level and receiver design that results in a "best" operating system. The most desirable system for this country and for the broadcast community will emerge after additional development and testing.

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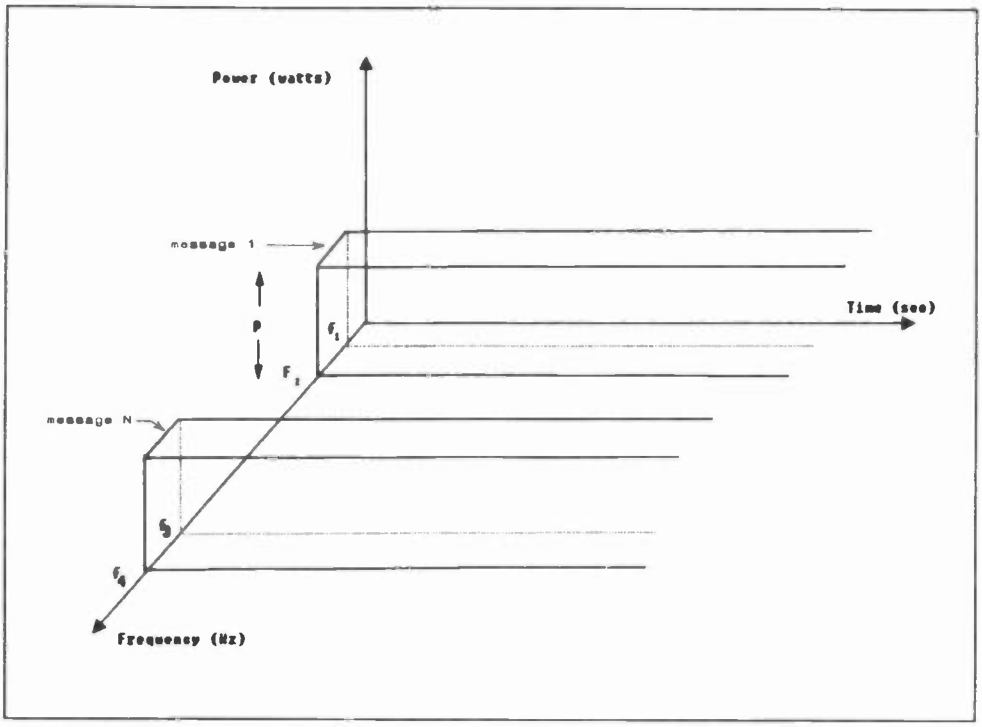

Fig. 1 Frequency, Time and Power Distribution Among Messages in FDM.

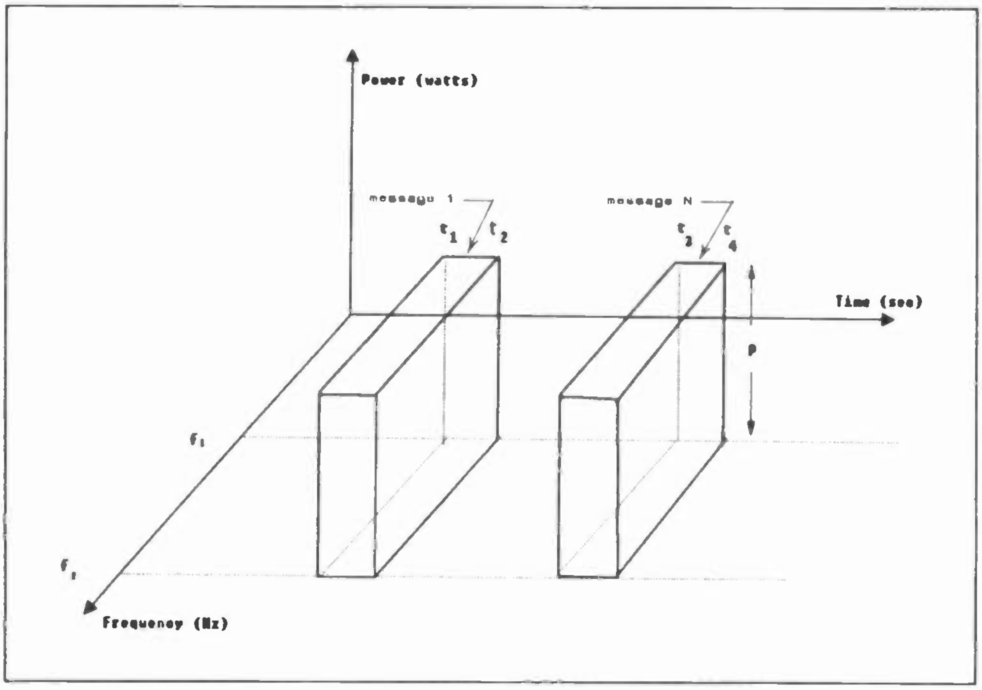

Fig. 2 Frequency, Time and Power Distribution Among Messages in TDM.

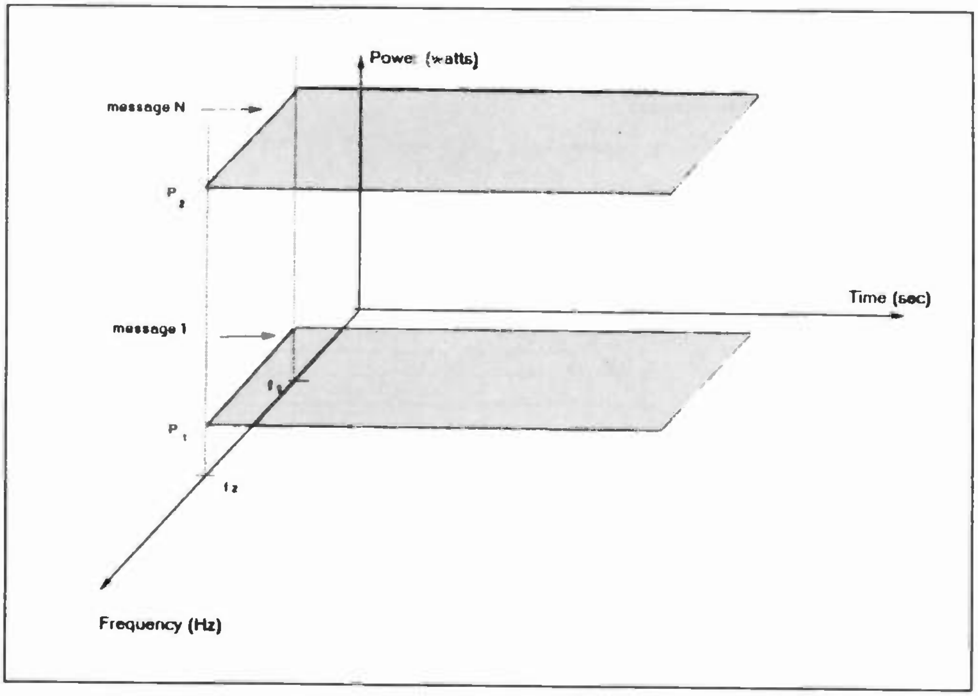

Fig. 3 Frequency, Time and Power Distribution Among Messages in PMx.

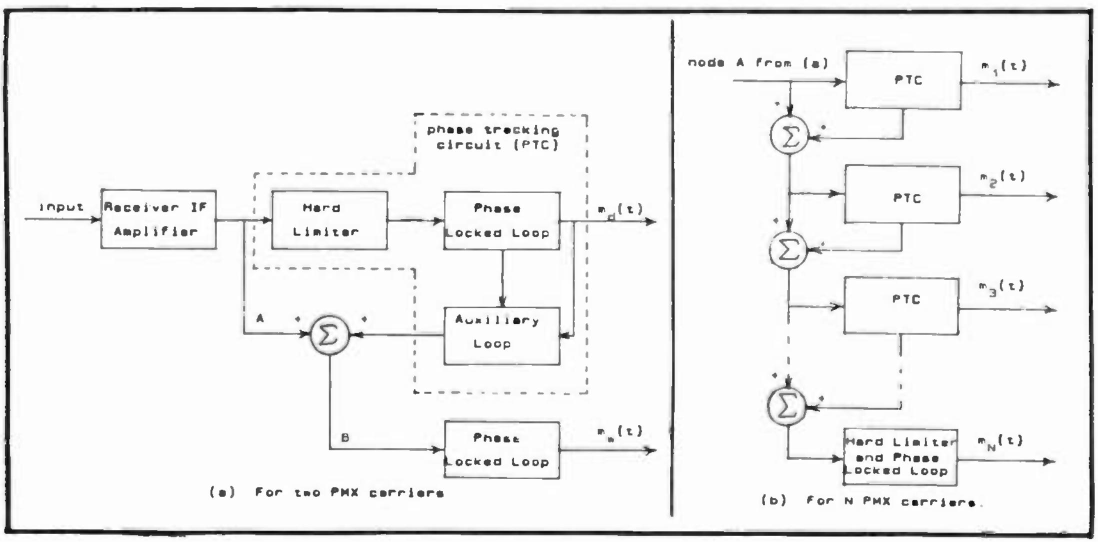

Fig. 4 Block Diagram of the Myers' Demodulator

TABLE I

FM Stereo Range Reduction for Various Values of Carrier Separation

y	x	$(R_{s,d}/R_s)$ X 100
10	0.414	95.3
15	0.137	98.4
20	0.043	99.5
25	0.014	99.8
30	0.004	99.9
35	0.0014	99.99

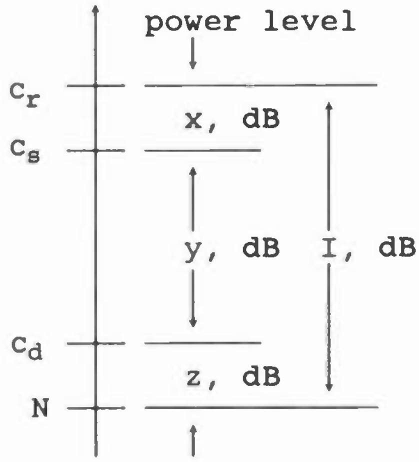

Fig. 5 An Indication of Power Levels Associated with Power Multiplexing

TABLE II

DAB Range Factors for Various Values of Change in SNR

M	$R_d/R_{d,e}$
+6	2
+5	1.78
+4	1.58
+3	1.41
+2	1.26
+1	1.122
0	1.0
-1	0.89
-2	0.79
-3	0.71
-4	0.63
-5	0.56
-6	0.5

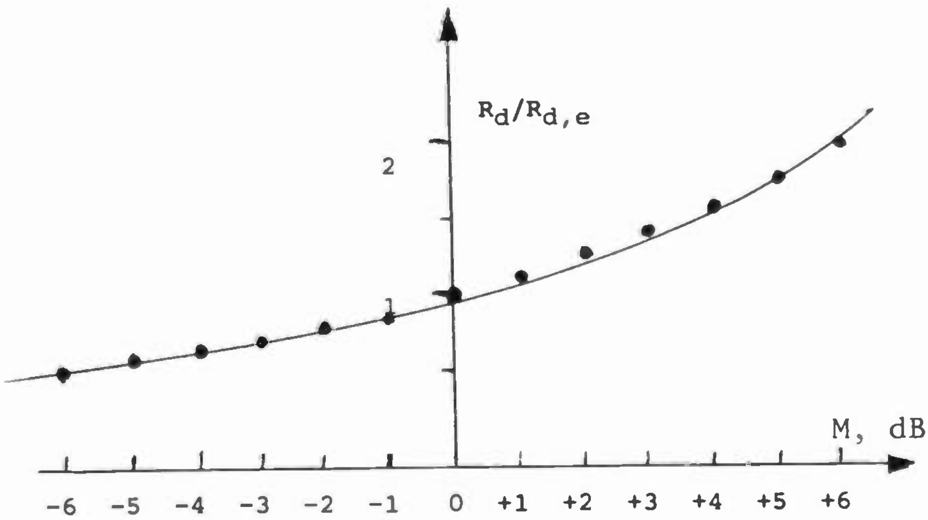

Fig. 6 CDAB Range Factors versus Change in SNR

MULTIPATH CANCELLATION TECHNIQUES FOR DIGITAL AUDIO BROADCASTING

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Abstract- Digital audio broadcasting has the potential for delivering programming to the public with no degradation from the original source. Radio broadcasting, however, has limitations in delivering signals to receivers which contain all the original information. One of the major sources of disruption of radio signals is multipath interference, occurring when the signals from the transmitter are received by the direct path and one or more reflected paths. Some strategies to eliminate the effects of this interference will be discussed.

Introduction

Digital audio broadcasting (DAB) promises to revolutionize audio broadcasting. Many believe that the improvements in audio quality is the reason for this revolution, and for AM broadcasters switching to DAB this may be the case. The delivered audio quality improvement for FM broadcasters switching to DAB will not be so great, in fact may not even be noticeable in a mobile environment. This is due to the fact that road and wind noise, and the acoustic environment and performance of most automotive speakers are presently the limiting factors in cars. DAB has the potential for a much more important improvement for the listener. This improvement is consistency and uniformity of service area.

FM broadcasting is an analog system. The signal quality gradually decreases as the signal level gets weaker. One important advantage of FM over AM broadcasting is that in FM the quality loss with weaker signals is a non-linear function. An FM signal will remain robust from strong signals areas to where the signal is relatively weak, while AM is progressively more susceptible to noise as signal strength decreases. When the FM signal decreases to the threshold level, the performance will rapidly deteriorate as the signal weakens below threshold. Digital broadcasting takes this one step further. With error correction, and a well designed RF coder-decoder system, the DAB signal will provide perfect reception until the system simply shuts off. There is no gradual worsening of the signal - but an abrupt end to the service area.

All radio broadcasting is susceptible to multipath interference, however the effects on FM and TV broadcasting are very significant for different reasons. TV broadcasting is susceptible to multipath because the television signal is very sensitive to multipath propagation, which produces the visual effect of ghosts. FM broadcasting is more robust and can operate with modest amounts of multipath interference, except for the many mobile FM receivers. Mobile receivers travel through all areas, and the users cannot simply reorient the antennas on their cars to avoid the problem. Digital audio broadcasting will suffer the same effects as FM due to the mobile nature of its audience.

Multipath

Multipath interference results from signals arriving at the receiver from more than one path. Usually, but not always, the direct path from the transmitter to the antenna is dominant. Additional paths from the transmitter via reflecting objects provide additional signals which are delayed from the direct path signals.

For an example, let us assume that the multipath environment consists of the direct signal, and a single reflected signal with an excess delay of T_d. For a single frequency transmitter, the signals received at a mobile receiver moving in a multipath environment would follow a standing wave pattern, ie., the received signal would increase and decrease in amplitude as the vector addition of the direct and delayed signals at the receiver. [1]

If the transmitter were a wide band emission such as a noise source, the wideband received RF frequency response would be a comb filter response, with many frequencies (spaced by $1/T_d$ Hz) attenuated by the effect of the reflected signal. If the receiver is mobile, then the excess delay varies and the attenuated frequencies vary when the vehicle moves. You can imagine how the situation can be complicated by multiple reflections, each with a separately changing delay, and relative amplitude when the receiver is moved.

Since the nature of multipath interference from multiple reflectors can become exceedingly complex, it is necessary to evaluate the propagation of radio waves with more than one path by statistical evaluation. The land mobile industry operates near virtually all the frequency bands proposed for DAB. Research shows that the distribution of signal strengths over small areas shows Rayleigh distribution[2] with the excess delays of the reflected signals ranging from -2 to +6 μs. [3 4]

An additional characteristic of the multipath environment is that the average rate of fluctuation of signal strength in a mobile receiver with multipath interference is dependent upon frequency of the transmission and speed of the vehicle according to the following equation:[5]

$$A = 0.003\ fv$$

where A is the average amplitude fluxuation rate in Hz, f is the carrier frequency in MHz and v is the vehicle speed in miles per hour. This means that for an automobile traveling 55 miles per hour, and a carrier frequency of 100 MHz that the signal level would go through one complete cycle from maximum to minimum to maximum in only 61 milliseconds, at 1500 MHz this period would only be 4 milliseconds.

Digital Audio Broadcasting Systems

Various systems have been proposed for DAB. They all rely upon several stages. The first stage in all systems is data reduction. CD quality audio requires approximately 700 kb/s in standard Pulse Code Modulation form for each audio channel. Various systems have been proposed to reduce this data rate. The present state of the art is to provide reduction to the 128 kb/s to 96 kb/s range while still retaining audio transparency. This substantial data reduction makes it possible to visualize an over the air channel which can broadcast audio programming.

The second stage is error correction coding of the rate reduced data. The data have additional redundant information added so that if some of the data are lost in transmission the remaining data provide enough information to regenerate the lost data. Additional to error correction is the ability of the coding system to detect that data has been so corrupted that the data cannot be regenerated at the receiver. There are methods which can be used to decrease the audio effects of loss of data if it is known that the data are defective. There are several different error correction codes which may be used, each design has strengths and weaknesses which should be matched to the problems of the transmission channel.

All DAB systems have a data latency. Since error correction coding and data reduction must operate on blocks of data, there must be a time delay in the program. Some systems seek to hold the data latency period to 30 mS or less. The eventual system to be implemented in the United States will probably have a data latency of between 300 mS and 1 S to provide for optimal data reduction and coding while still keeping clock times correct.

The third stage in some systems is the RF encoding and transmission. This translates the data stream from the error correction system to something which may be transmitted over the air. An example of a very simple RF coder is that originally proposed by CD Radio, Inc.[6] This system used Offset Quadrature Phase Shift Keying of a microwave carrier frequency. Simply described, each succeeding two bits of the data stream are used to form a **symbol** which sets the relative phase of the carrier frequency. The phase positions of the transmitted carrier are 0°, 90°, 180°, and 270° corresponding to the four states which two bits may represent. OQPSK is different from Quadrature Phase Shift Modulation by the fact that one of the two bits is delayed by one bit period from the other. This modulation system may be easily decoded with a phase locked loop. In this system the time to send one symbol is 7.8 μS. The bandwidth required to receive this signal is approximately 128 kHz.

In the European Broadcasting Union's *Eureka* 147 system several data reduced data streams from independent programs are distributed among many carriers spread across a several megaHertz band. Each carrier is QPSK encoded with two bits of one data stream, and each of the carriers are spaced at nearly the minimum possible frequency spacing while still permitting each channel to be independent, or orthogonal. Every 16th carrier is assigned to one program. The data stream is then distributed among the carriers which have been assigned to that program, so that as many as 30 carriers are used to transmit the data of one program at the same time. Each symbol time can then represent as many as 60 bits. The symbol time can then be as long as 468 μSeconds and still transmit the full data stream.

John Leonard and Glen Meyer's *Power Multiplexing* is a system to modulate a carrier in quadrature to the FM broadcasting carrier. This system has a short symbol time, and high modulated data rate.

The DAB system being worked on by William Spurlin, Peter Moncure and myself is a spread spectrum system, with the similarity to the *Eureka* 147 system that the symbol time is long, but designed to operate in frequency ranges already occupied by other services, such as conventional FM broadcasting.

Space Diversity

One strategy to overcome the effects of multipath interference in any RF system is to use space diversity. In concept this strategy makes use of two separate receivers, fed from two antennas spaced sufficiently far apart so that when one antenna is receiving a signal which is unable to be properly received due to multipath, the probability is high that the other receiver is receiving an acceptable signal. Since it is electrically possible to determine in most communications systems which of two receivers is receiving a better signal, the output is taken from the better performing receiver. This selection must, in a mobile environment be continually reviewed.

In a multiply polarized system, an additional advantage may be gained by polarizing the diversity antennas orthogonally.

Attempts at space diversity reception in FM broadcasting have been limited. The distance at which two antennas may be mounted on an automobile is too small limiting the improvement factor available. Automobile manufacturers are unwilling to mount additional whip antennas on cars, and the receiving equipment costs and complexity is great for diversity receivers. The optimal distance between antennas for full benefit from diversity reception decreases as the transmission frequency increases.

In a wideband system it is theoretically possible to recover eliminated frequency bands in one receiver by combining the main signals with the signals from the diversity antenna in a bandsplit system.

Ghost Canceling

In a stationary receiver, the excess delay and signal amplitude of the reflected signals remain relatively constant over time. Recognizing that the mechanism of multipath in this case is linear, and nearly time invariant, then it is possible to develop a system which will remove the reflected signals. Ghost canceling works well when implemented in the laboratory, and television broadcasters are busy developing an automatic system.

Ghost cancelers operate using an adjustable tapped delay line, and adders (subtracters). By adjusting the delay line tap so that the delay of the tap is equal to the excess delay of the reflected signal, and subtracting a portion of the signal at the tap from the signal at the input of the line, the resultant can be made to remove the effects of one reflection. By using additional taps and subtracters the results of each reflection may be removed. The major research effort is to develop a training signal which may be included in the vertical interval which can be used to set up these parameters on an adaptive basis.

Unfortunately, this technique is unlikely to be successful in the mobile environment, since the multipath characteristics at the receiver change with receiver position. The adaptive ghost canceler would always be searching for the correct parameters, and only find them when the vehicle stopped. It is probably impossible to develop an effective narrowband multipath corrector for FM broadcasting.
RF or IF multipath correction for broadband DAB systems could be implemented for stationary receivers, but the effectiveness is limited for mobile reception. The higher the frequency of the DAB system, the less effective a ghost canceling system can be. This is because the multipath signal fluxuation rate increases as the system frequency increases. It might be possible to implement a DAB ghost cancellation scheme which can provide some relief in a mobile environment at 100 MHz, but it is difficult to conceive of a system which could implement an adaptive algorithm at high UHF frequencies.

Frequency Dispersion

The decoder of the error correction system in DAB systems permits perfect reproduction of the audio signal in spite of some errors in the received data. If too many errors are received in one block, then the block is discarded. A discarded data block will result in an audible artifact. Multipath interference does not attenuate all frequencies simultaneously.

If the data stream is broken down into a number of independent narrow bandwidth channels separated in frequency from each other only one or a few channels will deliver incorrect data due to multipath interference at any time. If the same data were concentrated into a single high rate channel multipath cancellation would destroy much more of the data for two reasons. The delay of the multipath echos is long in comparison to the symbol duration. If a symbol takes 8 μS to send, and an echo comes along 4 μS delayed into the symbol period, the probability of that symbol being destroyed is much higher than if the symbol period is hundreds of μS.

The single channel high data rate system is likely to have part of its required channel bandwidth extend within a frequency band in which the multipath interference causes a cancellation. This cancellation will distort the detected data and cause a high data error rate.

To some extent these two effects are manifestations of the same problem. Essentially, each channel becomes more susceptible to multipath interference as the symbol rate increases in a channel. This problem is minimized by providing multiple independent channels, each with a low symbol rate.

The *Eureka* 147 system achieves the goal of low symbol rate by dividing the data stream for a program into many independent channels which are frequency interleaved with the other program's channels[7]. By spreading the program over many low rate widely dispersed channels *Eureka* 147 gains the benefits of frequency dispersion.

Time Dispersion

When mobile receivers are moved in areas of multipath interference the receivers alternately receive clear data, and areas with corrupted data. This fluctuation in data reliability follows the equations in the multipath section above. The data in a DAB program may be transmitted out of time sequence. If the data is time interleaved, and the interleaving separates the data within the same block enough in time to make the data errors independent, then an additional immunity from multipath interference is gained. To obtain independence of two consecutive symbols for an emission interleaved over P Symbols at a symbol rate T_s if: [8]

$$P \geq \frac{2.5}{2\pi} \frac{T_s}{F_o} \frac{c}{v}$$

where v is the velocity of the receiver, F_o the frequency and c the speed of light. As you can see the effectiveness of time interleaving improves with frequency and velocity of the receiver. Time interleaving is of limited use at 100 MHz and normal automotive speeds. At microwave frequencies this additional level of independence can partially compensate for the poorer propagation effects at higher frequencies, at the expense of additional receiver complexity.

Excess Symbol Time

If symbols are sent through a linear channel of a given bandwidth, there is a symbol rate above which the adjacent symbols interfere with each other. This symbol rate is known as the *Nyquist Rate*. If the actual symbol rate is reduced, providing guard delays longer than the excess delay of all expected multipath paths at the beginning and end of each symbol, then an optimized detector can sample the required symbol time and exclude the guard periods significantly improving the multipath interference rejection. This method trades off noise bandwidth, required channel bandwidth and spacing, and signaling rate for multipath resistance. This technique is effective for only relatively slow signalling rates. For the CD Radio, Inc. example above the expected multipath delays are nearly as long as the symbol time, making it a very unprofitable tradeoff, whereas in *Eureka* 147 the tradeoff is very favorable since the symbol time is so much longer than the maximum probable excess delay.

Conclusion

Multipath interference is a serious problem in any VHF, UHF or Microwave communications system. There are various ways to combat this interference for any modulation system, including diversity, and ghost canceling. Digital Audio Broadcasting provides the opportunity to eliminate many effects of multipath interference through error correction in combination with strategies unique to digital systems. Innovative system design has permitted the proponents of these systems to bring a high level of immunity to multipath interference to their systems.

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THE RADIOSAT℠ SYSTEM

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Abstract - MSAT satellites to be launched in 1994 can transmit many audio channels directly to cars with small antennas. Radio Satellite Corporation is developing the RadioSat℠ system to use MSAT to broadcast digital audio programs to cars throughout the United States. RadioSat℠ supports a wide array of integrated nationwide mobile services, including paging, two-way voice and data communications, and precision navigation, as well as nationwide digital audio broadcasts.

This paper provides comprehensive background information on broadcasting through MSAT. It reviews the institutional and regulatory history of MSAT and considers MSAT economics. It concludes with a description of the RadioSat℠ system.

INTRODUCTION

A unique new satellite will fly out of the Florida marshes into space in 1994. It is a product of a decade of efforts by government agencies and private industry in the United States and Canada to implement a North American Mobile Satellite Service (MSS) system named MSAT.

MSS has been under development in the United States for more than two decades. NASA demonstrated aeronautical, land and maritime mobile satellite services in a number of experiments using its Applications Technology Satellites in the late 1960's through the early 1980's.[1] Marisat, the first commercial MSS system, has been operating since 1976.[2] Marisat and all other existing MSS satellites are operated by INMARSAT, a London-based international consortium providing communications to ships at sea around the world.

Previous MSS satellites have had very limited radiated power, which has limited potential applications. MSAT satellites, which are designed to concentrate power into North America, each have a radiated power of about 500 kW – far higher than that of any other MSS satellites (Figure 1). The high radiated power of MSAT enables new services, such as commercial audio broadcasting.

The RadioSat℠ system under development by Radio Satellite Corporation was designed to provide consumer services that take advantage of the unique capabilities of MSAT. RadioSat℠ supports nationwide digital audio broadcasting with up to CD-quality, personal communications, and precision navigation services.

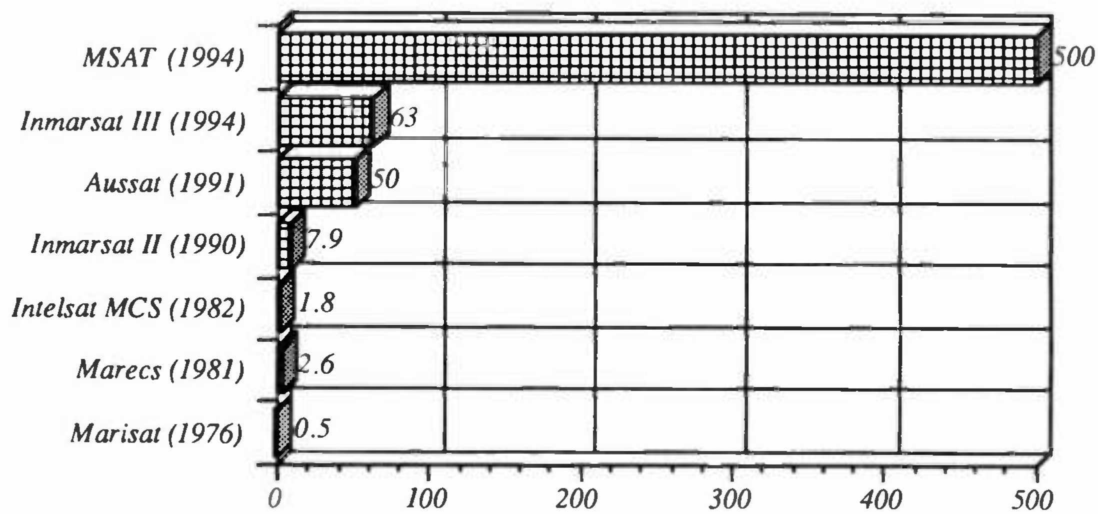

Figure 1. Effective Isotropic Radiated Power of MSS Satellites

MSAT Review

MSAT is the culmination of eight years of regulatory and institutional development. It is a cooperative venture between U.S. and Canadian government agencies and private industry.

MSAT has major economic advantages over the worldwide MSS system operated by INMARSAT. It uses large, multiple beam satellite antennas to concentrate coverage over areas with the highest demand. INMARSAT provides global coverage, requiring far less efficient antennas.

MSAT also has large economic advantages over a system designed to serve just one country. Joint implementation of the space segment by U.S. and Canadian companies eliminates the need for separate in-orbit spare satellites, since the U.S. and Canadian satellites can back up each other. Further savings are achieved in the ground segment by producing similar Network Operations and Telemetry, Tracking & Control (TT&C) centers for the U.S. and Canadian mobile satellite systems. Ground facilities also provide mutual backup and greater network access to users roaming over wide areas. The mutual backup of both the space and ground segments between the U.S. and Canadian systems assures high reliability.

The United States

In 1988, the FCC authorized the American Mobile Satellite Corporation (AMSC) to provide MSS in the U.S. This authorization followed 6 years of regulatory and institutional development (Table 1), and the first MSAT satellite will not be launched for another 4 years. This long formation period shows how difficult it is to implement new satellite services.

AMSC is a privately held corporation based in Washington, D.C. Its stockholders have committed $140 million in capital to the company. The AMSC stockholders are:

Hughes Communications Mobile Satellite Services, Inc.
Mtel Space Technologies Corporation
McCaw Space Technologies, Inc.
Mobile Satellite Corporation
North American Mobile Satellite, Inc.
Satellite Mobile Telephone Co.
Skylink Corporation
Transit Communications, Inc.

AMSC will own and operate the U.S. MSAT satellites and Network Control Center. It is a carrier's carrier. AMSC is allowed to provide service as both a wholesaler and retailer, though only on a non-discriminatory basis.[3]

1982	NASA Petition for Rulemaking
1983	Initial applications filed
1984	FCC Notice of Proposed Rulemaking
1985	FCC accepts applications
1986	Spectrum allocation
1987	$5 million deposits by applicants
1988	AMSC incorporated
1989	License issued to AMSC
1990	Satellite construction begins
1991	Interim service
1994	Launch; audio broadcasts begin

Table 1. AMSC Timeline

NASA will launch the first AMSC satellite in return for the use of some MSAT capacity for two years.[4] NASA plans to use MSAT to perform technology experiments and to enable government agencies to assess the usefulness of MSS to their operations. Government applications include public safety, aviation safety, communications for wide and remote area coverage (police and border control), monitoring of hazardous material transport, and other uses. After two years, agencies continuing with the service will become commercial customers of AMSC.

NASA sponsored an MSS development program at the Jet Propulsion Laboratory (JPL) throughout the 1980's.[5] JPL plans to demonstrate mobile satellite audio broadcasting through an INMARSAT satellite in mid-1991.

Canada

Telesat Mobile, Inc. (TMI), a subsidiary of Telesat, Canada's domestic satellite carrier, is the Canadian MSS operator. Telesat Canada is the leading stockholder of TMI; the rest of TMI is owned by various financial interests. TMI will own and operate the Canadian MSAT space segment, a Network Control Center, and a network of base and gateway stations.

The Canadian Department of Communications (DOC) has the responsibility for developing domestic telecommunications for Canada. DOC has made a $176 million commitment to MSS – $30 million for hardware development, $20 million for user trials, and $126 million for leased capacity from TMI for Canadian government needs.[6] DOC has developed advanced MSS technology at its Communications Research Centre.

AMSC and TMI signed a Joint Operating Agreement (JOA) on April 25, 1990 formalizing mutual backup and intercarrier leasing arrangements. The AMSC-TMI JOA is a landmark document formalizing the first regional MSS system. It promises significant benefits to both operators, including substantial reductions in cost and risk to each of them in the implementation of the North American MSS system.

Regulatory Review

The FCC considered using both UHF (821-825 MHz and 866-870 MHz) and L band (1545-1559 MHz and 1646.5-1660.5 MHz) spectrum for MSS. It eventually allocated L band spectrum (Figure 2).[7]

Consistent with recent trends in satellite communications regulation,[8] the FCC issued a broad license to AMSC. Throughout the MSS proceedings, the FCC emphasized its desire to encourage innovative use of MSAT. In the original Notice proposing an MSS Rulemaking,[9] the FCC proposed not to prohibit "any auxiliary or incidental common carrier service ... provided that the primary purpose of such satellites is to provide mobile radio services by satellite." The FCC reaffirmed this position when it authorized AMSC to construct, launch and operate a mobile satellite system.[10] The Commission stated in its Order establishing MSS licensing procedures that "[w]e will not at this time limit the types of services that may be offered by the consortium provided that the service is consistent with the MSS service definition."[11] MSS is defined as "A radiocommunication service ... [b]etween mobile earth stations and one or more space stations ..."[12]

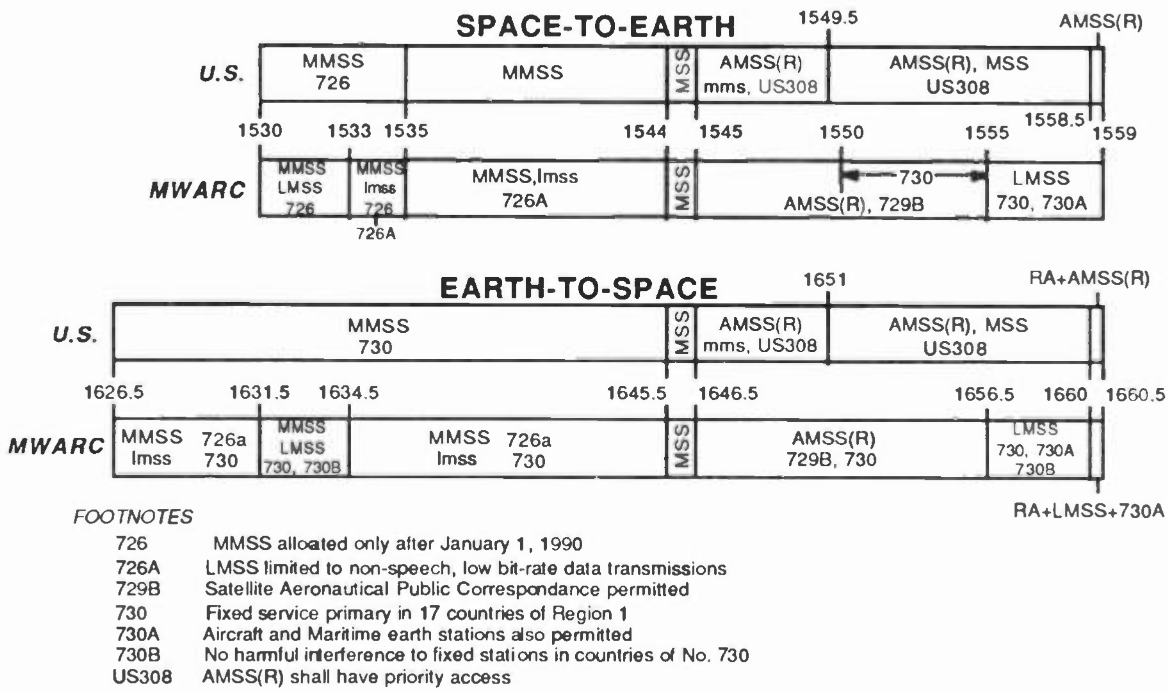

Figure 2. U.S. and MWARC MSS Allocations

Internationally, mobile satellite services have historically been broken into different allocations for Land Mobile Satellite Service (LMSS), Aeronautical Mobile Satellite Service (AMSS), and Maritime Mobile Satellite Service (MMSS). AMSS for en-route communications related to safety and regularity-of-flight has been defined as AMSS (R) and assigned separate allocations. Each of these services is a subset of MSS; they can all be provided in any MSS band.

The U.S. and Canada determined that domestic mobile satellite services are best provided through a single satellite system supporting land, aeronautical and maritime services. INMARSAT is now providing all of these services on a global basis through common satellite transponders. The provision of multiple services through a single system is most efficiently accomplished with a single contiguous allocation, with dynamic reallocation of capacity between the services performed by the system on demand. The United States thus allocated 13.5 MHz of L band spectrum in each direction to MSS, with AMSS (R) being primary or having priority in portions of the band.

The U.S. and Canada proposed common allocations for the services at the 1987 World Administrative Radio Conference on Mobile Services (MWARC). The MWARC adopted this approach in the MMSS band, reallocating MMSS spectrum to LMSS in three categories: Exclusive, Co-Primary, and Secondary (lmss) to MMSS. The MWARC also reallocated a portion of the AMSS (R) band to LMSS, but did not open up the remaining AMSS (R) band to other services. The U.S. and Canadian governments took reservations on that portion of the MWARC outcome. The U.S. and Canada will seek an extension of the combined allocation approach to the remaining exclusive AMSS (R) bands at the 1992 WARC. This approach now has widespread international support.

MSAT Economics

MSAT is, in essence, a mobile repeater that is 22,300 miles high. There are both negative and positive consequences to placing a repeater this high. The cost per channel of MSAT is much higher than the cost per channel of a terrestrial repeater. On the positive side, MSAT covers the entire country.

Historically, MSS systems have been used primarily for two-way voice services. These services generally support only a few dozen subscribers per channel. The high cost per channel of two-way voice service is recovered by charging high usage fees. Users will pay these fees in areas where service is not otherwise available.

Alternatively, an MSS system can recoup the high cost per channel by providing service to far more mobiles per channel than could be supported by terrestrial repeaters. This can be done by making use of the nationwide coverage of MSAT to provide services that support very large numbers of users on each channel.

Broadcasting exemplifies this second approach. The number of mobiles that can receive broadcasts is limited by the coverage area, which is hundreds of times greater for MSAT than for a terrestrial repeater. Thus each MSAT broadcast channel covers an enormous market area.

Mobile data services also can support very large numbers of users on each channel. A mobile satellite system operator can maximize revenue by integrating broadcasting and data services, such as paging, into low-cost car radios.

Mobile satellite broadcasting through MSAT is particularly attractive not only because of the high addressable market, but because of proven demand. A market survey conducted by Broadcast Investment Analysts (BIA) in conjunction with Strategic Radio Research of Chicago shows strong consumer demand for a 10 channel mobile satellite audio broadcast system.[13]

The RadioSat℠ system is capable of broadcasting CD-quality audio. However, CD-quality requires a very high data rate, which may be prohibitively expensive.[14] Fortunately, the BIA survey shows that "additional programming alternatives and expanded geographic coverage are far more important to consumers than the greater audio bandwidth and the dynamic range of "CD-quality" media."[15] Furthermore, high ambient noise levels in most cars make "CD-quality" superfluous. The RadioSat℠ system can operate with adequate quality at modest data rates using only a fraction of the capacity of a single MSAT satellite, with enormous economic and spectrum savings.

Propagation

Mobile satellite propagation characteristics are very different from those of terrestrial mobile communications systems. When viewed from the contiguous United States (CONUS), MSAT satellites are usually at an elevation angle of 30° or more. This high elevation angle permits line-of-sight communications. However, even minor obstacles such as trees or other foliage cause losses of 10 dB or more. As a result, mobile satellite propagation is dominated by shadowing.[16] Except in urban areas, shadowing losses are generally short term.

There is usually a strong line-of-sight component when the satellite is in view. Severe shadowing generally prevents reception whenever the satellite is not in view. Multipath effects on mobile satellite propagation are therefore minimal. Thus frequency diversity techniques, which counteract multipath fading but not shadowing, are of relatively little value.

Shadowing can be mitigated with time-diversity techniques. Interleaving should be effective for broadcast services, where large processing delays are acceptable.

THE RADIOSAT℠ SYSTEM

The RadioSat℠ system (Figure 3) under development by Radio Satellite Corporation (RSC) uses MSAT to provide integrated mobile communications services through low-cost car radios. It supports alphanumeric and voice paging, two-way voice communications (telephone and dispatch), digital audio and data broadcasting, and precision navigation. The system consists of leased capacity on an MSAT satellite, the AMSC Network Control Center (NCC) responsible for RSC's use of the satellite, a RadioSat℠ Network Center (RSNC) operated by RSC, gateways for two-way telephone conversations, base stations for two-way dispatch communications and for digital audio and high rate data broadcasts, and satellite car radios.

The MSAT Spacecraft

Hughes Aircraft Corporation and Spar Aerospace are jointly constructing three 3-axis stabilized satellites (Figure 3) for AMSC and one for TMI. AMSC satellites will be placed into geosynchronous orbits at 62°, 101° and 139° West longitude; the TMI satellite will be at 106° W longitude. All satellites are designed for 15 year lifetime. The first satellite will be launched in 1994.

MSAT uses K_u band frequencies for feeder links between fixed stations and the satellites. All L band satellite circuits are connected to K_u band feeder link circuits in the satellite. L band-to-L band circuits require two satellite hops via a gateway or base station. There is no satellite path for single hop L band-to-L band circuits.

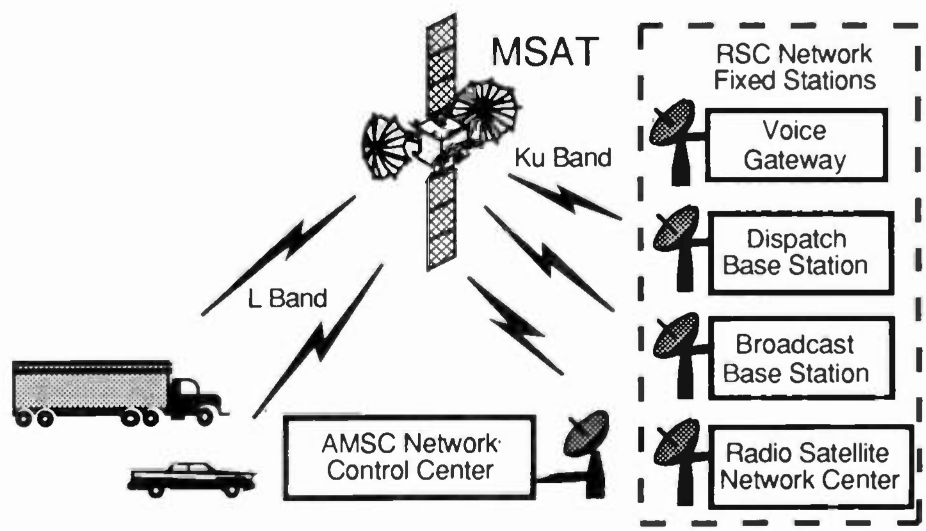

Figure 3. RadioSat℠ System Diagram

L band satellite antennas consist of separate transmit and receive arrays and reflectors. Separate transmit and receive antennas minimize passive intermodulation.[17] The reflectors are 6 x 5 meter elliptical unfurlable meshes.[18] Each antenna feed array produces 6 spot beams covering all of the Contiguous United States (CONUS) and Canada, along with 200 miles offshore. These 6 spot beams also cover Mexico, Hawaii, Alaska, Puerto Rico and the Virgin Islands.

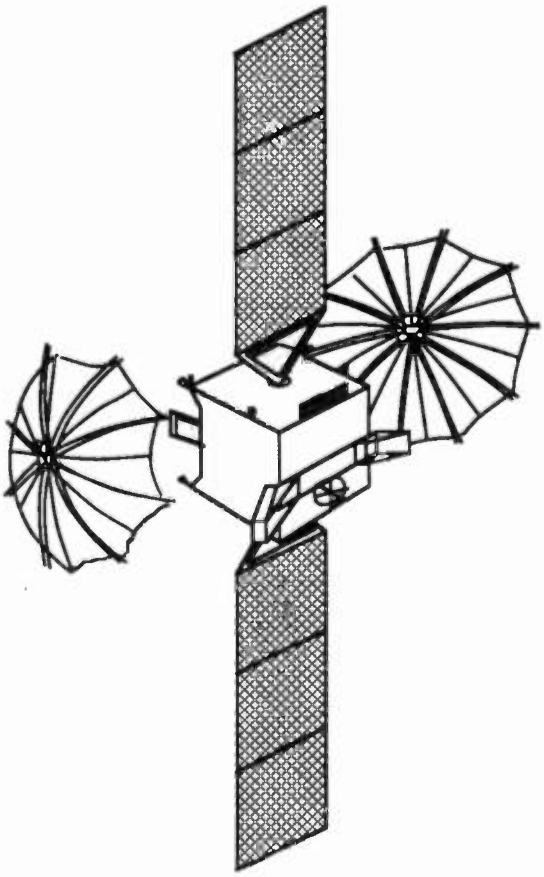

Figure 4. MSAT Spacecraft

Maximum aggregate linearized L band Effective Isotropic Radiated Power of all beams is 57.4 dBW at 16 dB Noise Power Ratio. L band G/T of each beam is 2.7 dB/K at edge-of-coverage in the basic coverage area. This G/T is about 30 times better than other L band satellite systems[19] because of the very large MSAT antennas. The high MSAT G/T makes MSAT an ideal platform for Electronic News Gathering.

Satellite car radios

Each mobile satellite terminal consists of an omnidirectional L band mobile satellite antenna assembly and a satellite car radio (Figure 5). The satellite car radio includes an AM/FM receiver. For optimal performance, separate AM/FM and L band antennas are used.

Satellite car radios simultaneously receive two channels: a Time Division Multiplex (TDM) data channel and an assignable channel. Both TDM and assignable channel transmissions can be broadcast to all mobiles, to groups of mobiles or to individual mobiles. This design gives each radio simultaneous access to all services.

The RSNC transmits a single TDM channel 24 hours a day through each satellite beam. This channel controls all mobile stations and sends low rate (under 2.4 kbps) data broadcasts such as alphanumeric pages, GPS differential corrections and integrity updates, stock updates, sport reports, travel advisories and emergency alerts. Each satellite car radio receives the TDM channel for the beam in which it is located at all times when it is turned on. TDM channels operate with a Forward Error Correction (FEC) code rate of 1/4.

The satellite car radio can receive transmissions from the RSNC, broadcast base stations, telephone gateways and dispatch base stations through the assignable channel. The assignable channel can receive any digital audio or high rate (2.4 kbps or higher) data transmission compatible with the RadioSat℠ network over the assignable channel. It is received simultaneously with the TDM channel. Digital audio, facsimile and high rate data broadcasts, telephone and voice dispatch transmissions and voice pages are received through this channel. It uses an FEC code rate of 1/2. One-way transmissions through the assignable channel are interleaved to mitigate the effects of fades; two-way transmissions are not interleaved to avoid the resultant delay.

TDM and assignable channels operate at variable data rates. TDM data rate varies from 1.2 kbps to 108 kbps; assignable channel from 2.4 kbps to 216 kbps.

Each satellite car radio requires a satellite RF electronics board and a RadioSat℠ Microchip (RSM –

see Figure 5). The RSM incorporates demodulators and decoders for both channels and provides data processing and control functions. The data processing and control portion of the RSM can set up and control two-way data and voice communications. The RSM includes a data coder and modulator for use with an optional data transmitter. An optional audio digitizer, compressor, coder and modulator chip can be added for two-way voice communications.

The RSM is the key to the design of the system. RSC is taking advantage of the dramatic increases in processing capabilities of new semiconductor devices to consolidate most of the processing required by its highly sophisticated radios onto a single chip. This chip can be mass produced for very low cost.

An optional Global Positioning System (GPS) microchip uses GPS broadcasts and navigation information sent through the RadioSat℠ TDM channels to estimate mobile position to within 2 meters.[20] GPS position estimates can be sent to dispatchers over RadioSat℠ data channels.

Performance requirements are modest (Table 2), consistent with mass production and distribution constraints. The mobile antenna can be a microstrip patch.

Transmitter Power[21]	2 watts
Mobile Antenna Gain	3 dBi
Mobile EIRP	3 dBw
Mobile G/T	-20 dB/K

Table 2. Satellite Radio Characteristics

RadioSat℠ Network Center

The RadioSat℠ network center assembles control information and incoming data from numerous sources (including audio channel frequencies, pages, differential GPS corrections, messages to individual mobiles, conversation requests, and channel assignments) into the TDM channel for each beam.[22] The network center encodes and modulates each TDM channel and transmits it to mobiles through the satellite. The network center also receives transmissions from mobiles equipped with data transmitters and forwards received data to appropriate destinations.

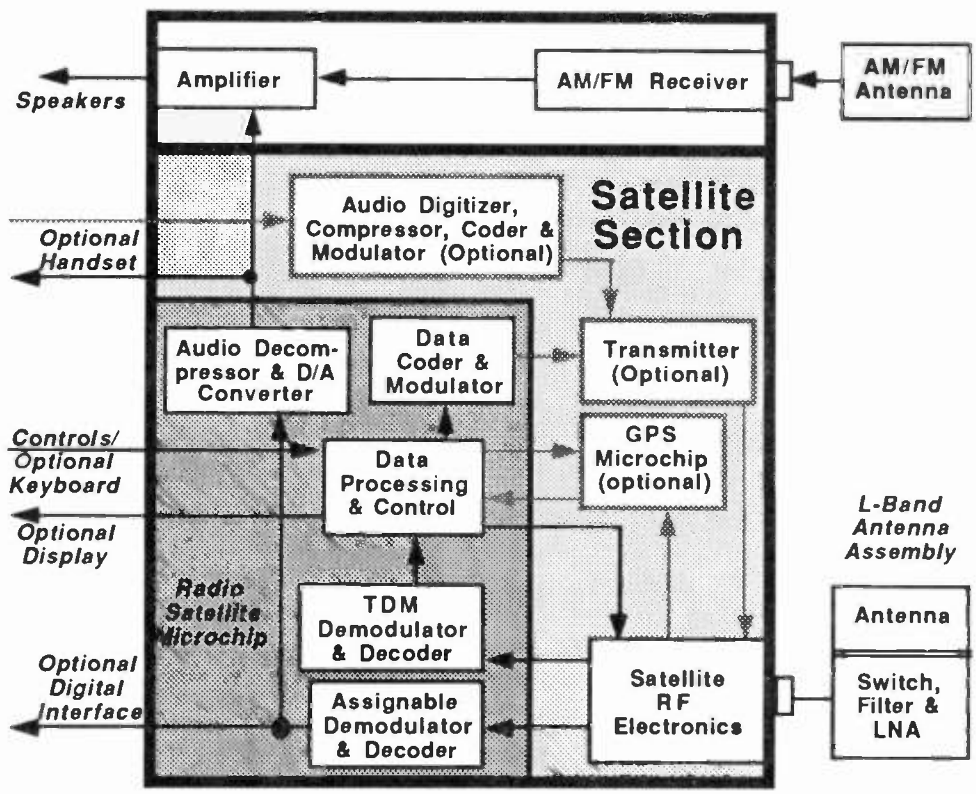

Figure 5. Satellite Car Radio Block Diagram

Broadcast Base Stations

Broadcasters operate their own base stations. Each base station transmits a single digital audio channel. Digital audio broadcast base stations digitize and compress audio signals, then encode and modulate these signals and transmit them at K_u band to an MSAT satellite. The satellite transponds the signals it receives from broadcast base stations to L band and broadcasts them to mobiles.

CONCLUSION

The launch of the first MSAT satellite in 1994 will permit broadcasters to provide nationwide service to cars for the first time. MSAT can also be used to provide many integrated communications services.

The RadioSat℠ system was designed to take advantage of the opportunities opened by MSAT. It will provide nationwide digital audio broadcasts to cars. It will also support two-way voice and data communications and precision navigation services. These services will all be provided through low-cost, highly integrated satellite car radios.

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12 47 CFR §2.1.

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15 Key survey results are disclosed in a letter from Debra Baker of BIA. *See* Attachment 2 of the Comments of Radio Satellite Corporation on the FCC's Notice of Inquiry on Digital Audio Radio, November 13, 1990, Gen. Docket No. 90-357. The telephone survey was conducted September 18-25, 1990. The 1002 respondents, selected randomly, reside in the continental United States. The margin of error of a typical finding in the survey was 2.5% at the 90% confidence level.

16 Goldhirsh, Julius and Wolfhard Vogel, "Propagation Degradation for Mobile Satellite Systems," Johns Hopkins APL Technical Digest, Volume 9, Number 2, 1988.

17 Hoeber, C. F., et. al. "Passive Intermodulation Product Generation in High Power Communications Satellites," AIAA 11th Communications Satellite Conference, Paper 86-0657, San Diego, March 11-20, 1986.

18 These antennas are much larger than those on any previous commercial satellite. However, unfurlable mesh reflectors of similar size are on several NASA Tracking and Data Relay System Satellites now in orbit and on NASA's Galileo spacecraft now on its way to Jupiter.

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SAFETY AND ENVIRONMENTAL CONCERNS

Wednesday, April 17, 1991

Moderator:
Warren Happel, Scripps Howard Broadcasting,
Cincinnati, Ohio

FCC TOWER ENFORCEMENT EFFORTS*
Richard M. Smith
Chief
Field Operations Bureau
Federal Communications Commission
Washington, District of Columbia

PCB CONCERNS FOR BROADCASTERS
Roland K. Kump
GE Company
Schenectady, New York

UTILITY PARALLELING EMERGENCY POWER GENERATOR REDUCES TRANSMITTER OPERATING COSTS
Harvey Arnold and Wayne Estabrooks
University of North Carolina Center for Public Television
Research Triangle Park, North Carolina

DISASTER PREPAREDNESS AND THE 1989 SAN FRANCISCO EARTHQUAKE: THIS IS NO DRILL!
Peter Hammar
Hammar Communications
San Carlos, California

COMPUTER ANALYSIS OF ON-TOWER RFR EXPOSURES*
William F. Hammett, P.E.
Hammett and Edison, Inc.
San Francisco, California

CONSTRUCTION OF A MULTIPLE USER FM FACILITY IN AN URBAN ENVIRONMENT
Larry M. Holtz
KGON-FM
and
Gray Frierson Haertig
Haertig & Associates
Portland, Oregon

*Paper not available at the time of publication.

PCB CONCERNS FOR BROADCASTERS

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ABSTRACT

In today's environment of fast communication, instant data transmission and information overload, it may be easy to overlook the presence of PCB's in the broadcast station.

This paper will review PCB issues which owners and managers of broadcast stations need to address to ensure that they are not only in compliance with regulations but manage to control risks and minimize potential liabilities.

INTRODUCTION

Polychlorinated biphenyls (PCB's) were synthetically manufactured in the laboratory in the 1920's and quickly found use in the manufacture of electrical equipment. Pure PCB has an appearance similar to Karo syrup - clear and thick - is chemically stable, has low electrical conductivity, transfers heat readily and does not burn. These properties made PCB ideal as an insulating fluid in transformers and capacitors.

PCB use increased dramatically in the 1930's and 40's in electrical equipment, heat transfer and hydraulic systems and found use in more unusual applications, such as inks, caulking compounds, carbonless paper and adhesives.

PCB's do not biodegrade quickly and do bioaccumulate in the fatty tissues of animals. These characteristics, combined with increasing environmental awareness in the 1960's and early 1970's and a few well publicized human exposures to PCB's and chlorinated biphenyl byproducts, resulted in Congress passing the Toxic Substances Control Act (TSCA) in 1976.

EPA REGULATIONS

The EPA began in 1978 to issue regulations concerning the manufacture, distribution in commerce, use and disposal of PCB's. Since PCB's were used in transformers, capacitors and may be found in small amounts in other liquid filled devices, the broadcast station owner and operator must exercise prudent PCB risk management. Compliance with EPA regulations is not an option but a requirement in today's environment of increased public scrutiny and enhanced EPA enforcement practices.

Askarel is the generic name for PCB's. In transformers, PCB's will normally be mixed with tri-chlorobenzene (TCB) which is a solvent to improve the fluid's viscosity and meet other design characteristics. The equipment manufacturer then developed his own tradename for this mixture containing a very high concentration (typically 600,000 ppm or more) of PCB's. Tradenames, such as Pyranol or Magvar, were used by GE, Inerteen by Westinghouse, Elemex by MacGraw, etc. A comprehensive listing of askarel tradenames and the associated manufacturer is contained in Table A of the "Broadcaster's Guide to EPA PCB Regulation Compliance" issued by the National Association of Broadcasters.

As the regulatory burden continued to grow with new EPA rule making, such as the "PCB Ban Rule" and the "Electrical Use Rule," the EPA became aware of several fires involving PCB's. The EPA has determined that when PCB and TCB are involved in fire situations where temperatures reach 600°C in the presence of oxygen, byproducts are produced. PCB will produce a byproduct-polychlorinated dibenzofuran (PCDF), while TCB produces polychlorinated dibenzodioxin (PCDD). The EPA has stated that dibenzofuran and dioxin are much more toxic than PCB's.

The EPA then set about writing a new "Transformer Fires Rule" in the interest of protecting firefighters and the general public from exposure to these PCB and TCB fire situation incomplete combustion byproducts.

Since electrical devices occasionally do leak and other devices fail causing leaks, the EPA decided to issue guidelines on how those releases of PCB's into the environment should be controlled and cleaned up. Those guidelines are contained in the "PCB Spill Policy" which was issued in 1987.

Most recently, the EPA has issued a new rule in an attempt to control all PCB's destined for disposal. That rule making was published in December 1989 and is called the "Notification and Manifesting Rule." This rule requires owners of PCB equipment destined for disposal to generate accurate manifests and ensure that the material reaches its intended destination in a timely manner.

The EPA expects that the owner and user of PCB containing equipment will become familiar with the EPA rules and policies and comply with them as they are written and interpreted by the EPA. If those rules and guidelines are not followed, another EPA policy originally promulgated in 1980 and modified and reissued in 1990 titled "The PCB Penalty Policy" will then apply. In addition, the EPA has gone on record to state that besides civil penalties, criminal action may be taken to enhance its enforcement efforts.

Compounding all these problems is the fact that PCB accidents, fires and spills make very good copy for the print media, and occasionally good stories for the broadcast media. A recent example would be the failure and subsequent fire involving a transformer just outside the New York Stock Exchange in December 1990. This fire caused the building to be closed and the New York Stock Exchange shut down for a number of hours on a trading day. Con Edison was called in to determine what the PCB concentration was in the transformer prior to the authorities allowing the building to be reoccupied. It turned out that no PCB's were found. Remember now, the fire was not in the building but in a transformer in a vault outside in the street.

PCB RISK MANAGEMENT

What should you do? If you haven't already done so, you should identify all PCB containing equipment in your facility and develop a good risk management plan. A PCB transformer is one which contains 500 ppm or more of PCB's in the fluid. An askarel (PCB) transformer will generally have a very high concentration well into the hundreds of thousands of ppm of PCB's. A capacitor containing PCB's will be pure PCB, that is, not mixed with any solvents to improve viscosity. Other liquid-filled equipment containing mineral oil as a dielectric fluid may contain small amounts of PCB's from a few ppm to several hundred ppm.

One of the EPA requirements is that there be no exposure risk to food/feed from PCB's. This exposure risk may include cafeterias or large vending areas close enough to your PCB containing equipment to meet the EPA requirement of providing a possible pathway between the PCB's and food. Should you have an exposure risk, the risk or the PCB transformer must be eliminated. The PCB "Electrical Use Rule" also requires detailed recordkeeping of your leak inspection program. If, during the course of the EPA required quarterly inspections, any quantity of PCB is discovered to be running off or about to run off the external surface of a transformer, a cleanup effort must begin within 48 hours of the leak's discovery. The transformer must be promptly repaired or replaced and inspections are required daily as a long as the transformer has an active leak.

PCB transformers and access to the transformers must have an EPA specified large PCB mark applied. The transformer must also be registered with the local primary fire response agency and if used within 30 meters of a commercial building, the building owner must also be notified. Combustible material must not be stored within 5 meters of a transformers or its enclosure, and the National Spill Response Center must be notified immediately in a fire-related situation where the transformer is ruptured and PCB's released.

A PCB transformer may not be untanked for repair and the disposal of liquids, cleanup materials or the transformer itself are strictly regulated by the EPA. The EPA requires that all PCB fluids 50 ppm and above be disposed of by high temperature incineration in an approved TSCA facility or by EPA approved alternate disposal methods. Since any significant release of PCB's or release of more than 1 lb. (reporting threshold) into the environment is considered illegal disposal, leaking equipment or improper disposal due to lack of knowledge to PCB content cannot only be very costly but could be viewed by the EPA as an attempt to evade the law.

PCB capacitors are also regulated more or less, depending on the PCB fluid content or volumetric measurements of the capacitor itself. Large capacitors, those containing 3 pounds or more of PCB fluid or greater than 200 cu. in., if not posing a food and feed risk, may continue to be used in a restricted access installation. The regulation states that the restricted access area must contain any releases of PCB within an indoor location. This may present a problem to some stations which will vent the power equipment room to the outside of the building. In this case, a PCB rupture or fire situation would most probably release PCB's outside the restricted access area.

Small PCB capacitors, those containing less than 3 pounds of PCB's or less than 100 cu. in. or 9 pounds or less and less than 200 cu. in., may also continue to be used and have no disposal restrictions unless they are leaking. Then they must be treated as solid PCB waste for disposal in a secure chemical waste landfill or a TSCA permitted high temperature incinerator.

Mineral oil filled transformers are classified by the EPA as PCB-contaminated unless the fluid is tested and found to have 500 ppm PCB or greater or less than 50 ppm PCB. If less than 50 ppm PCB, it would be classified as non-PCB. This means the transformer would not be regulated but disposal of the fluid from that transformer may be regulated, depending on the disposal method. If your transmitting equipment was built later than 1979, the high probability is that your transmitter will not have any PCB containing equipment. Even in this case, it would still be prudent to inspect for PCB capacitors and PCB's in the transformers. Older equipment should definitely be inspected for PCB transformers and PCB capacitors. If the nameplate of the transformer or capacitor does not clearly state what the dielectric fluid is, the transformer fluid can be tested to determine its PCB content.

The capacitors cannot be tested, and if there is any doubt as to the type of dielectric fluid, the manufacturer should be contacted. The safest course of action for capacitors would be replacement if any doubt exists.

If during the course of these inspections any leaks or stains are discovered, they should be promptly cleaned up using as a guideline the PCB spill policy. Should you have any capacitors in inventory, those capacitors should be inspected to determine if they require a PCB label and whether or not a sufficient quantity of PCB capacitors are in use or in inventory to require annual recordkeeping. One or more PCB transformers will require annual recordkeeping. The rectifier cabinet, exciters/modulator and power amplifier cabinets may all contain large and small capacitors. While it may seem expensive to dispose of some relatively small devices by transporting them in special containers to regulated landfills or incinerators, the cost is insignificant compared to the cost of cleaning up a broadcast station which has become involved in a fire-related incident which develops incomplete combustion byproducts from the PCB's or TCB's.

Should you decide to sell your used PCB transformers and capacitors, certain restrictions would apply. They must have been bought for use rather than resale prior to 1979 and they must be intact and nonleaking. Sale which would require export is not permitted. A PCB risk management plan based on a risk appraisal provided by a competent and PCB knowledgeable vendor is highly recommended. In the unlikely event of an EPA inspection, the new EPA penalty policy may result in significant fines for the most commonly found violations, such as failure to keep adequate records, incorrect recordkeeping, improper or lack of labeling, and active or historic leaks.

CONCLUSION

This paper was not meant to be a comprehensive summary of all the PCB rules and regulations which could possibly apply to owners and users of PCB containing equipment, but only to highlight areas where action should be taken in the interest of developing good PCB risk management practices.

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UTILITY PARALLELING EMERGENCY POWER GENERATOR REDUCES TRANSMITTER OPERATING COSTS

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ABSTRACT- Electric load management using on-site utility controlled power generation equipment is relatively new to television broadcasters. The local electric utility power provider is able to monitor and control peak demand by shedding large loads during periods of high electrical usage. The load of the broadcast transmission facility is managed using an on-site, remotely activated diesel generator and closed-transition utility-paralleling transfer switch. For savings to be realized, the system must be available to the utility for use during any peak period. Load management AC generating systems may offer significant savings and add to reliability of transmission facilities. This paper discusses considerations in design, installation, operational experiences, and maintenance of a load management system now in use at a high power UHF television transmission facility. A description of a load management system is given along with typical projected energy savings.

INTRODUCTION

The use of emergency power generators have been commonplace for most broadcasters to maintain program continuity during times of primary power failures. Upon loss of primary power, the emergency generator would automatically start, disconnect the load from the utility and switch the facility to the emergency generator feed. The system would switch back to the utility after utility power was restored. The generator was used only for emergency purposes and the system was designed so that it would be virtually impossible to have the station's emergency generator connected to the utility under any circumstances. This paper discuses the marriage of two technologies:

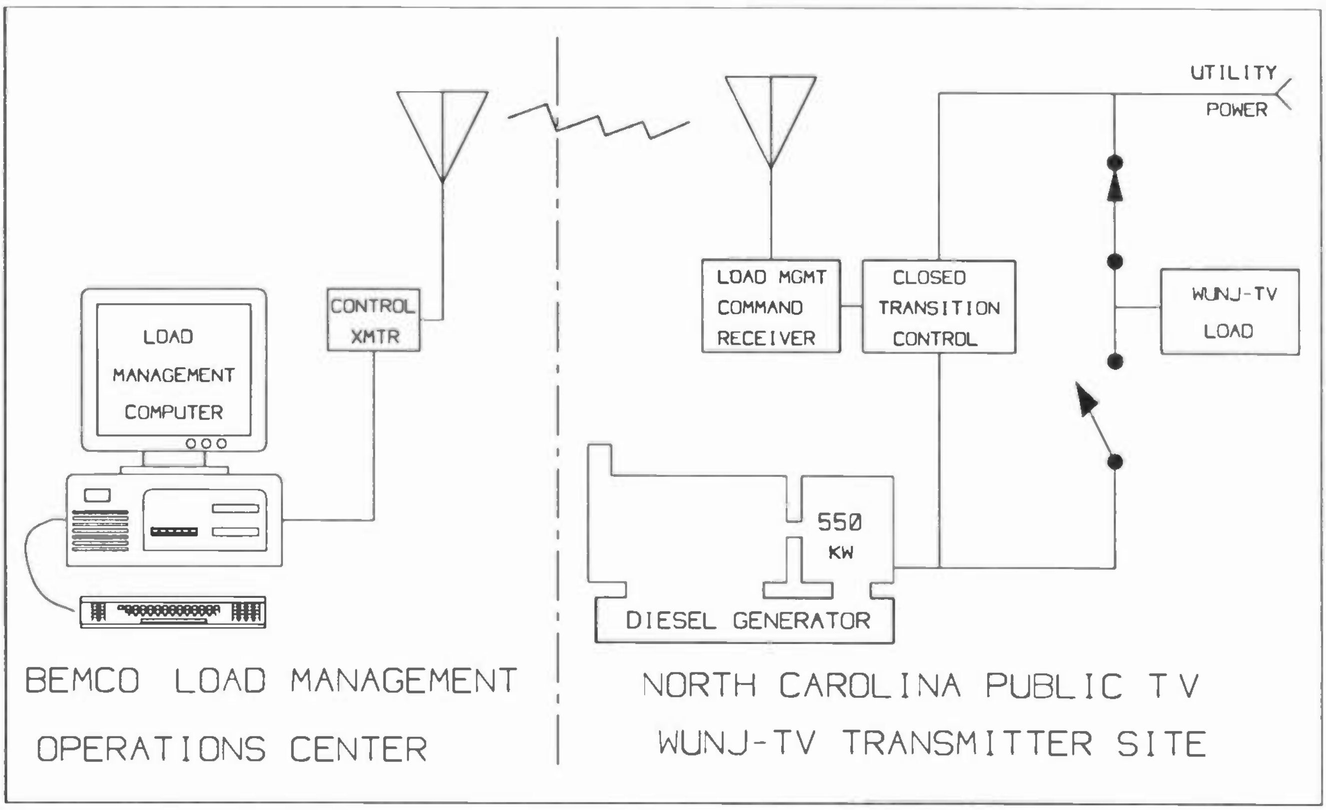
Closed transition switching and electric utility load management programs. Closed transition switching allows a smooth, uninterrupted transition between two power sources. Electric utility load management programs utilize on-site power generation to reduce the peak power demand during a specific time period. Combining these two technologies makes it possible for the electric utility to reduce peak system loads without interrupting the operation of the user.

Electric utilities have been using load management systems to disconnect high current residential loads such as water heaters and air-conditioner systems during peak periods for some time. This process usually involves disconnecting the load for a period of time via a radio controlled signal. Typical load shedding of 1-3 kW is possible for each participating household if the hot-water heater and air-conditioning system are disconnected during times of actual usage. This type of load management system is not suitable for broadcasters since there would be a disruption in service. A closed transition utility paralleling generator would allow the utility to shed the large load without any disturbance to the broadcaster.

The University of North Carolina Center for Public Television (UNC-CPTV) operates 10 full-service television transmitters.

Many of the transmitters are in rural areas served by the Electric Membership Cooperatives (EMC's). The majority of EMC's purchase the electricity they sell. When they exceed their contract demand, they pay a premium for the electricity purchased from their suppliers. These higher costs are passed on to the consumers of this power. It should be noted that many electric utilities are showing interest in point-of-use closed transition load management systems to reduce peak demand.

UNC-CPTV has been interested in load management systems for its high power television transmitters for some time, but was reluctant to commit large capital investments in equipment which required custom design and manufacture. Most electric utilities operating in North Carolina were not aggressively working to have these units in operation with broadcast facilities. It was initially believed that closed transition transfer switches could not guarantee a high level of reliability needed in broadcast operations or deliver smooth consistent transitions between the local diesel generator and the utility feed. Recently UNC-CPTV made the decision to incorporate a 550 kW load management system to support a new high power UHF Television transmission facility for WUNJ-TV Wilmington, North Carolina. The local electric provider, Brunswick Electric Membership Corporation (BEMC0) has assisted in the implementation of this project.



THE CLOSED TRANSITION SWITCHING SYSTEM

The closed transition switching (CTS) system differs from a standard open transition switch (OTS) in two ways. The first difference is that the CTS is designed to briefly parallel the utility and the generator. During this time, the generator is capable of back feeding power to the utility. In fact, large CTS systems are used when generators are paralleled for co-generation of electric power. The second significant difference between OTS and CTS systems is the ability of the CTS system to match the local generators voltage and phase to that of the incoming power. The switching system inhibits paralleling unless the voltage and phase are matched. The switching electronics contain governor and field control circuitry which slowly ramps up the load on the local generator. At this point the system is disconnected from the local utility. The CTS switch allows a smooth transfer back to the utility.

Closed Transition Switching System

COST SAVINGS

Brunswick EMC purchases power for approximately \$2.00 per kilowatt of demand, and approximately \$0.02 per kilowatt hour. The customer is billed approximately \$7.50 per kilowatt demand and \$0.06 per kilowatt hour usage. When the EMC exceeds their contracted demand, their cost per kilowatt of demand goes to approximately \$18.00 per kilowatt while the customer continues to pay only \$7.50 per kilowatt of demand. It is easy to see that if this continued for a long period, it would cause a significant loss to the EMC. This only happens approximately twice per month for a few hours, but can cost the utility thousands of dollars each month.

The value of the load management system is to allow the electrical provider to shed load during peak periods. The utility can save considerably on its power costs by not exceeding contracted demand. Frequently, they will pass the savings on to the customers.

PROJECTED COST SAVINGS FOR WUNJ-TV

TYPICAL MONTHLY RATE SCHEDULES
LARGE POWER SERVICE COMMERCIAL
SERVICE RATE (1)
(Non Load Management)

Basic Charge	\$18.50/month
Energy	\$0.06329 First 400 kWh/kW
	\$0.05029 Remaining kWh
Demand Charge	\$7.50 / kW

COORDINATED DEMAND CONTROL RATE (2)
(Load Management Program)

Basic Charge	$50.00 / month
Energy Charge	$0.0288 / kWh
Coincident Demand	$18.71 / kW
Noncoincident Peak Demand	$3.00 / kW

WUNJ-TV Calculated Monthly Power Bills Typical Monthly Energy Usage of 191,520 kWh Demand of 366 Kilowatts

Total Power Bill (No Load Management)—$14,298.24

Total Power Bill (With Load Management)-$5,789.78

Total Bill (Failed Load Management) $11,539.64

Potential Annual Savings $102,101.52

UTILITY PARTICIPATION

CTS momentarily connects the generator to the utility. Because of this backfeed, approval must be obtained from the utility. A properly installed system will not conflict with utility operation. Many utilities have some type of energy savings program in place, but the degree of cost savings vary. Some electric membership corporations can offer the greatest savings to participants since they usually purchase energy from other utilities. The EMC can save significant cost by closely limiting the monthly peak demand.

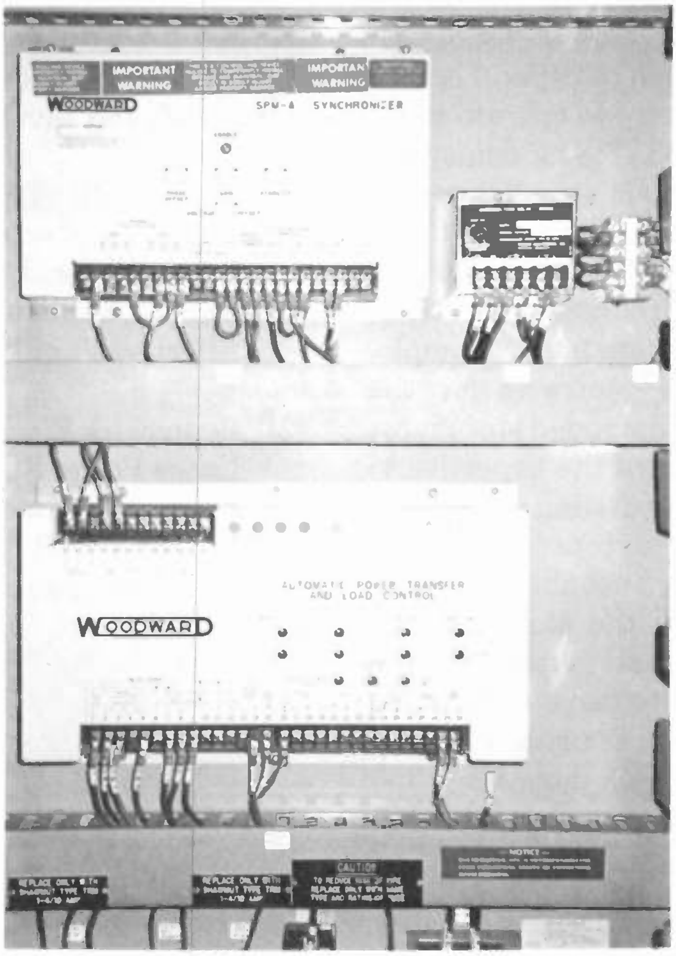
The cost of a closed transition load management system should be studied to determine if it makes financial sense to your organization. A typical closed transition switching system will cost about 25-35 percent more than that of a standard open transition emergency generator set for systems in the 100 kW to 1,000 kW range. Implementation of closed transition transfer switching load management may not be economically feasible if the connected load is less than 100 kW. The 100 kW point is usually the lower limit utilities judge as advantageous to offer significant savings in power costs. Some utilities offer creative finance plans because load management saves them money. Because many utilities can realize significant operational savings with load management they may be able to offer lease purchase options on the generator and switchgear at your facility. The reduction of energy costs may pay for the on-site load management system.

TRANSMISSION FACILITIES BEST SUITED

A properly designed and maintained closed transition transfer system is ideally suited for use at television and radio transmission facilities. Operation of the load management system at WUNJ-TV has been trouble free. The operation or reliability of the WUNJ-TV 140 kW UHF TV transmitter has not been compromised. Digital electronics associated with the transmitter site are not affected during load transfers. Although it is theoretically feasible, the possibility of a less than ideal switch to and from the load management system is of concern. Providing load management to an extremely sensitive operation such as a TV studio or data processing facility may require the use of an uninterruptable power supply system down stream of the CTS.

TESTING AND ANTICIPATED POWER FAILURES

An added advantage of operating an emergency power generator with utility paralleling capability is the ability to switch to the generator without any interruption. Testing of the generator system can be accomplished at any convenient time while operating under actual load conditions. The generator can be started and brought on line when an electrical outage is probable. When serious thunderstorms are anticipated, personnel would signal the system to start and transfer the load to the on site generator without disruption. This may protect the broadcast facility from lightning damage. The electric utility can also help to reduce lost air time by giving advance warning of scheduled power line maintenance.



Synchronizer and Power Transfer Electronics (Located Inside Closed Transition Switch)

PROTECTION FROM POWER LINE SURGES

It is important to protect the electronic circuitry in the closed transition transfer switch from power line surges, since a portion of the control circuitry remains connected to the incoming utility line at all times. Proper grounding and the installation of a fast-acting surge suppressor can reduce the chance of damage or malfunction due to power line surges.

UTILITY LINE REGULATION

Utility line regulation becomes an important factor when operating a load management system. Poor line regulation would not affect the transfer to the generator but can cause problems when transferring the load back to the utility. A disturbance may be noted during this transfer if the utility supply voltage drops significantly as the load is switched back to the utility. When the command is given to transfer the load back to the utility and the voltage is significantly higher due to poor line regulation, it may take time for the control system servos to match the generator voltage to the utility. The generator servos can take 1 minute per volt to match generator voltage to the utility line. The utility voltage will again decrease as the transfer switch returns the station's load back to the utility line. It is desirable that the power company load regulation be 2 percent or less. The electric utility may need to improve its regulation to allow for a smooth switch back to the line.

FUEL COSTS & SYSTEM MAINTENANCE

The electric utility usually operates the load management system approximately 3 hours at a time, three times per month. The cost of diesel fuel can be significant to users of large generator sets. A load management generator system delivering 380 kW will consume 35 gallons per hour. Monthly fuel costs at this rate of operation total $394 if the cost of diesel fuel is $1.25 a gallon. The cost of fuel and maintenance must be factored against the potential savings of energy cost.

The utility must be able to depend on the load management system to operate when it is needed. The period of peak demand is usually during the time of very hot or cold weather conditions. These are the times when equipment is susceptible to failure. For this reason, a comprehensive preventative maintenance program must be implemented. Maintenance contracts for the generator set are recommended since most broadcasters do not have the expertise to maintain the utility paralleling power generator properly. The electric utility does not save anything if they are not able to depend on the ability to shed the station's load when required. Normal generator and switchgear maintenance can be scheduled since the electric utility is usually able to predict the periods of peak demand.

CONCLUSION

The operation of the load management system at the WUNJ-TV transmitter facility has proven to be very reliable. The system has never failed to come on-line when needed. No operational difficulties have been experienced since the system was placed in operation seven months ago. Savings in power costs have significantly offset any costs for fuel or maintenance. Many broadcasters may be able to greatly reduce their operating costs and improve station reliability through the use of this type of load management system.

ACKNOWLEDGEMENT

The authors wish to acknowledge the support of many individuals working for North Carolina Public Television. Additional thanks is given to Brunswick County Electric Membership for their support in working with us to develop a system that performs well and saves money.

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DISASTER PREPAREDNESS AND THE 1989 SAN FRANCISCO EARTHQUAKE: THIS IS NO DRILL!

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Abstract: Much of the U.S. is earthquake-prone. Broadcast studios, transmitter sites, and program production facilities in quake areas are disasters waiting to happen. Managers can take basic precautions that will keep them on the air in the event of a quake. Even though some of these measures are simple, many people don't want to be bothered. However, San Francisco broadcasters know better, now that they have experienced the 1989 quake there. Earthquake lessons also apply to other natural disasters that can put a station off the air.

The Threat: Most broadcasters in the U.S. have never experienced a quake greater than 7.1 or 7.2 on the Richter scale. A real quake, i.e., 8.3 or greater, equal to the temblor that leveled San Francisco in 1906 or the 9.+ one that devastated Alaska in 1964, will be a nasty surprise wherever it happens. And it will happen. Everyone in the central Mississippi Valley and hundreds of miles either side of it is a "sitting duck" waiting for the next time the New Madrid fault snaps in southeastern Missouri. Some say the New Madrid makes the San Andreas fault look like a crack in the sidewalk. Since the New Madrid fault hasn't let go since 1812, however, when almost no one lived around the area that is now Greater St. Louis, few can believe how bad it could be. The Missouri area had *three* monster quakes measuring *more than 8.3* within a three month period during the winter of 1811-1812! All three quakes devastated parts of Missouri, Illinois, Indiana, Arkansas, Tennessee, Kentucky, and Mississippi, with damage as far away as Ohio and the Carolinas. The Mississippi River reportedly ran backwards each time, with massive flooding around what is now St. Louis. The river's course was permanently changed. According to geologists, monster quakes occur in cycles of 100 to 500 years, and can strike almost anywhere at any time. This fact is very difficult to believe for people outside of active quake zones such as California. The 7.1 quake that hit the Bay Area in October was clearly not "The Big One" that will inevitably hit somewhere in the U.S.

Some "tough-guy" general managers do not take basic disaster planning seriously, saying, in effect, "We'll take our chances that nothing will happen in our lifetime" (or at least during the next quarter!). They may reconsider that short-term business decision when they're standing in a pile of trashed equipment in the dark, realizing that basic and sometimes inexpensive methods could have kept them on the air. Every FCC license, in effect, says, "Serve the public", which means, among other things, "In a disaster, stay on the air any way you can" -- regardless of your format. If you're an independent running "Andy Hardy" movies or a public station with travel documentaries, you are providing a necessary service to some segment of the public in times of disaster, including helping calm the public's jangled nerves with entertainment. It is your responsibility to remain on the air. The steps you take will ultimately help your station's revenues and reputation. The basic "hardening" and disaster preparation techniques that get a station ready for an earthquake also let you cope with the aftermath of a tornado, a hurricane, a studio fire, flooding, an industrial explosion, or civil unrest.

Some of the following measures for stations and their personnel seem obvious, "no-brainers", such as keeping battery-operated devices handy and assigning a junior staffer to check once a week or month to make sure the equipment is where it is supposed to be, with fully charged batteries. Amazingly, in the quake of '89, some facilities found themselves without portable radios and TVs, and even without enough flashlights and batteries.

Here are some basic steps to protect your people, your studio, your transmitter site, and your mountain-tops. Make a detailed checklist, and assign a staffer or regular contractor to follow through on all measures. Take these seriously. If you don't, as San Francisco Bay Area broadcasters found out that October day, Mother Nature will see to it that you do, but on her terms!

1. Get the largest generator you can afford -- NOW! Loss of electrical power is the biggest problem in most disasters. Power and power-related problems plagued Bay Area broadcasters during and after the quake. Physical damage at most stations created fewer difficulties. No one likes to spend tens of thousands of dollars on a 150 or 500 kilowatt generator that may never be used. Ninety percent of all power failures last less than thirty minutes. Ninety-nine percent last less than an hour. It's that last one percent that drives CEs and GMs crazy.

2. Reinforce your fuel tanks and build in fuel supply redundancy. Several stations went off the air after the quake because they lost the fuel supply for their generators. Tanks spilled, fuel pumps failed, and fuel lines clogged. Solving potential fuel supply problems ahead of time is tedious but neither difficult nor expensive. Strap old tanks down or install new, purpose-designed, earthquake- and fire-resistant fuel storage systems. Check fuel often for purity. Diesel is prone to condensation, sediment, and algae build-up.

3. Maintain and regularly *stress-check* your generator and its fuel system under full load for at least one day per month. There are a hundred things that can go wrong with a diesel, gasoline, or propane generator that will keep a station off the air. Regular disaster simulation -- testing under stress -- helps uncover many bugs. Understand diesel engines. They are designed for constant full-load operation, not light intermittent duty. A full-load, day-long test will force hidden weaknesses to reveal themselves, such as a head gasket ready to blow, potential fuel pump failures, or oil and coolant problems. Do a twice-yearly, full-building shutdown test by pulling the main breaker for the building, letting the plant go dark, letting the generators come up automatically, and then seeing what's working and what's not, what needs resetting, and what's blown. Have a check list handy that tells what equipment needs to be reset. "Weeding out" marginal equipment is the key to reliability, not only before and during a crisis, but also during regular operation.

4. Arrange *ahead of time* with your local generator service and parts dealer for priority attention during a disaster. Had the disaster been any larger, Bay Area generator dealers would have been overwhelmed. Make sure you have a contractual arrangement -- and a personalized "inside track" -- with a reputable generator service company that will provide you with both instant response and a backup generator should yours fail entirely. Insist on having the home phone numbers of the key people in all of your vendors' organizations.

5. Make sure your backup power system is designed as *simply* as possible. Components in the generator and control systems should be "off the shelf". Parts not normally stocked locally should be replaced immediately, such as odd-voltage generator fuel pumps or unusual switches that have to be back-ordered from a warehouse out of state. Simplify your system and eliminate overly-complicated parts.

6. Rewire studio and transmitter electrical circuits to segregate loads for immediate and selective use of equipment and house power. The wiring design of many stations prevents selective load management during generator use. Segregate loads for easy reassignment via centrally located switch-box controls. Make equipment power as independent as possible from the house wiring. Include the elevators in your backup power plan. Intelligent wiring design makes your all-important "backups within backups" effective. If your built-in studio generator fails and you have to rely on a smaller backup unit brought in from a rental outfit, a simple flip of a few switches can easily route power to just those areas needed to keep a minimum operation going.

7. Improve your UPS systems. Besides helping you with brief, "routine" power outages, a good UPS set-up in a major disaster will keep the studio on the air from a few minutes to several hours while generators are started. A UPS will also protect gear from transients caused when the power company shuts down various grids to protect their equipment or comes back up unevenly.

8. Improve your emergency two-way communications. With phone service out, a station must be able to keep going by deploying its people efficiently. Always have **triple-redundant** backup emergency communications with the transmitter site and with vehicles as they go to and from the station, including ENG vans. Some engineers in the aftermath of the San Francisco quake were forced to rely on irregular telephone service -- including pay phones -- to communicate with people at the transmitter and at the station when their two-way radios and handy-talkies failed. Handy-talkies and mobile radios tuned to two-way frequencies should be placed **throughout** the plant and among selected station personnel **in their homes**. A staff member must be in charge of regularly checking all such equipment and their battery charge.

9. Get a structural inspection of your plant now, not "later"! Specifically with earthquakes in mind, have independent structural consultants with proven track records inspect towers, buildings, foundations, and deep-ground support for structural integrity. It's expensive, but this is a one-time cost that should be done now, not when you're staring at a mass of twisted steel with your station's revenue losses piling up. Learn what Bay Area stations that suffered tower damage know. Of course, some structural failures may be impossible to predict, but we should try.

10. Secure *ALL* equipment racks on the *TOP* as well as the bottom, no matter what their weight, load, or configuration. Many stations in the quake lost some equipment or air time when both light and heavy racks and cabinets fell over or actually completely tore loose from the floor due to lack of top-bracing. All interior structures should be secured by heavy steel bracing across their tops, in addition to bolting equipment racks to the floor. Top support bracing makes the equipment move with the building, not "oscillate" out of phase.

11. Besides dedicated areas at the station, equip an ENG truck as a studio back-up to *BECOME* the station and directly feed the transmitter in case the studio goes down. Most transmitter sites are "harder" and tend to "stay up" longer in a disaster than do studios. Have a contingency plan that provides maximum independence from your traditional methods of operation. A solution to a dark or fire-gutted studio facility may be sitting in your parking lot. If the transmitter is still operating, an ENG truck equipped in advance can act as a mobile studio and master control. The truck can normally function as an ENG unit, but in the event of any disaster that destroys or totally disables the studio, the ENG unit's standing order is to ACT AS THE STATION, with your news or other programming beaming directly to the transmitter. Equip one or two units with all the basic gear necessary to make basic sound and picture and microwave the signal to the transmitter. The process is especially easy for stations already using their transmitter site as a microwave hop.

12. Make sure key equipment is handy and well marked with simple instructions for use. Equipment such as the Emergency Broadcast System monitor, reset and power switches with protective covers, alarms, and two-way communications should be accessible and well marked with big, clear lettering that easily shows up with light from flashlights. Write out simple, basic operating procedures for all important equipment and keep these instructions nearby.

13. Keep battery-operated equipment handy, regularly checked out by an assigned staffer. Everyone should have simple battery-powered, consumer radios and TVs on site to check on themselves and the competition. Portable radios, TVs, flashlights, and electric lanterns, along with the dry cell batteries to run them, are always at a premium in an emergency. Most Bay Area stations had no battery-operated TVs on site and had no idea if their remote transmitters were on the air.

14. Don't forget the creature comforts! Be prepared for personnel to remain at the station for several days, since a disaster could make movement into and out of the city impossible. Emergency supplies, such as drinking and wash water, non-perishable food, eating utensils, buckets, and toilet paper are essential for people both in the studio and at the transmitter site. Along with first-aid kits, supplies should be well-stocked, clearly marked, and checked once a month. Designate someone such as an administrative assistant, a secretary, or an outside contractor to do this important work on a monthly basis.

15. Be sure at least two engineers can reach the transmitter and the studio at all times. People in markets with natural obstacles are dependent on bridges and tunnels to move around. In the San Francisco area, they have the Bay as well as the Marin and East Bay hills to contend with. In a disaster, engineering staff who live on the same side of a bay or hill as the transmitter and studio sites should go there automatically as soon as possible. Staff should assume they are needed, since most telephone and radio communications will probably be knocked out, as they were in parts of the Bay Area just after the quake. Everyone at the station should have PRE-ASSIGNED TASKS based on "if...then" scenarios worked out in advance. In large facilities, management should work out "game plans" to determine the off-duty people who are "on call" for disaster response at any given time. The plans should naturally supercede "normal" plans for small-scale emergencies.

16. Work out your emergency transportation arrangements now, not during a crisis. Off-road vehicles and helicopters are an absolute necessity in getting equipment in and around a disaster area, both for news crews and for engineers maintaining equipment. Vehicles guaranteed at your disposal can make the difference between staying on the air to serve the public or staying dark. Special transportation is essential to get your people to the transmitter site, to reach emergency parts and equipment sources, and to get to the hospital in case of injury. You may think that, since your studios and transmitters are both located downtown, you won't need a four-wheel drive vehicle to get around in a disaster. Imagine the condition of the streets of Chicago after a tornado has ripped through the downtown area. A four-wheel-drive truck may be the only way to get past huge pieces of debris in the street to get that vital transmitter part to the site. If the station doesn't own such a vehicle, make sure a staffer who does can be reached at all times to help out on a contractual basis.

17. Practice makes perfect. In technical procedures, drill not only your engineering and maintenance staff but also on-air "combo" talent and even sales people and secretaries. Anyone at the station could end up running the board if your regular staff is injured. You will need all the help you can get. Drill everyone on simple chores such as turning off and resetting sensitive studio and office equipment before power is restored. Regularly distribute, update, and interpret procedures in staff meetings for ALL personnel.

18. You're on your own. One of the greatest lessons of San Francisco was: "Except for the charity of your fellow broadcasters and a few citizens who may be hit as badly as you are, you're on your own." You can't think of every possible contingency for future emergencies, but the big '89 San Francisco earthquake shows the need to try. Common sense -- and a sense of urgency -- will pay you and your station big dividends when the next big disaster strikes -- AND IT WILL!

CONSTRUCTION OF A MULTIPLE USER FM FACILITY IN AN URBAN ENVIRONMENT

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Portland, Oregon

Abstract
The broadcaster, in attempting to construct a large scale project in an urban environment, will face many hurdles. The successful completion of the project depends on how successfully he surmounts these hurdles. Many worthwhile projects have failed because the engineers in charge did not appreciate the nature of the problems that they had to solve and the nature of the resources needed to solve them. It took not only innovative technical solutions but a change in our attitude as engineers and a commitment to citizen and civic involvement in order to successfully complete the new 603' KGON tower in Portland, Oregon.

HOW WE GOT TO BUILD IT

For many years KGON had been dissatisfied with its FM transmitting facilities in the Healy Heights area of southwest Portland, Oregon. It leased space at the 150' level on another FM broadcaster's tower along with four other tenant FM stations. This 260' tower is located among several others in the middle of an upper-income residential area. In the immediate vicinity were 8 FM's, 1 AM, 1 TV, hundreds of land mobile users, and telephone, electric and natural gas utility communications facilities. (Subsequently 2 FM's, the AM, and the TV left the area.) KGON was affected by Docket 80-90. In order to preserve its class C status, KGON needed to raise its antenna at least 62 feet, which was not possible on this tower.

KGON and NIER

In 1985 the FCC adopted a standard for non-ionizing electromagnetic radiation (NIER) based on the standard issued by The American National Standards Institute as set forth in C 95.1 - 1982, "American National Standard Safety Levels with Respect to Human Exposure, 300 kHz to 100 gHz.". We suspected that radiation from all the users on this tower would combine to exceed 1000 μw/cm^2 at 100 mHz, the upper limit set in the standard. The City of Portland was considering adopting an even more stringent standard of 200 μw/cm^2 at 100 mHz. We knew from both broad band and narrow band measurements made by the EPA in the summer of 1986, that not only did the aggregate radiation from the tower exceed the City of Portland proposed standard, but, that in some places, KGON's contribution exceeded the standard.

KGON had the option of opposing adoption of the new city standard as several broadcasters did, but decided that the prudent course was to support the proposed standard and seek a technical solution to comply with it. We decided on this course of action for several reasons. First, we thought that the adoption of a more stringent standard was inevitable. Multnomah County, the county surrounding Portland, had adopted a 200 μw/cm^2 standard several years earlier for facilities outside of the city, and there was a strong lobby among citizens groups in Portland for an even more stringent standard of 100 μw/cm^2. Second, we knew that lower ground level radiation would not only reduce nuisance complaints from neighbors about interference with electronic appliances, but would reduce the amount of interference to our own equipment. And finally, it was good public relations. By supporting a lower NIER standard, KGON could demonstrate to the neighbors and to the city its commitment to health, safety, and neighborhood livability.

Our goals were clear; improve our signal; reduce NIER; preserve our Class C status under Docket 80-90; improve our transmission reliability; and acquire control of our facility. KGON purchased a 280' tower currently in use by another FM broadcaster. It was located 350' south of KGON's current tower. We purchased this tower with a view to strengthening it to meet current building codes, and installing a new FM antenna to be shared by KGON and the former tower owner.

The Neighbors and the City

Many obstacles faced us before construction could commence. The greatest of these was neighborhood relations. There has been a long history of animosity between tower owners and users and the neighborhoods in which they are located. The neighbors refer to the tower farms as electronic jungles, and view their presence with great trepidation, the spate of publicity concerning the health effects of NIER doing nothing to assuage their fears for their own health. They have been plagued by malfunctioning VCR's, TV's, radios, computers, telephone answering machines, garage door openers, and the occasional dental filling. They hear rock and roll on their toasters. After years of complaints and questions, neither the tower owners and users nor the city had given them any satisfaction.

In frustration they turned to the neighborhood associations for organization and to the land use process for clout in dealing with what they viewed as rampant, unregulated growth in the number of towers and the number of tower users. The Healy Heights Neighborhood Association (HHNA) represents neighbors in the immediate Healy Heights area. HHNA and several other neighborhood associations banded together to form a larger group, the South West Hills Residential League (SWHRL). Since several neighborhoods were dealing with communications tower issues, a special committee devoted entirely to the question of towers was formed within SWHRL. The neighbors and their associations became a well organized and vocal opposition during Planning Commission and City Council meetings These groups vigorously lobbied City Council members to do something about towers and tower users.

In return, tower owners viewed land use decisions as often arbitrary. As more broadcasters were upgrading their facilities and new spectrum was being developed in the Land Mobile service, tower users wanted some assurance that the city would support their needs and that they would be governed by a known set of regulations. Increasingly the city saw the issue of communications towers as a headache that wouldn't go away. In response to this, the City of Portland placed a moratorium on the construction of all communications towers, during which time it would attempt for the first time to codify the land use requirements affecting communication towers.

In order to build a communications tower in the City of Portland, the property owner must first obtain a Conditional Use permit (CU) from the Planning Bureau. After public notification, a hearing is held in which testimony from the applicant, the public and the planning bureau staff is heard. A hearings officer then decides whether the application will be granted. If granted, the applicant's use of the property will be subject to a series of conditions imposed as part of the CU. Decisions of the hearings officer may be appealed to the City Council. The comments of the affected neighborhood association are particularly important as is the recommendation of the planning staff.

Therefore, the first major hurdle KGON needed to overcome, was garnering the support of a very hostile neighborhood and receiving a positive recommendation from their neighborhood association. Of utmost importance was initiating a line of credible communication in the neighborhood and overcoming 20 years of bad feelings and distrust.

The second major hurdle was convincing the Planning Bureau and the City Council that our project was a solution to their planning problem.

Engineers With an Attitude

The success of a project is more often determined by the attitude of those undertaking it than by technical elegance, and it would be our attitude that would determine the success of this project. It was apparent to us that the situation was delicate and that it would be easy for us to destroy any chances for success in this project by insulting the already alienated neighbors. We decided to examine the attitudes that had engendered alienation between the tower operators and the neighborhood. All too often we have watched ourselves sabotage our own best efforts because of an inappropriate attitude towards others or an unwillingness to deal with people in an honest and adult fashion. In short, we decided it was more important to be successful in our endeavor than to be right.

The attitudes we identified were:

- Arrogance and misplaced feelings of superiority.
- A belief that we are uniquely capable making certain decisions for others because we possess specialized knowledge .
- A belief that our rights outweigh the rights of others.
- Closed mindedness.

These attitudes betray themselves in the following behaviors:

- Trivializing and belittling other people's problems.
- Defensiveness and unwillingness to accept criticism.
- Unwillingness to accept responsibility and take appropriate action.
- Unwillingness to admit lack of knowledge, and the need to be right, always!

These behaviors:

- Cause others to react defensively, destroying communication.
- Cause situations to become polarized.
- Make people feel like their beliefs and ideas are unimportant and that they are unworthy.

Make the engineer unable to see the actual problems and solutions.

And, worst of all, destroy the engineer's credibility.

One Bad Example...

The negative effects of poor attitude were clearly revealed to us by the activities of another Healy Heights tower owner who provides space for a large number of land mobile tenants. His tower is 140' tall and is within 50' of the nearest residence. Like many towers in the Land Mobile Service, his tower is covered by a tremendous number of antennas and is somewhat unsightly. That, and its proximity to residences, causes some grumbling amongst the nearer neighbors for aesthetic reasons.

As this owner's business matured, he added many more users and consequently started receiving complaints of interference to TV reception and to other electronic appliances. While the high powered FM broadcasters in the neighborhood were responsible for a large share of the overall interference problem, the operation of several high powered low band paging systems at this site caused particular and easily identifiable interference to neighborhood televisions. The operator chose to ignore the complaints for many years, claiming in the face of clear evidence that other operators were responsible for the interference. He made little effort to aid neighbors in correcting any interference problems and generally acted as if he had the right to act in any fashion he so desired.

When this operator's tower approached maximum capacity, he decided to replace it with a 300' - 400' self supporting structure. He drew up plans for a new facility without seeking any neighborhood involvement. Though it was plain that he needed expansion space, he later represented to the neighborhood association and the city that his motive for seeking a larger tower was only to address the aesthetic and interference concerns of the neighborhood. He also misrepresented the degree to which he had sought out neighborhood input to his plan. The association came out strongly against the plan for a number of reasons, many of which might well have been resolved had he sought their advice instead of presenting his plan as a *fait accompli*. Only after submitting his plan to the city did he make an attempt to correct a few interference problems by purchasing better television receivers for some of the neighbors. The neighborhood viewed this action as cynical and were not swayed. The city hearings officer denied his CU application. After the hearing it came to light that one of the reasons he needed additional tower space was that he expected to receive a construction permit for a low power television station. He well knew that the neighborhood was strongly opposed to any additional broadcast stations in the area. Shortly after the hearing he distributed a letter amongst the neighbors which impugned their integrity and accused them of not playing fairly.

By apparent misrepresentations, he destroyed his credibility. He acted arrogantly to the neighbors by telling them what the solutions to their problems were and by stating that he did not feel the neighbors had a right to exercise any control over his use of his property. He trivialized and belittled their concerns by ignoring interference. And he insulted them by impugning their integrity. His appeal was roundly rejected by the City Council. He was so determined to be right that he still has 140' tower with no room.

And Another

It may be instructive to look at a situation in which the broadcast industry has been a victim of these same attitudes and behaviors. A case in point is the recent struggle between broadcasters and the FAA over receiver induced third order intermodulation interference in some avionics receivers. The FAA has unilaterally decided that, as a part of its mission to assure aviation safety, it has a right to regulate not only physical intrusion into the airspace but electromagnetic intrusion as well. The FAA considers the power, location, and height of a proposed FM broadcast facility and relates it to that of other FM broadcast facilities in the area and makes a decision as to whether the proposed facility is likely to induce third order intermodulation interference in any avionics receiver. If the FAA decides that interference may be induced, it then refuses to allow the proposed facility to be built. The criteria by which these decisions are made have been developed as a matter of policy rather than law, without benefit of public testimony. The FAA has been hesitant about revealing these criteria. Though the broadcast industry has repeatedly attempted to negotiate, the FAA feels that there is nothing to negotiate, since they contend that they are the sole arbiters in this matter.

The FAA has acted arrogantly by refusing to reveal its criteria. It believes that it alone is qualified to make decisions regarding the effect of receiver induced third order intermodulation interference in avionics receivers. It has acted in a closed minded fashion by not seeking public input and alternative solutions to the problem. By claiming that its problems are of higher priority, it trivializes and belittles broadcasters' problems

Broadcasters are frustrated at having worthwhile projects stymied by the FAA without recourse. They are angered by the FAA's unwillingness to provide enough technical information for engineers to design installations that will meet FAA requirements prior to submission. They are left to presume, either rightly or wrongly, that the FAA's criteria are so unsubstantiated that they cannot withstand scrutiny by others. This undermines the credibility of the FAA engineers in the eyes of the public. Broadcasters are further upset by the FAA's unwillingness to consider alternative solutions to the problem, such as receiver quality requirements. But mostly they are incensed at being controlled by an organization that refuses to let them participate in the regulatory process. The FAA's intransigence in this matter may well lead to deciding this issue in congress or in the courts, a decision that is apt to be disliked by all parties and viewed as a failure on the part of the FAA to properly fulfill its mission.

Learning from the Mistakes

Amidst an abundance of bad examples, both personal and general, we elected to learn from these mistakes and approach our project with a set of attitudes that we felt would assure our success. These attitudes are:

- A recognition that everyone has a right to his own beliefs.
- Others' rights are no less important than ours.
- An assumption that others' actions and intentions are honest.
- A recognition that all interested parties can make a positive contribution.
- A belief that, given all the information, people are capable of making a reasonable decision.

These attitudes translate into a set of behaviors. They are a commitment to:

- Be candid in all our dealings.
- Be solicitous and receptive to others' ideas,
- Seriously consider and not trivialize people's concerns,
- Be inclusive in problem solving,
- Work towards solutions that satisfy all parties,
- Recognize that KGON has the responsibility to correct the problems it creates.

Putting the Team Together

An understanding of these attitudes and behaviors evolved into a commitment on KGON's part to a collegial process amongst ourselves and to community involvement as corporate policy. Dan Hern, vice-president and station manager, and Andy Sutcliffe, our corporate public relations director, joined the project. For legal counsel they located and retained Christopher Thomas, a former Portland city attorney. Mr. Thomas is an effective mediator, with much experience in city and neighborhood politics. His biggest asset is his commitment to negotiated solutions to neighborhood problems.

Because KGON's parent company has no corporate engineering department, full responsibility for seeing the project through to completion was handled in house. We began with organizing and delegating responsibilities. The KGON technical staff consisted of Larry Holtz, chief engineer and Clinton Locey, assistant chief engineer. It was decided that Gray Frierson Haertig, a broadcast consultant who had assisted KGON with some projects in the past, would be hired for the duration of the project, to help with the increased work load.

Larry Holtz, acted as the project director. His primary responsibilities were:

- Budget preparation and management
- Corporate communications,
- Contract supervision
- Pre-construction public relations
- General project oversight
- General project design

Gray Haertig, acted as the project manager/engineer. His primary responsibilities were:

- Day-to-day construction liaison and coordination between the technical staff, the various contractors, and outside consultants;
- On-site supervision of all construction
- General project engineering and design
- Tenant coordination and liaison
- Construction phase public relations

Clinton Locey's primary responsibilities were:

- FM antenna research
- Ground level RF radiation predictions
- FM combining system research
- Design and installation of the computer based transmitter site control system
- Maintenance of the studio and existing transmitter facilities during the tower construction

Although each engineer had primary responsibilities, most designs were a product of everyone's efforts. Final decisions were made by consensus.

Working with Our Neighbors

Even though it appeared that the tower ordinance then under draft at the city might allow us to move to our newly acquired 280' tower without seeking neighborhood approval, to do so would not honor our commitment to citizen participation.

To this end, we contacted the Board of Directors of the Southwest Hills Residential League, and they requested that we work with their tower subcommittee. This subcommittee was comprised of doctors, scientists, and other professionals and non-professionals who lived in the neighborhood.

After explaining to the neighborhood group of our need for the increase in antenna height; the inability to do so at our current location; and our lack of success in finding a new site; we revealed our recent acquisition of the 280' tower and our tentative plans for it. We asked for their comments, and the initial response of the committee was quite negative. Their years of frustration had left them feeling that the only agreeable solution was to have all the communications towers removed from the hill. We indicated that, while we were unwilling to cease operation, as new property owners and neighbors we would strive to establish a working relationship with them and explore the possibility of reaching a solution agreeable to all. We invited them to air their concerns and grievances. Even though they were wary at first, as it became evident that we were willing to listen and weren't categorically excluding any of their ideas from consideration, they warmed to the process.

Their initial concerns centered around three main issues:

Health effects of living in high RF fields
Nuisance interference to electronic appliances
Unlimited growth in the number of towers and the inability to monitor the number and kind of users

Their first concern was of paramount importance. They were aware that the EPA measurements had shown that the ANSI standard for NIER was being exceeded at certain locations in the neighborhood. Knowing that Multnomah County had legislated a standard that was 1/5 of the ANSI standard, they were unwilling to accept radiation levels above this. There was sentiment for an even lower level in the neighborhood.

Their second concern was a problem they lived with day in and day out. They considered interference a major infringement of their right to enjoy the same pleasures that other citizens of the city enjoy.

Their third concern was long term. Those living in the neighborhood for many years felt that there had been a continuing aesthetic deterioration in their environment. They wanted to see this trend checked and perhaps reversed. Never knowing who their electronic neighbors were denied them the remedies, such as personal negotiation, that most citizens have available to resolve problems.

We recognized that the solution to the first two problems was essentially the same; a reduction in ground level RF radiation.

By moving to the 280' tower and using an antenna especially designed to minimize downward radiation, we knew that we could significantly lower our contribution to the overall RF level in the neighborhood. In fact, we were confident that the combined radiation from KGON and the former tower owner's station could be brought below 200 $\mu w/cm^2$. Four stations would remain on the tower that we were vacating. Because of their contribution, overall radiation in the neighborhood would stay very high. The reduction in radiation would probably not be enough for the neighbors to see any great change in nuisance interference.

Moving to this tower would not address their third concern at all. The number and kind of towers would remain the same, and additional tower users in the neighborhood would have few options for space other than new construction.

At this juncture, KGON could have decided that it had fulfilled its commitment to neighborhood participation, and, that as far as KGON's operation was involved, it had addressed the concerns of the neighbors. With a clear conscience we could have proceeded with our plans for the 280' tower. We elected, however, to continue meeting with the neighborhood committee. Because we were now long term neighbors, we were afraid that if we proceeded without actually solving some of the problems, our actions would be viewed as cynical, and our relations with the neighbors would sink back down into the mire.

The most direct way to accommodate the neighbors desires for fewer towers in the neighborhood was to consolidate as many users onto one tower as possible. To that end, we investigated relocating all six of the FM broadcasters onto the 280' tower using either a master antenna system or individual antennas. We also retained a structural engineering firm to evaluate the 280' tower's ability to support additional antennas.

We reported back to the committee that the structural engineer had determined that, even with elaborate reenforcement, the 280' tower was unable to support additional antennas. We also reported that it was impossible to combine all six FM stations into one antenna on a 280' tower and still meet the proposed city NIER standards. We suggested that all of their primary goals might be achieved by constructing a taller and stronger tower with a master antenna for all the FM stations. They requested that we look into this possibility and report back our findings.

First we decided what amount of ground level radiation would be acceptable. Even though the proposed city ordinance contemplated an NIER standard of 200 $\mu w/cm^2$, we knew that there was public sentiment for 100 $\mu w/cm^2$, a standard that might well be adopted in the ordinance. The proposed standard would specify a cumulative maximum regardless of source and, responsibility for correcting problems would likely be divided among property owners, rather than NIER generators. Because we could not assure that we would be the only high level NIER source, we decided the prudent course was to design assuming that the total NIER from our tower should not exceed one half of the standard. This led to a very conservative 50 $\mu w/cm^2$. We assumed that all 6 FM stations might eventually be part of the master antenna system, bringing the total combined horizontal and vertical ERP to 1050 KW.

We then retained Benjamin Dawson of Hatfield and Dawson, Consulting Engineers, to assist us in analyzing a series of antenna designs and tower heights in order to determine their effect on ground level NIER. We established that a minimum tower height of 600' was required for the antennas that might be suitable for multiplex operation. We finally decided on a circularly-polarized, 3 element Jampro spiral antenna. Although this antenna had never been used for FM broadcasting, it was well proven in both UHF and VHF television service. It possesses the 10 mHz bandwidth and power handling that we needed to accommodate all of the FM broadcasters on Healy Heights. It is also distinguished by an exceptional inherent omnidirectionality and uniform axial ratio. We decided to use minimum antenna gain consistent with producing 100 KW ERP using a 35 KW transmitter output power. The antenna was designed with 1/2° beam tilt, 20% first null fill, and 5% second null fill.

Though we had no assurances that other broadcasters might be interested in moving, we knew that three were struggling with Docket 80-90 concerns of their own. We believed the superior antenna performance we had to offer and the 400' increase in HAAT would be compelling inducements to move.

Yes, We Want a Big Tower, But...

We reported our findings to the committee. They were intrigued with the idea of one big tower replacing most of the others, but were understandably concerned about the aesthetic impact of a 600' tower in their neighborhood. We offered to further research tower types. They asked us to provide them with specific information regarding NIER.

Because the proposed city tower ordinance discouraged the use of guyed structures, we gathered pictures and information on a variety of existing designs for self-supporting, tapering "Eiffel Tower" type structures. We believed that these would meet with the greatest acceptance by the committee.

We created a set of NIER prediction charts for use by the neighbors. By entering the distance of his house from the tower and its elevation above sea level, a homeowner could read the theoretical NIER from the chart. This chart was created by first establishing an overall vertical radiation pattern for the antenna by averaging the predicted vertical radiation factors at each depression angle for all six of the frequencies. Actual power densities were predicted using a free space calculation based on the slant range distance from the antenna radiation center and the combined horizontal and vertical ERP multiplied by the antenna vertical radiation pattern power factor at the appropriate depression angle. The calculated power density was then augmented by a factor of 2.56 to account for reinforcement by in-phase ground reflections in some locations. This is a methodology suggested by EPA research [1].

This information was presented to the committee at the next meeting. They were encouraged by the material and felt that perhaps a resolution to the years of neighborhood conflict were at hand. They urged us to prepare a final design that could be presented to the Board of Directors of SWHRL. Throughout our discussions, both SWHRL and KGON kept the Portland city planners appraised of our progress. As bureaucrats anxious to keep all parties happy, the planners encouraged the work of resolving issues among broadcasters and residents.

Old Problems Solved, New Questions Raised

There were, however, a number of concerns the neighborhood committee wished us to bear in mind as we prepared our proposal:

First, because there wasn't any technical way for us to assure that our design would eliminate nuisance interference in the neighborhood, they wanted a plan to correct problems that might arise.

Second, they still had troubling aesthetic concerns about the presence of a 600' tower in the neighborhood. Tower color and obstruction lighting were also important.

Third, they stressed the primary importance of consolidation. The tower must be able to support the necessary antennas. And, they wanted proof of our efforts to get other tower users on the hill to move to our facility.

Fourth, every effort must be made to reduce wind noise from the tower and equipment noise from the transmitter building.

Fifth, there were safety and liability concerns about ice falling from the tower after ice storms.

Sixth, they wanted us to recognize that we intended to operate what is essentially an industrial activity in a residential neighborhood. They had concerns about landscaping, traffic, and the general appearance of the property.

Seventh, they wanted us to mitigate as much as possible the noise, dirt, and traffic associated with construction.

And Eighth, they knew that radio towers are frequently attractive nuisances, beckoning gawkers, lovers, graffiti artists, teen age drinkers, and fools seeking a thrill by climbing the tallest thing around.

Zeroing in on the Design

We were particularly impressed with the tower designs of Skilling Ward Magnusson Barkshire, Inc., a structural engineering firm in Seattle, Washington. We contacted SWMB to prepare an initial design. Our goal was to have a design complete enough to produce an artist's rendering of the tower for presentation to the neighborhood group and for inclusion in our application for a conditional use permit from the city.

We worked with Mr. Ramon Upsahl and developed the design. Termed a "verandeel space frame", the structure consists of three tubular steel legs gracefully tapering from a 5' diameter at the base to a 1.5' diameter at the 505' level. These legs are joined by horizontal tubular members every 75'. The legs and horizontal members are filled with ultra-high strength concrete. It is essentially a concrete structure with external reinforcing. The structure contains no diagonal members, producing a very open and airy appearance. The legs are on 75' centers at the base, tapering to 20' centers at the 505' level. All structural joints below 300' are welded and not bolted to minimize visual impact. The tower is surmounted by a 98' tapering pole around which the Jampro spiral antenna is wound.

The tower is designed for exceptional strength. In addition to meeting the extremely stringent requirements of EIA standard RS-222D, we required that the 28,000 pound antenna and pole not rotate more than 1/2° in a 55 mph wind. The structure is designed such that all of the available antenna mounting space above 300' can be fully loaded with typical microwave dishes and land mobile antennas without exceeding its safe wind loading. For a thorough description of the structural design criteria, see *New Tower Construction Techniques*, Ramon D. Upsahl, P.E. [2].

SWMB commissioned Dr. Clifford F. Mass, a professor of meteorology at the University of Washington, to study tower icing. Using geographical information and thirty years of meteorological data, he predicted maximum probable ice accumulations. By correlating this with wind speed and direction during the post-icing thaw, he was able to predict probable ice fall patterns.

Using computer modeled wind tunnel data, SWMB determined the design wind velocity. This data was also used to minimize wind noise.

With this tower as a centerpiece, we proceeded to design a facility that could accommodate not only all 6 FM broadcasters in the Healy Heights area, but all of the land mobile and point-to-point microwave operators, as well.

We then prepared an application to the city for a conditional use permit to build this facility. This application contained a number of self-imposed conditions that reflected many of the concerns raised by the neighborhood committee. We promised to:

Survey the residences within a 1000' radius of the tower before construction commenced to determine how many were receiving interference and of what kind. We additionally promised to do a follow-up survey after the tower was fully operational and to correct any problems that stemmed from our operation.

Limit the number of high powered broadcasters to those then operating in the neighborhood.

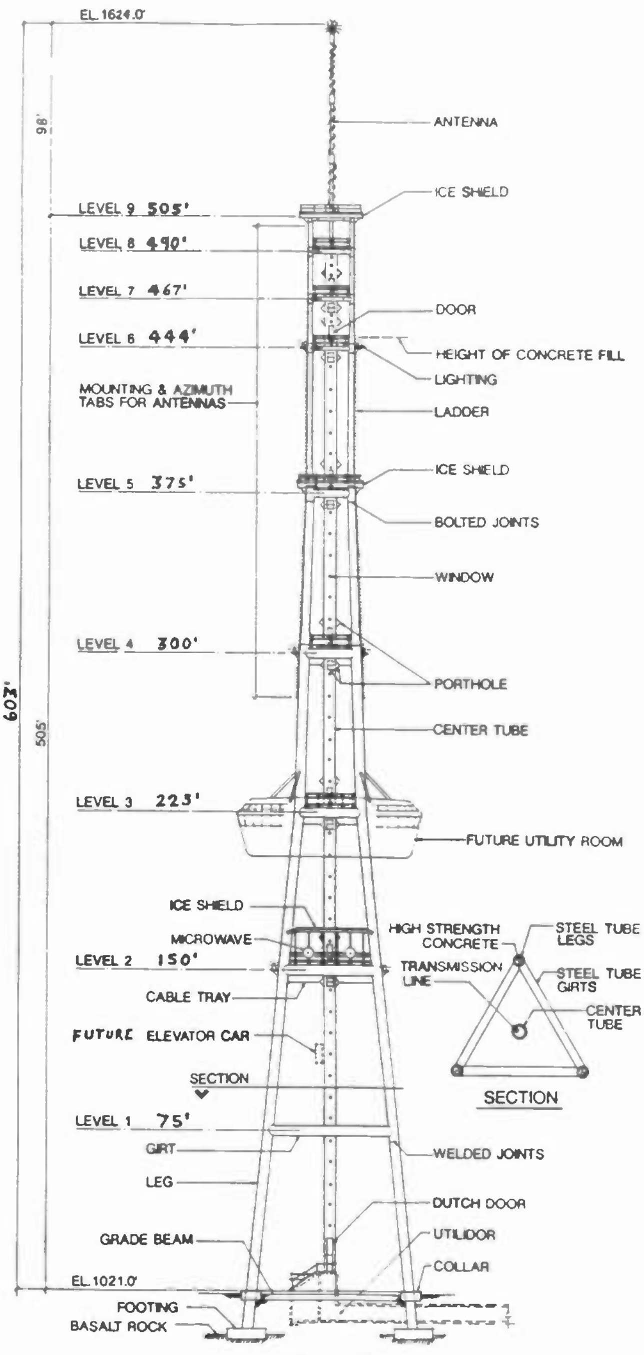
Provide documentary evidence of our effort to get tower users to relocate to our facility.

To guarantee that the NIER level would not exceed 50 $\mu w/cm^2$ at the property lines, and to retain a registered professional engineer to make these NIER measurements.

To meet or exceed the residential requirements of the City of Portland's noise ordinance.

Locate all land mobile antennas above the level of the then highest land mobile antenna in the area (approximately 150').

Provide the city with a frequently updated list of all users at our facility.



The Solution Realized

We made a presentation of our plans to the Board of Directors of SWHRL and received their endorsement. In a series of six meetings spaced throughout one year, we were able to significantly repair twenty years of hard feelings and bad communications with the neighborhood, and receive their encouragement to build a new 603' super tower in their back yards.

The city planning staff saw our joint proposal as the solution to a long standing problem which they had been unable to resolve. At the conditional use permit hearing, the planning staff endorsed the project, but a few neighbors and the owner of the land mobile communications tower spoke in opposition. Unfortunately, the hearings officer agreed with the opposition and rejected our application. But knowing we had clear backing, we appealed the decision to the City Council, where again, SWHRL and the city planners urged approval. The Council agreed and approved the conditional use.

Land Mobile

Though land mobile users were considered in our initial project, further investigation revealed that, if a significant number of existing Healy Heights land mobile users were consolidated on our tower, it would constitute a sizeable tenant base and revenue source. We designed the facility to accommodate the projected number of new and relocated land mobile tenants.

KGON decided it would rather hire an outside management firm than manage the land mobile business in house. To that end, we developed a Request For Proposals and contacted thirteen potential managers. We had eight interested responses and three serious proposals. With the assistance of Ray Trott, a land mobile consultant in Irving, Texas, we evaluated the proposals and selected Motorola C. & E. of San Diego, California, as site manager.

Motorola already provides two-way service for the majority of tenants in the Healy Heights area from facilities on another tower. They guaranteed to relocate all of their tenants to the KGON tower as they took over the land mobile management. As a part of their agreement with KGON we required that they satisfy any and all neighborhood interference complaints arising from their operation and prohibit low band paging operations on the tower.

WHAT WE BUILT

The ultimate design of the facility was the result of satisfying both the objectives of the radio station and the neighborhood and of meeting the conditions imposed in our conditional use permit. The tower's unusual design required the development of several innovative methods of accommodating broadcast and communications equipment.

Tower Platforms and Antenna Mounts

At each of the levels where there are horizontal members (girts), a platform fills the triangular-shaped area between the girts. The platforms are made of a fiberglass reenforced plastic grating to eliminate the generation of intermodulation products at rusty joints between steel grating and structural members. Two of the platforms extend 4' beyond the tower perimeter to serve as ice shields for antennas below them. A separate ice shield is provided for the 150' level. The handrails at each platform and their vertical supports are strong enough to mount typical land mobile antennas and microwave dishes up to 8' diameter. The platforms allow mounting transmitters, receiver pre-amplifiers, and filtering equipment on the tower. A 60 ampere 208/120 volt, 3-phase power service runs the full height of the tower and breaks out at each platform in a weather tight circuit breaker panel. These features allow the elimination of long transmission line runs, enhancing equipment performance and lowering costs. The tower structure is reinforced at the 225' platform to accommodate the possible future addition of a two story triangular, enclosed equipment room with 90' long outside walls.

Additional antennas may be attached to tabs welded at 5' intervals on the three tower legs between the 280' and 500' elevations. Additional tabs are are offset from these to provide attachments for diagonal bracing. Access rungs on the legs allow climbers to reach these antennas from the tower platforms.

Center Tube

A 6' diameter steel tube hangs from the 490' foot level in the center of the tower and descends to ground level. All the transmission lines are run in the center tube, thus hiding them from view and protecting them from weather and vandalism.

At each platform several large bulkheads protrude from the center tube allowing the lines to exit with sufficient bending radius. The bulkhead penetrations use standard Microflect® waveguide boots and cushions for a weather proof seal. A ladder equipped with a climbing safety device is located in the tube providing access for courageous non-climbers to the lines and all seven platforms. Lights and steamship-style portholes illuminate the inside and combat claustrophobia. A sliding steel door opens at ground level and at each platform.

Standby Antennas

At midpoint on the tower are individual standby antennas for each station, to be used during maintenance or failure of the main antenna, transmission line, or combiner system. Because four bay, half-wave spaced antennas are used, all stations may operate at full power and still meet the 50 μw/cm^2 ground level NIER requirement. The power density at the main antenna will be below the ANSI standard as well, allowing repair work to performed. The center tube acts as an effective RF shield to workers as they climb through the area of the operating standby antennas.

Tower Color and Obstruction Lighting

During the design process, the Healy Heights neighbors were interested in pursuing a neutral tower color instead of the standard seven alternate bands of aeronautical orange and white. KGON contacted the regional FAA administrator to discuss the possibilities. The FAA indicated that they might consider a variance if requested by the city and if medium intensity white strobe lights were used during the day. We contacted Brighter Ideas, Inc., who were kind enough to lend us a strobe light. We placed the light high on our 280' tower and operated it for several days. The neighborhood association urged people to stop by and observe the light's operation. KGON representatives were present to answer questions about the strobe's operation and its potential use on the 600' tower. Feelings were mixed, but those people who observed from a distance decided that the flashing light was a little too obnoxious. After a few days the neighborhood association voted to avoid use of daytime strobe lights on the 600' tower and instead tolerate the standard orange and white banding.

Conventional FAA dual lamp beacons are installed at the top and on all three legs at the 300' level. Side obstruction lamps are installed on all three legs at the 150' and 450' levels. Each side light position is equipped with two lamps, the second lamp switching on if the first one fails. Excepting the top tower beacon, all lamps are accessible for replacement from the tower platforms. A sophisticated monitoring and control system designed by Hughey and Phillips warns when any one bulb fails and alarms when any light position is dark because of the failure of two lamps. This status information is monitored continuously by the facility's remote control system.

Transmission Line Tunnel

Intersecting the bottom of the center tube is an underground concrete tunnel, 8' high and 9' wide, running 135' back to the equipment building. Unistrut® type anchoring channel is imbedded in the tunnel walls and ceiling at 4' 4-1/2" intervals to allow the easy attachment of transmission lines and conduits. The tunnel is equipped with a 36" cable tray attached to one wall to carry small (< 1-5/8") flexible transmission lines and low voltage control cables. A fire wall is installed at the building end of the tunnel to prevent the center tube from acting as a 490' chimney in the event of fire. Large transmission lines penetrate this wall through a custom designed bulkhead that employs standard Microflect® waveguide boots and cushions. The cable tray penetration is fire stopped using the Sealbag® system. These consist of sturdy plastic fabric bags filled with a compound that swells and sets when subjected to heat. These are packed in the cable tray, on top of and around the cables, where it penetrates the fire wall. They are easily removed allowing for the installation of additional cables and transmission lines.

Tower Elevator

Provisions are made for a future tower elevator. This will run on a dual tubular track attached to the outside of the center tube. The elevator will park in its own underground room that leads off the end of the tunnel. The design employs a rack-and-pinion drive, thus eliminating hoisting cables and their attendant maintenance problems and potential dangers. The elevator is not just a convenience but maximizes the potential of the tower by making extensive tower mounted equipment a real possibility.

Transmission Lines

We selected 9-3/16" rigid copper coaxial transmission line manufactured by MYAT, Inc., even though both 8" line and dual 6-1/8" lines had the necessary power handling capability. In the event of catastrophic antenna failure or severe lightning strike, we wanted to minimize the chance of transmission line failure. Frequently, dual lines and split antenna feeds are employed to allow partial power operation while repairs are being made to the antenna system. Since occupational safety and health laws would not allow repair of the antenna while a portion of it were in operation, we decided that dual lines were an unneeded complexity. Even though this was first time MYAT had manufactured 9-3/16" line, we chose them because of price, a willingness to work closely with us during the design phase, and the quality of manufacture we had seen in their smaller lines.

We had the factory reserve several lengths of line. As the installation proceeded, we telephoned them the dimensions of custom lengths and they were on site in one or two days. After installation, Don Aves, their chief design engineer, came to Portland to assure that the line was performing as designed. It was refreshing to call the factory and frequently have the president answer the phone. The manufacturing quality of the delivered line was impeccable, it was delivered on time, there were no air leaks, and each line section had been swept on a network analyzer. Interestingly, even though the copper tubing was manufactured in Germany by Cablewave's sister company, Kabelmetal Electro A.G., Cablewave does not manufacture this line.

The line features watchband spring inner conductor connectors to minimize wear as the inner conductor moves with temperature changes. The bullets are captive at one end of the line. The vertical section of the line is hung with conventional sliding spring hangers, two hangers per 17-1/2' section. The horizontal run in the tunnel is hung using unique two point hangers developed by the KGON engineering staff in collaboration with MYAT, Inc. Because the line is not exposed to wind, lateral restraints are not required. Broadcasters wishing to use 9-3/16" line should be aware that there is no EIA standard for it. While all the manufacturers seem to use the same bolt pattern and O-rings at the flanges, bullet dimensions are different, leading to the need for custom-made adapter bullets when attaching one manufacturers line to equipment manufactured by others.

Cablewave 3-1/2" and 3' semi-flexible coaxial lines are used for the auxiliary antennas. MYAT 3-1/8" unflanged rigid line is used exclusively for the indoor transmission line runs.

Combining System

The FM broadcast stations are multiplexed together using a balanced constant impedance ladder combiner manufactured by Shively Laboratories. Computer modelling has simplified the very complex tuning procedure these combiners require and allow several design parameters to be optimized. We selected Shively because of their good track record in designing and tuning this type of combiner. Not only were they willing to guarantee fairly strict specifications, but they could show us several installations where these specifications were being met. Each module in our combiner system consists of a 3-1/8" input hybrid, two filters using four 24" square cavities each, a 6-1/8" output hybrid, and a 6-1/8" output patch panel. (The module closest to the antenna has a 9-3/16" output hybrid and patch panel.) The patch panels allow KGON personnel to manually remove any combiner module from the main transmission line. Each module can handle a 40 KW input power. Each station has its own module except for a lower powered educational station. It uses a single five cavity filter, employing 18" cavities, that feeds the wide band input port of the system.

The entire combiner system was erected and tuned in the factory and then dismantled and shipped on a dedicated truck to Portland. The KGON staff placed the modules and did some preliminary erection. Shively dispatched two engineers to Portland to finish the erection, tune the combiner system and perform final acceptance testing. KGON scheduled things so that it was possible to operate the entire system at full power with three stations while the Shively engineers were on hand. Though the engineers had scheduled four days in Portland, they were able to spend one day sight seeing and leave a day early. The measured results easily met the guaranteed specifications. The input VSWR is below 1.1:1 over ± 150 kHz for every station, with one station enjoying a 1.01:1 VSWR. The group delay is below 50 nanoseconds over ± 150 kHz. The insertion loss variation is less than ±.1 db over ±150 kHz. And the isolation between any two inputs is 60 db or greater.

Shively also provided a combiner monitoring system. This consists of a module for each station that monitors incident power, reflected power, and VSWR for both the combiner input and output. Analog voltages proportional to these parameters are available simultaneously on the back panel for use by the individual stations. Contacts are also available that close when any of six user settable alarm points is exceeded. There is also a master display panel that shows the status of each of the reflectometers, the input switching, and the output patch panel.

At the input to each combiner module is a motorized coaxial switch that switches the station's transmitter output between the combiner module input and the standby antenna. This switch is controlled by an interface panel designed and constructed by the KGON engineering staff. It allows tenant stations to control the coax switch remotely. It also allows KGON personnel to control the switch either remotely or locally. In addition, it takes information from the combiner monitoring system and automatically removes a station from the combiner system if a problem is detected. In the event of either an automatic transfer or a transfer initiated by KGON personnel, tenant control of the switch is locked out to assure the safety of workers and equipment. The interface provides a dry contact opening to control transmitter high voltage interlocks, and status information on the position of the coaxial switch.

Transmitter Building

The building has two 2800 square foot stories with a daylight basement and a single story extension of 1000 square feet. It was decided early on in the process that KGON would maintain complete control of the combiner system and restrict access to it. To this end, KGON has a large room on the top floor that contains all of its transmitting equipment, the computer controlled site monitoring and control system, and the entire combining system. The transmitters and associated racks are built through a wall in order to separate ventilation systems and minimize noise. The front panels of the equipment are flush with the wall leaving the main body of the equipment in its own room. A 12' workbench faces the transmitters.

Each tenant has its own suite for its equipment, each with its own outside entrance. On the top floor are two 15' x 20' tenant spaces. On the lower floor are two more 15' x 20' tenant spaces, and one 20' x 35' space. Each of the tenant spaces is provided with a water supply and drain suitable to operate a 50 KW water cooled dummy load. On the lower floor is a 35' x 40' space reserved for land mobile and non-broadcast microwave operations. The tunnel terminates in this room, thus minimizing transmission line lengths for land mobile users.

KGON provides toilet facilities and a break room for use by all the tenants. Because engineers are occasionally trapped by inclement weather at this site, the break room is equipped with kitchen facilities and a day bed.

To minimize the headache of lost keys and to maintain a record of building users, all outside doors are controlled by a centrally operated access control system with numeric keypads near all the doors.

A cable tray system runs throughout the entire building. It allows small transmission lines and low voltage cables to be run between all of the suites on both floors. The cable tray system interconnects with the tray in the tunnel. The tray is suspended from the ceiling on cradles made of Kindorf® channel. Large transmission lines from each of the tenant's suites is hung from this channel using slide hangers.

Fire Detection And Suppression

Each suite has a separate electrical service and ventilation system provided by the tenant. There is a 2 hour fire separation between all suites, and all penetrations between suites are protected by the Sealbag® fire stopping system. Each suite has its own fire detection system and CO_2 flood fire suppression system. The fire detection system uses a configuration of redundant detectors to prevent false alarms. A separate system of heat detectors controls the CO_2 flood system. A 30 second delay before release of the carbon dioxide gas, during which a digitally synthesized voice announces the danger, allows personnel to escape. When a fire is detected or the CO_2 flood system is activated, all power is removed from the suite, turning off all the equipment and ventilation, and the fire department is automatically summoned. Spring dampers in the ventilation system seal the room. A strobe flashes outside the door of the suite where a fire is detected, and all the doors in the building are unlocked to enable the fire department to find the fire and have easy access to it.

Emergency Power Generation

The 1000 square foot addition on the first floor houses a small Faraday-shielded room for sensitive equipment test and repair as well as the electrical service entrances and a 500 KW natural gas fired emergency power generator. This generator has sufficient capacity to run the entire facility without load shedding. Each tenant has its own automatic transfer switch and can start the generator independently. We chose natural gas firing because of the difficulty and high cost of installing and maintaining either underground or above ground diesel storage facilities for a facility of this size. The engine requires 100 cfm of gas at 15 psi at full load. Because the gas utility could not guarantee this pressure in mid-winter, we installed a variable speed gas compressor on site.

Of particular concern in designing the power generation system was the question of meeting the very strict city noise ordinance. The nearest residence is only 75' from the generator. We housed the generator indoors, in a concrete room that is built into a bank. Special construction techniques were used in the roof to minimize noise transfer and ventilation in the room goes through sound traps. Natural gas engines are inherently less noisy than diesel fired units. The engine exhaust is run through a "critical" design muffler. At full load, the exhaust noise is a mere burble. The engine is cooled by city water passing through heat exchangers, thus eliminating radiator fan noise and the need for large quantities of cooling air. A natural gas detector is mounted in the room to shut off the gas supply in the event of a leak, and an emergency engine shutdown switch is provided next to the entrance for the fire department.

Transmission Plant

KGON elected to employ a fully redundant FM transmission plant with identical new Broadcast Electronics FM-35B transmitters, dual 23 gHz PCM encoded digital STL's, and a redundant audio processing line. The output of the off-line transmitter is automatically routed to a calorimetric dummy load with the filament switch controlling the water flow valve, and a flow switch in the dummy load water supply connected in the transmitter plate interlock circuit.

Ventilation

Because summer temperatures rarely exceed 100° F in the Portland area, we elected to only ventilate the transmitters and not provide air conditioning. Temperature rise in the transmitter room may not exceed 10° above ambient with both transmitters operating at full load. The transmitters operate freely into the room and the ventilation system maintains the room temperature. This makes for a less complicated system and eliminates the back pressure problems that frequently plague ducted transmitter installations. A constant flow system is used with room temperature being controlled by the amount of recirculation. The room operates at a positive pressure relative to the outside at all times, minimizing dust infiltration. Both outside air and recirculated air is filtered by 2" pleated filters followed by 98% rated bag filters. Outside air is mixed with recirculated air in a plenum before the filters. This allows the warm inside air to dry the outside air, preventing fog from waterlogging the filters.

The combiner room and the work bench area are fully air-conditioned. The air receives the same filtration as the transmitter room and the system operates at a positive pressure. The two equipment racks located between the transmitters take their air from the air conditioned space using rack mounted blowers and are exhausted back into this space. The other tenants provide their own ventilation systems; their leases with us require systems similar to ours.

Power Conditioning

A 400 amp 3 phase Islatron® power conditioner serves the KGON installation. It uses a metal oxide varistor bank followed by a "tracking filter." The clamp voltage of the MOV's is held intentionally quite high so that they only clamp very large voltage spikes. Since MOV's tend to deteriorate every time they absorb energy, this increases the life of these components. The "tracking filter" is designed to handle smaller spikes. It consists of a series inductance formed by wrapping bus bars in a ferrite material. This acts as a low pass filter increasing the rise time of transients. This is followed by shunt capacitor which can absorb energy or release it back to the line as determined by a smart switching circuit. Supposedly this circuit compares the actual voltage to a heavily low pass filtered version of the input power to determine whether energy must be removed or returned to the line. The manufacturer claims that this unit will hold the instantaneous output voltage to within 2 volts of what it should be.

The power for all rack mounted equipment, computer equipment, and the STL's is run through a 5 KVA Ferrups® uninterruptable power supply manufactured by Best Power Products. Unlike most UPS's, the Ferrups inverter is off line-when on normal power. This improves efficiency and reduces premature failure of inverter components. Both normal and emergency power is filtered through a ferro-resonant transformer, assuring an harmonic distortion of less than 5% and tight voltage regulation on either normal or emergency power.

Summary

Our effectiveness as engineers depends on how well we recognize the nature of the problems which we must solve and our ability to solve them. Frequently the solutions hinge on our understanding of the effects of our attitude and our willingness to change. The successful completion of KGON's new 603' tower is a direct result of our recognition of the importance of community involvement and our willingness to change self-defeating attitudes. The ultimate design of our facility is the result of satisfying our objectives as well as those of the neighborhood.

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And the numerous vendors, distributors and subcontractors that make our jobs so much easier and a project of this magnitude possible.

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AUDIO SYSTEM DESIGN AND MEASUREMENT

Thursday, April 18, 1991

Moderator:
Margaret Bryant, WMAQ-FM, Chicago, Illinois

FAST RESPONSE AND DISTORTION TESTING
Richard C. Cabot
Audio Precision
Beaverton, Oregon

THE FUTURE OF ANALOG AUDIO CARTRIDGES
William Franklin
Fidelipac Corporation
Moorestown, New Jersey

A DIGITAL AUDIO CART MACHINE FOR BROADCAST*
Robert Easton
360 Systems
Tarzana, California

AUDIO PROCESSING FOR RADIO IN THE DIGITAL DOMAIN*
William Gillman
Gentner Electronics Corporation
Salt Lake City, Utah

APPLICATIONS FOR DIGITAL AUDIO TAPE MACHINES IN RADIO BROADCASTING
Mel Lambert
Media and Marketing
Studio City, California

A WORLD'S FIRST STUDIO ACOUSTIC DESIGN
Brian McGettigan
Radio New Zealand
Wellington, New Zealand

*Paper not available at the time of publication.

FAST RESPONSE AND DISTORTION TESTING

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The paper will discuss methods to test the amplitude response, phase response and distortion of audio equipment and links. A new approach is described which is considerably faster than existing methods in common use. The signal may be transmitted and acquired in one second, producing minimal disturbance to broadcast programs and allowing quasi-real time measurements. Results may be displayed as obtained or viewed off line if desired. Measurement results on typical equipment will be compared with those obtained using conventional techniques.

INTRODUCTION

Some broadcast devices and links are never tested because of the high cost of broadcast system down time. Many others are inadequately tested because of the substantial test time required. Conventional testing with manually operated equipment can take as much as an hour for a thorough evaluation. This can be reduced substantially with computer controlled equipment. If run after the station is finished with normal programming for the day the burden becomes strictly one of the broadcast engineer's time. However, if the facility operates 24 hours per day any testing requires suspension of programming for the duration of the testing. Clearly, the less time required, the better.

AUTOMATED SYSTEM ONE PROCEDURE MEASUREMENTS

To measure amplitude, phase and distortion at several frequencies the conventional approach is to step a sinewave generator from one frequency to the next. The amplitude and phase of the fundamental and the amplitude of the harmonics are measured at the output at each new frequency. There are practical limits to how fast each measurement can be made, setting a limit on total measurement time.

A complete broadcast proof of performance in accordance with the U S Federal Communications Commission guidelines can be completed in 3 minutes with an Audio Precision System One. This testing includes response and distortion measurements versus frequency at several different modulation levels, separation and noise checks.

EBU SEQUENCE MEASUREMENTS

In an effort to speed testing and allow standardization between government telephone and broadcast systems, the CCITT and the EBU standardized have a test procedure. The standards specify two sequences of test tones, one for monophonic links and one for stereophonic links. The monophonic sequence lasts 31 seconds, the stereophonic sequence 33 seconds. These are used for testing various parameters including response, distortion, phase, crosstalk, noise and compander action. The sequence begins with a frequency shift keyed (FSK) preamble which indicates which sequence is being sent and the originator of the test. The tones are one second in duration each, the noise test is eight seconds long.

This test has the advantage of being standardized by an international body, making it easier to test links which cross national boundaries. Its biggest problem is that all sinewave tests are made to last one second, regardless of what they are intended to measure. Some parameters, such as the level measurement required for frequency response, can be measured in substantially less than 1 second with little expense. Other parameters, such as distortion, are very expensive to measure in less than one second. This is especially a problem with stereo measurements, often requiring manufacturers to build duplicate measurement hardware for the two channels to meet the specified timing. Little attempt is made to combine tests where a single stimulus could have been used for more than one measurement. Only the interchannel phase and frequency response tests are performed simultaneously.

Figure 1 shows the results of an EBU sequence measurement using an Audio Precision System One. This measurement requires an optional EBU sequence receiver board and software developed by RTW of Germany. This board decodes the preamble information and supplies some additional filtering required for the individual harmonic measurements. The system speed is sufficiently high that all performance measurements are made with the standard measurement hardware. The software which controls the system for these measurements is capable of transmitting and receiving the standard EBU sequences or up to 98 custom sequences defined by the user.

MULTITONE MEASUREMENTS

One way to speed a complex set of measurements is to use parallel processing. To do this, all test frequencies are applied simultaneously as shown in Figure 2. To obtain the desired data each one must be measured individually at the output with a dedicated bandpass filter, voltmeter and phasemeter. The cost and complexity of multiple bandpass filters and rms converters would be substantial, especially if high accuracy was required. The sharpness of filter required at high frequencies to separate distortion products from original stimulus frequencies would work against the use of inexpensive 1/3 octave type filters.

This high performance frequency selective analysis is exactly that supplied by an FFT. The FFT operates with sampled data, acquires a block of input signal, and transforms it to the frequency domain. The block length is typically chosen to be a power of two, but this is not required with some FFT algorithms. The transformation yields one frequency point for each pair of input samples, giving 8192 frequency points from 16384 input samples. Since a fixed number of points will be acquired for analysis the stimulus signal may be generated digitally to exactly match the acquisition length of the FFT analysis. This block orientation of stimulus and analysis produces a fixed frequency resolution of the stimulus frequencies matching the bin centers of the analysis.

Frequency resolution limitations

To prevent discontinuities when the stimulus waveform repeats, all components in the waveform must have an integer number of cycles in the block. Therefore all tone frequencies are integer multiples of the block repetition frequency. For a 48 kHz sample rate and an 8192 point stimulus/analysis block this will produce a 5.86 Hz frequency resolution. All desired frequencies must be quantized with this resolution. Therefore the repetition frequency becomes both the lowest frequency which can be generated and the closest possible spacing of tones. For example, if 5 Hz, 10 Hz, 100 Hz and 105 Hz are desired stimulus frequencies, the actual frequencies will be 5.86 Hz, 11.72 Hz, 99.6 Hz and 105.47 Hz. These

```
IDENTIFICATION = RTW      SEQUENCE = 01                03.04.1990   10:52:19
```

Ref.Level	Left	Right
1020 Hz	[dBu]	[dBu]
	-2.83	-2.82

Frequency [Hz]	Left [dBr]	Right [dBr]	Phase [deg]
1020	-11.97	-11.96	-0.1
40	0.01	-0.03	-0.1
80	0.02	0.02	0.0
200	0.02	0.02	0.0
500	0.02	0.02	-0.1
820	0.01	0.01	-0.1
2000	-0.00	-0.01	-0.2
3000	0.04	0.03	-0.3
5000	-0.00	-0.03	-0.3
6300	0.04	-0.01	-0.2
9500	0.03	-0.03	-0.1
11500	-0.02	-0.08	0.0
13500	-0.01	-0.04	0.1
15000	-0.03	-0.06	0.1

Frequency [Hz]	Level [dBr]	Dist. [dB]	
1020	8.98	-74.5	A ak2
1020	8.99	-74.6	B ak2
PAUSE			
60	9.00	-75.3	A ak3
60	9.01	-78.3	B ak3

Frequency [Hz]	Level [dBr]	Crosstalk [dB]
2040	-11.99	-62.18
2040	-12.00	-68.55

Linearity [Hz]	Left [dBr]	Right [dBr]
800	5.98	5.99
800	-6.05	-5.98
800	5.98	5.99

NOISE A	-77.8	dBqpsmax
NOISE B	-75.2	dBqpsmax

Figure 1 EBU stereo transmission test sequence results.

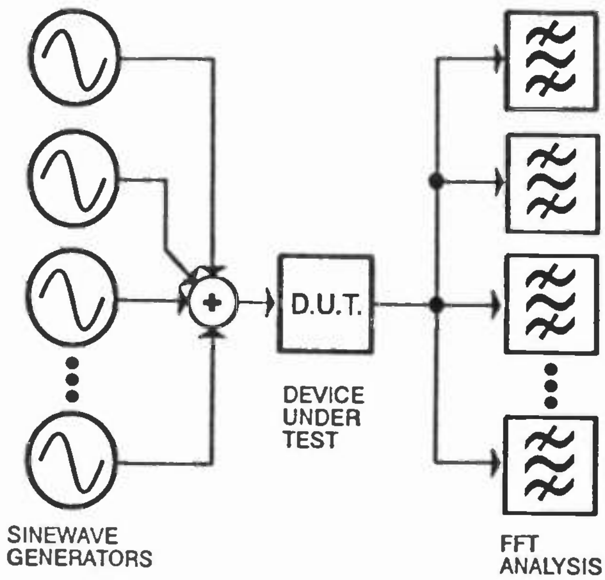

Figure 2 Block diagram of multitone test setup.

frequencies were obtained by rounding to the nearest allowable frequency value. The software allows selection of truncation or rounding of frequency values when creating the waveform data.

Measurement of response

Figure 3 shows the spectrum of a 60 tone (1/6 th octave) test signal. All frequencies are contained in the signal simultaneously. By using data at only the frequencies of the original stimulus a conventional frequency response graph is obtained as shown in Figure 4. The measured points are interpolated to obtain the complete curve.

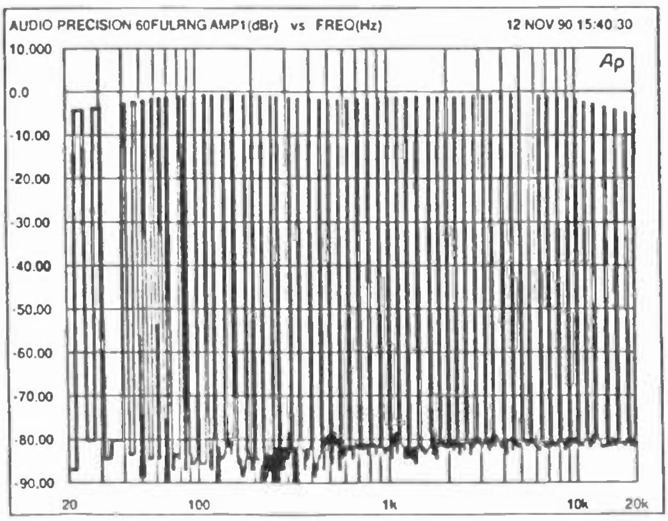

Figure 3 Spectrum of 60 tone (1/6 th octave) test signal.

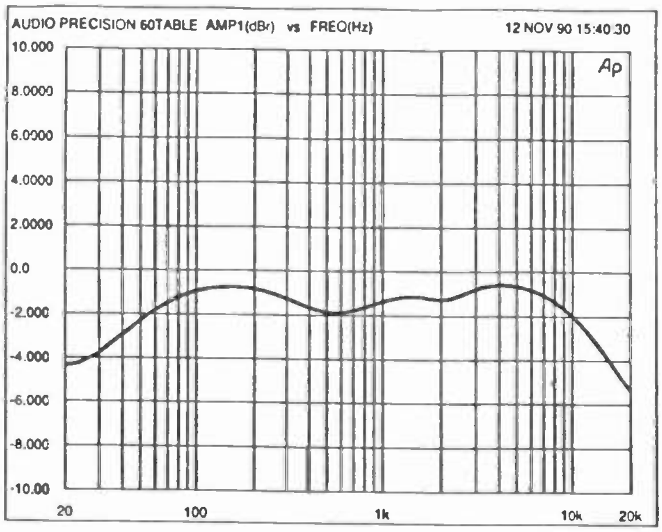

Figure 4 Frequency response plot obtained by measuring amplitude at stimulus frequencies only.

Since the frequencies used in the stimulus may be user selected, detail may be concentrated in frequency ranges of special interest. If performance in the low frequency and high frequency regions are known to be the problem areas the test signal might consist of 5 tones from 20 Hz to 100 Hz, a tone at 1 kHz and 5 tones from 10 kHz to 20 kHz.

Measurement of distortion

Nonlinearities in the system under test will produce harmonics of the stimulus sinewaves. If stimulus frequencies are selected which are not integer multiples of each other the harmonics of these tones will occur at unique places in the spectrum. The high frequency-resolution of the FFT allows the amplitude of each harmonic to be measured separately. A plot of the harmonic levels for a 5 tone test signal is shown in Figure 5. These amplitudes could be root-sum-square combined to obtain a THD figure. If the harmonics are allowed to overlap it is possible for harmonics with opposing phases to cancel, lowering distortion readings. If the nonlinearity has sharp discontinuities it will generate high-order harmonics. Low frequency stimuli will then have harmonics which extend all across the audio band. This makes selection of other stimulus frequencies difficult if harmonic overlap is to be avoided. Most nonlinearities produce a distortion spectrum which falls off with increasing harmonic order. This allows an upper limit on harmonic order to be used when calculating which frequencies will cause overlap.

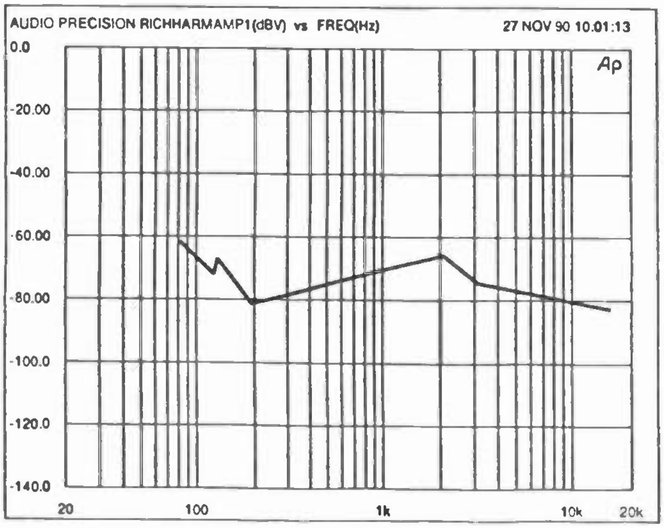

Figure 5 Distortion plot obtained by measuring amplitude at harmonic frequencies only.

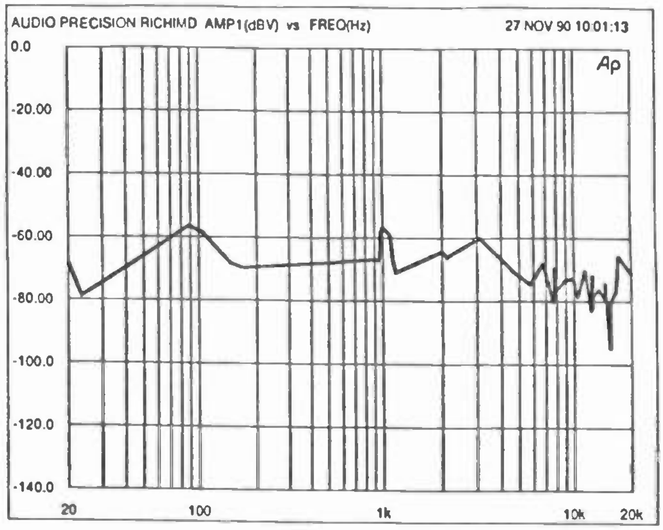

Figure 6 Distortion plot obtained by measuring amplitude at intermodulation frequencies only.

This is not to say that the harmonic amplitudes above this upper limit will be zero but that their amplitude is negligible when compared to the components already included.

If the system has nonlinearities there will be intermodulation products between all combinations of tone frequencies. If the nonlinearity can be modelled as a power series, the intermodulation frequencies can be predicted. These intermodulation components will appear at frequencies above, below and between the stimulus frequencies. The calculation of these intermodulation frequencies gets very complex when many stimulus tones are included and when the nonlinearity is of high order. This causes frequency selection problems similar to that from harmonic distortion. In practice the problems of intermodulation frequency avoidance limits the usable stimulus frequencies much sooner than the harmonic avoidance issue. A plot of intermodulation distortion levels for the 5 tone signal used earlier is shown in Figure 6.

It is important to note that the distortion measurements obtained with this technique are not directly comparable to those obtained by single sinewave THD+N testing or by conventional IMD testing. There are several reasons for this. The crest factor of a multitone stimulus will always be higher than that of a single sinewave. For the same peak signal amplitude, the amplitude of each individual tone will be lower than a single sinewave at that frequency. The resulting distortion products will be different in the two cases.

Measurement of noise

By picking stimulus frequencies which leave gaps in the harmonic and intermodulation spectrum it is possible to measure noise with this same test. The FFT analysis merely has to display the levels in the otherwise empty bins to obtain a plot of noise floor spectrum. An example of this for the five tone stimulus used for the previous measurements is shown in Figure 7. These amplitudes could be root-sum-square combined to obtain a single figure. The squared and summed value must be multiplied by a constant representing the number of bins used in the computation, the bandwidth of the bins, and the bandwidth of the measurement to yield an accurate wideband noise figure. If enough frequency points are used it is possible to factor in weighting filter gain vs frequency when computing noise, yielding a weighted noise measurement.

Measurement of phase

The FFT phase values will reflect the phase of each component in the received signal relative to the start of the acquired data. If absolute phase information is desired it is easily obtained from a single channel of data. The phase of each component in the multitone test stimulus is known since the waveform is software created and digitally generated. If the stimulus components have been specified to all be in phase, the measured phase will be the phase shift through the device under test. This approach may produce a high crest factor stimulus, as described below. If non-zero phases have been specified for the stimulus components,

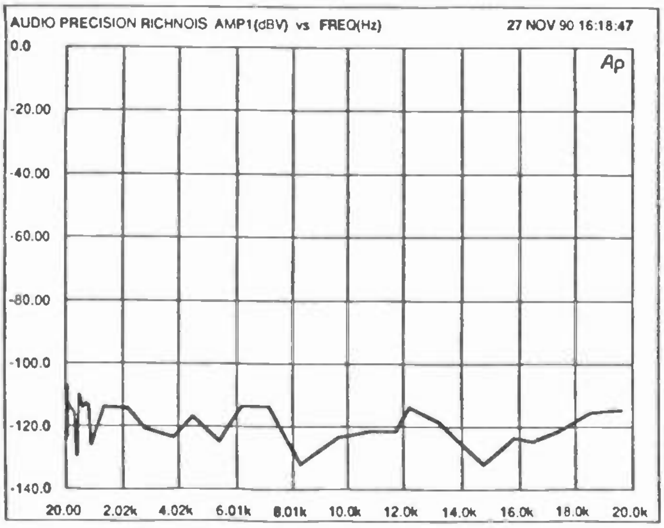

Figure 7 Noise plot obtained by measuring amplitude at empty bin frequencies only.

it is simple to have the software subtract the stimulus phase from the measured data to obtain the correct values. When measuring systems with time delay, this delay will be reflected in the measurements. Since the added phase shift will be proportional to the product of time delay and frequency it is straightforward to compensate for it if necessary.

If a stereo measurement is made, interchannel phase may be obtained by subtracting the phase data of one channel from the other. If the two channels were stimulated with the same signal it is not necessary to perform any delay correction nor are the original stimulus phases important. This is because the measurement depends on differences between the two channels and any correction applied to both channels will cancel in the subtraction.

Crest factor effects

Since all stimulus tones are a multiple of the block repetition frequency their phases are locked together. If the worst case phase values were picked for the tones all of the individual sinewave peaks would line up in time once during the block of samples. The peak amplitude of the resulting signal would be equal to the sum of each components amplitude. The rms value of the signal is the root-sum-square of each components amplitude. For equal amplitude tones the peak value increases as the number of tones added while the rms increases as the square root of the number of tones added. Crest factor, the ratio of peak to rms, would therefore increase as the square root of the number of tones. A single sinewave has a crest factor of the square root of two (1.414). For a 60 component signal the worst case crest factor would be 10.95.

Careful selection of sinewave phases can significantly reduce this value. A random number generator may be used to select phase values and will typically produce a crest factor much lower than the worst case. Other algorithms exist to randomize the phases for minimum crest factor. The time domain view of a 60 tone signal with random phases is shown in Figure 8. Note that it appears somewhat like program material or noise.

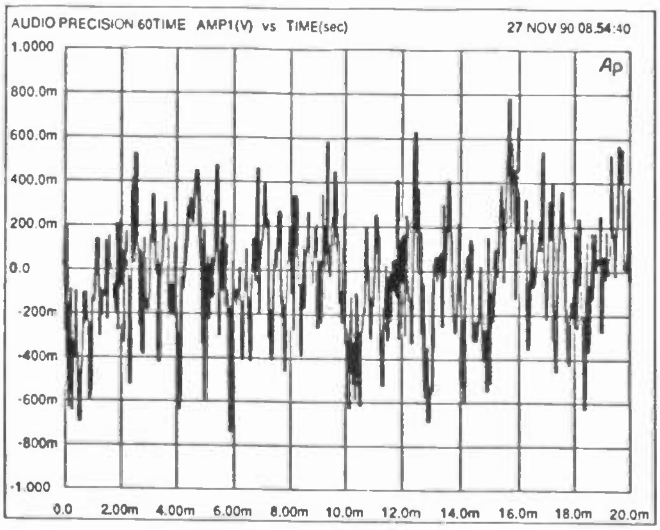

Figure 8 Time domain view of 60 tone test signal with random phases.

It is interesting to note that the crest factor of a multitone test method in which the frequencies were not locked to a common submultiple could never reduce the crest factor below the theoretical worst case. As the sinewave drift in and out of phase there would be cancellation and reinforcement of peaks. On occasion all of the sinewaves would line up with their peaks occurring simultaneously and the resultant peak signal amplitude would be the sum of all of the individual peak amplitudes. Although the occurrence of such events would be rare, they would happen and could clip the system under test.

Effects of time shifting

If the system introduces any frequency shifting, as would happen from speed error in a tape recorder, the stimulus and acquisition blocks would no longer line up. This would shift the stimulus frequencies out of the FFT bin centers and require the use of windowing in the FFT. Windowing causes leakage between neighboring bins and, depending on the window used, amplitude inaccuracies. If the frequencies are not closely spaced, this loss in resolution may be acceptable. If not, the data would need to be corrected for the time shift. Software is being developed to time shift the acquired data using interpolation/decimation techniques allowing correction for speed errors. Wow and flutter will produce sidebands on the signal components which will limit the distortion measurement residual for closely spaced frequencies.

Program material simulation

Multitone test signals are much more like program material in the time domain than traditional sinewave stimuli. The waveshape of the 60 tone signal is shown in Figure 9. The software which creates the stimulus waveform allows the amplitude of each component to be separately specified. This capability may be used to shape the stimulus spectrum to simulate the program material. Another is to follow a specific pre-emphasis curve as might be done in broadcast applications. If the test signal spectrum matches that of typical program material it will more accurately predict the behavior of compressors and other processing equipment in use.

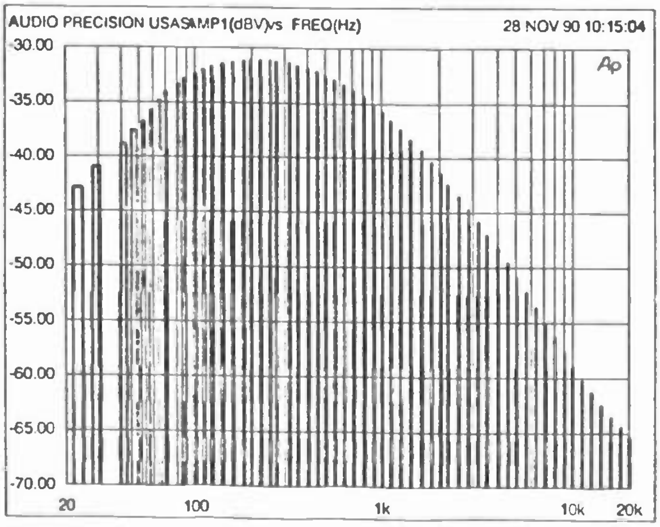

Figure 9 Spectrum of 60 tone (1/6 th octave) test signal when weighted to simulate program material.

Measurement time vs acquisition time

Signal acquisition and measurement are separate operations. This fact may be used reduce system down-time even further. The link must be down for the length of time required to transmit and receive the stimulus, but the measurement computations, comparison to limits, etc. may be made off line. The minimum time required to transmit the stimulus is generally longer than the stimulus repetition length in order to let the system settle to steady state operation. The settling time is based on both the lowest frequency component in the stimulus and on the desired measurement resolution between components. Three cycles of this minimum frequency are usually adequate for practical systems. For a 2.93 Hz frequency resolution this would require approximately one second. If the low frequency limit is raised and resolution sacrificed there will be a corresponding reduction in simulation/acquisition time. Since the stimulus data and acquired data are treated as blocks, the time required to does not change with the number of tones used, only the block length.

The FFTs and post processing required to obtain response, phase, distortion, noise and later comparison to limits typically requires a few seconds. Unlike the acquisition time, the post processing time will scale with the number of data points being measured and displayed.

CONCLUSIONS

A new multitone fast measurement technique has been described. The measurement can be made in approximately one second and yields results comparable to conventional measurement techniques.

ACKNOWLEDGMENTS

Bob Wright wrote the waveform creation software. Bob Metzler provided many interesting discussions of applications issues and helped greatly in evaluating new software. Bill Rich deserves special thanks for an excellent job preparing illustrations.

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THE FUTURE OF ANALOG AUDIO CARTRIDGES

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ABSTRACT _ Based on specifications established by the NAB for Cartridge Tape Recording in 1964 and 1975, both hardware and software equipment have progressed in design and innovation to become the mainstay in day to day sound presentation in broadcasting. This paper will deal with the recent innovation in cartridge machine and tape cartridge designs, as well as review the driving forces in audio reproduction formats, and the current offerings of digital audio equipment.

INTRODUCTION

Today there is a healthy business in the cartridge machine industry. There are about a dozen manufacturers supplying equipment and tape for the endless loop format worldwide. Within the last four years, all of the major companies have introduced new models with extra features and applied technology. The winner in this competitive market is the end user. Never have specifications and features been as potent as the recent offerings of cartridge equipment.

Several new tape cartridge designs have been introduced in the last four years as well. Advanced magnetic particles used in current tape formulations combined with improved plastic assemblies have transformed what was originally intended as a 4-track consumer sound source into the professional around the clock broadcasting format of today. Yet all of these cartridges are compatible with the various models and vintages of equipment and storage space most radio stations have installed. Obviously, such a large installed base and library must be a testimony to the reliability of the cartridge format.

The digital age of audio is here and there are several formats including tape, hard disk and solid state memory. Some formats are already in the second and third generations. All of the new digital units address a certain niche of audio production. And with these various digital formats come new operating procedures and technical skills, new repair and maintenance techniques, different interfacing and differing costs of operations. Significant additional costs of integrating any one of these new digital formats into a broadcast operation hinder acceptance of any new format. Critical factors relate to the cost effectiveness and/or performance benifits that any digital format may offer over the audio cartridge. Technology for technology's sake does not enhance market share or the bottom line.

CARTRIDGE MACHINE REPRODUCERS

The cartridge machine is a dedicated purpose piece of equipment. It provides immediate access of program, can be controlled remotely and works effectively and efficiently everyday. Audio cartridges can be used for a variety of applications such as music reproduction, spot announcement, I.D's and jingles, commercials and listener requests. By utilizing its own dedicated control track, the system provides all sorts of simple interfacing possibilities from idiot lights to automatic sequencing, to more advanced logging and billing information applications. This format has evolved around broadcasters' need for a recordable medium with ease of operation and high fidelity sound reproduction at an economical cost.

The cartridge machine/cartridge format has its strong points and weak points. Technological improvements in hardware provide solutions to many of the weaker points of the system. New audio cartridge machines exploit the latest I.C. technology. Lower noise devices, LSI's, VCA's and host of dedicated function I.C.'s make electrical circuit designs simple and allow for more features. Analog audio circuitry designs have definitely not stood still in the last ten years. We see head pre-preamps and the use of lower noise devices offering substantial gains in system signal to noise ratios. Dedicated noise reduction systems both external and internal also lend significant sonic improvements to the analog tape format.

Tape transports employ advanced D.C. servo designs and solenoid configurations which make occasional dragging carts a possible cause for some maintenance and alignment, but no longer a total system weakness. Adjustable head bridges and improved hold down systems stabilize the cartridge to machine interface allowing excellent consistency and repeatability. Smart status lights, timers, dual tape type units, all of these features can be found among the latest cartridge machine offerings. These features are the result of years of input from users, and are made possible by advances in electronic componentry, engineering design tools and more sophisticated manufacturing techniques.

The reliability and serviceability of modern cart machines makes it easy to keep machines in fine tune condition. The days of point to point wiring are gone. Socketed devices and trouble shooting indicators simplify a technician's life. Replacing heads or pinch rollers, and level adjustments are all within the grasp of even the most junior technicians. The analog cartridge format is so highly refined, there seem to be few if any good reasons to abandon it.

However, this is not just a case of "If it ain't broke don't fix it", as some would say. The modern cartridge machine format offers a combination of advantages not provided by any other single piece of equipment; such strong points include user friendly ease of operation, versatility, recordability, instant access, durability and dependability, ease of maintenance, economy and excellent audio quality.

CARTRIDGE LUBE TAPE

The real starting point for back coated lube tape was a consumer tape media for 4-track and 8-track formats, optimized at 3 3/4 I.P.S. with many major tape producers supplying the demand. As consumer demand disappeared for the 8-track so did the tape. Today cartridges utilize specialty tape products designed specifically for the unique requirements of the broadcaster. Although new magnetic tapes have been introduced, formulations maintain compatibility with older products in the field while providing the listener significant sonic benefits. Several cart machine manufacturers now offer convenient methods of utilizing both normal and high output tapes interchangeably.

It is important to point out that sub-stantial improvements in sound quality can be gained by upgrading to higher output tape stock. This can be accomplished for a very small increase in cost, particulary if compared to the cost of a completely new format.

A quick comparison of the tape specifications used in the orginal cartridge media compared to high performance products offered today shows that dramatic improvement can be obtained. Moreover, further improvement in coating procedures and magnetic formulations are likely.

CARTRIDGE DESIGN

The housing which carries the magnetic media interfaces with the machine and contributes to total systems performance. Originally carts were designed for monophonic reproduction. When stereo arrived, new precision specifications tightened up dimensional tolerances since the internal tape path is a critical concern. The cart, too, evolved; such features as tension control, improved flutter, new engineering plastics, consistency and stability enhance acoustical performance and life expectancy from the tape media.

DRIVING FORCES IN AUDIO REPRODUCTION FORMATS

In trying to foretell the future of audio formats, one does not necessarily need a crystal ball. History shows us how our present formats developed. Since tremendous R&D investment is required to convert technology into hardware, large consumer markets afford the best avenue to recoup new product development expenses. In the case of 1/4" audio tape, some 40 years ago both the consumer and professional markets standardized on this format. Broad acceptance created a very affordable and readily available format. Of course, for professional applications the requirements were refined, but it is the consumer market which allows manufacturers to take advantage of economies of scale; this reduces the cost of electronic hardware components as well as the cost of the media.

Coupled with the manufacturers' point of view is the music industry's interest in distribution of its software product. So, as the consumer behemoths support new audio formats, new product technology eventually trickles down to the broadcaster. Today you see hardware manufacturers such as Sony and Matsushita purchasing huge software organizations perhaps in a effort to control future audio formats, and possibly release the stranglehold of artist rights advocates on consumer digital recording hardware. It appears that these unions may elimate some of the obstacles to new formats coming to market.

The costs of manufacturing, media, recording and mass duplication also play a major role and will be a greater influence as hardware and software resources are concentrated in the same hands. This fact is especially pertinent to DAT which is much more expensive to duplicate because it must be performed in real time rather than at high speed. Thus, this would suggest that DAT will not become a popular low cost format.

As audio technology evolves toward digital techniques, one also sees a convergence of computer and non-computer requirements. The computer represents another consumer product which offers manufacturers the low costs of components and subassemblies for recording and storage of audio. The interfacing of these device into a practical, user-friendly configuration may be our destination.

DIGITAL AUDIO SOURCES

There are currently a number of digital audio systems available. Some are highly developed at this time and others preliminary product annoucements without release dates. Most available digital audio systems were designed for a specific application. The majority are for multitrack or sound editing. These systems and workstations, unfortunately, don't offer much interchangeability but do handle audio production work nicely with digital techniques. These features simplify our jobs once operation is mastered. Presently, we enjoy the co-existence of digital production and analog reproduction over the air waves . And of course, a digital mastering source results in a first generation analog source with maximum audio fidelity.

OPTICAL RECORDING

Magneto-optical recording technology seems to be the most promising longer term. The recordable CD would fit comfortably into our studio enviroment today. However, equipment and media aren't readily available and the costs are extremely high. This new format is in its infancy and will require many generations over many years to get all the bugs out, much less to make it suit broadcast applications. At this point, magneto-optical disks are slower than hard disks. Unclear standards deter drive makers commitment to mass production. "Write-once" and earasable disks represent just 2% of Japanese production. Indications are a few Japanese firms contemplate expansion but decisions to begin will be based on consumer demands[1] and plants will not be operational for several years or more. This is a format to watch if pricing comes down and we see what applications and interfacing are developed. It should be mentioned however, that it is not necessarily in the interest of the owners of copyrighted software to make it easier for the consumer to produce high quality copies of software.

HARD DISK SYSTEMS

Digital hard disk based systems have been around for years. The newer systems main application is for multi-track and sound editing. Minimum requirements for storage space are on the order of 360 mega bites. Hard disk recording systems require lots of memory. Larger hard disks, in the 300 to 600 meg range, offer faster access. Pricing on this type of drive is falling and experts predict a surplus of hard drives within the year.[2] Several of the weak points of the hard disk system are on the mechanical side. They are noisy and have 80-some heads which adds to the question of reliability.[3] Handling digital audio makes hard drives work continuously. Hard drives crash and backup is a must. An entire weeks'/months' commericals on one disk is not a reassuring way to go. The cost of backup is part of the cost of operation. Service of a large hard drive can be a very lengthy process with down time on the order of a week or two.

A 600 meg hard disk costs about $3000 dollars and is coming down.[4] As stated earlier, the systems currently available best address production applications and each uses new and varying interfaces. At this time, hard disk does not represent a practical replacement for the analog cart.

RAM BASED SYSTEMS

The ultimate recording system would have no moving parts or mechanical components, would be very fast and simple to use: Random Access Memory solid state I.C.s. Unfortunately price is a major problem with RAM, currently about $40 dollars per meg. Prices will drop, but today at $120 per spot, that's a lot of cartridges.

There are a couple of plug-in type systems available which use a mechanical/connector interface but repeated handling degrades life expectancy even if the chips remain intact. One means of cost reducing digital systems is to lower the frequency bandwidth enabling a longer recording with the same amount of memory; but you trade off sound quality and pay for digital audio anyway.

R-DAT

At its introduction, R-Dat digital audio was intended specifically for consumer applications. But it was professionals who jumped on its possiblilities only to find that the machines fit limited professional requirements. Many newer R-Dat units now incorporate certain professional features. There is even an add-on unit which makes a consumer R-Dat operate somewhat like a cart machine. One of the shortcomings of this format is the serial nature of the two hour cassette. Most operations do not handle long pre-programmed audio sequences. Accessing spot location in the R-Dat is rapid but not without presentation problems. And as with any new technology and format, new skills and maintenance are required. The digital audio tape system, while robust, is susceptible to dirty heads and misalignment. The audio mute feature blanks out audio completely when serious errors are encountered.[5] Possible program interruption and dead air time limit the desirability of this format for on-air applications. On the other hand, analog carts suffer only partial signal loss due to a dirty head or wrinkled tape but continue to provide audio.

New A to D converters in outboard boxes designed to improve the R-Dat sonic quality are available for about $3500. Obviously, there is room for improvement in digitial audio processing. The debate over digital audio architecture is still on going.[6] "Is 16 bit accuracy enough or is 18 or 20 bit resolution better?" Newer devices improve digital means of signal representation. The main point is that the dust hasn't settled on any one digital audio scheme. A single digital recorder is just that, a recorder designed for one application. Pay attention to the experiences of other end users in their attempts to integrate the digital recorder.

Summary

The future of audio for broadcast is definitely going to be digital. However, I don't think it will take the form of any current digital format. The risk to a manufacturer to develop a digitial broadcast application recorder is substantial and the market small. Perhaps, we will find ourselves not with one universal format but two, as in video, where no single format owns exclusive rights to all applications.

To develop a total new system with its specific application and total cost/performance equal to analog cartridge technology is a challenge. Moreover, the changing state of technology makes putting a finger on a time-table difficult. Engineering design cycles span years from concept to mass production. The standardization and broad base of installed analog cartridge equipment is likely to maintain this format and is the most popular on-air medium for the next 5 to 7 years. The latest generation of cart machines operating with higher output tape is arguably the eastiest, most cost effective means to improve broadcast sonic quality for some years to come. For the forseeable future, I would avoid the risky and expensive plunge into the deep end of digital audio pool.

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APPLICATIONS FOR DIGITAL AUDIO TAPE MACHINES IN RADIO BROADCASTING

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Abstract: As an affordable medium for providing 16-bit digital quality in a variety of audio recording and replay functions, Digital Audio Tape has seen growing acceptance throughout the radio broadcast community. DAT also provides additional features and functions not offered by conventional analog reel-to-reel and cart machines, including accurate location and cueing to selected PNOs and Absolute Time position.

INTRODUCTION

There can be no denying that digital technology continues to impact just about every aspect of the broadcasting industry, from portable recorders to direct broadcast into the home — a development that can only be a few years into the future. And it is the rare AM or FM station that does not use Compact Disc as its preferred source of music playback — even if the material is first transferred to reel-to-reel and NAB cartridge, for compatibility with existing formats and/or automation systems. Within the radio production studio, hard-disk recorders and random-access editing systems offer enhanced flexibility for creating jingles, commercials, station IDs, and so on. High quality can be further ensured by using digital STLs, distribution networks and back-hauls to carry the mono or stereo signal into and out of the station, and between production centers.

But if Compact Disc has whetted out appetites for the enhanced signal-to-noise, distortion, wow&flutter and frequency response performance of 16-bit digital, cost-effective hardware for recording and replaying broadcast audio signals is still restricted. The currently available choices include:

1. Digital open-reel recorders (either DASH- or PD-format) costing upwards of $20-30k, but which provide razor-blade editing, selectable 44.1 or 48 kHz (and, on some models, 96 kHz) sampling frequencies, and timecode synchronization.
2. Digital audio processors and companion half-inch or U-Matic video recorders, costing between $17k and $25k for a basic record/replay system.
3. Consumer-format digital processors and companion VHS/Betamax VCRs, costing between $3k and $7k for a base system.
4. Digital Audio Tape (DAT) portable and studio machines, costing between $1.6k and $12k, some with IEC-Format timecode, others designed for sample-accurate editing with variable crossfade times, and other features.
5. Disk-based systems, including hard-disk and erasable magneto-optical systems, which can cost between $15k and $50k, including PC controller, dependent upon the amount of storage capacity, control flexibility and user features.

IMPACT OF DAT MACHINES

Because of their relatively low cost and advanced operational features, DAT machines have more of an immediate application within the broadcast environment than many other digital formats. In a growing number of situations, the enhanced technical performance, reliability, fast cueing, simple operation, two-hour record capacity and cost effectiveness, have made portable and studio DAT machines the natural choice for improving the overall audio quality and efficiency of a radio station's output.

In addition to selecting DAT to provide enhanced record/replay quality for critical recordings, such as classical performances, there are additional features of that make DAT machines even more appropriate to the broadcast environment. Along with the PCM audio material recorded onto the tape, a DAT player also records various location and timing information within a dedicated subcode section. Just as commercial Compact Discs contain index points that identify the start of each track or music section, so a series of Start IDs can be recorded onto a DAT tape along with the audio material to mark chosen sections. Many DAT machines provide high-speed, direct access to these Program Numbers (PNOs) and Absolute Time positions (in hours, minutes and seconds). Newer DATs also feature remote control ports which, via suitable software, provide direct cueing to Absolute Time locations to an accuracy of ±1 DAT Frames (3.33 milliseconds).

It is this ability of a DAT machine to quickly cue themselves to a Start ID or Absolute Time location that opens up a virtual galaxy of opportunities for the broadcast community. Unlike recording studios, a radio facility needs to be able to access literally hundreds of discrete audio events — be they music elements, commercials, station IDs, or whatever — any of which might be replayed in random order throughout the day. (Because most professional CD players and DAT machines are also equipped with compatible digital input and output ports, first-generation air tapes can be made directly from commercial Compact Discs.)

And within the broadcast production studio, DAT machine can provide access to a library of mono/stereo music beds and narration elements that can be replayed as necessary to build up the new commercial, jingle, station ID, and so on.

SUMMARY OF POTENTIAL APPLICATIONS

Within the broadcast environment, there exist many tasks for which DAT machines represent an ideal choice, including:

•• Automated random-access to libraries of DAT tapes, to provide fast access to up to 120 minutes of stereo material stored on multiple machines in both automated and operator-assist facilities, including commercials, jingles, station IDs, PSAs and music, as a potential high-quality replacement for NAB cart machines. Via suitable software, a menu of available titles can be accessed from a CRT, and used to generate a customized play list of tape selections. These sequences can then be set up to replay continuously, or triggered as individual events against a master clock.

•• Satellite Networked Broadcasting, to provide computer-programmable, random-access playback of master material at origination studios, for streamlining the playout of music tracks and linking voice messages, as well as commercials and news bulletins (where appropriate). In addition, DAT machines can be used to automate the downloading of digital-quality material from an originating service (including full remote control of START/STOP functions at recording site, plus monitoring or record quality during transfers.

•• Fast and convenient inload and archive to/from disk-based Digital Audio Workstations, from production tracks to the edited and mixed masters, including remote control of all DAT functions during the input and mastering of stereo music, narration and other elements.

•• Remote control of record/replay functions during the production of radio commercials, PSAs, jingles, etc., to allow overdubbing against music beds and voice tracks, or to provide duplicate copies of a final mono/stereo production.

•• Syndication via DAT, with integrated master/slave control for simultaneous duplication of several dozen DAT tapes simultaneously, including monitoring of record levels and other system parameters during analog/digital transfers.

DIGITAL HARDWARE REQUIREMENTS

Recurrent within these and other broadcast applications, there are four primary requirements that must be satisfied by any recording/playback system:

1. A high degree of reliability.
2. Full remote control of primary transport and mode functions.
3. Fast and accurate cueing to a designated tape location.
4. Quick start-up times from STOP and PAUSE-PLAY modes.

And, in this day and age of computerized billing and traffic control, we might choose to add integrated logging of all playback events against a master clock, for documented confirmation that each scheduled commercial or audio source did indeed air at the correct time.

Reliability: Given that DAT machines have now been available in professional formats for over three years, the first criteria of reliability is well proven. Many decks feature built-in error indicators, and one or two even provide front-panel display of interpolated error rates for each DAT head. (As with any tape-based system, regular head cleaning and use of good-quality cassettes can dramatically reduce oxide shedding and head-clogging problems.) Newer-generation systems also feature serial control interfaces that allow each deck to be polled for their current error-rate status.

Remote Control: Many of the second-and third-generation professional DAT machines feature some sort of remote-control capability, ranging in complexity from simple switch closures for basic STOP, START, RECORD, PAUSE, FFD/REW and SKIP FORWARD/BACKWARD functions, for example, to a medium-complexity parallel port that might offer up to 30-40 discrete control functions via combinations of switch closures and/or TTL-type level changes. (Although infra-red remote controls are convenient within a small production-studio environment, their usefulness is limited within an air studio or automated facility.)

One drawback with the majority of remote-control schemes, however, is that they cannot provide sufficient combinations of commands to handle more than a basic set of pre-determined circumstances, and they do not provide data *from* the device being controlled. (In other words, once a machine has been issued a command, it is assumed that the instruction was received and acted upon.)

Bidirectional nine-pin serial ports provide two, complimentary forms of remote control:
1. *System Commands* from a central remote controller or computer-based automation system that cause the deck to enter a designated transport mode, or perform a system function, such a RENUMBER or MUSIC SCAN. Such commands can also instruct the DAT machine to locate at high speed to a designated PNO (Program number), and enter PLAY-PAUSE mode, ready for a subsequent PLAY commands to begin playback.
2. *Interrogation Commands* that cause the deck to output a variety of data, including its current PNO and Absolute Time locations, digital replay level, error-rate, and so on.

Fast Cueing: When instructed to locate automatically to a designated PNO, most DAT machines perform the high-speed search at 250- or even 400-times normal playback speed, which means that any point on a DAT tape can be reached within about 30 seconds (usually less). In addition, if the deck is equipped with a serial interface capable of outputting Absolute Time data, then via suitable operating software individual decks could be cued to within a DAT frame (3.33 msec).

Quick Start Times: Accurately and quickly locating to a chosen tape position — either a pre-recorded Start ID, or a H:M:S:F Absolute Time location — is only part of the story. Once that chosen location has been reaches, and the DAT machine cued in PAUSE-PLAY mode, it should output audio within a reasonably short time of receiving a PLAY While the start-up time of a typical DAT machines is of the order of 400-750 msec, for many applications this is sufficiently fast for replaying music, jingles and commercials. If tighter cueing is necessary, then the remote controller or automation system can be set up to "pre-roll" the tape slightly ahead of time, so that it is outputting audio at precisely the correct time.

It should be recalled that most CD players have an average start-up time of between 100 and 250 msecs, while NAB cartridges, dependent on their vintage and level of mechanical adjustment, offer start times of between 150 and 500 msecs.

IMPLEMENTING SERIAL CONTROL

Fortunately, there now exists a new generation of DAT machines that offer bidirectional serial control of just about every transport and programming function, and allow external remote controllers and computer-based systems to interrogate the deck for a variety of extremely useful location and diagnostic information.

For example, the DAT machines can be instructed to output the current transport mode and/or replay level. Such information would allow the controlling software to check that the designated machine in a programmed sequence actually entered PLAY mode, for example, and could also check that the correct levels are being output. (Or it might even use this data to control a VCA-based level control section to ensure correctly matched audio balances.)

DIGITAL AUDIO TAPE: A Technical Summary

DAT is a generic term used to describe two-channel studio and portable recorders that utilize either rotary- or stationary-head transports, and a small, cassette-style tape housing. "R-DAT" and "S-DAT" is the correct nomenclature to distinguish between these different designs.

First-generation R-DAT decks share a common ancestry with consumer machines; current machines now offer features and functions designed specifically for professional broadcast applications. S-DAT machines utilizing similar sized tape housing are still being developed; the recently announced DCC (Digital Compact Cassette) consumer format from Philips will utilize a stationary-head transport design.

R-DAT (or, more usually, "DAT") machines include video-style drum assembly normally fitted with two record/playback heads, and which rotates at a rate of 2,000 rpm. (Four-head decks provide off-tape or confidence monitoring, plus enhanced editing and other features.) Drum diameter on first-generation machines is 30 mm; both larger and smaller diameters — particularly for portable recorders — are also possible. Tape width is 0.15-inch, slightly wider than analog Compact Cassette. Speed is 0.32 inches/sec, about 1/6th that of a cassette.

Tape is laid around the rotating head drum for 90 degrees, in a spiral or helical wrap. This configuration minimizes stress on the DAT tape, and also allows for high-speed forward and rewind functions without having to unwrap tape from the drum.

As can be seen from Figure 1, during recordings each head "writes" alternating or interleaving, helical-scan diagonal tracks that lay one beside the other on tape at a 6.23 degree angle. Track width is only 13.591 microns, about 1/10th the diameter of a human hair. Because the gap alignment differs between the two heads differs by ±20°, each effectively ignores information written by the other.

The complete DAT signal includes **Subcode**, ATF (Automatic Track Finding) and **PCM Audio** areas. Two ATF sections provide redundant information to ensure correct head tracking, and do away with the need for control tracks. The two redundant Subcode sections allow various **Start IDs** to be recorded and erased as necessary; also recorded here is **Absolute Time** data and **Program Numbers** for fast cueing. Start IDs mark the beginning of a tape cut or musical section, and enable fast search location at up to 400x normal playback speeds.

Also included within the PCM Audio area are other subcode data that identify the **Sampling Frequency** and **Quantization Format**, number of channels, anti-copy status, and the presence/absence of emphasis.

As with any tape-based system, DAT is subject to several kinds of playback errors. Random errors appear because of crosstalk between tracks, incompletely erased signals, and mechanical instability; burst errors result from debris or scratches on the tape, or from clogged heads.

A Double Reed-Solomon Code ensures highly reliable detection and correction of errors; the system can completely restore all original PCM data even if up to 0.1 inches of the track is missing, and can compensate for data loss of up to 0.3 inches by interpolation.

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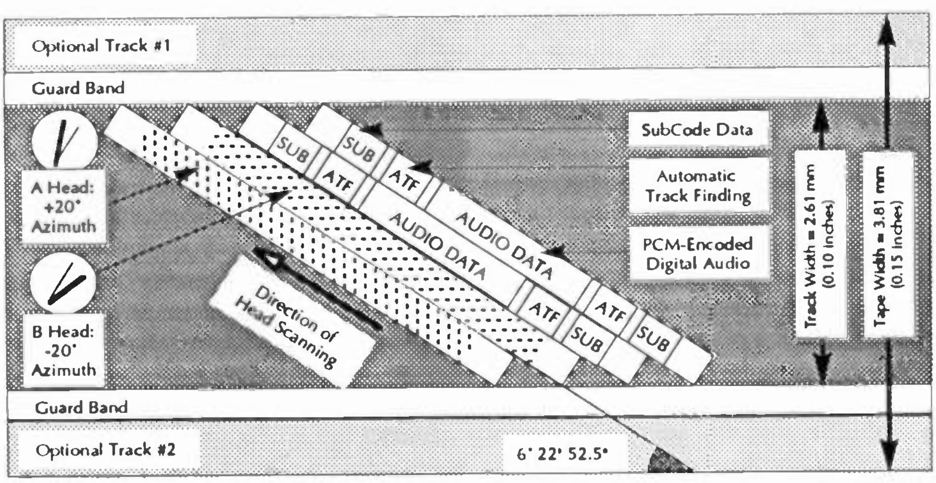

Figure 1: Format of helical-scan Digital Audio Tape, showing alternating diagonal tracks and subcode data areas.

Of the various serial interfaces schemes that might be used to control professional DAT machines, two protocols are beginning to impact the radio broadcast community: ES-Bus and P2. Developed jointly by EBU (European Broadcasting Union) and SMPTE (Society of Motion Picture and Television Engineers), the ES-Bus Control Protocol enables integrated control of multiple DAT machines, controllers and automation systems connected on a serial network. The P2 protocol, on the other hand, is intended for simpler point-to-point or single-controller systems; it utilizes the same connector and electrical characteristics As ES-Bus, as well as similar bidirectional command sequences.

Both of these interface schemes enable full remote control of DAT players, in addition to allowing a variety of information — such as transport status, PNO and Absolute Times, plus level/error data, for example — to be passed back from an assigned transport to the central controller.

In a "typical" broadcast application, as shown in Figure 2, the master controller might be connected via an ES-Bus-compatible bidirectional interface to up to as many as a dozen (or more) serially-equipped DAT machine loaded with the station's play list of music cuts, commercials, jingles, IDs, and so on. The controller can be set up to initiate START/STOP sequences, according to the system requirements, and also cueing the currently off-line machines to a required location, so that they are ready to start when the current segment ends.

During duplication of syndication DATs, or while preparing pre-sorted "submasters" of a station's play list or current jingle package, the serial controller system can be set up to record, let's say, 15 seconds of the dubbed material, then rewind each tape and check the error rates of each machine in turn, If these values are within a

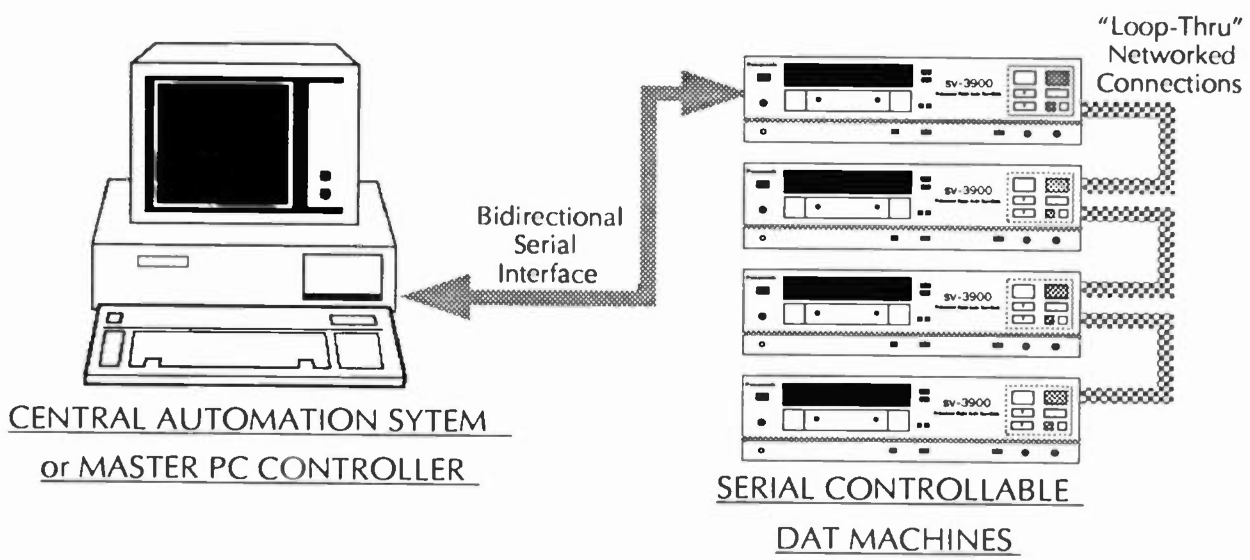

Figure 2: Use of a bi-directional interface to control multiple DAT machines on a serial network.

Under computer-programmed control, a bank of serial controllable decks would provide fast, random-access to libraries of pre-recorded DAT tapes containing libraries of music cuts, commercials, stations IDs, and other audio material. Information from the DAT machines enable automatic logging of each replay event.

For automated satellite-delivery systems, the audio output from each machine can also be continuously monitored via the serial port, to ensure reliable sequencing of music and commercials.

pre-set tolerance window, the system could then rewind all tapes loaded into the various DAT machines, and begin the final analog/digital copying process. During these transfers, the transport status of each slave machine could also be checked sequentially, to ensure that each copy has been made in its entirety.

For controlling a collection of DAT machines during duplication and other dubbing applications, various ES-Bus commands allow selected decks to be assigned to a Group; a single STOP, START or RECORD instruction will cause all machines assigned to a specific group to respond simultaneously. During satellite distribution of syndicated programing, for example, Telco lines could be used to carry serial instructions to the individual DAT machines being commanded from the upload site. Once all machines are on-line, loaded with tape, and ready to roll — possibly following a short check of record/replay quality — then a single command would trigger recording of possibly several hundred decks assigned to capture the incoming material.

ENHANCED SOFTWARE CONTROL

It goes without saying that serial-controllable DAT machines are only as intelligent as the software controlling them. ES-Bus and P2 protocols, for example, are easy to implement, comprising several dozen commands made up of two or more bytes, plus a checksum, bundled into asynchronous data packets running at 38.4 kbps, with either even or odd parity. Basic P2-compatible transport commands fall into a simple two-byte command category, while more complex "Go-to-PNO" and interrogation commands, plus specific commands to individual or groups of machines under ES-bus networked control, might consist of up to 10 bytes of data.

Most PCs and automation controllers can be set up to output ES-Bus and P2-compatible commands from their serial ports. To allow reliable operation with extended cable runs, balanced RS-422 interface is the preferred serial configuration; it is also possible to implement single-ended RS-232C I/Os by rewiring the appropriate connectors, or possibly using a simple interface coupler.

Controlling software can be written in a variety of languages, including assembler or machine code. The majority of broadcasters, however, will probably utilize high-level languages, such as Quick-BASIC, Pascal, Think and Lightning C, etc., which are easier to develop and debug. Use of high-level languages also greatly simplifies the development of suitable screen display and keyboard/mouse input routines.

TOWARDS THE FUTURE: INTEGRATED BROADCAST SYSTEMS

With an increasing number of broadcasting components now being offered with comprehensive serial-control interfaces, it is inevitable that more and more radio stations will be looking to provide integrated system control of various audio record/replay and system functions. Once a master PC or custom-designed microprocessor controller has been developed, it is reasonably easy to develop the appropriate software utilizing industry-standard protocols to implement full remote control of the various system elements. Programmable signal processing equipment, including equalizers, signal delays and routers, equipped with serial control or even MIDI interfaces, will allow additional components to be controlled from a central PC-based system.

Digital Audio Tape represents a cost-effective, programmable medium for bringing digital-quality recording and playback to the radio broadcast industry.

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A WORLD'S FIRST STUDIO ACOUSTIC DESIGN

Brian McGettigan
Radio New Zealand
Wellington, New Zealand

Radio New Zealand's existing building built in 1930 was outmoded and due for demolition by the owners in April 1990.

On September 1st 1989 Radio New Zealand purchased an existing building used for Kodak film processing. The design team were briefed and instructed to have all the operational control rooms and studios ready for occupation by Easter 1990 just eight months away. Thirty one control rooms and studios designed and built in eight months as well as the office space for 160 people. A tall order!

A team of myself as Acoustic Consultant with Architect Grant Copland and Mechanical Services Consultants set to work with eight months to plan and build a complex that under normal circumstances would have taken up to two and a half years.

We were given until December to have the first control rooms constucted.

To achieve this, six control rooms and eight voice booths each with double isolated walls and ceilings weighing between five and ten tonnes, were built 100 kilometres away and transported complete with all wiring and control desks, lifted and positioned by crane and gently eased into the building.

The facilities included a music recording complex in which we have introduced to the broadcasting world a new dimension in music studio development, with one studio able to be used for functions that traditionally would have required at least three studios. Because of the technique used, we were not only able to reduce the space required by the three studios to that of one, but were also able to reduce the size of the remaining one studio by 30%. Required studio floor area was thus reduced by a factor of approximately 4 whilst still retaining full functionality, resulting in a considerable decrease in the capital cost of the complex.

The Variable Acoustic Studio system has enabled us to vary the reverberation time in the new studio from 0.5 to 1.5 seconds with any section able to be live or dead with varying amounts of diffusion. All changes can take place in a matter of minutes under computer control, with changes also stored on computer so that repeat recordings of tracks can later be re-recorded in an identical acoustic environment.

INTERACTIVE TELEVISION

Thursday, April 18, 1991

Moderator:
William Loveless, Bonneville International Corporation, Salt Lake City, Utah

INTERACTIVE TELEVISION TECHNOLOGIES: AN OVERVIEW *
Dr. Diana Gagnon
Consultant
New York, New York

THE ANSWER TO TV RESPONSE SYSTEMS
Fernando Morales
TV Answer
McLean, Virginia
and
Harold L. Kassens and Howard T. Head
A.D. Ring, P.C.
Washington, District of Columbia

THE INTERACTIVE NETWORK
Robert Brown, Ph.D.
Interactive Network, Inc.
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T-NET INTERACTIVE TELEVISION*
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CABLE AND INTERACTIVE TELEVISION*
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Cable Television Laboratories, Inc.
Boulder, Colorado

*Paper not available at the time of publication.

THE ANSWER TO TV RESPONSE SYSTEMS

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TV Answer
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Harold L. Kassens and Howard T. Head
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Washington, District of Columbia

Abstract - TV Answer has developed and is currently testing a new system of interactive television which offers a wide range of services to the home viewer. The system works equally well whether the home receiver is connected to cable, satellite, rabbit ears or a roof-top antenna system.

The TV Answer system does not require the use of two-way cable or telephone lines to provide the interactive capability. A TV Answer home unit at the television set provides all required circuitry and computer logic to control the receiver, and is linked to a nearby "base station" by a narrow-band channel at 218.25 MHz, using PPM in a half duplex mode. Cellular technology permits reuse of this channel as required. At each base station, communication with TV Answer Headquarters is established by satellite using a VSAT.

This paper describes the functioning of the system, and illustrates the method of providing the control and communications.

Introduction

In December 1987, TV Answer filed with the FCC a petition for rule making (RM-6196) requesting an allocation of 500 kilohertz of spectrum from 218.0 to 218.5 megahertz, in order to establish a two-way interactive television viewer response system. In its filings with the Commission, TV Answer has set forth the details of operation of its system. On January 10, 1991 the Commission granted the TV Answer petition to the extent of issuing a notice of proposed rulemaking. The Commission proposes to allocate the 500 kHz band for an "Interactive Video Data Service (IVDS)" and to set forth proposed rules under which the service will operate. The TV Answer system will operate within the confines of these proposed rules.

System Operation

In some respects the TV Answer network is similar to a cellular phone network. Individual cell sites or "base stations" are installed to provide service to a market area. Each cell site has a maximum service capacity of 10,000 households. At the heart of each cell site is a VSAT (Very Small Aperture Terminal) two-way satellite earth-station, which provides two-way communications between TV Answer national headquarters and each cell site. Headquarters sends TV program listings, interactive commercials, order forms for goods and services, news about TV Answer, updated memory card information and order confirmations to each cell site.

The VSAT and transmitter/receiver at the cell site provide a link between headquarters and the home response unit or "box". This loop also relays viewer responses from home units to their respective cell sites. In this way viewers' orders are placed and other information such as down-loaded TV listing information, memory updates and order confirmation information are exchanged between the cell site and the home response units.

Viewer responses to commercials, service offers or product orders are collected by the cell site receiver and are relayed via Hughes satellite to TV Answer headquarters where they are processed. Appropriate collection, ordering and billing activities are performed either by TV Answer or by other parties related to the viewer's responses. The same down-link information received by TV Answer headquarters is also available to program originators. Using their own VSAT earth-stations, interactive service providers such as television networks, shopping services, food sellers or pay-per-view providers can receive viewer responses at the same time as TV Answer headquarters.

Each cell site contains, in addition to its satellite and computer equipment, a transmitter with output power variable up to twenty watts capable of operating at any designated frequency between 218.0 and 218.5 MHz, a high gain receiver covering the same frequency range and also transmitting and receiving antennas mounted at an appropriate height above the cell site.

The in-home response unit is a transmitter-receiver-computer device which operates at the same frequency as its associated cell site with a power which is also variable from one to twenty watts. The unit is normally located atop the TV set and contains a quarter-wave whip antenna. The computer controls the television and other related inputs and transmits response information based on commands from the viewer using the television remote control or a hand-held, infra-red trigger-driven joystick device.

In response to queries from the cell site which appear on a viewer's screen, and in conjunction with control codes transmitted from the cell site, the TV Answer unit in the home awaits the selection by the viewer of an appropriate choice using the hand-held device. The choice is then stored in the internal memory of the response box. At a predetermined time relative to a reference established by the outgoing codes, each individual response box transmits a sine-squared pulse approximately 50 microseconds in duration at the proper RF frequency. The exact time of transmission is chosen so that pulses are received sequentially at the cell site from all responding boxes.

Prior to the actual use of the response units in the query mode, a calibration cycle establishes both the presence of the response box in the system and the round trip time delay which occurs because of propagation factors and the distance of the response box from the cell site. At the start of this calibration cycle, a unique, box-specific code is transmitted from the cell site resulting in a responding transmission from an individual response box independent of the user input. The response unit will respond to the calibration inquiry provided it is powered-up and is receiving the outbound data and associated codes. The total round-trip time from the initiation of calibration to the reception of the response pulse is measured, transmitted to the box, and stored within each unit for future use in establishing the response sequence of all boxes in the system.

Software has been developed which can be used during the calibration cycle to adjust the actual transmitting power level of each individual response unit. Although the peak power which can be transmitted from each unit is presently designed to be 20 watts, this power can be varied under control of the software within the unit. In order to use the minimum transmit power necessary to establish a reliable link, the calibration cycle tests for responses from each home unit beginning with the lowest possible output power. If no confirmation of receipt of signals is received at the cell site, on the next calibration cycle the power will be raised to the next step and calibration will be attempted again. Once a valid response pulse has been received and confirmed, the home unit will transmit all future responses at the selected power level and a new power level can be established only during a recalibration cycle. If a link is not established at any power level, the response box is internally disabled until manually reset.

Response Pulse Propagation

The ultimate performance of the entire TV Answer system is determined primarily by the ability of equipment at the cell sites to receive and decode the response unit pulses. Experience with TV Answer's experimental system has shown that the practical limit of receive threshold is approximately -90 dBm (31.9 dBu) given the typical noise environment found in urban areas. Considerable research and extensive field experience have shown that the losses encountered over actual propagation paths are substantially greater than those encountered in free space. These losses, due to such factors as atmospheric absorption, obstructions, multipath and diffraction, have been found to produce attenuation over even relatively short paths of at least 20 dB over free-space losses. These losses increase substantially when a small indoor antenna is employed atop the box and the propagation path from the transmission antenna to the receive antenna must penetrate building walls. Using even a conservative estimate for building attenuation of 20 dB and additional path losses of 20 dB for a moderate length path at 218 MHz, the practical limit on the range of a TV Answer unit with 20 watts is approximately 10-12 miles using a well designed cell receive site.

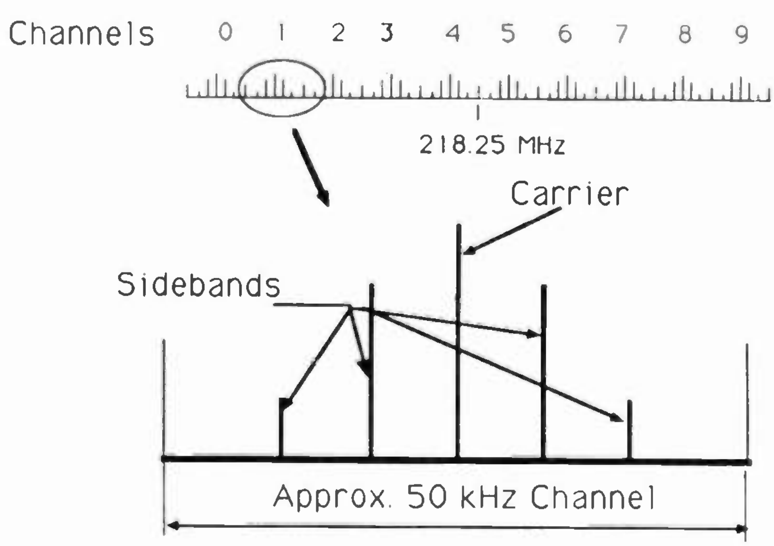
Channel Allotments & Timing

In requesting the allocation of 500 kilohertz of spectrum space for TV response systems, TV Answer contemplates that the band could be divided into ten 50 kilohertz channels. Assuming, for example, a maximum modulation of 20,208 baud, the Carson rule would indicate a total bandwidth of each channel of 40.416 kilohertz (See Figure 1).

The TV Answer system is designed so that each "frame" consists of 1083 bits or 53 milliseconds. Of this period, 27 milliseconds are allocated to transmissions from the cell site (to correlate and interrogate each home unit) and 26 milliseconds are allocated to all the home units assigned to each cell site (530 bits). Thus the time is shared almost

Figure 1

COMMUNICATION CHANNEL DESIGN CRITERIA



CARSON RULE Example:

Bandwidth=2* highest modulated frequency + 2* peak FM deviation

=2* 20208 + 2* 0 =40416 ------> Approx. 50 kHz

Figure 2

DATA TRANSMISSION FORMAT

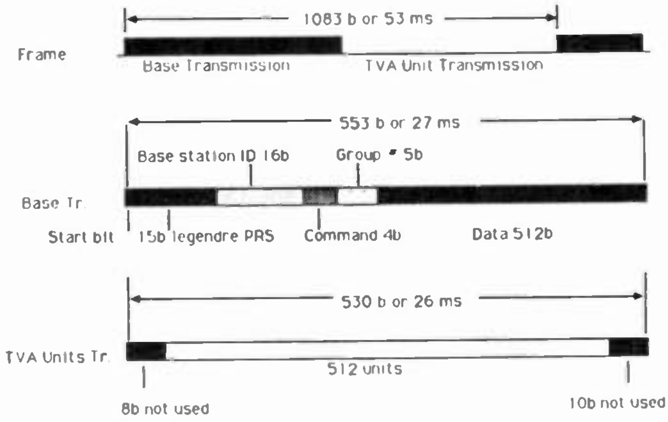
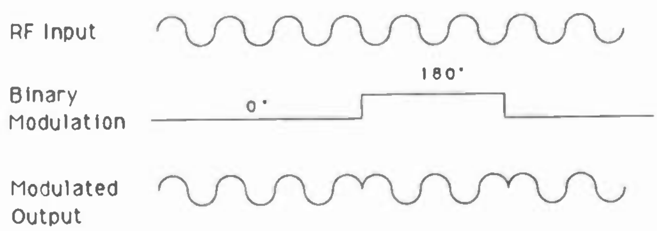
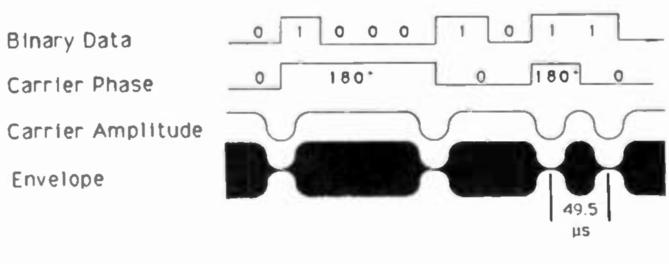


Figure 3

PHASE MODULATION



DATA TRANSMISSION



equally for each frame between the cell site and all of its own home units. (See Figure 2).

As shown in Figure 3, binary information is used to modulate the r.f. carrier of the cell unit transmitter. A change in carrier phase of 180 degrees indicates a change in transmission of zeros to ones, or ones to ones--and a change in envelope amplitude at the transition time.

Figure 4 shows the individual assigned carrier frequency of each cell site and its associated home response units and the timing for the system.

The modulating wave-form of the home unit response pulse is sine-squared and is shown in Figure 5. The pulse has been shaped in order to limit the generation of sidebands and to concentrate spectral energy as close as possible to the fundamental operating frequency. As was indicated in connection with Figure 3 above, the 180 degree phase shift principle used to transmit zeros and ones can also be used in the home unit to send digital replies.

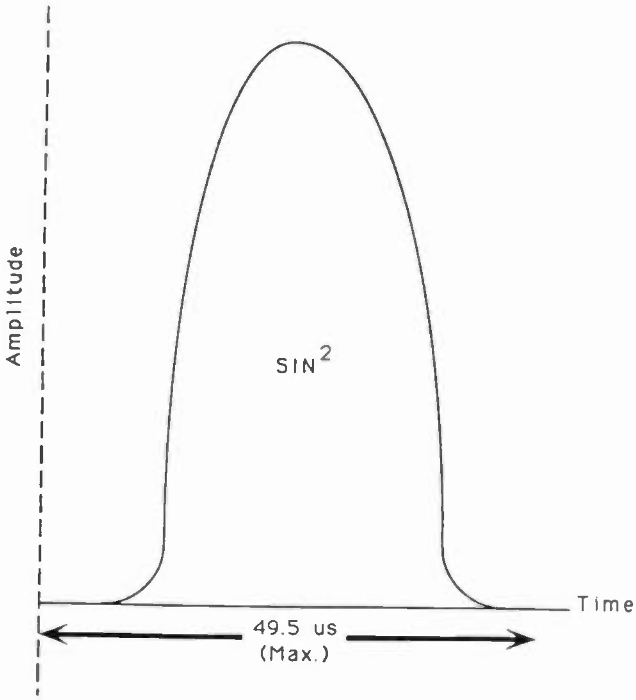
Figure 4

FREQUENCIES AND TIMING FOR THE SYSTEM

CHANNEL		0	1	2	3	4	5	6	7	8	9
FC (CARRIER FRECUENCY)	MHz	218.025	218.075	218.125	218.175	218.225	218.275	218.325	218.375	218.425	218.475
T (MASTER PULSE TIME)	µs	49.5	49.5	49.5	49.5	49.5	49.5	49.5	49.5	49.4	49.4
F MOD (BAUD RATE)	kHz	20187.5	20192.1	20196.8	20201.4	20206.0	20210.6	20215.3	20219.9	20224.5	20229.2
BW (RF BANDWIDTH)	kHz	40375.0	40384.3	40393.5	40402.8	40412.0	40421.3	40430.6	40439.8	40449.1	40458.3
FRT (TRANSMIT FRAME TIME)	µs	27393.2	27386.9	27380.6	27374.4	27368.1	27361.8	27355.5	27349.3	27343.0	27336.8
FRR (RECEIVE FRAME TIME)	µs	26253.9	26247.9	26241.8	26235.8	26229.8	26223.8	26217.8	26211.8	26205.8	26199.8
TF (TOTAL FRAME TIME)	µs	53647.1	53634.8	53622.5	53610.2	53597.9	53585.6	53573.3	53561.1	53548.8	53536.6
FRAME RATE	s	18.6	18.6	18.6	18.7	18.7	18.7	18.7	18.7	18.7	18.7

Figure 5

RESPONSE PULSE WAVEFORM



Channel 13 Interference Considerations

The 216-220 MHz portion of the spectrum is located immediately above the frequencies allocated for Channel 13 television broadcast stations. Due to the inadequate adjacent channel selectivity characteristics of most television receivers, some limitations must be placed on the strengths of other signals within 6 MHz of each individual television allocation.

During the proceeding establishing the Automated Maritime Telecommunication Service (AMTS) (presently authorized in the 216-220 MHz band), the FCC examined the issue of adjacent channel interference to Channel 13. FCC laboratory measurements were conducted in 1975 to ascertain the susceptibility of television receivers to signals in the 216-222 MHz band. Using these measurements, a methodology for predicting the extent of potential interference areas surrounding AMTS land-based control stations was developed. A detailed description of the method established for computing these interference areas is contained in the Commission's 1982 report, "Guidance for Evaluating the Potential for Interference to TV from Stations of Inland Waterways Communications Systems."

Since the 1975 measurements of receiver performance of signals in the 216-222 MHz band were based on only five receivers, and other adjacent channel selectivity measurements are based on even older receivers operating at different frequencies, TV Answer undertook a comprehensive measurement program to ascertain the adjacent channel selectivity performance of more modern receivers.[1] Twenty-eight modern TV receivers were tested under a variety of signal conditions in order to most accurately simulate in the laboratory as many conditions as possible which may be experienced in the field. The tests are similar in nature but more extensive in scope than those conducted by the FCC in 1975.

The primary focus of the tests was to establish the threshold at which interference became "just perceptible" due to a single CW or pulse modulated signal in the band 216-220 MHz in the presence of a Channel 13 signal. Due to the low power and itinerant nature of the TVA response units, the generation of intermodulation, cross modulation and other more complicated interference phenomena was deemed unlikely. Using the results of the measurements of the 28 receivers, and assuming free-space propagation, reveals that in order to protect 90% of the receivers located at the Grade B contour and tuned to Channel 13 would require a cell site transmitter with a radiated power of 1.05 watts. At the Grade A contour the power would be 6.6 watts and at the city grade contour 26.4 watts would be acceptable to avoid interference. Powers between these contours could be adjusted accordingly.

Signals From The Home Units

Transmissions from the home units have characteristics which are different from those of the base station. The transmissions are pulses modulated on the 218 MHz carrier; thus, whereas the base station transmissions approach a continuous-wave condition, the pulse transmissions are brief and have a different appearance from that of CW on a TV screen.

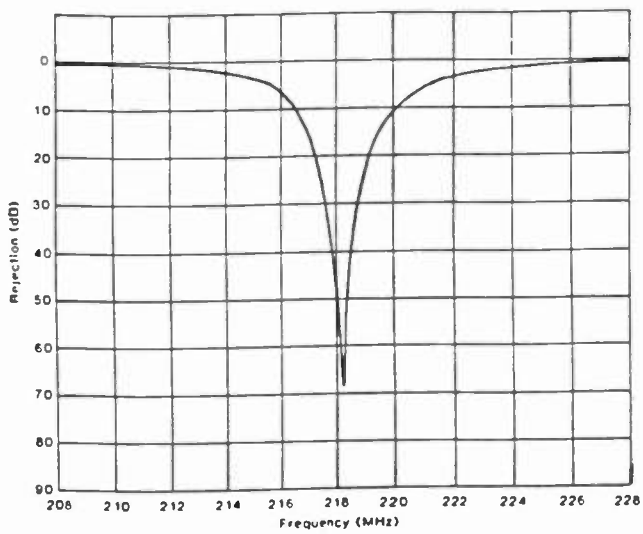
The tests by Bednarek on the relative visibility of both CW and pulse transmissions at 216.25 MHz concluded that a peak pulse power 4 dB higher than CW was required to produce equal visibility on the television screen at the "just perceptible" level. Further tests conducted by TV Answer using 24 non-technical participants confirms the 4 dB difference.

The duty cycle of the pulse transmissions plays an important role in the visibility of the pattern on the television screen. First, a single transmission from the TVA home unit appears to last only a second. Next, the TVA subscriber may be expected to generate pulses only on occasion and not constantly. Finally, any effect of the 218 MHz pulses will be felt only by viewers whose TV receivers are susceptible to this type of signal who are not connected to cable, and who are watching an over-the-air picture.

Filters

TV Answer contemplates the use of "notch" filters at the television receiver input

Figure 6



MEASURED FILTER REJECTION

Prepared for
TV ANSWER, INC.

A. D. RING & ASSOCIATES, P.C.

terminals wherever needed to reject signals at 218.25 MHz. A graph of the response of one such filter which has been tested and found suitable for this purpose is shown in Figure 6. Viewing tests revealed no perceptible effect on the Channel 13 picture. These filters are quite simple and can be readily installed even by untrained personnel.

Conclusion

In its January 10, 1991 meeting, the Commission stated that it believed that interactive video data systems will provide the public with a convenient method for interacting with all forms of video programming, including commercial and educational broadcast television, cable television and direct broadcast satellite service. As a result of its several years of testing its system, TV Answer is confident that it can more than adequately satisfy the Commission's objectives for the proposed new service.

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THE INTERACTIVE NETWORK

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Mountain View, California

Abstract- Described is a system which will allow the consumer to have interactive TV or radio in his home without the requirement of adding two-way cable, or being continually connected to his telephone. Participants are able to play competitively across the nation for prizes and rankings, as well as give opinions on issues, and receive information services such as news, sports, weather and stocks.

INTRODUCTION

Interactive Network (IN) has developed a Control unit which will allow consumers to interact with their favorite TV programs. This may take the form of playing along with sporting events, for example in football predicting what the next play call will be, or in baseball predicting what the batter will do, or playing along with game shows actually competing with the contestants on the show and with other players across the nation for prizes and rankings. The device can also be used to obtain opinions on issues, vote on how a soap opera should end or who won a political debate, and for delivering information services such as news, weather, sports and stocks.

Up until now, it has generally been required that the consumer be continually connected to a communications link such as a two-way cable system or a telephone line, in order to achieve true interactivity. This may be reasonable for a small number of users or in a restricted market area, but it is not practical if one wishes to quickly reach the mass market of the U.S. The nature of what is felt to be the most interesting interactive applications, however, does not require that the return path be continuously connected. Instead, one can have interactivity with a store and forward approach requiring only one short burst of return path data at the conclusion, yet still achieve a sense of interactivity during the programming, and allow for real competition among the participants.[1]

Basically, the Control unit receives a constant flow of digital information. This information is used during game play to score the user and give him instantaneous feedback on how he is doing throughout the event. The digital link is also used to down load new game software, and can even be used to down load a completely new operating system into the unit. Once the player has completed a session, he connects the unit to the phone line and in a short (10 second) packet data phone call the results of what he has done are uploaded into the IN Central Computer System. Once the results have been collected in the computer, winners can be determined and in a few minutes the names and scores of the winners are displayed on the player's Control unit screen.

Outbound data is sent from the IN Computer via phone lines to a network of FM stations and then transmitted on a subcarrier to the Control unit. The results are sent back to the computer via a Tymnet packet data system. In addition to the FM subcarrier network, there is an entirely separate method of delivering the same data to the user. This method is on the Vertical Blanking Interval (VBI) of the PBS TV system. This second network is used to service those consumers who may not be in a major market area covered by FM, or who may be in a fringe area. Data is sent via phone line to PBS in Washington D.C. and inserted on their national distribution feed. In order to receive the VBI signal, the user must use a second piece of equipment called a Booster.

THE HARDWARE

The Control Unit

The Control unit has many of the elements of a lap top computer, such as the CPU, memory, keyboard, display and modem. In addition it has three unique features. One is a digital FM subcarrier data receiver with forward error correction capable of receiving 9600 bps. The second is an IR receiver also operating at the same data rate. The third feature is security. At the heart of the unit is a hardware encrypted microcomputer chip which scrambles both the address lines and data lines so that the program and data in memory is not decipherable. Also, all of the information coming into and going out of the device is encrypted by software. A block diagram of the Control unit is shown in Figure 1.

The Booster

The Booster is essentially a repeater. It receivers VBI data and relays it to the Control unit. It is comprised of a TV tuner, IF, demodulator, teletext decoder, CPU and IR transmitter. One full line of the VBI is used and it is modulated at the standard teletext rate of 5.727 Mbits per second. Up to four bits of forward error correction is achieved per line (field), and up to two missing lines in eight can be re-constructed using block error correction. The net throughput is 9600 bps which equals that of the FM network. A unique IR link had to be designed in order to reliably relay the data to the Control unit. The technique floods the room with IR so that the positioning of the Control unit is not critical.

The Central Computer

The Central Computer is a fault tolerant, real time Stratus machine. It is tasked with the job of receiving data from live production consoles, canned files and computer programs and prioritizing and packetizing this

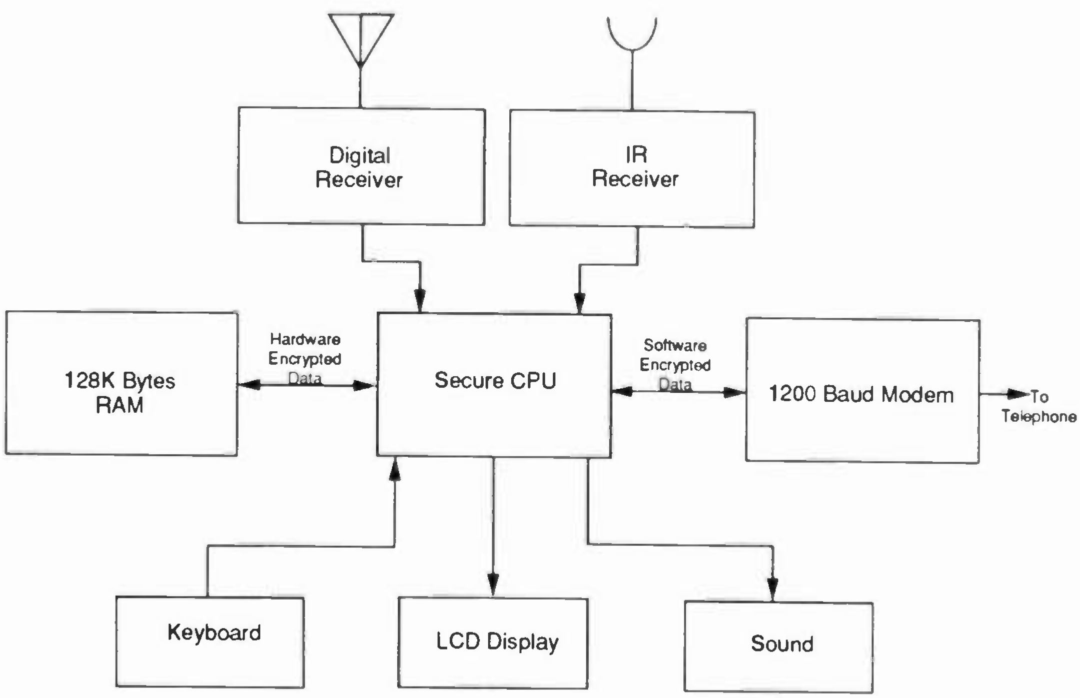

Figure 1 INTERACTIVE CONTROL UNIT BLOCK DIAGRAM

data in order for it to be sent over the network. It also receives packet data calls from the users, determines winners and broadcasts the results, including a histogram of scores, back to the users. In addition, it archives results to a database.

THE NETWORK

Figure 2 displays a block diagram of the complete network system. As previously mentioned there are two separate methods to deliver the same outbound data to the user. Also shown is the return path via the user's telephone line.

FM

Data is sent from the Central Computer to FM radio stations over telephone lines. In some cases conditioned analog lease lines are used with 19.2 Kbps modems on each end. Since the data must be sent in real time with minimal delay, it was necessary to disable error correction which would have required the modems to resend bad blocks. However, forward error correction was implemented. For the other FM stations, data is sent over Pacific Bell's Advanced Digital Network (ADN) which allows transmission at up to 56Kbps with minor time delays. A monitor system is installed at each FM station for the purpose of transmitting error rate information back to the Central Computer, using a reverse channel. The received data is modulated on either the lower or upper subcarrier of the FM radio station.

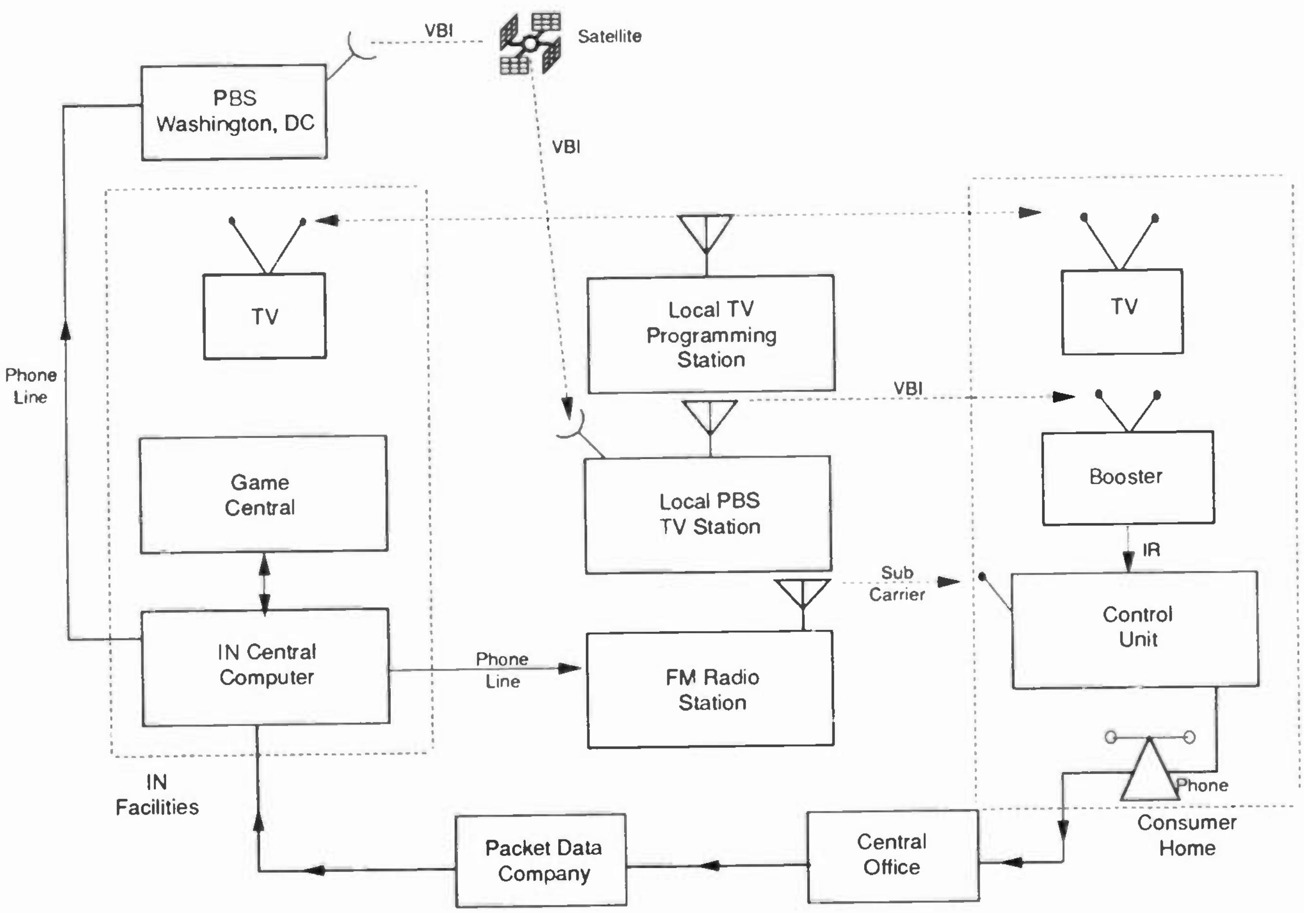

Figure 2 INTERACTIVE NETWORK SYSTEM

VBI
The same data is sent, in a similar fashion, to PBS in Washington, D.C. where it is inserted on their network satellite feed. This data is received by each PBS local TV station. Appropriate stations have data bridges which remove the VBI data from the network feed and re-insert it into the actual video being transmitted.

Telephone
At the completion of an event, the results of game play are stored in the Control unit. This data is sent back to the Central Computer during a five to ten second packet data phone call. The user plugs his telephone line into the unit which has a built in modem. The device automatically dials the nearest Tymnet gateway node. A link is established to the Central Computer, and the data packet is transmitted.

PRODUCTION

Production of a live event requires real time generation of data which is simulcast with the TV or radio program. This data goes out over the FM and VBI network. It may consist, for example, of the result of a play in a football game or the question and answer during a quiz show. The timing of this data is critical in order to allow the user to have enough time to input his prediction, but not too much time so that he can answer after the fact. Any variable, or unpredictable time delays in the network will cause serious problems.

Synchronization of the data is also problematic. This is particularly true for a pre-recorded event where the local station may change the length of a commercial, or even pre-empt part of a program. In these cases, techniques are required to allow re-syncing of the simulcast data.

RESULTS

The Interactive Network System described in this paper has been operational since April of 1990. It functioned in a Beta test mode from April through September for 200 participants in the Sacramento, California area. During this time approximately 22 hours of interactive sports programming (including football, baseball and basketball) was offered per week. The amount of interactive game show programming was approximately 7 hours per week, while stand alone games such as trivia and word puzzles amounted for about 200 hours per week.

Though operation of the system was not completely error free, most of the difficulties occurred at the beginning of the test and were either operational errors, or software bugs. All of these problems were corrected by test end. The operational problems dealt with synching and time delays as mentioned above, as well as some human errors. All 200 testers were able to receive data, operate the Control unit, play games, score and receive game results.

1 This system is protected under U.S. Patent #4,592,546.

AUTHOR INDEX

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