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42nd Annual Broadcast Engineering Conference

LAS VEGAS, NEVADA

1988 PROCEEDINGS

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National Association of
NAB
BROADCASTERS

Michael C. Rau

Vice President
Science and Technology
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(202) 429-5339

March 8, 1988

Dear Industry Engineer:

"Broadcasting and Democracy: The Winning Ticket," the theme of the 1988 convention, reflects the vital role that broadcasters play in the American election process. As engineers, we face the challenge of maintaining the technical quality of broadcasting, to ensure that broadcasting maintains the quality of democracy.

The papers contained within these proceedings present information on new ideas, technologies and methods that will augment your technical knowledge and skill. We encourage you to read and study these papers. Consider this study an investment in your future -- and the future of your industry.

On behalf of NAB's Science and Technology department staff, I am pleased to present these 1988 Engineering Conference Proceedings.

Best regards,

Michael C. Rau

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THE AM SPLATTER MONITOR

Thomas G. Wright
Delta Electronics Inc.
Alexandria, Virginia

INTRODUCTION

The AM Splatter Monitor is designed to evaluate a transmitter's level of AM interference while avoiding some of the limitations of existing measurement techniques[1]. The development of the AM Splatter Monitor is a result of the National Radio System Committee's (NRSC) desire to more effectively measure the performance of radio stations in evaluating the NRSC preemphasis standard (NRSC standard)[2]. One goal of this committee is the reduction of AM interference[3] (splatter), particularly second adjacent channel interference, using a high performance, 10 kHz low pass filter[4]. The AM Splatter Monitor is ideally suited to evaluate the effects of this filter on transmitted interference.

This paper discusses the nature and effects of splatter and describes the uses of the AM Splatter Monitor compared to other spectrum measurement techniques.

SPLATTER DEFINITION

Splatter is one of those intuitive but ill defined concepts with which every broadcast engineer is acquainted. To the author's knowledge, no technical reference dictionary defines the term. For the purposes of this paper, let splatter be defined as the undesired portion of a station's output spectrum caused by modulation. Using this definition, power supply hum sidebands and carrier harmonics along with co-channel and adjacent channel interference are not splatter. Second adjacent channel interference is splatter. Also, note that the exact definition depends upon the engineer's opinion about which parts of the transmitter output spectrum are desirable. This is natural because the engineer's expectations are partly determined by regulation and partly by existing technology both of which may change. Finally, splatter is defined for the station's output spectrum which includes both the transmitter's output spectrum and the far field spectrum.

THE SPLATTER PROBLEM

Splatter is a problem because it interferes with reception of other stations. Nearly everyone has experienced the steam-locomotive-like sound of second adjacent channel interference while trying to receive a weak AM radio signal, especially at night. This effect is due to splatter from another radio station. The presence of such splatter does not necessarily indicate a violation of FCC emission limitations rules[5] because the receiver's automatic gain control brings up the splatter along with the weak signal.

Several secondary effects of splatter are harmful to the AM broadcaster. The existence of splatter from thousands of radio stations raises the general noise level of the AM band and thereby reduces the quality of AM broadcast programming. Also, splatter is energy wasted because the splatter sidebands are never audible to the station's listeners. In fact, the signals which cause splatter may intermodulate in the transmitter to produce distortion components within the desired portion of the spectrum and, therefore, distortion in the received signal.

SPLATTER SOURCES

The primary cause of splatter is higher frequency audio components at the transmitter's modulator input[6]. These higher frequency audio signals are translated directly into splatter by the normal process of modulation. A typical source of these audio signals is an improperly filtered clipper in the audio processor. Fortunately, the better audio processors incorporate a low overshoot filter to eliminate these clipping products.

Other sources of splatter are overmodulation, improper use of the transmitter's protective clippers, distortion and noise in the modulator, incidental phase modulation (IPM), and improperly operated AM stereo. In the case of incidental phase modulation, the resulting phase modulation sideband pairs would not, if left undisturbed, affect receiver envelope detectors. However, these sidebands are disturbed by every tuned circuit all the way through to the detector, especially the asymmetrical skirts of the IF bandpass. So some of this sideband energy is converted to AM sidebands which are detected as distortion. This is why reduction of IPM by proper transmitter neutralization improves the sound of AM stations.

MEASUREMENT TECHNIQUES

The regulations governing emission limitations[7] do not specify the monitoring equipment to be used or the frequency of measurement but specify only that the broadcaster must not violate the internationally agreed upon spectrum limits. Thus, strictly speaking, the broadcaster must guarantee at all times that he is not violating these limits. In practice, however, the spectrum is checked only periodically, perhaps once a year, using a rented or borrowed spectrum analyzer or wave analyzer and the assumption is made that the spectrum is acceptable at all other times. Until now, this was the only practical recourse available to the broadcaster due to the high cost of the necessary measurement equipment and the requirement for competent technical people to operate the complex equipment.

Other equipment readily available, such as communication receivers and field strength meters, are not suitable for close-in spectrum measurements because they lack the necessary dynamic range and selectivity. Even a high quality spectrum analyzer has the limitation that as it sweeps through the measurement band, it looks at only a small segment of the spectrum at any given time. Thus, the spectrum analyzer would not record the existence of a burst of splatter at other segments of the measurement band[8].

THE AM SPLATTER MONITOR

The AM Splatter Monitor is a dedicated, specialty device primarily intended for full time measurement of the spectrum segments between 11 kHz and 100 kHz away from the carrier on both sides of the carrier. The AM Splatter Monitor measures splatter level and any spurious emissions which fall within this spectrum segment. The AM Splatter Monitor is an economical device designed to fit within the budget of an AM broadcast station. Figure 1 is a front panel view of the AM Splatter Monitor.

Because splatter level normally decreases with frequency away from the carrier, the AM Splatter Monitor measures the most important segment of spectrum associated with splatter. This same segment of spectrum is where the changes in splatter level occur. These changes are due to factors such as shifts in modulation level, changes in program material, audio processor adjustments, and tube aging. The AM Splatter Monitor has an alarm that may be set to detect such changes. The station can use this alarm through a remote control system to immediately signal the occurrence of a splatter problem.

The AM Splatter Monitor is normally installed in a rack at the transmitter site to continuously monitor the transmitter's output

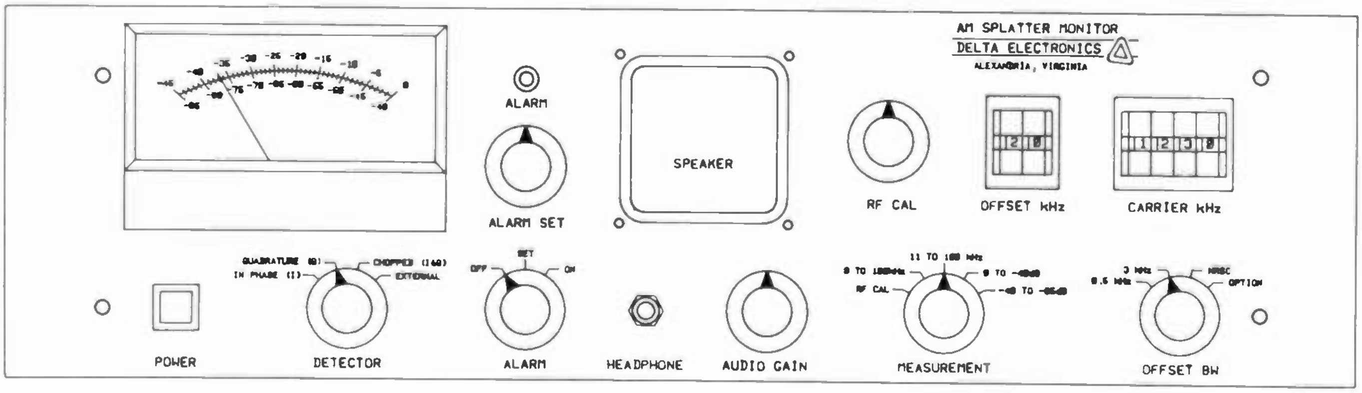

FIGURE 1

spectrum. Although the regulations regarding emission limitations require field measurement to assure compliance[9], the intervening elements between the transmitter's output and the far field are usually quite linear so continuous monitoring of transmitter's output is a reasonable indication of operational compliance. The AM Splatter Monitor is portable, and may be removed from the rack for field monitoring to assess compliance of the close-in spectrum (within 100 kHz) to emission limitations rules[10]. The unit may also be used for field monitoring in the strong signal areas of other AM stations to investigate interference complaints. For these purposes, the AM Splatter Monitor derives power from an automobile's cigarette lighter jack (+12V) and receives its RF input signal from an optional, active antenna.

Figure 2 is a simplified, functional block diagram of the AM Splatter Monitor. The reader is encouraged to refer to this figure while reading the following description.

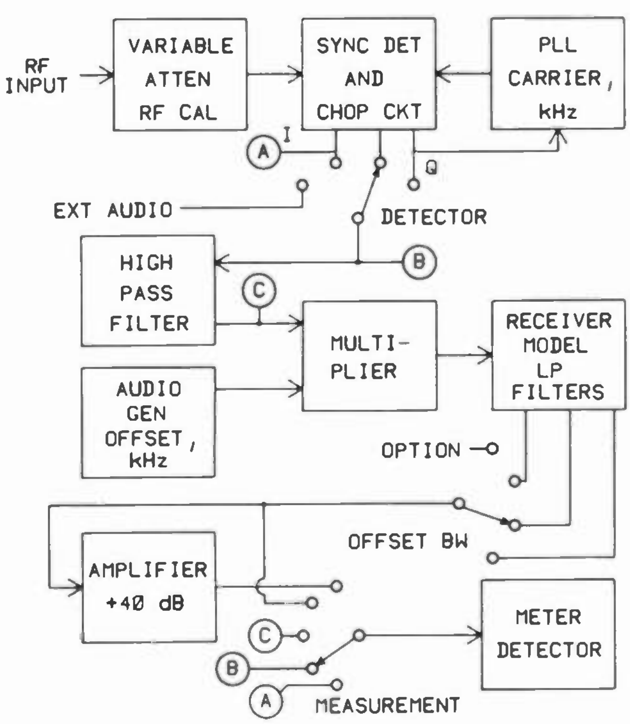

FIGURE 2

A 3 1/2 digit thumbwheel switch labeled CARRIER kHz adjusts the operating frequency of the AM Splatter Monitor from 450 kHz to 1700 kHz. Simple crystal and jumper changes allow operation at either 9 kHz or 10 kHz channel spacing. When tuned to 450 kHz, the AM Splatter Monitor can be connected to the 450 kHz IF output of a synthesized receiving, taking advantage of the AM Splatter Monitor's synchronous detectors either to evaluate receiver performance or for off the air monitoring. In monitoring applications, first evaluate the receiver using an amplitude modulated, signal generator and also evaluate the signal generator directly with the AM Splatter Monitor.

The AM Splatter Monitor uses high performance in-phase and quadrature synchronous detectors. The output of each synchronous detector is available on the rear panel. When used to measure splatter, the in-phase synchronous detector measures the splatter due to distortion and clipper products and the quadrature synchronous detector measures splatter due to incidental quadrature modulation which is related to incidental phase modulation. A measure of the overall splatter level requires a combination of the in-phase splatter and the quadrature splatter. A low frequency chopper circuit performs this function. The engineer uses the DETECTOR switch to select the in-phase detector, the quadrature detector, the chopped combination of these two detectors, or an external audio input depending upon his measurement needs. The external audio input is used to analyze the audio source material fed to the transmitter's modulator.

A five position switch labeled MEASUREMENT selects the function of the AM Splatter Monitor. In the first switch position, the AM Splatter Monitor measures the DC portion of the in-phase synchronous detector for calibration of the RF input level (carrier level). The second switch position selects measurement of all signals within 100 kHz of the carrier. The meter, when reading the in-phase detector or the chopper circuit, will typically read several dB down as it measures the desired modulation and splatter. The demodulated signal is audible from the front panel speaker or by use of headphones. In this switch position, the external synchronous detector outputs are used for receiver evaluation or off the air monitoring as mentioned above or are used with an FFT spectrum analyzer[11].

In the third switch position, a sharp high pass filter[12] is inserted so that the meter reads only that portion of the spectrum between 11 kHz and 100 kHz on either side of the carrier. This is a measure of the total splatter produced by the radio station and, unlike a swept spectrum analyzer, all spectrum components of interest are always available for measurement.

In the last two switch positions, the AM Splatter Monitor measures a selected spectrum segment of the total splatter signal in two ranges. The top meter range is elected by the fourth switch position and measures down to 45 dB below the calibration reference. The fifth and last switch position selects the bottom meter range which measures between 40 dB and 85 dB below the calibration reference. The segment of the spectrum selected is determined by a thumbwheel switch labeled OFFSET kHz and by a bandwidth switch labeled OFFSET BW. The OFFSET BW switch selects an equivalent receiver model and the OFFSET kHz thumbwheel determines how far that equivalent receiver is tuned away from the carrier on both sides of the carrier. For spectrum analyzer like applications, the OFFSET BW switch is set to the 0.5 kHz position yielding an RF bandwidth of 1 kHz which matches the step size of the OFFSET kHz thumbwheel. In the 3 kHz switch position, the AM Splatter Monitor responds like a typical narrow band radio. In the NRSC position, a wide band receiver is modeled with NRSC deemphasis. The switch position labeled OPTION allows selection of a customer determined receiver model contained on an optional plug in assembly.

A typical example of the use of the AM Splatter Monitor is monitoring the splatter produced on the second adjacent channels, 20 kHz away from, and on both sides of, the carrier. The OFFSET kHz switch is set to 20 for the required 20 kHz frequency offset. The OFFSET BW switch might be set to the 3 kHz position to measure the level of total splatter energy received by a typical narrow band receiver. According to the emission limitation rules[13], the maximum acceptable splatter level for this frequency is 25 dB below the carrier reference so the MEASUREMENT switch is set to the fourth position for measurements down to 45 dB below the carrier reference. The AM Splatter Monitor's meter must not read above the 25 mark on the top scale (-25 dBc).

The ballistics of the meter's detector circuit are set to match the integration factors of the human ear. Therefore, the meter reading for the example given above is equivalent to the interference level perceived by a listener. This is, of course, exactly the desired measurement if we assume that the purpose of the whole exercise is reduction of objectionable interference. The question may arise, however, of whether this measurement will agree with a measurement derived from some other measurement method. Will a spectrum analyzer, for instance, read the same as the AM Splatter Monitor? The answer is a qualified yes.

In the case of fixed sidebands due to test tone modulation, the peak detector of the spectrum analyzer responds the same as the quasi-peak detector of the AM Splatter Monitor. This is how the AM Splatter Monitor is calibrated. Modulation using the pulsed USASI noise source as recommended in the NRSC standard[14] yields the same readings provided that the spectrum analyzer is used either in the fixed frequency mode or with peak hold over a long time period. The same situation occurs with real modulation so that the qualification mentioned above is that the AM Splatter Monitor produces conservative readings, that is, if anything, higher readings than a sweep spectrum analyzer. Therefore, the AM Splatter Monitor is, in some ways, superior to a spectrum analyzer for this special application.

SUMMARY

Splatter is unwanted spectrum components due to modulation which interfere with other stations and cause distortion of the desired signal. The AM Splatter Monitor measures the level of splatter and can be used to identify and correct the sources of splatter. The AM Splatter Monitor is primarily intended for continuous monitoring of the transmitter output to indicate operational compliance to the regulations governing spectrum limitations[15]. It is also useful for field measurements in strong signal areas. The AM Splatter Monitor's alarm can be used to remotely alert the occurrence of splatter problems. The high performance, synchronous detectors in the AM Splatter Monitor make this unit useful for a variety of other applications.

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[1]Harrison J. Klein, P.E., Modulation, Overmodulation, and Occupied Bandwidth: Recommendations for the AM Broadcast Industry, (NAB September 1986) p. 22

[2]National Radio Systems Committee (a joint National Association of Broadcasters and Electronic Industries Association group), Interim Voluntary National Standard, (January 1987)

[3]op. cit. NRSC p. 1

[4]Ibid p. 7

[5]Federal Communications Commission, Rules and Regulations, Part 73, #73.44

[6]op. cit. Klein p. 23

[7]op. cit. FCC

[8]op. cit. Klein p. 22

[9]*op. cit.* FCC
[10]*Ibid*

[11]*op. cit.* Klein p. 23

[12]Courtesy of Bob Orban, Orban Associates, Inc., 645 Bryant Street, San Francisco, CA 94107

[13]*op. cit.* FCC

[14]*op. cit.* NRSC p. 9

[15]*op. cit.* FCC

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A NEW LOW PROFILE ANTI-SKYWAVE ANTENNA FOR AM BROADCASTING

Basil F. Pinzone, James F. Corum, Ph.D. and Kenneth L. Corum
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ABSTRACT

A new technique for controlling the elevation plane pattern of broadcast antennas is presented. Practical results, measurements and theoretical predictions are evaluated and discussed. It is concluded that the new Corum structure is a viable candidate for solving skywave problems and for providing low profile radiators where traditional antennas are impractical.

Introduction

A novel technique for the control of the elevation plane pattern of broadcast antennas has been developed at Pinzone Communications Products' Newbury, Ohio laboratory and the antenna test range at its rural Windsor, Ohio engineering experimental station. This remarkable new technology makes possible the practical implementation of anti-fade and anti-skywave antennas.

Nighttime skywave interference has continued to be a major plague for the AM broadcast service since the early days of radio. For most broadcasters, skywave radiation presents two problems. First, it represents wasted radiated power. Secondly, because of the legally protected contours of existing stations, high angle radiation severely limits a broadcast station's market place, or primary coverage area. From a pragmatic point of view, if the interferrence-causing high angle radiation could be significantly reduced, in many cases, the transmitter power and ground wave coverage could be dramatically increased with no additional interference to other stations. The potential market place economic impact is obvious.

High angle radiation is also a plague to clear channel broadcast stations. The groundwave daylight service area, for a clear channel station, may exist out to a considerable range. However, the nighttime skywave signal can be greater than the groundwave signal well within the daytime coverage area. When the groundwave and skywave signals are of the same magnitude, they can phase out one another, resulting in serious fading and irritating audio distortion in the detected signal at the consumer's receiver. The outer edge of the primary service area, caused by this self-interferrence phenomenon, is called the "fading wall". Its physical elimination is only possible by the reduction of high angle radiation.

Historical Perspective

The history of the anti-skywave problem makes for fascinating reading. Stuart Ballantine, in the second of his two historic papers from Harvard in the 1920's, according to Laport, ". . . disclosed a hitherto unknown fact: there was an optimum height for a vertical radiator for obtaining maximum groundwave field strength."[1]

Laport continues, "Further study of the optimum height antenna disclosed eventually that the conditions of maximum groundwave and best antifading characteristics were not obtained with the same height. . . . The optimum choice for antifading over land was experimentally established at about 190 degrees."[1] The 225 degree tower gives the maximum groundwave but its high angle lobe produces a non-negligible fading wall.

Laport, in his fascinating 1952 publication, observes that, "By 1934, the modern broadcast radiator had evolved to its present state."[1]

Summarizing the state of affairs in the early 1950's Laport concludes, "Diligent research and experiments have been conducted for other possible broadcast principles that might equal or surpass those disclosed by Ballantine."[1] As we know today, the results (though often significant) have been of marginal utility.

Practical Requirements

Over the past several years a renewed interest in anti-skywave antennas has been exhibited by broadcasters, the NAB and consulting engineers. The noteworthy papers publishing the separate approaches of Biby and Prestholdt indicate the creative effort put forth to arrive at acceptable alternatives to expensive radiators of heroic proportions.[2,3]

Perhaps the clearest verbalization of the necessary technical requirements which challenge the creation of any realistic Anti-Skywave Antenna was put forth by Richard Biby in his 1986 NAB Engineering Conference technical paper:

> "In order to be really economically viable, an 'Anti-Skywave' antenna design concept must be able to take the typical 90 degree vertical tower, with a conventional buried copper wire ground system, make minimal changes thereto, and end up with decreased nighttime interference and improved groundwave signal strength. All the while, the system should remain non-directive, but still offer the possibility of being made directive in the horizontal plane if such were needed."[2]

To this we would also add, because of the Sommerfeld attenuation function, the structure must produce only a vertically polarized groundwave. What is needed is some way to increase the vertical current moment of the radiating system (the integral of $i \bullet dl$).

This has been done, and Pinzone Communications Products, Inc. has just such a solution available.

The patented Corum Antenna provides a splendid candidate to simultaneously surmount all of the above engineering requirements.

It can be used to retrofit existing towers, at ground level, and produce an enhanced elevation plane directivity previously unavailable to design engineers.

Elementary Considerations

Normal mode helices have been of interest since Pocklington's famous 1897 paper. One particularly intriguing idea is to take a self resonant normal mode helix, pull it around into a closed multiply connected region and let the resulting structure, which has been called a "Corum Element", combine the tuning and matching networks with the radiating structure itself.[4,5] The radiation resistance is now in series with the coil inductance, and this combination is shunted by the helix turn-to-turn capacitance. The impedance transforming nature of this lumped circuit equivalent is well known, and it also has the advantage of transforming a relatively small feedpoint current into a stepped up current passing through the radiation resistance.

There exist a variety of techniques available to predict the behavior of simple antennas. Because of the geometrical complexity of our structure, traditional moment methods are not only cumbersome, but require inordinate computer time and yield little insight into the physics of the antenna. We have found the Kron/Gobau Diakoptic technique much more promising. We can attest to the oft heard complaint that, "The method of moments is little more than a numerical experiment. It generates no analytical formula or expression by which to gauge how the result might change with a change in configuration . . . insight can only be gained by running the experiment again and again." Consequently, we favor an analytical model for the physical insight which it provides.

The field theory analysis of the basic Corum element is fairly straightforward. Since the structure is a slow wave self resonant helix, it is reasonable to assume a superposed sinusoidal distribution of electric and magnetic current, where the electric current is given by

$$(1)\quad J(r') = I_o \mathrm{Sin}(n\phi')\ \delta(\mathrm{Cos}\ \theta')\ \frac{\delta(r'-a)}{a}\ \hat{\phi}'$$

the coordinates having their usual meanings (see figure 1), and the magnetic current is found from

$$(2)\qquad I_m = \mu\omega(\pi b^2/s)\ I_o\ \mathrm{Cos}(n\phi')$$

where a is the major radius of the torus, b is the helix radius (the minor radius of the torus) and s is the turn-to-turn spacing. In these expressions, n is a mode number for the current distribution on the structure. The radiated fields are determined in Reference 4 as:

$$(3a)\quad E^e_\phi = -\frac{\beta_g a Z_o I_o}{2r}\ \cos n\phi\ J'_n(\beta_g a \sin\theta)\ e^{j(n\pi/2)}$$

$$(3b)\quad E^e_\theta = \frac{n\beta_g a Z_o I_o}{2r}\ \sin n\phi\ \frac{J_n(\beta_g a \sin\theta)}{\beta_g a \tan\theta}\ e^{j(n\pi/2)}$$

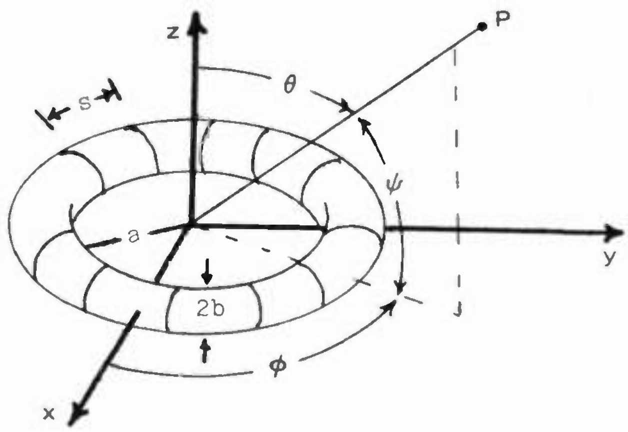


Fig. 1. Geometrical configuration.

Several variations of the basic configuration are now possible. Note the presence of the azimuthally directed (or horizontally polarized) electric field components. These must be eliminated for groundwave AM broadcasting. By contrawinding the helix [References 4,5,7], the azimuthal component of electric current is cancelled out and one is simply left with what is commonly known as a poloidal flow of electric current. (See figures 2 and 3.) A toroidal flow of electric current would be in the azimuthal direction (parallel to a thread in the center of the doughnut), while a poloidal flow is up over and around the anchor ring, or minor axis of the torus. This produces the phi-directed effective magnetic current of equation (2) above. Occasionally, this is called a caduceus winding.

$$E^m_\theta = -\frac{\beta_g a I_m}{2r} \cos n\phi \; J_n'(\beta_g a \sin\theta) \; e^{j(n\pi/2)} \tag{3c}$$

$$E^m_\phi = -\frac{n\beta_g a I_m}{2r} \sin n\phi \; \frac{J_n(\beta_g a \sin\theta)}{\beta_g a \tan\theta} \; e^{j(n\pi/2)} \tag{3d}$$

The superscript e indicates a field component attributable to the electric current and m to the magnetic current. $J_n(x)$ is the usual Bessel function of order n and β_g is the phase constant appropriate for the slow wave helix.

It should be clear that the structure is basically a low Q leaky resonator. The resonator loss resistance arises from the radiation resistance, the skin effect and proximity effect losses. The skin and proximity effect losses may be calculated in the usual manner for coils. The radiation resistance may be gotten by a Poynting integration of the radiated fields.

These fields are not unlike those produced by the superposition of a resonant electric loop and a "magnetic frill" or circular slot antenna. [Neither of which is really practical for AM broadcasting - the loop produces a horizontally polarized ground wave and the slot requires the construction of a heroic ground plane. The theory of these structures is of present interest.] However, because of the slow wave nature of the Corum helix, the physical size of a self resonant structure has been considerably reduced. The self resonant electric loop and annular slot require a circumference on the order of a free space wavelength. The Corum Helix is self resonant at only a fraction of this size.

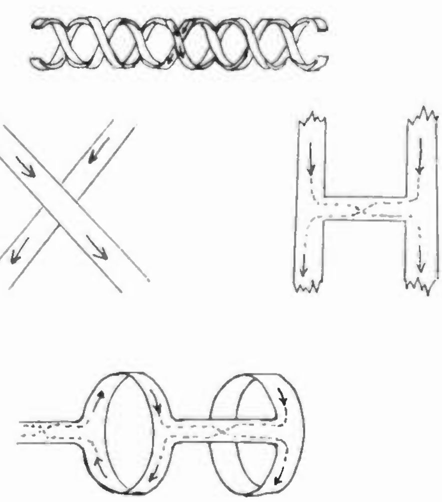


Fig. 2. Fundamental structure for the rings in a contrawound helix.

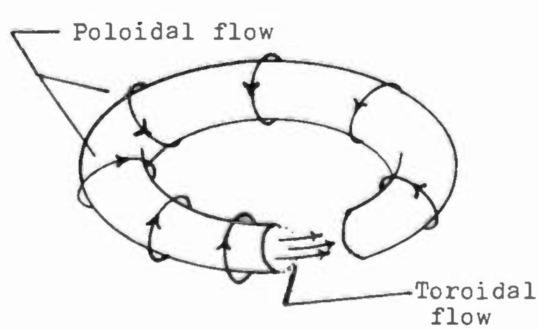


Fig. 3. A "poloidal" flow of electric current is equivalent to a "toroidal" flow of effective magnetic current.

The radiated fields are then described only by expressions 3(c) and 3(d). Note that "Figure Eight" patterns can be obtained. Further, and this is important, if one divides the toroidal helix into 4 or more segments, it can be fed as Smith's omnidirectional cloverleaf antenna, so familiar in FM broadcasting. [See references 4 and 8.] The resultant magnetic current distribution will be uniform, (n = 0), and the radiated field is described simply by equation (3c). See figure 4.

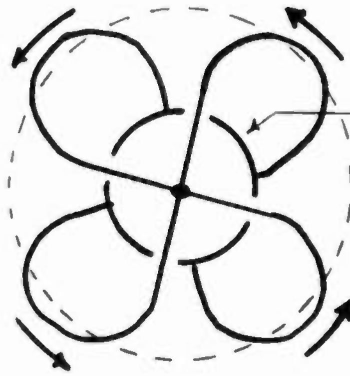


Fig. 4. Smith's "cloverleaf" arrangement for producing a uniform azimuthal current distribution.

THE CORUM ELEMENT

We now have a Low Profile, slow wave **vertically polarized**, self resonant, omnidirectional (in the azimuthal plane) radiator with a substantial feedpoint impedance. Consequently, the structure has considerable desirability as a stand alone electrically small antenna at frequencies where ground wave propagation or ground effects are important.

For example, in the fundamental mode (n=0), a power flow calculation gives the predicted vertically polarized elevation plane field pattern, in RMS mV/m at 1 mile, per kilowatt radiated, as the expression

$$(4)\quad E_\theta = -\,412.7\sqrt{P_{KW}}\; J_1(\beta_g a \sin\theta)\quad .$$

In order to experimentally test the validity of the analytical model above, a variety of structures have been fabricated at frequencies from 150 KHz to several GHz. Typical measured and calculated elevation plane electric field strength patterns are shown in figure 5. The measured vertically polarized azimuthal plane pattern is displayed in figure 6. The theoretical and measured feedpoint impedances are shown in figure 7.

Needless to say, we have considerable confidence in the relatively simple analytical model presented in the previous section. And now we come to the exciting news concerning anti-skywave antennas.

THE CORUM ANTENNA

The above element, when used alone, may produce rather high angle radiation. See figure 8. This fact, however may be used to great advantage. In order to produce low angle radiation, a standard tower may be surrounded with the Corum Element and phased to produce increased groundwave and decreased skywave radiation. See figure 9.

This new structure is now marketed under the name "Corum Antenna". Since the system's phase center is at the tower's spatial position, the resultant radiation pattern is simply the sum of the Corum Element pattern, above, and that of a vertical conductor of height G, given by the well known expression

$$(5)\quad E_\theta = E_o\,\frac{\cos(G\sin\Psi)-\cos G}{(1-\cos G)\cos\Psi}\quad \text{mV/m @ 1 mi}$$

where E_o is the RMS field intensity produced by the tower in mV/m at 1 mile, per kilowatt radiated (186.5 for a stub; 194.5 for a quarter wave tower; 236.2 for a half wave tower; 275 for a 5/8 wave tower; etc.). In practice, a multiplicative efficiency factor would be employed to account for ground system and structure losses.

Elevation Plane Patterns

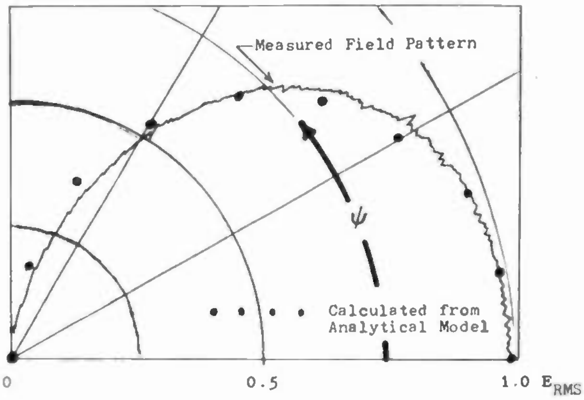
This remarkable structure provides the design engineer with a new flexibility in pattern synthesis. The element may be used in a variety of applications to dramatically tailor the shape of radiated patterns.

As a demonstration of the utility and flexibility of this new technology, let us plot the elevation plane patterns of several configurations. In the following we let

K = the electrical circumference of the Corum Element in wavelengths ($K = \beta_g a$)

P = the fraction of the total power radiated by the central element

G = the electrical height of the central element ($G = \beta_o H$).



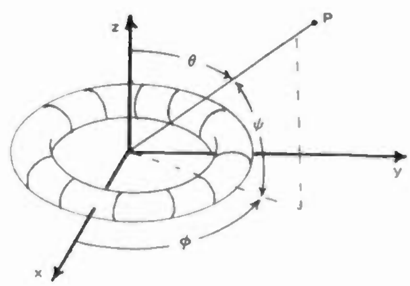


Fig. 5. Experimentally measured electric field strength pattern (vertical polarization) from a contrawound Corum Element lying in the x-y plane. The black dots are the values calculated from the theoretical model. The pattern is omnidirectional in the azimuthal plane.

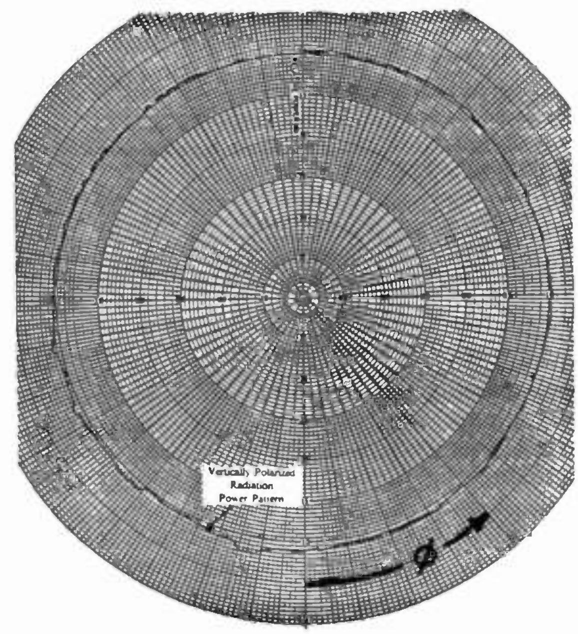


Fig. 6. Measured azimuthal plane radiation power pattern. The Corum element was in the x-y (azimuthal) plane and the polarization state was vertical.

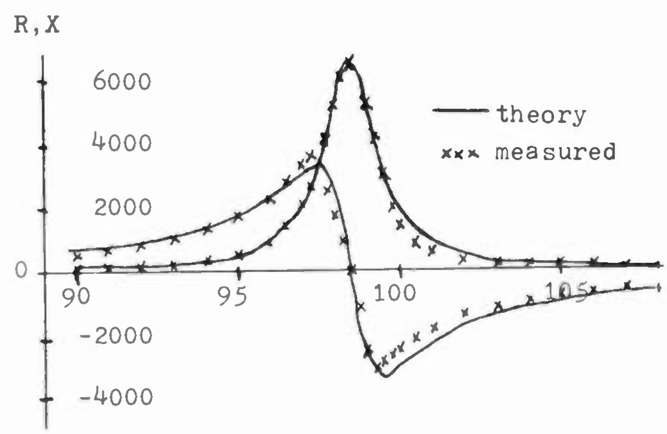


Fig. 7. A comparison of the measured and theoretical impedance verses frequency for a single Corum helix antenna designed for 98.5 MHz.

In figure 10, we calculate and plot the elevation plane electric field pattern for the case where the electrical circumference of the Corum element is 1 λ_g and the power is split 55% to the tower and 45% to the Corum element. The radiated power is 1 KW. The Corum element is constructed of 2 inch diameter copper tubing and produces a vertically polarized omnidirectional pattern in the azimuthal plane. The RMS field strength at one mile is 241.5 mV/m when losses are included (256.0 mv/m @ 1 mile in the loss less case), or equivalently, 388.6 mV/m @ 1 Km including losses (411.9 mV/m @ 1 Km for the loss less case).

In figure 11, we compare Gihring and Brown's optimum antifade tower (190 degrees high), the standard 225 degree tower, and a new antenna made possible by combining the 225 degree tower and a cophasal Corum element with

$$K = 4.25$$

$$P = 0.945$$

The new antenna is clearly superior at all critical skywave angles.

In figure 12, we present a comparison of the optimum antifade tower (190 degrees), the standard Quarter wave tower (90 degrees), and a short stub (12 degrees, in this example) surrounded by a Corum element with

$$K = 1.0$$

$$P = 0.54$$

fed 180 degrees out of phase with respect to the stub. Losses are neglected.

Table 1 gives the actual numerical values of the various elevation plane patterns and compares the classical performance with what the new technology makes possible.

Remember to compare apples with apples. The results are very dramatic at angles above (and below) 30 degrees. These numbers are only typical, and have not even been optimized.

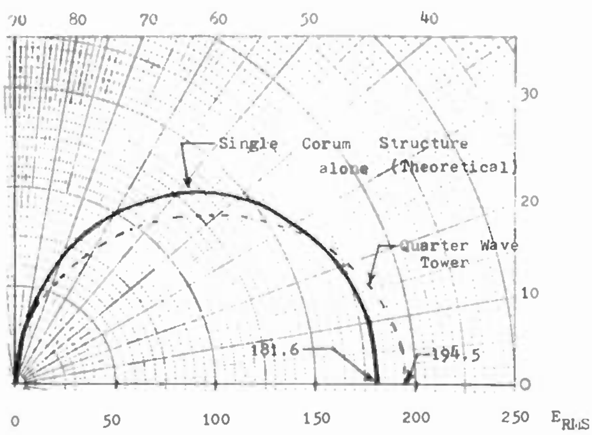


Fig. 8. Theoretical elevation plane of a single Corum element alone. (Elevation plane pattern of a quarter wave tower for reference.) E RMS in mV/m at 1 mile for 1 KW radiated.

Is there any reason why the center tower is even necessary ? As with concentric annular slot arrays at microwave frequencies, one may array concentric Corum elements. In figure 13, we show just such an array. In this example, we portray the case where the inner element has K = 1 and the outer element has K = 2. The radiated power is 1 KW and the calculation is at a frequency of 620 KHz. There are 52 rings in the outer element displaced 10 feet apart, with a major radius of 80 feet. The wire diameter on the outer

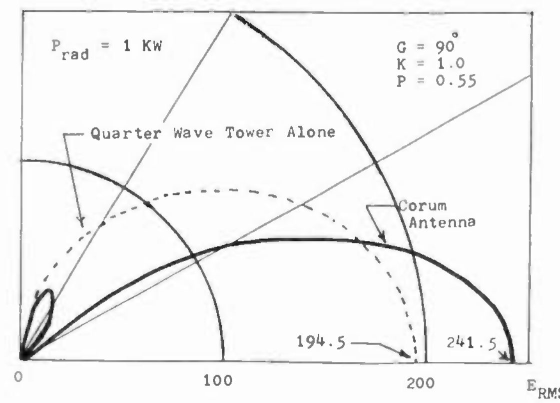


fig. 10. Elevation plane pattern for a quarter wave tower surrounded by a Corum element (losses included). In the loss free case, the RMS field strength rises to 256 mV/m at 1 mile, per kilowatt radiated.

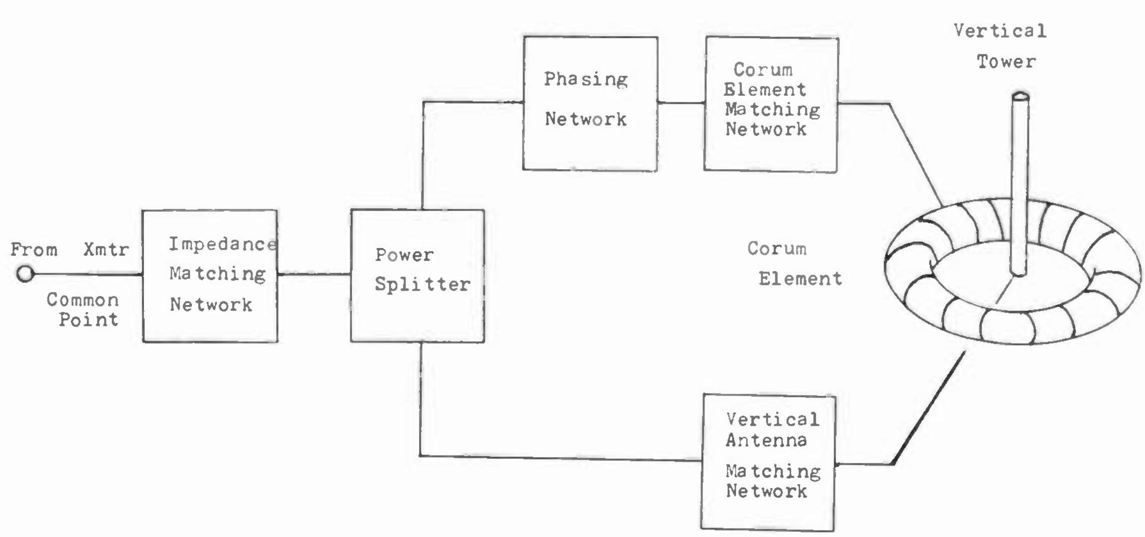


Figure 9 . Block diagram of Corum array feed system.

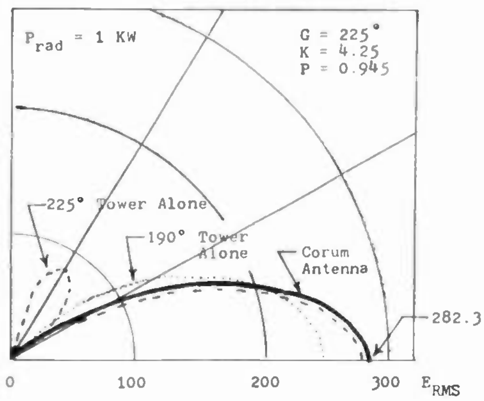


Fig. 11. Elevation plane pattern comparing the optimum antifade tower (190 degrees), Ballantine's optimum ground wave tower (225 degrees), and a 225 degree tower surrounded by a Corum Element. Note that the high angle lobe of the 5/8 wave tower can be phased out. Other combinations of K and P are also effective.

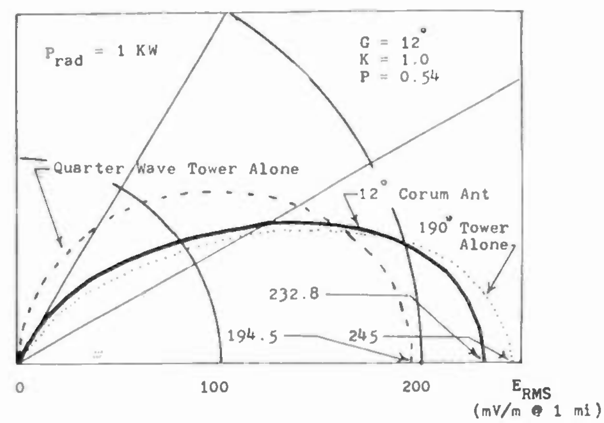


Fig. 12. A remarkable configuration in which a Corum Element surrounding a 12 degree tower not only out performs a quarter wave tower, but is competitive with a full sized optimum antifade tower (190 degrees).

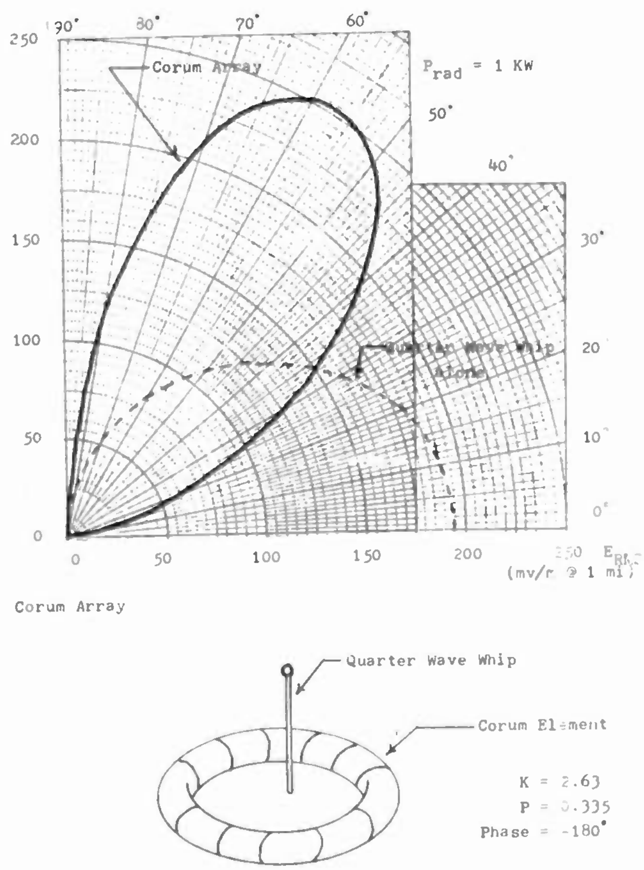


Fig. 14. Utilization of a Corum Element to produce high angle radiation from a standard quarter wave whip antenna.

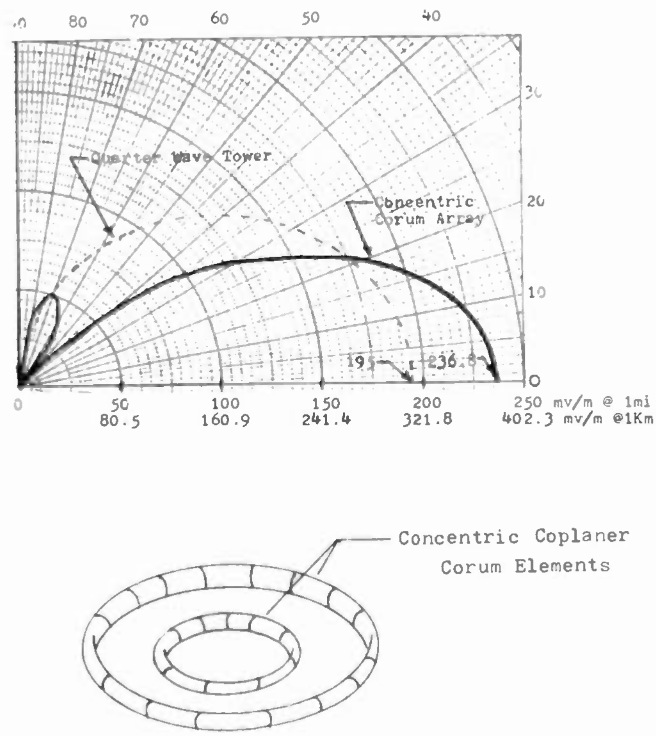


Fig. 13. A new element available for antenna design. At 620 KHz the structure is less than 60 feet high and 80 feet in radius. The ground wave field strength is comparable to that of a half wave Tower.

element is taken as 0.75 inches. The RMS field strength at one mile (including structural losses) is 236.8 mV/m (249 mV/m in the loss less case). The power is split 55% radiated by the inner element and 45% by the outer element. The electric field pattern is omnidirectional and vertically polarized. A variety of elevation plane patterns may be obtained by varying the electrical circumferences and power division. And, of course, this element may be used in a larger phased array.

Occasionally, one desires high angle radiation. In figure 14, we present the case where a Corum element may be fabricated to produce high angle radiation from a standard quarter wave whip antenna. In this case the appropriately fed surrounding element elevates the pattern maxima to about 60 degrees above the horizon. The pattern may even be made switchable for mixed high/low angle radiation for communication with elevated platforms and with other ground stations. Again, the figure has not been optimized, and superior configurations may exist for this application.

By the way, it should now be obvious how we have been able to increase the current moment without adding more tower height.

(6) $E_z \sim i \bullet dl = i_z^e dz + i_\phi^m a d\phi$

As with a monopole and an annular slot, the vertical electric field is increased by the equivalent magnetic current.

What about ground screens and counterpoises ? Certainly, one may employ a Corum Element in place of an extensive ground system to produce the same ground wave signal. This comes about simply because more signal is pulled out of the sky and concentrated along the earth. This latter technique might be of interest to stations lacking sufficient real estate for a full sized ground screen.

Conclusions

All this sounds too good to be true. One should always be on guard against extravagant claims made by theoreticians and antenna manufacturers. Common sense and past experience should provide some guidance, even with a new technology.

In this regard, one must bear in mind the fundamental limits associated with antennas. One does not get something for nothing. Chu's fundamental limit, which relates the lowest achievable Q of a loss less structure to its maximum physical dimension, is still in force. The Corum Element is a remarkable structure, but it has been wrestled from Nature at the price of reduced bandwidth. Fortunately, even at the low end of the AM broadcast band, we are in that happy state of affairs where it is possible to operate with Q's as high as 20, and practical configurations can be fabricated. The problem is not as acute at higher frequencies.

In summary, then, we believe that our patented structure is not only the best available candidate for an anti-skywave antenna, but is, perhaps, the only viable alternative for many situations.

It satisfies the pragmatic requirements placed above on an anti-skywave antenna. It has five significant advantages:

1. The Corum Element is constructed at or near ground level.

2. It may be used to retrofit most existing arrays.

3. Each element of the phased array now has a much narrower elevation plane pattern.

4. Arrays of concentric Corum Elements may be used, trading off going out in horizontal extent for going up in tower height, to achieve directivity.

5. It permits pattern tailoring which would otherwise be impossible with traditional radiators.

Unless we have committed serious error, we believe that by finding an alternative to the annular slot, we have brought to light broadcast principles that "equal or surpass those disclosed by Ballantine".

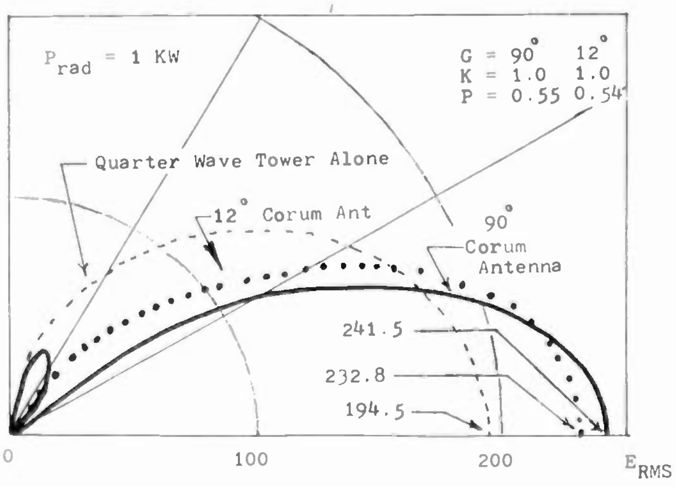


Fig. 15. A comparison of a quarter wave tower, a 12 degree Corum antenna and a 90 degree Corum antenna.

TABLE 1 COMPARISON OF ANTENNAS FOR ANTI-FADE OPERATION

PSI	12 Degree Tower	90 Degree Tower	180 Degree Tower	190 Degree Tower	225 Degree Tower
0	186.500	194.500	236.200	245.000	275.000
10	183.646	190.199	222.438	228.567	242.664
20	175.178	177.824	185.528	184.915	159.922
30	161.367	158.809	136.374	127.950	60.359
40	142.652	135.108	87.312	72.996	-22.758
50	119.622	108.715	47.438	30.771	-71.163
60	92.994	81.261	20.619	5.182	-83.583
70	63.581	53.791	6.182	-5.362	-68.834
80	32.272	26.727	0.776	-5.326	-37.941

CORUM ELEMENTS:

G = 12°	90°	225°
K = 1.00	1.25	4.25
P = 0.54	0.63	0.945
Phase = -180	-180	0

F(psi)	F(psi)	F(psi)
232.823	235.148	282.310
221.004	222.595	249.530
188.022	187.798	167.628
140.717	138.534	74.942
88.686	85.329	7.791
42.019	38.626	-18.585
8.683	6.118	-13.362
-7.602	-9.080	1.086
-8.372	-8.988	6.364

Comments: K is the electrical circumference of the slow wave Corum element in velocity inhibited wavelengths. P is the fraction of the total input power supplied to the center element.

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BROADBAND TRANSFORMERS FOR AM DIRECTIONAL ARRAYS

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ABSTRACT

Broadband transmission line impedance matching transformers can improve the bandwidth, stability, and ease of adjustment of AM directional feeding systems. This paper will present information on the design and application of two types of transmission line transformers, the 4:1 and 9:1, for AM broadcast use.

I. INTRODUCTION

The increased usage of AM stereo and a desire by the AM broadcaster to obtain the best possible sound quality has made it necessary to optimize the frequency response of AM directional arrays. Arrays with low operating base impedances and extremely low power towers are often found to have bandwidth and stability problems. Limited bandwidth will appear as signal clipping and distortion at the receiver. Stability problems will require additional maintenance to keep the array in tolerance. Another problem associated with these arrays is that if proper consideration has not been included in the phasor system design, array tuning may be a "no win" situation. In the past, multiple sections have been used to keep the circuit Q low. However, these represent additional components and in many cases, the difficulty in adjusting the system is not enhanced by improved performance. Transmission line impedance transformers may be used to increase adjustability, stability and the bandwidth of such systems.

II. TRANSMISSION LINE TRANSFORMERS

Transmission line transformers have been in use for over 20 years primarily in the HF and VHF range. They differ from conventional transformers in that at high frequencies the energy from input to output is coupled through the dielectric of the coiled transmission line wound on the core, not the core itself. In the LF and MF range, transmission line transformers may be treated as a conventional autotransformer connected in a "transmission line" configuration. Transmission line transformers are catagorized by their impedance transformation ratio, with two of the most common being the 4:1 and 9:1.

A. THE 4:1 TRANSFORMER

Figure 1 shows the schematic diagram of a 4:1 transmission line transformer. The defining circuit parameters are:

$$Z_{in} = 4\ Z_{Load} \quad (1)$$

$$V_{in} = 2\ V_{Load} \quad (2)$$

$$I_{in} = I_{Load}/2 \quad (3)$$

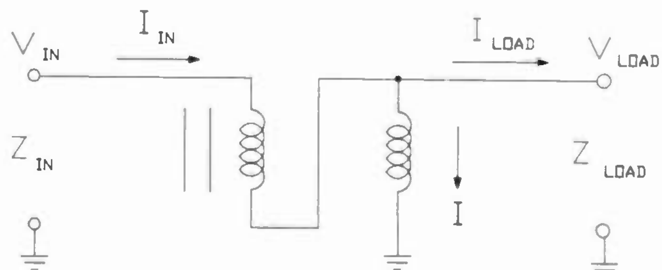

Figure 1. Schematic of 4:1 Transmission Line Transformer

1. Construction

The 4:1 transformer may be bifilar wound on a suitable toroid or other shape core. Care must be made in allowing proper insulation and current carrying ability of the windings. Also sufficient mechanical support is necessary for the core.

2. Design Parameters

The two most important design constraints for the transmission line transformer are:

1. The core magnetizing inductance (L_m) should be at least 10 times the magnitude of Z_{load}.
2. The number of bifilar turns used

at the rated current shall not exceed B_{max} for the core. If this is not done, three deterants will result: Non-Linearity (saturation), generation of harmonics, and inefficiency.

For a toroid core:

$$L_m = 40\pi N^2 \mu_c \left[\frac{A_e(cm^2)}{\ell_e(cm)}\right] \mu H$$

$$B = \frac{V_{rms} \times 10^8}{4.44FNA_e} \text{ gauss}$$

Where:

N= Number of turns per winding

μ_c= Core relative permability

A_e= Core effective cross-sectional area

ℓ_e= Core average magnetic path length

F= Lowest operating frequency in Hz

V_{rms}= Voltage on winding

3. 4:1 TRANSFORMER MODELS AND PHASE SHIFT

Figure 2 shows an equivalent circuit model of the 4:1 transformer. Figure 3 is the same model except a mathematical model for the ideal transformer is included. Equation 7 gives the current phase shift across the transformer which agrees within 5% of actual measured values.

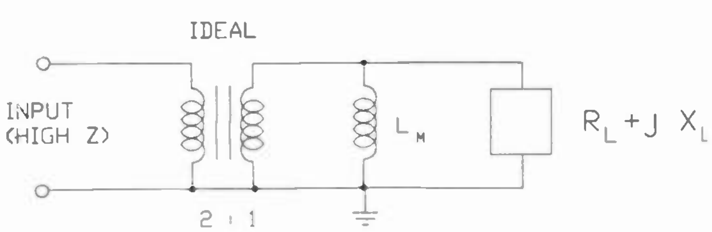

Figure 2. Equivalent circuit with ideal transformer.

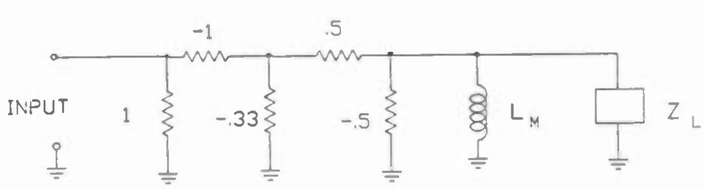

Figure 3. Equivalent circuit with a mathematical model for the ideal transformer.

For $|Z_1| << X_m$ $\quad \beta \approx 0°$

Actual measurment shows $-5° < \beta < 0°$

$$\beta = -\tan^{-1}\left[\frac{R_1}{X_1 + X_m}\right] \text{ degrees} \qquad (7)$$

$$X_m = 2\pi F L_m \text{ ohms} \qquad (8)$$

$$Z_1 = R_1 + jX_1 \qquad (9)$$

B: THE 9:1 TRANSFORMER

Figure 4 shows the schematic of a 9:1 transmission line transformer. The defining circuit parameters are:

$$Z_{in} = 9\ Z_{load} \qquad (10)$$

$$V_{in} = 3\ V_{load} \qquad (11)$$

$$I_{in} = I_{load}/3 = I \qquad (12)$$

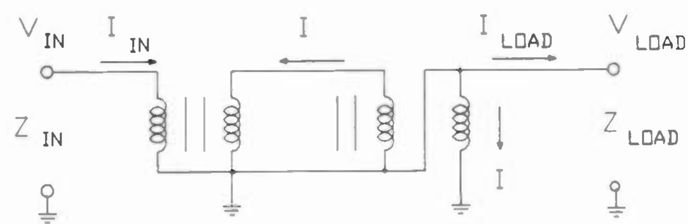

Figure 4. 9:1 Transformer Schematic

Construction and design of the 9:1 transformer follow that of the 4:1 transformer except two separate cores are required for the 9:1 and each must be bifilar wound.

III. USE OF TRANSMISSION LINE TRANSFORMERS IN AM PHASOR DESIGN

The purpose of this paper is to show that transformers can be used in phasor systems to increase adjustability, bandwidth and stability. A three tower inline phasor was designed twice, first with no transformers, and second with a transformer in the lower power tower's power divider and antenna tuning unit. Field ratios are 1:2:1 and a progressive phase shift of 90° . In any array, moding should be done to find the most desirable parameters. However, this particular array has no other pairs that work.

Design parameters:

F = 800 kHz   P = 1000 W.

#1 $1.0\angle 0°$

#2 $2.0\angle 90°$

#3 $1.0\angle 180°$

$G_1 = G_2 = G_3 = 70°$

$S_{12}=S_{23}=80°$

Three uniform cross section, guyed towers, 18" face
Operating base impedances and power distribution

Tower	Base Z (Ω)	Power (W)
1	36.4 - j14.8	295.7
2	21.0 - j50.0	659.7
3	5.8 - j66.3	44.6

Figure 5 shows the phasor design without transformers, and Figure 6 is a design with transformers added. Network phase shifts in Figure 6 have been changed to compensate for a power divider phase shift change with a transformer added.

Looking first at the power divider of Case 1, Tower 2 which carries over 60% of the total power is fed directly off of the power divider buss. Power division for Towers 1 and 3 is achieved across two 35uH "shunt" coils. With the two "shunt" coils adjusted for proper power division, the power divider phase shifts are:

#1 -34.2°

#2 0°

#3 -35.9°

$Z_{pd} = 28.9 + j38\,\Omega$

These numbers do not appear to present problems with respect to adjustability and matching, but let us try to adjust the power coil of tower 3. Since a pure 50 ohm load does not usually exist while tuning, a V.S.W.R. of 1.4:1 was terminated on each transmission line. By adjusting the power coil of tower 3 we can easily adjust the output ratio ±40%. However, this introduces a phase change of 18°. The common point impedance remains remains relatively constant with adjustment of tower 3, but will have a more appreciable change with adjustment of tower 1.

This shows that this power divider's performance is not as desirable as it could be. Now we will look at Case 2 with a 4:1 transformer and a 10uH inductor used for the power divider of tower 3. The power divider of tower 1 and 2 remain unchanged. The phase shifts and the input impedance at the design center are:

#1 -34.2°

#2 0°

#3 -16°

$Z_{pd} = 29.7 + j26.8\,\Omega$

At first look we see no dramatic improvement except the phase shift of the power divider for tower 3 was cut nearly in half. Next we will perform the same adjustment as in Case 1, that is, adjusting the power coil of tower 3 for ±40%. Care should be taken when using this power divider because the power of tower 3 cannot exceed 1/4 the power in tower 2 unless a matching network is placed on tower 2's load to raise the input resistance to the desired amount.

Adjusting the coil for tower 3, as in case 1, we find we can adjust output ratio ±40%, but the phase shift only changes 5°. Input impedance varies only ±3 ohms with adjustment. This shows that this power divider offers nearly independent control of power and an increase in common point stability. One final note on the power divider, care should be used in determining the "shunt" coil size. A shunt reactance approximately equal to the load resistance is optimum, and smaller inductors will tend to have high circulating currents and limited adjustability.

1. Bandwidth

Bandwidth of the complete array will be discussed in two parameters. First the change in common point impedance versus frequency and second, a change in each array element ratio and phase versus frequency. Immediately, one can see that in Case 2, the "Q" of the matching "T" network for tower 3 will be much lower than that of Case 1. The design constraints used in determining the network configurations are:

1. All network phase shifts are kept as low as possible so that excessive Q is not introduced into the system.
2. Since Tower 2 has considerably more power than the other two towers, the matching network for that tower is optimized to provide a "good" match to the transmission line for sideband load impedances, (±10 kHz was used for this case).

Table 1 shows the result of a circuit analysis performed on the entire system. A coupled circuit model was used for the antenna array. Note, this model is frequency dependent, and must be computed

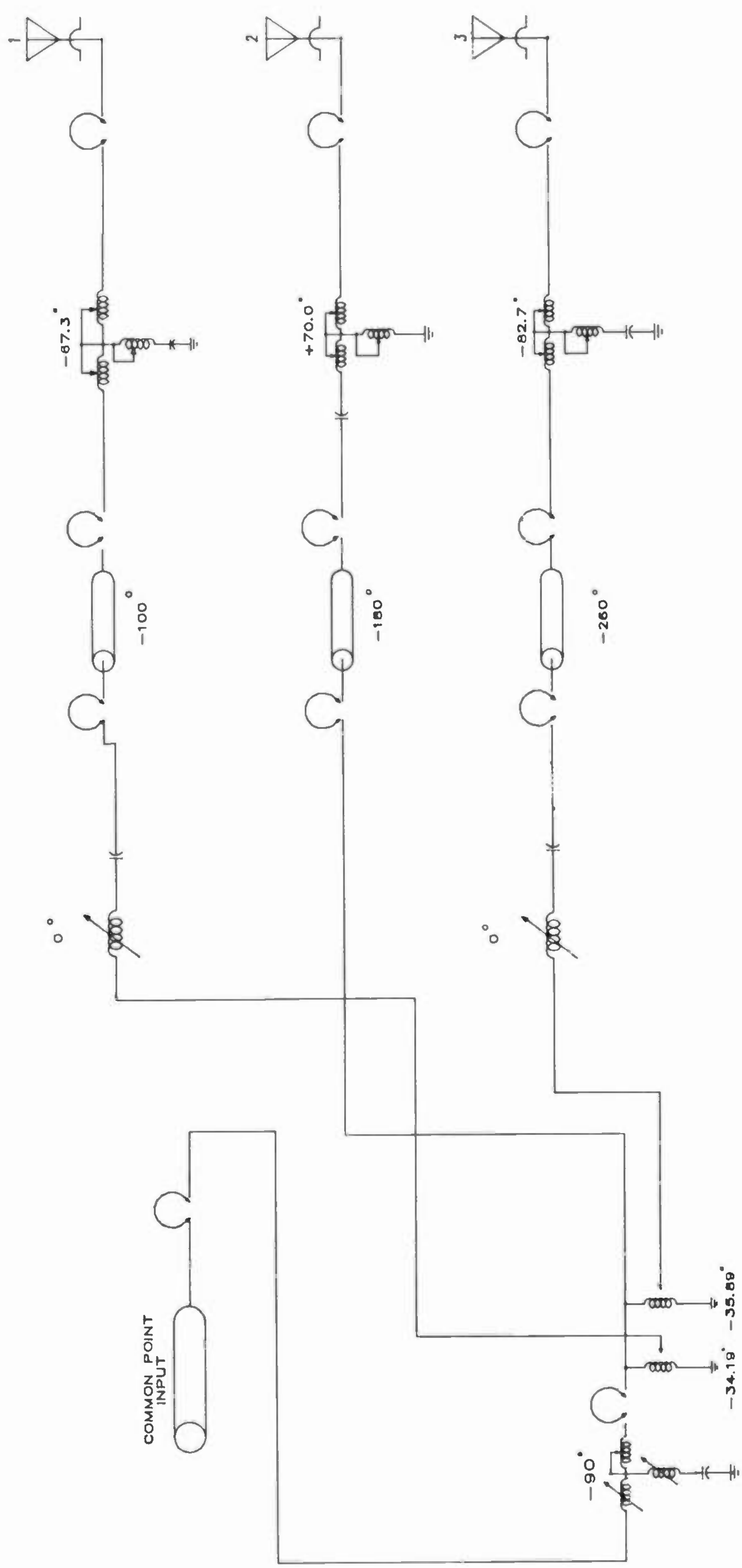

FIGURE 5:
3 TOWER PHASOR

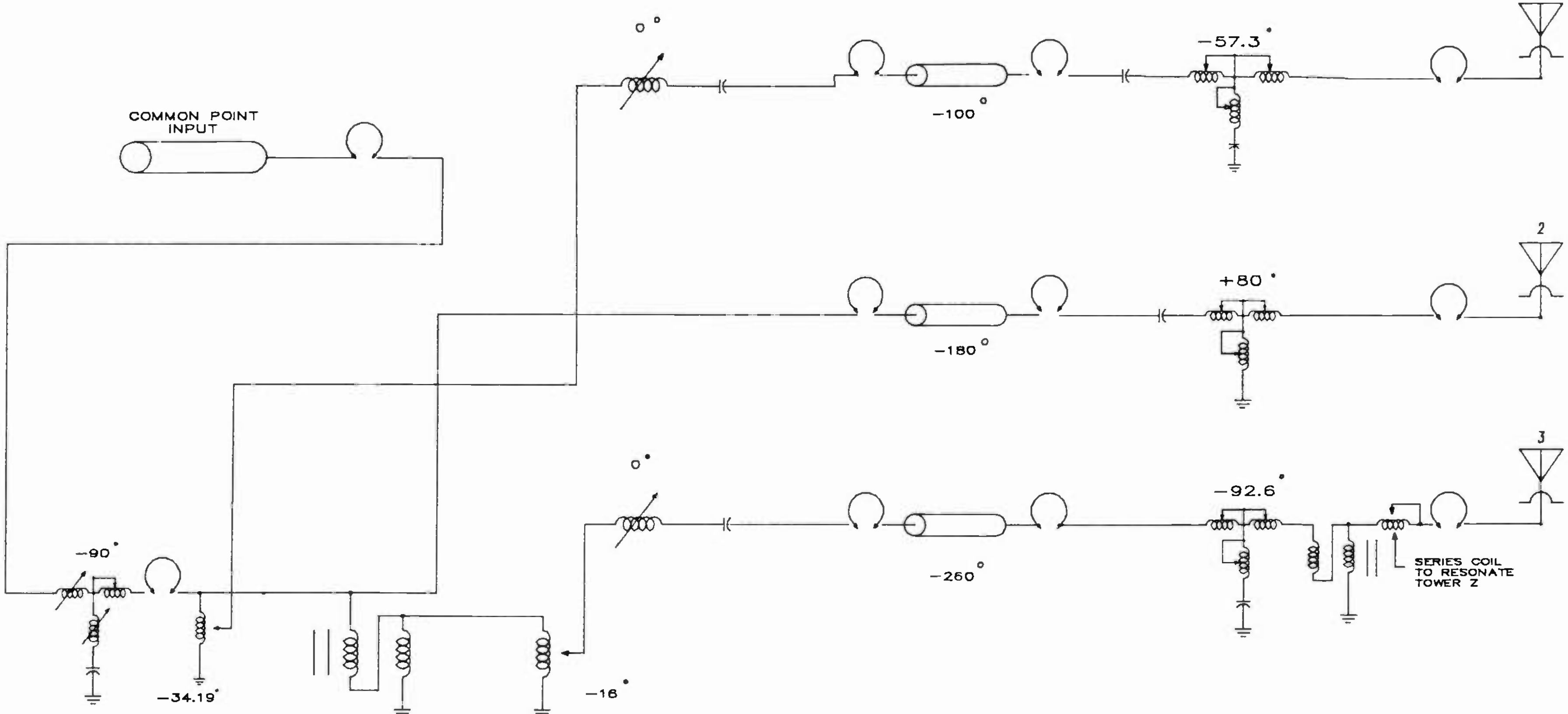

FIGURE 6:
3 TOWER PHASOR
WITH TRANSFORMERS

for the sideband frequencies as well as the carrier frequency. As stated before, the transmission lines are seldomly matched exactly to their characteristic impedance. For this analysis, a constant V.S.W.R. of 1.3:1 at the carrier frequency was placed on each transmission line. LTU "T" networks accomplish this with slight change from the design center, and series phase control networks were adjusted to compensate for the phase change of the transmission lines.

Table 1. Bandwidth Anaylsis

Case 1 (No Transformers)

790 kHz:

$G_1=G_2=G_3=69.1°$

$S_{12}=S_{23}=79°$

Tower	Normalized Loop Current	Base Impedance
1	$1.00\underline{/0°}$	43.7 - j22.9
2	$2.14\underline{/74°}$	21.2 - j52.9
3	$1.20\underline{/174°}$	2.1 - j65.8

$Z_{cp} = 55.1 + j2.5\Omega$

800 kHz (Carrier):

$G_1=G_2=G_3=70°$

$S_{12}=S_{23}=80°$

1	$1.00\underline{/0°}$	36.4 - j14.8
2	$2.00\underline{/90°}$	21.0 - j50.0
3	$1.00\underline{/180°}$	5.8 - j66.3

$Z_{cp} = 50 + j0\Omega$

810 kHz:

$G_1=G_2=G_3=70.9°$

$S_{12}=S_{23}=81°$

1	$1.00\underline{/0°}$	31.8 + j.86
2	$2.30\underline{/101.5°}$	22.0 - j43
3	$1.28\underline{/190.1°}$	9.9 - j63.1

$Z_{cp} = 43.7 + j6.7\Omega$

Case 2 (with transformers)

790 kHz:

Tower	Normalized Loop Current	Base Impedance
1	$1.00\underline{/0°}$	41.5 - j22.7
2	$2.11\underline{/78.7°}$	20.9 - j54.1
3	$1.04\underline{/173.5°}$	2.74 -j69.1

$Z_{cp} = 53.2 - j0.24\Omega$

810 kHz:

1	$1.00\underline{/0°}$	44 - j17.1
2	$2.02\underline{/98.2°}$	21.5 - j54.0
3	$.970\underline{/189.6°}$	10.3 - j39.4

$Z_{cp} = 46.1 + j5.0\Omega$

Comparing the results obtained in Table 1, the following conclusions may be made regarding the bandwidth of the two systems:

1. Case 2 with transformers yields a more desirable common point impedance response at ±10 kHz.

2. The loop current ratio and phases of Case 2 do not vary as much as those in Case 1. This also represents that the sideband "driving point" impedance's of Case 2 are changing less rapidly than in Case 1. These impedances are also shown in Table 1. One reason for this is the fact that if the transformers $X_m >> |Z|_{load}$, the phase shift will be independent of load variation.

As a final note to this bandwidth study, if the transmission lines are assumed to be perfectly matched to 50+j0 at the carrier frequency, both the common point impedance response and the array response will show improvement. At this point in phasor design, the circuits and transmission line lengths can be modified by iteration to yield an "optimum" design. However, the purpose of this analysis is to compare, as equally as possible, a phasor system with and without transformers, without consideration of any other possible system improvement.

IV. CONCLUSIONS

This paper has demonstrated transmission line transformers may be used in the AM broadcast band to improve the ease of adjustment, bandwidth and stability of an antenna array. Vector Technology has successfully used both 4:1 and 9:1 transformers at power of up to 5 kW. We have identified situations where transmission line transformers will solve a myriad of problems and intend to pursue this approach in our phasor design in the future.

Figures 7a and 7b are photographs of some actual transformers Vector Technology has built and installed. In Figure 7a the transformer is part of the power divider while in 7b it is part of a large antenna tuning unit.

V. ACKNOWLEDGEMENTS

My sincere thanks to Ted Schober of Radio Techniques for giving Vector Technology the opportunity to use these devices on stations that he represents. His counsel and suggestions have added immeasurably to the success of this program. To Russ Mundscheck, Director of Engineering for WEAZ and WFIL in Philadelphia, for the vision in seeing the potential for these devices as well as his practical suggestions. To Melvyn Lieberman, President of Vector Technology, Inc., thank you for giving me the opportunity to prepare this paper and for your many helpful suggestions in the preparation of this manuscript.

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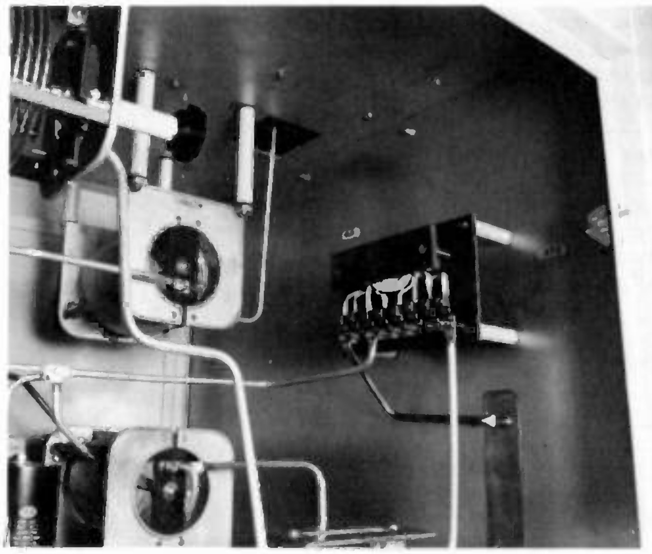

Figure 7a. Transmission line transformer used in power divider.

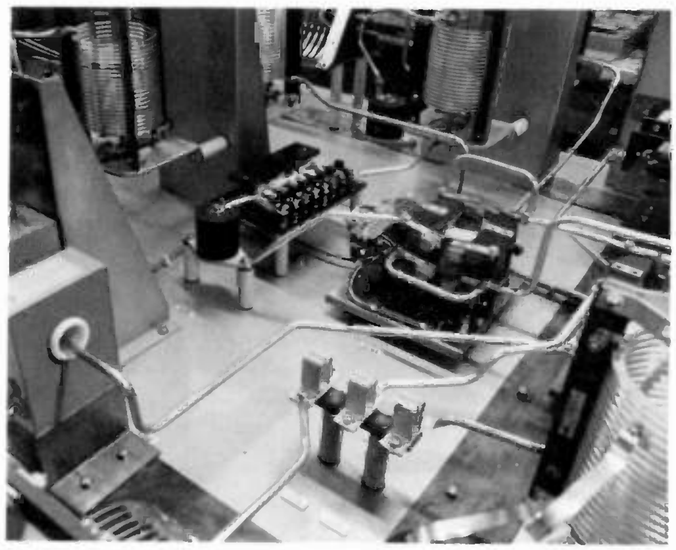

Figure 7b. Transmission line transformer used in antenna tuning unit.

A LOW-COST, HIGH PERFORMANCE AM NOISE BLANKER

Oliver L. Richards
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Worcester, Massachusetts

The system performance of AM radio is being improved noticeably by the development of AM stereo and the wider audio frequency response offered by the new National Radio System Committee (NRSC) standard. The combination of AM stereo with the NRSC standard offers the audio frequency response required to provide satisfactory stereo imaging with the potential for actually improving adjacent channel interference.

AM is also rather sensitive to manmade and meterological noise. Until standards similar to the International Electrotechnical Commission (CISPR) specification are promulgated in this country, the only solution is to treat the noise at the receiver.

This paper describes a noise blanker for AM band entertainment radios which is capable of virtual elimination of impulse noise caused by automotive ignitions, SCR power controls, meteorological, and other sources.

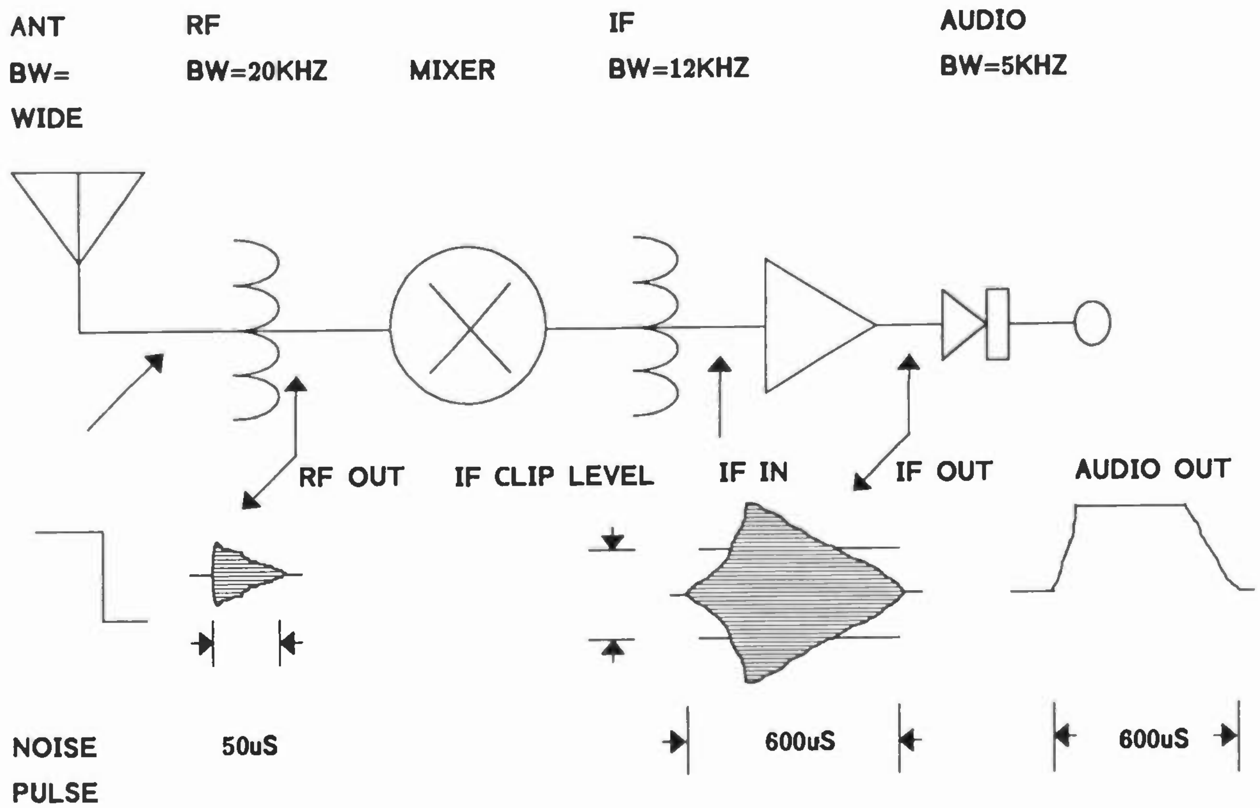

FIGURE 1

Noise pulse progressing through a typical AM receiver

The system is simple, requiring no complicated audio frequency phase-shift networks. This allows it to be almost entirely realized in a low-cost monolithic integrated circuit.

Background

Noise blanking has been employed almost from the beginning of over-the-air communication. The first reference seems to be in an 1899 British patent. Although it was a two-diode arrangement, which would more properly be called a "noise limiter", today it illustrates the pervasive and continuous nature of this kind of noise.

In subsequent years various approaches have been taken to reduce impulse noise in receivers. Most systems were developed for communication or navigation equipment. These receivers had both the requirement to extract a usable signal under the widest range of conditions and sufficient value to support whatever complexity and increased cost that might entail.

Consumer AM receivers are under much more severe cost constraints and perhaps surprisingly require much better noise suppression to actually achieve entertainment rather than annoyance.

Noise blankers can be divided into two basic categories. The first, audio blanking, detects the incoming noise pulse in the RF or IF, then blanks the audio. This has the disadvantage that nothing is done to protect the receiver RF and IF from overload and the noise pulse is stretched to a value equivalent to at least the period corresponding to the entire selectivity of the receiver (see Figure 1). This produces a long and variable-length audio pulse. Since the pulse is stretched to a length, determined by the IF passband, the pulse contains audio components that are less than the desired maximum audio response which is also determined by the IF passband. Under conditions of high noise energy, the pulse is stretched to an even greater period since the gain of the receiver will amplify the start and finish point of the initially stretched pulse, and the IF saturation will clip off the peak. Saturating an IF designed for limiting can also cause bias shifts which may require increased settling time. Under very-high pulse levels, the receiver AGC may also be activated to the point that the desired signal is heavily attenuated.

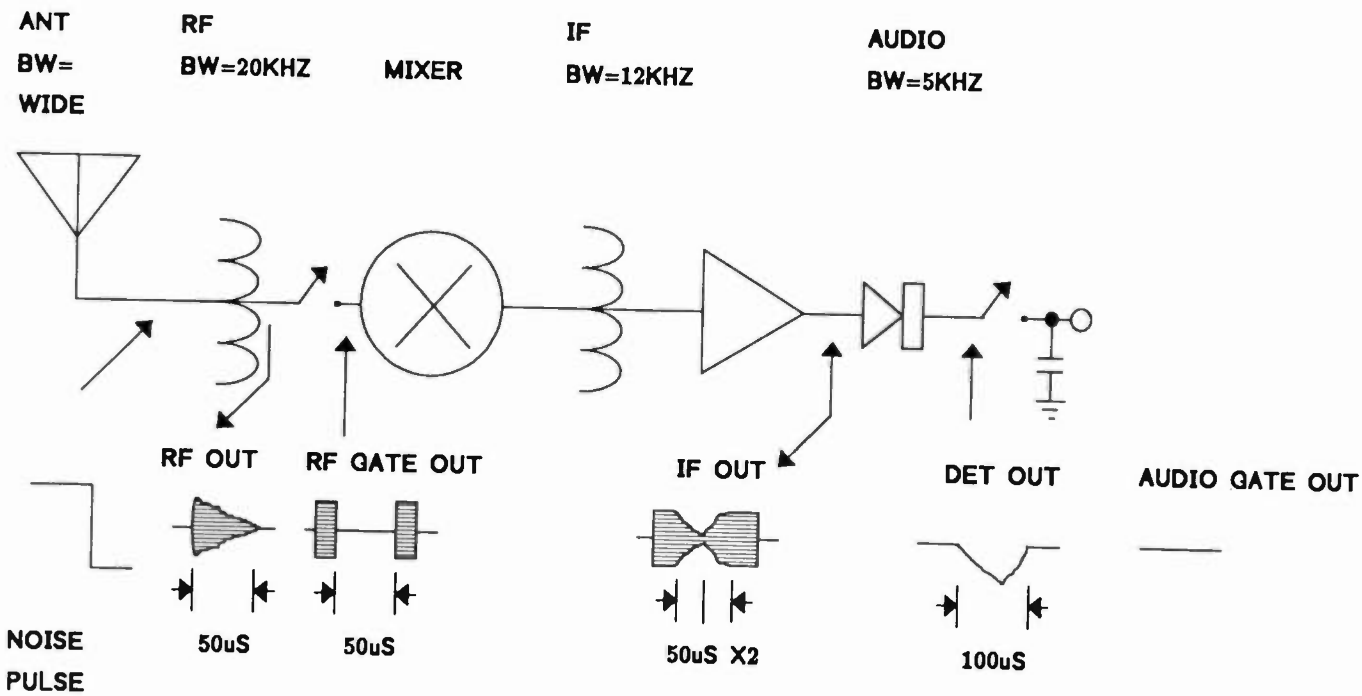

Figure 2

Lamb blanker with improvements

Audio blanking does have the advantages of simple timing requirements and good noise elimination under unmodulated carrier conditions. However, the long blank period (typically 600 us) produces audible damage to the audio.

The second category of noise blanking is known as Lamb noise blanking. This more advanced system puts the switch in the RF or IF, ahead of most of the gain and selectivity; thus, overcoming the stretching and AGC blocking described earlier. Variations of this system are widely used in communication and navigation gear today (see Figure 2).

A New Insight

The new system described here can be viewed as an improvement of the Lamb noise blanker to achieve the increased performance and cost requirements of an entertainment receiver. The Lamb noise blanker can blank the incoming noise pulse at a point in the receiver RF that has wide enough bandwidth to permit a narrow blanking pulse which is effectively faster than the audio response and the IF bandwidth. This does, however, leave a hole in the carrier (and modulation) of the desired signal. This hole can only approach 100% negative modulation (zero carrier), is beyond the bandwidth of the IF, and therefore, filled by the finite decay and rise time of the IF selectivity.

The IF selectivity stretches the hole in the carrier to a value of twice the RF blanking period. This is comprised of a logarithmic decay period during the blank state of the RF switch and complimentary logarithmic rise time that starts at the end of the RF blank time.

The new system recognizes the synergistic relationship between RF blanking and the brief noise created by the hole and the role of audio gating. The holes introduced into the carrier are an annoying crackle which would be unacceptable in entertainment receivers. Due to the short period of the hole, a sample-and-hold gate can be employed in series with the audio, operating at slightly more than twice the RF gate period, removing the crackle created by the holes. The audio gate timing is determined by the RF bandwidth, not the IF bandwidth as would be the case in audio-only blanking. The short nature of this gate period and its sample and hold implementation, place all of the sound product created by the audio blanking well beyond the highest frequency that can be reproduced by the IF and can, thus, be filtered out in the audio stages without loss of the desired frequency response as set by the IF bandwidth.

Conclusion

This system has been realized in a single monolithic integrated circuit fabricated with standard BiMos 4 processing and is housed in a standard 18-pin dual in-line package.

References

James J. Lamb
"A Noise Silencing IF for Superhet Receivers"
QST, February 1936

International Electrotechnical Commission
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1985 Article 14 and 16

ADVANCED MEANS TO MEASURE AND MINIMIZE FM MULTIPATH

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INTRODUCTION

Multipath interference to FM Stereo Broadcast Reception is the single most important problem that all FM broadcasters must deal with.

Any RF distortions originating in the station's transmitting system will not only increase the distortion and cross-talk in off-the-air reception, but will also greatly aggravate the incidence and severity of multi-path receiving problems.

In an effort to minimize this naturally occurring trait of VHF-FM broadcasting, I have made exhaustive investigations into practical ways to locate the sources, measure, and minimize the susceptibility of the FM broadcast signal to multi-path.

Borrowing techniques from VHF television engineering is necessary to make proper measurements, locate problems, and achieve the necessary optimization and wide-banding of the RF system in order to improve FM performance.

OVERVIEW

This paper presents practical ways to optimize the Radio Frequency portion of your facility. This will reduce source generated multi-path reception complaints and give you the added bonus of enabling your transmitter to operate with virtual digital audio transparency.

Basic areas of FM signal optimization

1. Transmitting System

2. Composite Signal

3. Channel Cleanliness

The three most important meters that you use to judge your transmission system performance are inaccurate and are seriously mis-leading you as to your actual signal quality, multi-path resiliency, and listenability.

These 3 meters are:

1. Your monaural modulation monitor as used for measuring unwanted synchronous AM content in your FM signal. (very inaccurate as all readings are displayed after a 6 dB per octave de-emphasis filter and sometimes after a 15 KHz low pass filter.)

A direct and wide-band non-filtered monitoring method must be used to minimize amplitude modulation problems at all frequencies.

2. Your transmitter output reflectometer as used as an indicator of RF filtering and antenna quality. (In reality an acceptable VSWR on your carrier is not a guarantee of the quality of your all-important sidebands.)

Bandwidth and group delay must be measured and optimized over all sideband frequencies in order to guarantee the quality of the modulated signal.

3. Off air reception utilizing a directional yagi antenna feeding a high gain fully limiting pre-amp and stereo modulation monitor.

Relying on this ideal off-the-air setup for your quality control does not give you a true reading of deficiencies with your carrier signal or monaural quality.

OVERVIEW OF "MULTI-PATH"

During so-called "multi-path" problems, the listenability of the received signal suffers from:

1. Momentary loss of the carrier.

2. Amplitude modulation of the carrier.

3. Out-of-phase (delayed) modulation.

4. Interference to the desired signal.

Distortion or loss occurs with-in the receiver during multi-path conditions.

1. If the incoming carrier is completely canceled by an equal strength, out of phase reflection, the internally generated noise from with-in the receiver is momentarily detected. (usually the problem when listening very near to the reflecting source.)

2. Reflections cause two or more modulated RF signals to arrive at the receiving antenna. Their vectors add or subtract, causing the incoming carrier level and phase to vary.

These changes create distortion problems as the IF, limiter and detector stages conduct at various different points on the incoming complex RF waveform.

Any variations in signal level originating in the transmitter will add to the cancellations occurring in the field causing a minor problem to become worse.

#3 The stereo detector is constantly upset as the direct and long delayed (reflected) signal(s) alternately emerge from the detector in various phases and with sidebands out of place. This causes distortion and intermodulation. (usually the problem when listening several thousand feet or more away from the reflecting source).

If the reflection has a significantly long time delay, many other interesting problems are suddenly created that can effect one or more critical stereo and SCA modulating frequencies.

4. An otherwise acceptable signal can be severely damaged by the presence of minute amounts of interference on the same frequency.

MEASUREMENTS OF SYNCHRONOUS AM

Synchronous AM is amplitude modulation components in the FM carrier that are directly related to (in sync with) the intended FM modulation.

Any amplitude (AM) variations with modulation of the FM carrier signal will greatly aggravate the incidence of multi-path receiving distortions in the field. Both the "picket fence" effects in the mobile and the "fuzzy, hard to tune in" problems on the typical home receiver equipped with little or no antenna.

Many engineers rely on the AM noise detector incorporated into their FM modulation monitor and have not optimized their facilities for minimum AM content because they believe that they do not have a problem.

FM modulation monitors give an entirely inaccurate reading of the amplitude modulation component of the FM stations carrier. This is because the FCC rules and regulations once permitted the measurement to be made through a standard FM 75 uS 6 dB per octave de-emphasis filter (a 15KHz low pass was often included). The presence of theses filters in the meter produce a very attractive and very achievable although technically false AM content figure.

It is necessary to take the RF sample after all RF filtering (harmonic, notch and bandpass) and feed it into a 100 MHz scope with a 50 ohm load at the scope! Your AM content can be seen directly as variations of the carrier level.

A second method is to build a simple diode detector or use one designed for video monitoring, and feed this directly into any wideband oscilloscope or AC voltmeter.

AM content measurements should not be made off air because the RF receiving amplifier will limit out any AM content before it gets to the accompanying monitor.

Full stereo program modulation or a complex audio wave fed into the wideband input of your exciter must be present for system tuning for minimum synchronous AM content.

Be extra careful when you are feeding into a narrow band-pass filter, if you modulate too hard, you may overheat it.

TRANSMITTER TUNING

Theoretically, FM has an infinite number of sidebands and is infinitely wide. For practical purposes, we need to transmit only the sidebands that contain significant amounts of energy and information.

The practice of peaking everything up is not a good idea when attempting to broadcast a clean, pure, wide-band RF signal. While you will achieve excellent carrier efficiency, you will also have a severe AM problem as the carrier level will vary with the program material.

The modern FM engineer needs to understand and adapt many of the transmitting techniques of television.

Specifically:

1. Wideband over-coupled transmitters to maintain sideband quality.

2. Sweeping the transmission system to minimize bandpass response ripple.

3. Minimizing group delay.

4. Monitoring done with oscilloscopes and transmission line sampling diodes.

6. Extremely low reflected power on long transmission line runs.

In an effort to produce high efficiency FM transmitters, considerable broadband quality was lost. It is only with the re-introduction of TV type over-coupled transmitters and filters, improved wide-band solid state exciters and drivers, that we will be able to transmit the full quality that is possible with optimized over the air VHF-FM stereo broadcasting.

With some advice from your transmitter manufacturer, you may want to over-couple the tuned circuits in your transmitter. This will increase the bandwidth but reduce their ability to generate high RF voltages. The loss of some drive may require adjusting the grid bias. This is also a good idea as hard class "C" bias will produce incidental phase modulation if it is driven by a stage or circuit with AM components present. I believe it better to operate closer to class B where the final is just beyond cut-off with no drive.

All of these adjustments will result in a much more transparent transmitter, but at a loss in overall efficiency. This can cause the final to produce more heat, and you may want to convert to the direct measurement of power output.

Make sure that all tube type amplifier stages are properly neutralized. An amplifier gets noisier and narrower as it gets closer it gets to oscillation. (Especially watch for variation with modulation in the screen current).

If your transmitter has an older design high-Q narrow band tube type driver stage, I recommend that you update it with a wide-band solid state driver.

PARALLEL TRANSMITTER OPTIMIZATION

Parallel transmitters provide added reliability as well as being a time proven way to generate very high RF power. However, special care must be taken to minimize AM content both individually and again when combined for normal operation.

It is not too difficult to optimize each side, but when combined there can be problems. The practice of feeding both units from the exciter output with a single "T" connector leads to serious interaction between the input stages of the transmitters. A second problem comes when a coil and variable capacitor are put in one leg to correct the phasing of the two RF paths so that the outputs combine in phase to produce maximum output power and minimum reject port power.

To prevent interaction between the two transmitters, either a 3 dB pad must be inserted in the line feeding each driver stage or utilize a balanced hybrid in place of the T and the pads.

Because exciter & driver power is precious, especially in the wideband mode, I recommend the 90 degree hybrid method.

To phase the two transmitters I use a co-axial "line stretcher" (also called a "trombone section") as it does not affect line impedance or loading.

With the above improvements, I have eliminated the tuning, drift and synchronous AM problems normally associated with parallel transmitter operation.

REFLECTOMETER MEASUREMENTS

It is erroneously believed that your combiner, filter, and antenna are all working perfectly if the VSWR (reflected power) is low. Actually it is the sidebands that determine program content and quality and it is the sideband response and quality that must be measured and optimized.

Even with acceptable VSWR at your significant sideband frequencies, there still may be a problem with group delay distortion which is present in any tuned circuit, such as a multi-station filter.

OPTIMIZATION OF RF FILTERING, COMBINERS, AND MULTI-STATION COUPLERS

Many FM stations have found it necessary to share or co-locate at the same site or area. To prevent catastrophic interaction or intermodulation the use of selective radio frequency filters in the RF line between the transmitters and antenna(s) are necessary.

The negative aspects of these filters, if not properly designed, tuned, operated and especially maintained will be signal distortions due mainly to group delay.

Group delay is the time differential between one frequency and another in a modulated signal. It is usually expressed over a given bandwidth that includes the center carrier and the far reaches of the significant upper and lower sidebands

When a signal with group delay is eventually demodulated, the content and time alignment of these sidebands and carrier must be correct or their will be distortion in the detected output.

Group delay figures should be kept as low as possible.

As a station operating through a multi-station combiner, I recommend that filters and multi-station combiners continuously exceed the following specifications under all operating conditions.

Group Delay	<25 ns at +/- 75 KHz
	<50 ns at +/- 150 KHz
Bandwidth (ripple)	<.1 dB at +/- 75 KHz
	<.2 dB at +/- 150 Khz
	<.5 dB at +/- 200 KHz

CHANNEL CLEANLINESS

Consider that the normal shadows and multi-path nulls in FM carrier strength in a high rise metro situation continuously reduces your signal by 20 dB and occasionally by 40 dB. The result to your listeners receiver is that the predicted City Grade signal of 3.16 mV will null down to between .03 and .31 mV. Normally, in the absence of anything else on your frequency, .1 mV is enough for good stereo reception.

There are four things that can dirty up a channel:

1. Intermodulation products
2. Noise
3. Distant co-channel stations
4. Cable-TV leakage

It takes only a few micro-volts of interference under your signal to create multi-path type reception complaints.

RF INTERMODULATION

This is when two (or more) stations mix together and produce spurious carriers at mathematically related frequencies.

Example: 100 MHz and 102 MHz get into a non-linear device such as any FM transmitter final stage.

The 100 mhz and it's harmonics with-in the transmitter mix with 102 and it's harmonics to produce signals at: 104 MHz, 98 MHz and 106 MHz, and at other frequencies that probably do not get through the mixing transmitters harmonic filter.

```
102 + 100 = 202 MHz
102 - 100 =   2 MHz
204 - 100 = 104 MHz--
204 + 100 = 304 MHz
200 - 102 =  98 MHz--
200 + 102 = 302 MHz
306 - 200 = 106 MHz--
306 + 200 = 506 MHz
```

Since the harmonics are involved in the mix, the modulation bandwidth will also be multiplied a like amount resulting in a very wide "spurs" that can be several channels wide.

Three or more carriers can also mix, but because of conversion losses it is unlikely that such a complicated mix will cause problems.

CO-CHANNEL BROADCAST

Interfering carriers propagating into your market from a distance can sometimes cause local reception problems.

There is usually not enough signal to enable the other station to pop out on top of you, but it can cause additional noise and distortion during multi-path nulls. This problem will vary with, weather, class of channel, and whether the other station is on the hot or cold side of a major building or geographical feature that is affecting your signal.

CABLE TV

Cable TV cable commonly will radiate signals in the FM broadcast band (and elsewhere).

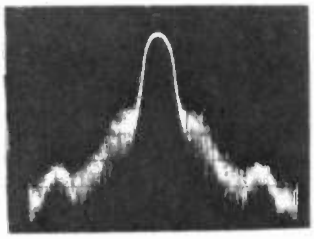
These signals can be from carriage of the station itself, or various imported non-broadcast services. Many cable systems also carry a marker beacon used for system alignment at or near 100 MHz. Make sure that your local cable company relocates all signals away from your on-air frequency.

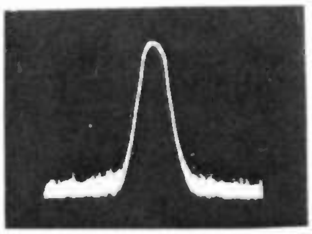
TRANSMITTER NOISE

Any amplifier or oscillator produces a certain amount of wideband noise.

When an excessive amount is generated in the broadcast transmitter due to poor neutralization, parasitic oscillations, or serious lack of drive, it can affect the reception of other stations.

Because of the artificially high back-ground noise level being radiated by the offending transmitter, FM reception on nearby channels can be poorer than expected for a mile or more around the antenna site.





Spectral display of noisy and clean unmodulated FM transmitter carriers.

OPTIMIZATION OF SCA

Until three years ago, subcarriers were very heavily regulated by the FCC and had demanding technical specifications that resulted in minimal interference and crosstalk into the host station. Since then the FCC has de-regulated the SCA specifications, and SCA companies have been able to redesign their SCA system in their own best interests.

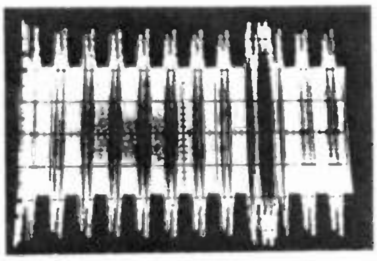
It is very difficult to quantify and document multi-path related SCA problems since it is not present under pristine laboratory conditions. It becomes detectable and obvious only on the streets and receivers in the real world.

Many FM stations have successfully carried a SCA subcarrier signal for years, as a way to supplement their income.

Others believe that the extra dollars are not worth damaging the listenability of their signal even slightly.

It is up to each station to evaluate the effects on it's listenability by a proposed, ongoing, or upgraded carriage of a SCA.

1. From a puritanical standpoint, the best SCA is no SCA.
2. If you have an SCA, keep or move it as far away from the sensitive stereophonic subcarrier as possible. I recommend SCAs be operated no lower than 76 KHz if you want to emulate digital quality for your station.
3. Make sure that there is no amplitude modulation of a FM or frequency shift SCA.
4. Make sure that the SCA is spectrally clean.
5. When given the choice, select an old style narrow deviation/shift SCA.



Oscilloscope display of a SCA subcarrier with an undesirable AM component.

AUDIO / STEREO MECHANICS

To avoid using two separate radio channels for stereo sound transmission, and to allow compatibility with existing monaural radio receivers, a matrix system was developed that allows normal monaural reception of a mixture of left and right channel information on existing radios through the use of multiplexing a single carrier.

The L-R components are transmitted on a subcarrier that is mixed in with the monaural audio. A stereo subcarrier detector then decodes the L-R information and uses the monaural L+R and L-R channels to re-assemble discrete Left and Right channels.

The system creates a few technical problems.

The multiplex system brings an increase in noise level due to the addition of the noise and interference in the AM double sideband stereophonic subcarrier region between 19 and 57 KHz.

Also the presence of a 19 KHz pilot signal is necessary to accurately detect the 38 KHz stereo subcarrier.

Simple harmonic distortion will cause non-harmonically related audio products to be thrown into the Stereophonic and SCA region causing interference to stereo and SCA reception. Intermodulation distortion causes mathematical addition and subtraction of all the frequencies present.

An example would be a SCA occupying spectrum between 62 and 72 KHz intermodulating with the 19 KHz pilot and appearing (at a reduced level due to conversion loss) between 43 and 53 KHz and 71 and 81 KHz.

The components between 43 and 53 KHz fall into the stereophonic subcarrier region and are then detected by the 38 KHz AM detector as garbage in the 5 to 15 KHz range.

MECHANICS OF SUBCARRIERS

Since all practical modulation contains alternating frequencies, and subcarriers consist of specific frequencies, we need to examine the wavelengths to calculate the distances where multi-path cancellations of the modulating frequencies will occur.

We can use the following formula to convert frequency to wavelength:

$$\text{Frequency (KHz)} = \frac{984{,}000}{\text{Wavelength in feet}}$$

There are several frequencies and their corresponding wavelength distances that we need to look at:

Freq	Use	Full	Half	Quarter
19 KHz	PILOT	51,790'	25,895'	12,948'
38 KHz	STEREO	25,895'	12,948'	6,474'
57 KHz	SCA	17,263'	8,632'	4,316'
67 KHz	SCA	14,687'	7,344'	3,672'
92 KHz	SCA	10,696'	5,348'	2,674'

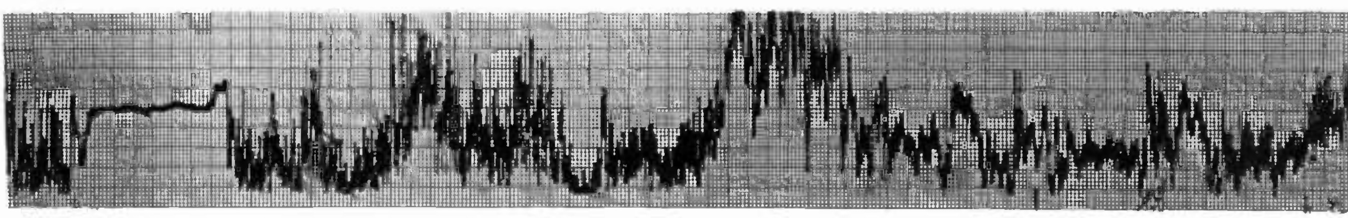
Most of these distances are significant when you remember that you have to allow for the round trip delay of the reflected signal both to and from the reflector as compared to the direct path to your receiver.

By using these distances, you can roughly determine at what distances from a major reflecting object you will experience problems.

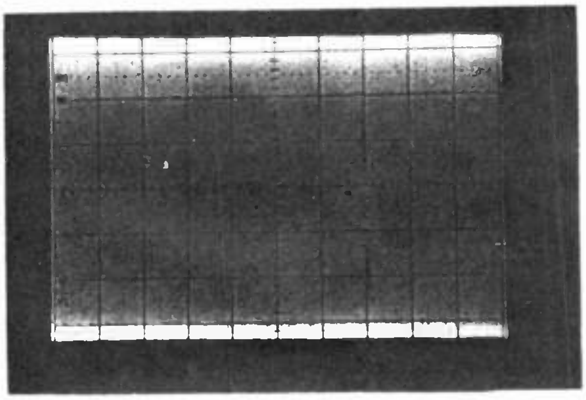
Stereo pilot problems at 12,948 feet in front of a major reflecting source.

L-R interference at 6,474 and 12,948 feet in front of a reflector.

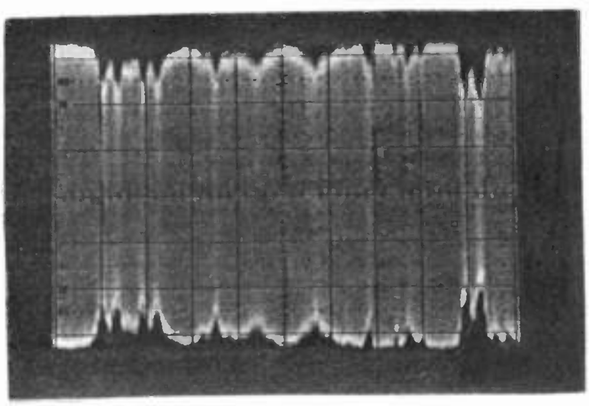
67 khZ SCA interference at 3,672 feet.



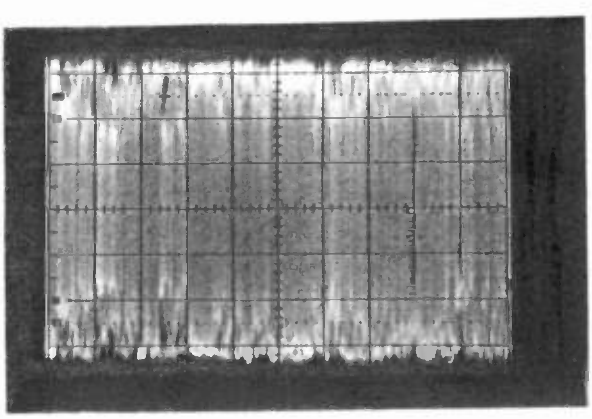
Mobile chart recording of FM field strength variations over a five minute period in a typical metro environment. Trace is calibrated from 1 to 10 mV. Variation in receive level exceeds 20 dB. Stable period was recorded when the automobile was waiting at a traffic light.



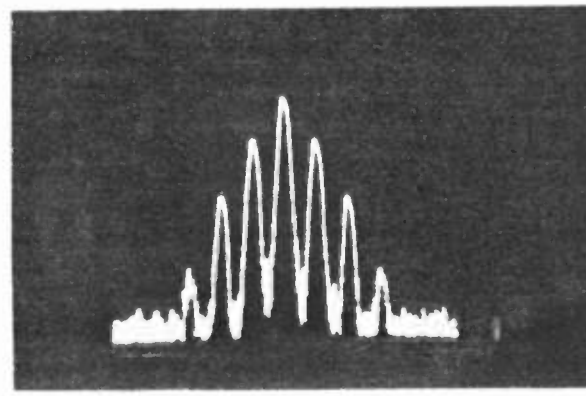
Carrier signal from properly broad-banded FM transmitter with full FM modulation, as displayed on 100 MHz scope.



Carrier signal with considerable synchronous AM content due to poor broadband performance.

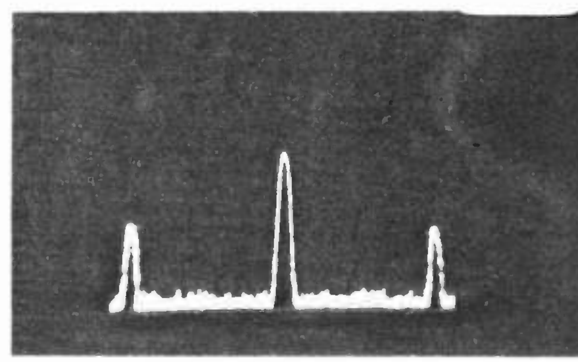


Carrier signal with severe AM problems.

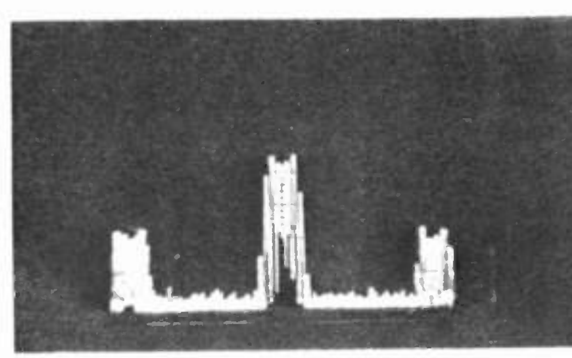


Spectral display of a FM carrier as modulated by a 19 KHz pilot at 10% injection (7.5 KHz Deviation).

Note the three significant sidebands on each side (spaced every 19 KHz). Additional sidebands would appear if the deviation were greater.



FM carrier with 92 KHz SCA. (first sidebands are located at + & - 92 KHz from center frequency)



FM carrier with 1 KHz modulation at 10% (7.5KHz Dev.) and unmodulated 92 KHz SCA at 10 % injection. Note how each modulating frequency creates additional symmetrical sidebands around every other frequency and that there is significant energy beyond + & - 75 Khz

RECEIVER PROBLEMS

The early FM stereo automobile receivers were not very successful from a signal to noise and multi-path interference standpoint. To improve the second generation automobile stereo receivers, a variable blend feature was incorporated by the manufacturers in order to improve stereo signal to noise.

The "stereo-blend" radios begin to blend (mix to mono) at between 1 and 5 mV of carrier level, and is in full blend at and below the .5 mV level.

Unfortunately, most metro FM listening is done in this range of carrier levels.

As the carrier level rapidly crosses the threshold, the "blend" transition becomes very irritating to the mobile listener. This is especially true if the left and right channels contain separate mid and high frequency information.

The results of this are often but erroneously referred to as signal fading.

Music dubbing, control room operators, and engineering must take into account the effects of variable blend as part of their audio quality control program.

MONITORING TRUE "AIR" QUALITY

Off air monitoring should be done with the same type of antenna as the average listener uses. I feel that this method of monitoring is far superior to the use of a large yagi antenna, high gain pre-amp and modulation monitor method that most stations use.

Is your off air signal good and listenable in stereo when using a line cord antenna, how about a 20" piece of wire or the ultimate home antenna, the 300 ohm "T" stuffed behind a table, shelf or tacked to the wall.

Several things will happen when you switch to actual off air monitoring, first, if you have any synchronous AM you will have fuzzy reception. You will know if your SCA is damaging your stereo signal under less than perfect conditions. Maybe your stereo light will even go out occasionally... in other words, to duplicate the receiving conditions of 99.9 % of your listeners.

CHECK-LIST...............................

1. Clean wideband transmission system:

   A. Synchronous AM measured and minimized to <-40 dB
      i. Wide-band transmitter tuning
      ii. proper driver and final neutralization
   B. Wide-band, low group delay filters, combiners, etc.
   C. True wide-band antenna

2. Correcting for stereo receiver irritations of "Variable Blend":

   A. Keep stereo audio in phase
   B. Air mono versions of early pseudo stereo material
   C. In severe cases, transmit pre-blended HF audio during drive times

3. Optimize stereo parameters:

   A. No harmonics of stereo pilot or stereo subcarrier
   B. SCA, if present must be super-clean, narrow band, with-out any amplitude variations and at or above 76 KHz.
   C. Do not synthesize mono recordings into stereo, when recombined to mono they don't always mix)
   D. Phase correct source material

5. Channel cleanliness:

   A. Intermodulation products
   B. Co-channel imports
   C. Cable TV leakage
   D. Wideband noise

CONCLUSION

Unfortunately, there is not a total cure for multi-path related signal problems.

In order to minimize reception problems associated with over the air listening, and to remain competitive with other entertainment mediums, the FM broadcast engineer must optimize and maintain his signal to the very limit that present technology will allow.

DESIGN CONSIDERATIONS FOR MULTI-STATION FM ANTENNAS

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For a variety of reasons, many FM stations are getting together and planning master antenna systems. There can be two (2) stations or twelve (12) stations, but regardless of the number there are certain things that are the same, such as; what type of combiner to use, how many and what type of antenna to use and how do we protect the system. As in all such projects there are always some things that we can not do anything about, such as; where we can put the antenna on an existing tower or how high the FAA will let us place the antenna. We will only be addressing three (3) portions of the antenna system in this paper; antenna, combiner and monitor/protection system. Keep in mind, however, that we will have many other considerations, such as; building space required, air conditioning requirements, electrical service requirements, etc.

MULTI-STATION ANTENNA

Since the antenna will determine the power handling requirements for the system, this is the place to start. If the antenna is to be installed on a new tower, we have eliminated a problem already since we can build the tower to support whatever antenna we choose. If the antenna is to be mounted on an existing tower, we will have to either use an antenna that the tower can accommodate or plan on *"beefing it up"*. In either case we will have to make the decision of how many bays of antenna we will have. Since the number of stations plus the number of antenna bays *(gain of the antenna)* will determine the power handling requirements of the antenna feed system, as well as the transmission line to the antenna, we must first determine how many bays of antenna will be required to do what we want regarding signal strength vs distance from the tower. The larger number of bays will provide lower signal strength in the first ten (10) miles and provide the highest gain, hence lowest transmitter power which results in lower power handling requirements in the feed system. Keep in mind that the more bays of antenna we have, the more tower we must have to support it. The height of the antenna center of radiation will also determine the close-in signal strength. The lower the center of radiation the more signal in close. Figure 1 shows a ten (10) bay antenna at 1000' *(solid line)* and 1500' *(dashed line)*, a ten (10) bay antenna at 1000' *(solid line)* and a twelve (12) bay antenna at 1000' *(dashed line)*. These plots run from 1 mile to 100 miles on a logarithmic scale. Vertical divisions are 5 dBu on a linear scale. These plots are for antennas with no beam tilt or null fill.

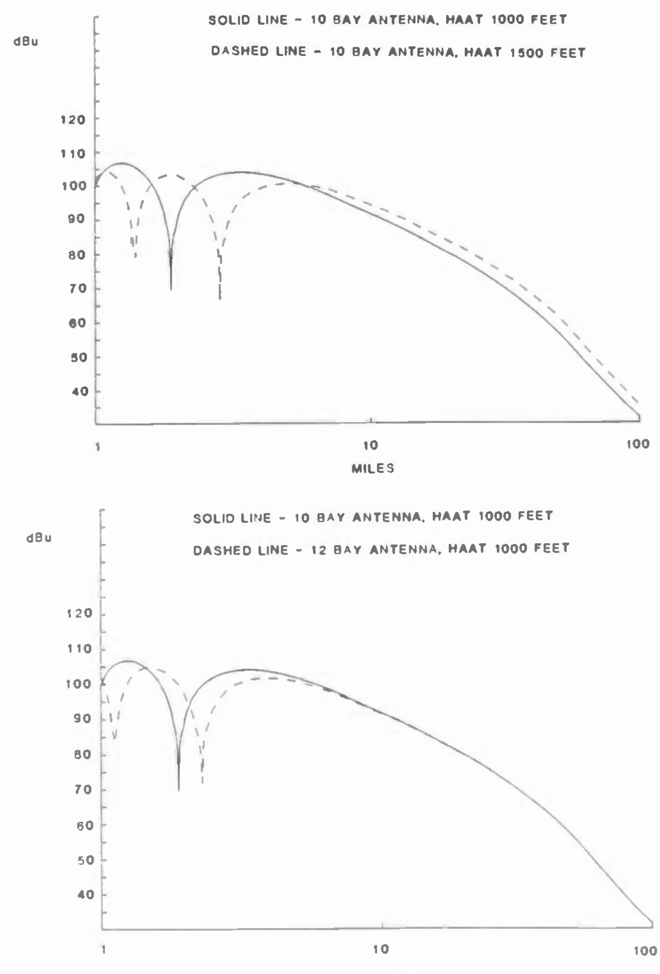


Figure 1

Since we are talking, in most cases, about six (6) to ten (10) stations, we will probably need a panel type antenna to handle the power due to the large number of elements we can divide the power into. This is important so that we can use the smallest size semi-flex type cable for the feed lines in order to fit them inside the supporting structure. Large cables require a large bending radius while the smaller size allows sharper bends to fit it all in. If we are mounting the antenna panels on an existing tower that has a large face we will have to go to four (4) panels per bay to achieve a good circular azimuth pattern and this will require more feed lines to find a place for. The circularity of the pattern will be determined by the size of the tower and the number of panels around, so keep this in mind when specifying this spec. If we are talking omni directional, +/- 2dB is considered omni by most engineers.

Do we want beam tilt or null fill? A glance at Figure 1 will tell us that we will gain nothing beyond about ten (10) miles by adding beam tilt. Null fill is another story and 10% null fill would be good and will not effect the pattern otherwise.

Because we want our antenna to have good circular polarization we want to spec axial ratio. This spec will determine what kind of antenna we will end up with. A cavity backed radiator type antenna will have a better axial ratio since the vertical and horizontal components are more equal in magnitude and phase. For this type of antenna we can expect a 2 dB ratio. If we want a *"flat panel"* type, the spec should be more like 4 dB.

As was pointed out earlier, each system is different and must be addressed on an individual basis. Let's assume the following conditions: eight (8) Class C stations *(100 KW ERP)*, new 1000' guyed tower with the antenna top mounted. Since we have a new installation we have the option of using one larger transmission line or split the antenna into two (2) halves and use two (2) smaller transmission lines. The latter will allow us to use either half of the antenna in case of a failure in the other half. As this is to be a first class operation let's use this dual line system.

Now that we know what we want, we can make some calculations and put sizes on the transmission line. The gain of a ten (10) bay antenna with 10% null fill will be 7.23 dB and as we will have eight (8), 100 KW stations on the system we will need approximately 19 KW per station or 152 KW total power. Since we are using a two (2) line system each line will have to deliver 76 KW to the antenna. In order to handle this power safely we must use two (2), 6 1/8", 50 ohm transmission lines. Our antenna will have 60 elements to divide the 152 KW into which gives us 2.53 KW per feed line. This power can be handled with 7/8" semi-flex transmission lines. From all of this data we can formulate the specifications for our antenna which we will submit to the manufacturer of our choice for bids. A set of specifications for this antenna is shown in Figure 2.

ANTENNA SPECIFICATIONS

Antenna Type	10 Bay CBR Panel *(2, 5 Bay Arrays fed in Phase)*
Azimuth Pattern	Omni Directional +/- 2 dB
Null Fill	10% First Null
Beam Tilt	None
Impedance	50 Ohms
Axial Ratio	< 2 dB
Polarization	Circular
Polarization Sense	Right Hand
VSWR	< 1.15/1.0 89-107 MHz 1.1/1.0 At Each Listed Carrier +/- 200 KHz
Radomes	As Required
Gain	To Be Measured
Power Input *(Total Antenna)*	152 KW Average 2,432 KWPeak

Figure 2

MULTI-STATION COMBINER

Now that we have our antenna specified we need to do the same for a combiner to feed the antenna system. As there is more than one type of combiner we must select the type to be used. Because we are going to be combining eight (8) stations, our choices are down to two (2); Bandstop *(Notch)* Constant Impedance or Bandpass Constant Impedance. These two (2) systems are shown in Figure 3.

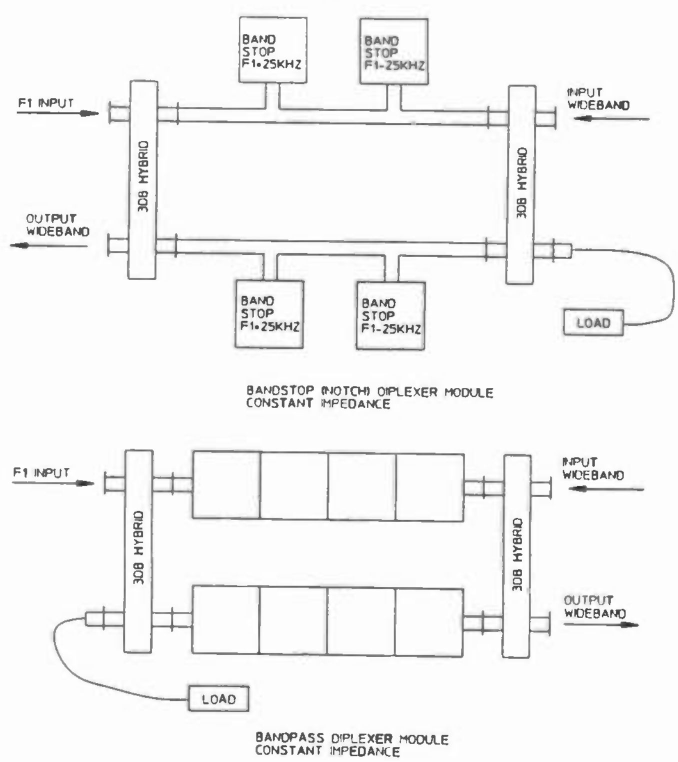


Figure 3

In the bandstop unit the notch cavities between the hybrids are tuned at, or near, the carrier frequency of the station feeding the module which enters at the narrowband input port marked F1 input. The power is divided equally by the hybrid and sent down the upper and lower legs of the diplexer. When the power reaches the cavities it is reflected back toward the input hybrid where it emerges at the wideband output port. Any power not reflected back to the input goes to the rightmost hybrid where it combines and goes into the reject load. All other stations feed similar modules and their wideband output is fed into the F1 module at its wideband input port at the top of the rightmost hybrid. Then its power is split and sent down the upper and lower legs of the diplexer. It passes the cavities as they are tuned to F1 only, recombines in the input hybrid and exits along with F1 at the wideband port. This system is considered a series system since all the signals pass through all modules between a station's input and the output of the diplexer.

In the bandpass system we have the two (2) hybrids, input and output, but now the reject cavities are replaced with bandpass filter units. The filter units are tuned so that they pass the input frequency of the station and have a response such that they are very wide and flat for +/-200 KHz and down 25 dB at +/-800 KHz. The modules for all stations are identical except for the frequency that the filter units are tuned to.

F1 enters the diplexer at the narrowband input port which is the upper port of the leftmost hybrid. It is then split and sent to the upper and lower filter units where it passes to the wideband output, or rightmost hybrid where it combines and exits the module at the wideband output port *(lower right most hybrid port)*. This port is connected to the next module's wideband input port. Any power reflected by the filter units goes back to the input or narrowband hybrid and is sent to the reject load on the lower port.

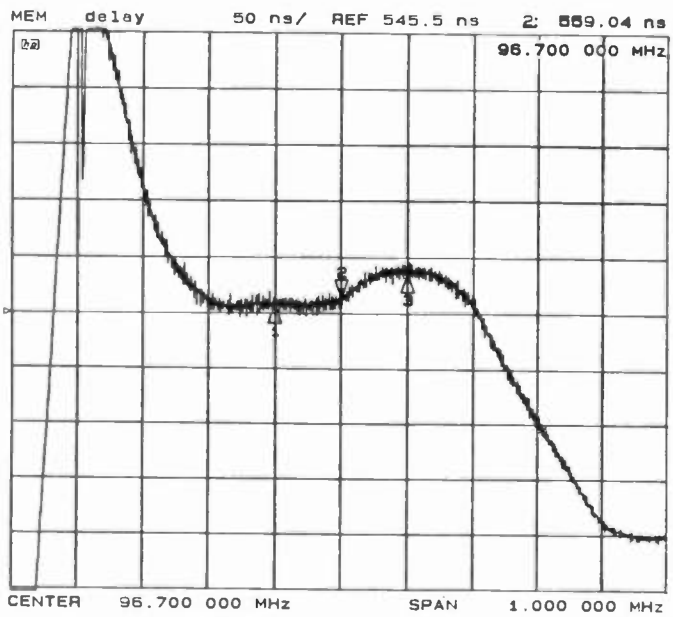
The signal from the F1 wideband output enters the F2 module at the wideband input port, splits and is sent toward the input end of the module, but since it is tuned to F2, the F1 power is reflected back to the wideband output hybrid and exits along with F2 at the wideband output port of the F2 module. This is repeated for all station modules in the system. The bandpass system is in essence a parallel system since the signal of each station only passes through its own module with the exception of the output hybrid.

The bandpass system has another point going for it and that is signals from nearby stations that could get into the system and cause an intermod product are filtered out of the system. In a reject system these signals can enter the system and cause intermod products and transmit them via the antenna.

In any multi-station transmission system intermodulation products are likely to develope. This is caused by power from one transmitter getting into another transmitter which acts as a mixer. The usual intermod product is a third order product and is composed of the fundamental of one transmitter mixing with the second harmonic of the transmitter creating the product. This product is spaced from the carrier of the generating transmitter by the difference in the two transmitter fundamental frequencies and can be radiated from the antenna if they can reach that point. The bandpass filter has two (2) ways in which it deals with this. The isolation from one narrowband input port to any other input port *(the path RF must take to get into another transmitter to generate the product)* is the sum of the input hybrid isolation and the response of the filter. Assuming the hybrid isolation is 30 dB and the rejection due to filter response is 25 dB *(minimum)* we will have 55 dB minimum isolation from input to input. Since our filter units are symmetrical, the rejection on the other side *(intermod frequency)* is the same or 25 dB. This 25 dB is added to the input to input isolation for a total of 80 dB minimum attenuation of the intermod product, assuming the combiner does all the work. No transmitter that we are aware of is 100% efficient as a mixer so there will actually be >80 dB rejection of the intermod product. This example is for 800 KHz spaced transmitters and if the spacing is greater the rejection will be greater. Interestingly, the FCC specification for intermod products is 80 dB below the unmodulated carrier *(reference)* when the transmitter is modulated with normal program material.

Group delay is a parameter that is being specified now in combiners and filters in general that are being used in transmitting systems. Group delay is specified as a deviation of delay in nanoseconds over a frequency span of KHz. Group delay will effect crosstalk and for those using SCA's this will manifest itself as main channel audio in the SCA channels. Main channel into stereo sub carrier can result with its attendant distorted. A new worry for those going into FMX is the effect of group delay on the phase relationship of the in-phase and the quadrature channels of the 38 KHz sub-carrier. This will be evident in receivers which are or are not equipped for FMX. While there has been little work done to put a number on how much group delay can be tolerated, we have some decent guide lines to use in establishing a specification. Measurements made on transmitters show that we can get +/-25 nano-seconds over +/- 200 KHz. Figure 4 is a plot of group delay through a 25 KW transmitter operating at full power.

Achieving this value of group delay is easily obtainable in all but the most closely spaced channels. In the case of 800 KHz spacings it requires equalizers on the input of the modules to achieve the +/-25 ns spec. In the overall combiner the cost of this will be minimal or nonexistent if there are no 800 KHz spacings. There are special cases where this specification has been relaxed in favor of a tighter response specification to meet unusual circumstances. Since we want our combiner to last through technology advancements in the future +/-25 ns over +/-150 KHz has been accepted as the standard. Even though a system with more group delay may not cause audible distortion, the future may change this and our system will be ready for it.

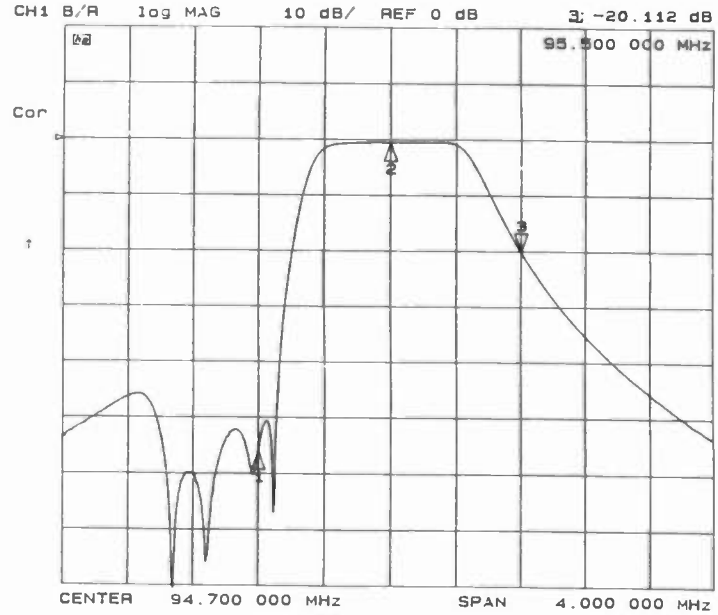


TRANSMITTER GROUP DELAY

Figure 4

In a reject combiner system the +/-130 KHz bandwidth *(approximately)* results in group delay of about +/-100 ns. This produces deviations four (4) times greater than those obtainable using state of the art techniques.

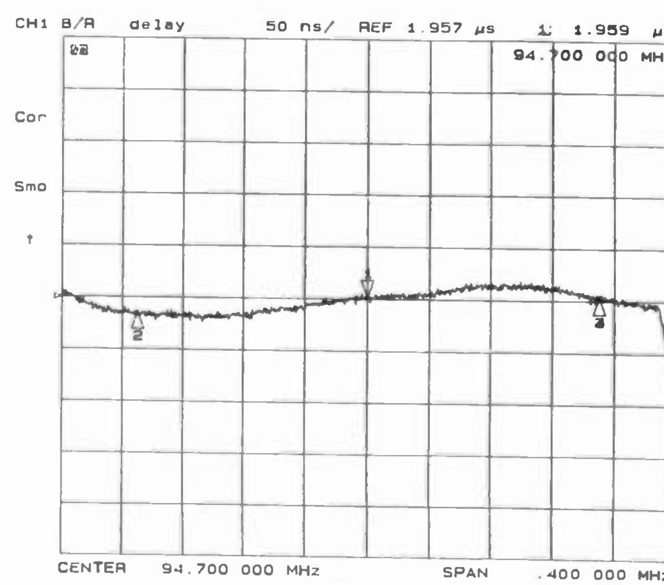
In the bandpass combiner, group delay is the sum of the narrowband input group delay and the wideband input group delay of all modules between the input and the combiner output. If we only look at an individual module we will see a symmetrical response and group delay for the module only. When we put the entire system together we should check the response and group delay through the entire combiner *(narrowband input to combiner output)* so that we see the real picture. If we have modules tuned to 800 KHz on either side of another module the group delay will rise rapidly on that side. Figure 5 illustrates this condition. If the module was only measured for response it would be symmetrical. In this instance there is a module following this module that is 800 KHz lower in frequency and when we measure the response through the entire system we see the effect.



RESPONSE

Figure 5

Under these circumstances it is possible to utilize a group delay compensation module. A group delay compensation module consists of a hybrid and two (2) cavities used as notch cavities. Since group delay is additive, the inverted response subtracts from the standard response effectively cancelling the group delay deviation to within the +/-25 ns. Figure 6 is a measured plot of the group delay from the module narrowband input port to the combiner output of the module shown in Figure 5 after a delay compensation unit was installed.



GROUP DELAY

Figure 6

As can be seen the group delay is well within our +/-25 ns specification. Figure 7 shows a drawing of a module with and without the group delay equalizer cavity units. This group delay improvement is at the expense of an additional 0.25 dB attenuation in that station's module. In large systems, the insertion loss can be high on the first module *(low power end)* due to the cumulative total of all wideband losses. This will be a rare condition since there will not normally be that many close spaced stations.

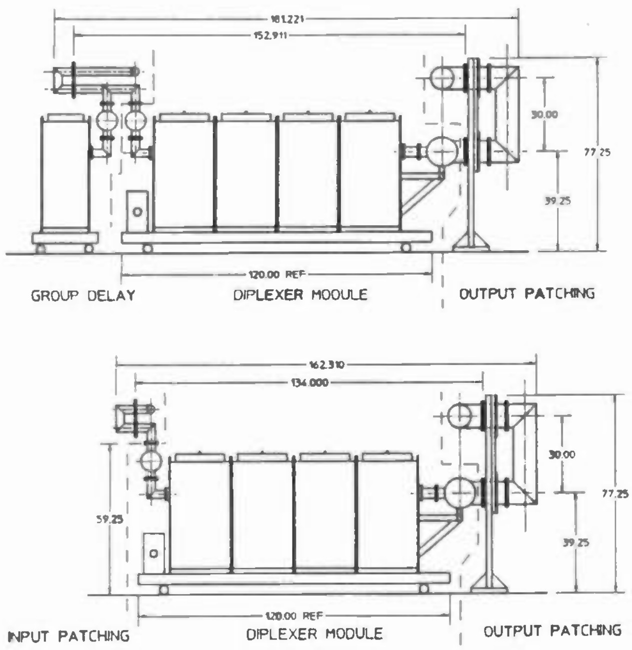
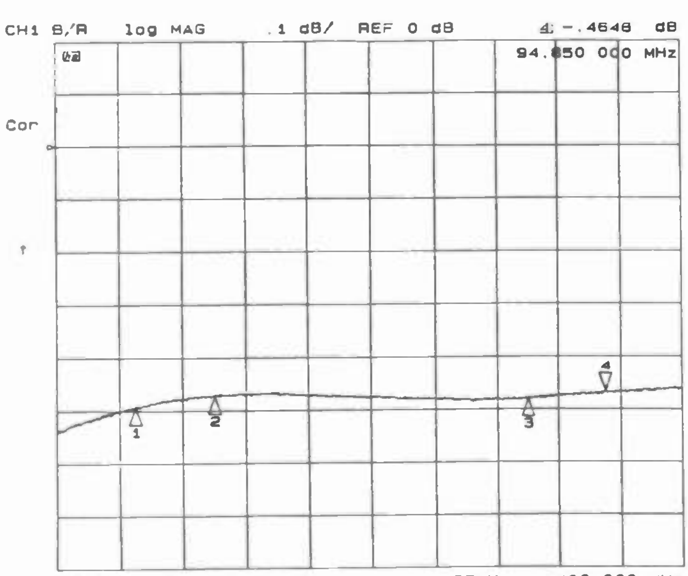


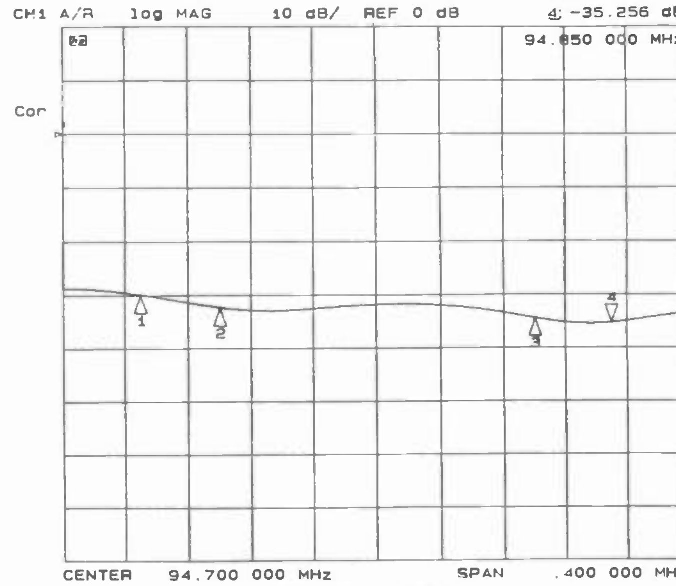
Figure 7

Figure 8 is a measured plot of the insertion loss of a module with group delay correction from its narrowband input through the entire combiner and Figure 9 is the measured return loss through the same path and indicates a VSWR of <1.05 across the channel.



INSERTION LOSS

Figure 8



RETURN LOSS

Figure 9

Due to the many advantages of the bandpass system, it is our system of choice so let's put some modules together in a system. Figure 10 shows five (5) modules hooked up in a complete system with two (2) of the station's modules requiring group delay correction. If the correction is not required or if a station chooses to take the delay as it comes, these units are modular and are mearly removed. In this layout the RF path is from right to left. The right side is referred to as the low power end. This drawing shows patch panels which allow a module to be patched out and that station put into the wideband port of the system which is normally terminated for emergency operation. The input patching is desirable, however, the output patches can be replaced with a *"long link"* just as well.

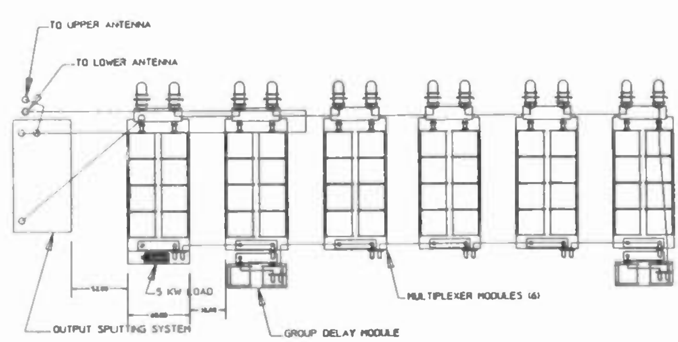


Figure 10

From this data we can write our specifications for the combiner and Figure 11 is what a set of specifications might look like.

COMBINER SPECIFICATIONS

Number of Inputs	Eight (8)
Frequency Range	88-108 MHz
Frequency Response, Each Channel:	
0 +/- 55 KHz	+0.0dB<-0.1dB
0 +/-100 KHz	+0.0dB<-0.2dB
0 +/-200 Khz	+0.0dB<-0.3dB
+/-800 KHz	$\geq$-25 dB
Isolation, All Inputs To All Other Inputs Across +/-100 KHz	55 dB Minimum
Insertion Loss[1]	0.6 dB Maximum
Power Input, Each Module (3 1/8" Coax)	25 KW Average
Power Output (9 3/16" Coax)	200 KW Average
Nominal Impedance, Input & Output	50 Ohms
VSWR, Each Module At Carrier[2]	1.05
+/-100KHz	1.10
+/-200KHz	1.15
VSWR, Broadband Port[3]	1.30
Group Delay +/-150 KHz	+/-25 Nano Seconds
Cooling	Convection

[1] Measured from module input to combiner output at antenna patch panel.

[2] Measured from module input to combiner output terminated at antenna patch panel.

[3] Measured at each module input patch panel with that module bypassed and termination at the antenna patch panel.

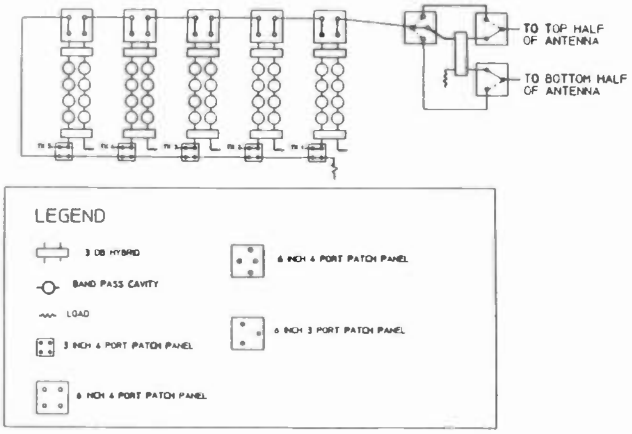
Figure 11

ANTENNA PATCHING

Since we have an antenna which is in reality two (2) antenna systems fed in phase, we must provide two (2) outputs from the one (1) combiner output to excite the antenna system, as well as, provide means for feeding the single combiner output to either top or bottom antenna for emergency use. This is done quite simply by feeding the combiner output into a pi/2 hybrid via a patch panel. In the normal mode the power is fed to the hybrid input port and the output ports to each antenna transmission via patch panels. Since the hybrid output ports are 90 degrees apart, one (1) antenna transmission line must be 90 degrees longer than the other at mid band to feed both antennas in phase. While this length will only be correct at one frequency, the beam tilt that results at frequencies below and above mid-band will be undetectable due to the width of the beam. An additional length of line equal to the difference in height of the feed points of the two (2) antennas must be taken into consideration. This can be done in the same correction section as the 90 degree correction.

In the emergency mode the pi/2 hybrid is patched out and the combiner output patched directly into whichever antenna is operable. Since the (2) two antennas *(or half of the antenna if you wish)* is not designed to take the full combiner power output, provisions will have to be made to reduce each transmitter to half power. This will be discussed in the monitor/protection system.

Figure 12 shows five (5) modules with the antenna patching system. The normal mode is shown by solid lines and the emergency modes are shown by dashed lines at the patch panels. By moving two (2) patches it is possible to bypass the hybrid and feed either the upper of lower antenna half.



SCHEMATIC OF 5 MODULE BANDPASS MULTIPLEXER

Figure 12

MONITOR/PROTECTION SYSTEM

The monitor/protection system for out multi-station FM antenna system can be as complex as desired. The main purpose is to protect the system in the event of a failure so that damage is kept to a minimum or prevented altogether.

The most efficient monitor/protection system makes use of a computer that keeps track of the system's configurations and has parameters for the various configurations, so, if the system is going to be overpowered, the transmitters can be prevented from operating at too high a power. For example, the system is patched so that all of the combiner output is fed to the upper antenna half. In this case all transmitters will be forced to half power by the system interface. If by some reason one of the transmitters is not switched to half power, the system will trip that transmitter through the interlock circuit, preventing damage to a transmission line or antenna.

The system can be programmed to make decisions and take the action necessary, providing electric coax switches are used in place of manual patches, or, it can use an auto dial modem to call a control point when something abnormal occurs and personnel can be dispatched to the site to take whatever action is required. This feature can be used to send status reports at predetermined intervals even though there is no problem. Regardless of whether we have a control point or the computer makes the decisions, we will want a print-out at regular intervals for a system record. We should also have the stations notified via the individual station's remote control system when a failure or problem occurs.

Protection of the antenna system should be accomplished by a power measurement of forward and reflected power. We should use a thermal type of power meter since it must read the total power of all the stations. Bird Watchers are unsatisfactory for this purpose since they use a diode detector that would require a bandpass filter and have a unit for each station, or, rely on one station to trip the system. The problem with this system is, if that station is off, you have no protection.

There are many options available when you use a computer based monitor/protection system, however, it is easy to make it so complex that not only will the cost be very high, but due to the complexity of such a system, you can end up with a *"tail wagging the dog"* type of situation. Regardless of which way you go with the system, you should use industrial grade PC components and computer.

An important consideration is to know if and when the computer has failed and you do not have any protection. This can be done by having the computer do another task such as reset a countdown timer. If the timer is not reset before it times out you can assume the computer has failed. If this happens the system should be shut down and the stations notified via the remote control systems. Some stations will not want to be shut down because of a computer failure. In this case, while it is not recommended, the system can be dropped to half power and all stations notified via the remote control systems.

If a station is shut down for any reason, the system should be such that an engineer must go to the site to verify the problem and take corrective action and then reset the system to allow the transmitter to come back up. If this is not done and the transmitter is repeatedly brought up by remote control, damage could occur to the system. The purpose of the protection system is to keep this type of thing from happening.

Figure 13 is a photograph of one monitor/protection system now in use on an operating system. The rack on the left houses the power meters for the two (2) transmission lines, a back up Bird Fastwatcher monitoring one station through a bandpass filter, the sampling unit and the computer. The right hand rack houses the Bird Watchers used to sample the forward and reflected power to each module and is used in their normal configurations as a back up system.

Figure 14 is a photograph of a six (6) station combiner system. This system uses group delay equalizers on two (2) modules. While this system uses patch panels on the output of each module, it is not necessary provided some means is provided to bypass the module such as a long link described earlier.

Figure 15 is photograph of the antenna patch panel described earlier to provide the necessary facilities to use half of the antenna system.

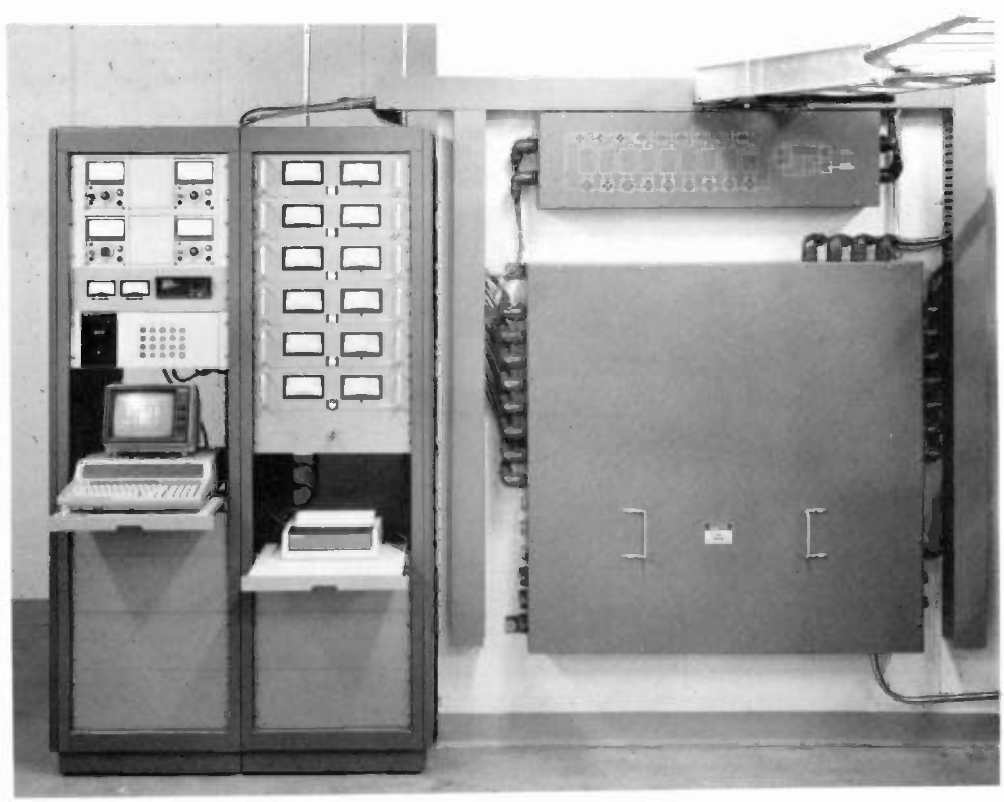


Figure 13

Figure 14

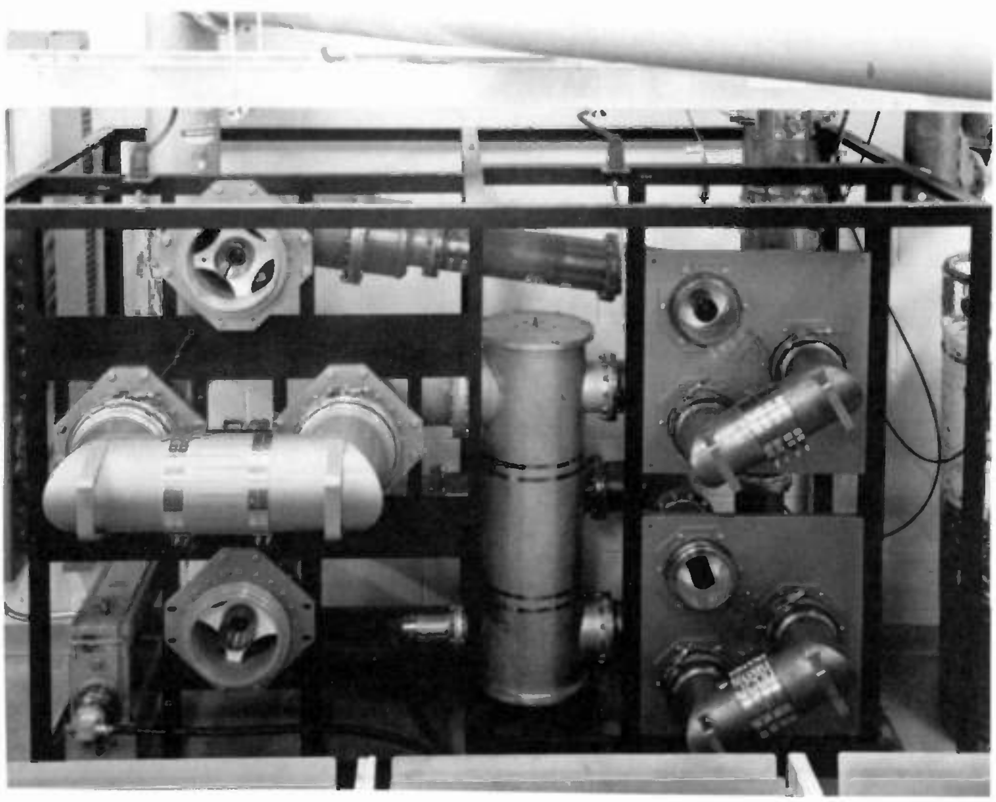


Figure 15

TECHNIQUES FOR MEASURING SYNCHRONOUS AM NOISE IN FM TRANSMITTERS

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INTRODUCTION

This paper presents an explanation of synchronous AM noise caused by FM modulation of limited bandwidth systems. Synchronous AM noise is presently one of the "hottest" topics among FM and TV-Stereo broadcast engineers. The causes of this type of incidental AM modulation in the presence of FM modulation are reviewed with emphasis on the practical application of synchronous AM noise measurements to optimize transmitter tuning.

The value of synchronous AM measurements in evaluating the transmission system as well as test equipment selection, system set-up, and interpretation of results are explained. The type of equipment and measurement procedure used can lead to incorrect results. The author explains how to avoid these "pitfalls" so that the station engineer has confidence that the measured results are correct.

It is hoped that this paper will be a valuable reference that broadcast engineers can use to improve the operation of their transmitter systems.

TWO TYPES OF AM MODULATION

The perfect FM transmitter would have an absolutely constant output, regardless of FM modulation or power supply variations. In practice, there is always some residual amplitude modulation of the FM transmitter output. There are two types of AM modulation that are of interest to the FM broadcast engineer:

1. Asynchronous AM modulation is measured without FM modulation and is primarily related to power supply ripple or filament supply imbalance. This is the only type of AM noise measurement that is required by the FCC.

2. Synchronous AM modulation (incidental AM) measured with FM modulation is related to the tuning and overall bandwidth of the system. Synchronous AM noise is not a concern of the FCC.

Asynchronous AM

Residual amplitude modulation of the transmitter output, due primarily to power supply ripple, is measured with an AM envelope detector. Most FM modulation monitors include an AM detector for this purpose. This detector should include 75 microsecond de-emphasis on its output. The residual AM noise in a properly operating FM transmitter will be at least 50 dB below the level which would represent 100 percent amplitude modulation of the carrier. If the transmitter is unable to meet the 50 dB requirement, the problem can usually be traced to a power supply component including the filament supply or to line imbalance in a three phase system.

Synchronous AM

Synchronous AM is a measure of the amount of incidental amplitude modulation introduced onto the carrier by the presence of FM modulation. This measurement is very useful for determining the proper tuning of the transmitter. Since all transmitters have limited bandwidth, there will be a slight drop-off in power output as the carrier frequency is swept to either side of the center frequency. This slight change in RF output level follows the waveform of the signal being applied to the FM modulator causing AM modulation in synchronization with the FM modulation. The concept is similar to the slope detection of FM by an AM detector used in conjunction with a tuned circuit.

Both types of AM noise measurements are made directly at the transmitter output (or an accurate sample of its output). No amplifying or limiting equipment may be used between the transmitter output and the AM detector since non-linearities in this equipment could modify the AM noise level present. Since the transmitter cannot be fully amplitude modulated, an equivalent reference level must be established indirectly by a measurement of the RF carrier voltage.

Refer to the instructions of the detector manufacturer to determine this reference level. Generally, the reference level is determined by setting a carrier level meter to a specified reading or to obtain a specific DC voltage level at the output of the detector diode without modulation.

WHY IS SYNCHRONOUS AM IMPORTANT?

Measurement of synchronous AM gives the station engineer an idea of the overall system bandwidth and whether the passband is positioned correctly. Tuning for minimum synchronous AM will assure that the transmitter passband is properly centered on the FM channel.

How Does Tuning Affect The FM Sidebands?

The higher order FM sidebands will be slightly attenuated in amplitude and shifted in phase 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 usually be minimized by centering the passband of the amplifier around the signal being transmitted. This will cause the amplitude and phase 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 passband is to tune the amplifier for minimum synchronous AM modulation while applying FM modulation to the transmitter.

How Good Should Synchronous AM Be?

Synchronous AM of 40 dB or more below equivalent 100% AM, is considered to be acceptable. Higher levels of synchronous AM will cause increased "chopping" of the signal at the receiver near limiting threshold under weak signal "fringe area" conditions and can exacerbate multipath problems. Excessive synchronous AM is also an indirect indication of passband induced distortion problems that degrade stereo performance and SCA crosstalk.

Many of the older multi-tube transmitter designs presently in use will have as much as 5% (-26 dB) synchronous AM when simply tuned for best power output and efficiency even though the asynchronous AM (without modulation) may be better than -50 dB. Some of the newer single tube transmitters can be adjusted for 50 dB or more suppression of synchronous AM. The synchronous AM level of virtually any FM transmitter can be improved by proper tuning techniques. An approximation to the overall system bandwidth can be related to the synchronous AM as shown in Table 1.

TUNING YOUR TRANSMITTER FOR PEAK PERFORMANCE

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.

APPROXIMATE SYSTEM BANDWIDTH AS RELATED TO SYNCHRONOUS AM

SYNCHRONOUS AM (below equivalent 100% AM) (with ±75 kHz @ 400 Hz FM)	APPROXIMATE BANDWIDTH OF TRANSMITTER (-3dB)	RF LEVEL VARIATION AT RECEIVER LIMITER (%)	(dB)
-30 dB	580 kHz	3.16%	.28 dB
-35 dB	760 kHz	1.77%	.17 dB
-40 dB	1.0 MHz	1.00%	.09 dB
-45 dB	1.4 MHz	.56%	.05 dB
-50 dB	1.8 MHz	.32%	.03 dB
-55 dB	2.4 MHz	.18%	.02 dB
-60 dB	3.4 MHz	.10%	.009 dB

TABLE 1.

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 halfwave 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. Quarterwave 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 setpoint 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.

MINIMIZING SYNCHRONOUS AM

After the correct loading point has been set, FM modulate 100% (±75 kHz) at 400 Hz and fine-adjust the transmitter's input tuning and output tuning controls for minimum 400 Hz AM modulation as detected by a wideband envelope detector (diode and line probe). The input matching and output loading controls should need no further adjustment at this point. It is helpful to display the demodulated output from the AM detector on an oscilloscope while making this adjustment. The output of the AM envelope detector should be connected to the vertical input (Y input) of the scope while the sweep is triggered by a sample of the 400 Hz audio tone fed to the external trigger input. This is called the "AM WAVEFORM" measurement. Note that as the minimum point of synchronous AM is reached, the demodulated output from the AM detector will double in frequency from 400 Hz to 800 Hz, 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. This effect is illustrated in Figure-1.

SYNCHRONOUS AM WAVEFORMS AND CALCULATIONS

DIRECT MEASUREMENT OF SYNCHRONOUS AM NOISE USING A HALF WAVE PRECISION ENVELOPE DETECTOR AND OSCILLOSCOPE.

$$\text{VOLTAGE RATIO} \left[\frac{\text{AC}_{p.p} \text{ VOLTS (AC MODULATION)}}{\text{2X DC VOLTS (RECTIFIED CARRIER)}} \right]$$

$dB = 20 \ LOG_{10}$(VOLTAGE RATIO)
(BELOW 100% EQUIV AM)

%AM = 100 x (VOLTAGE RATIO)

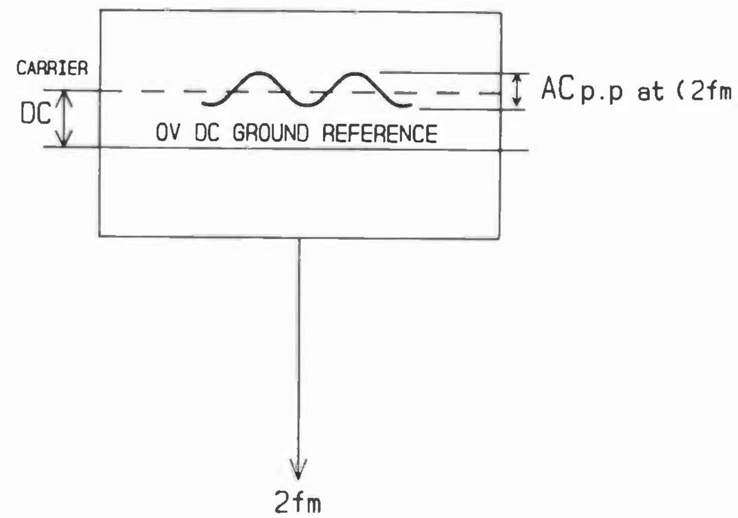
EXAMPLE:

RECTIFIED CARRIER DC = 940MV
AC MODULATION AC = $4.6MV_{p.p}$

$$\text{VOLTAGE RATIO} = \frac{4.6\times10^{-3}}{2\times940\times10^{-3}} = \frac{4.6\times10^{-3}}{1880\times10^{-3}} = .002447$$

$dB = 20 \ LOG_{10}(.002447) = -52.23dB$
$\%AM = 100 \times (.002447) \approxeq 0.25\%$

SCOPE DISPLAY OF HALFWAVE ENVELOPE DETECTOR OUTPUT



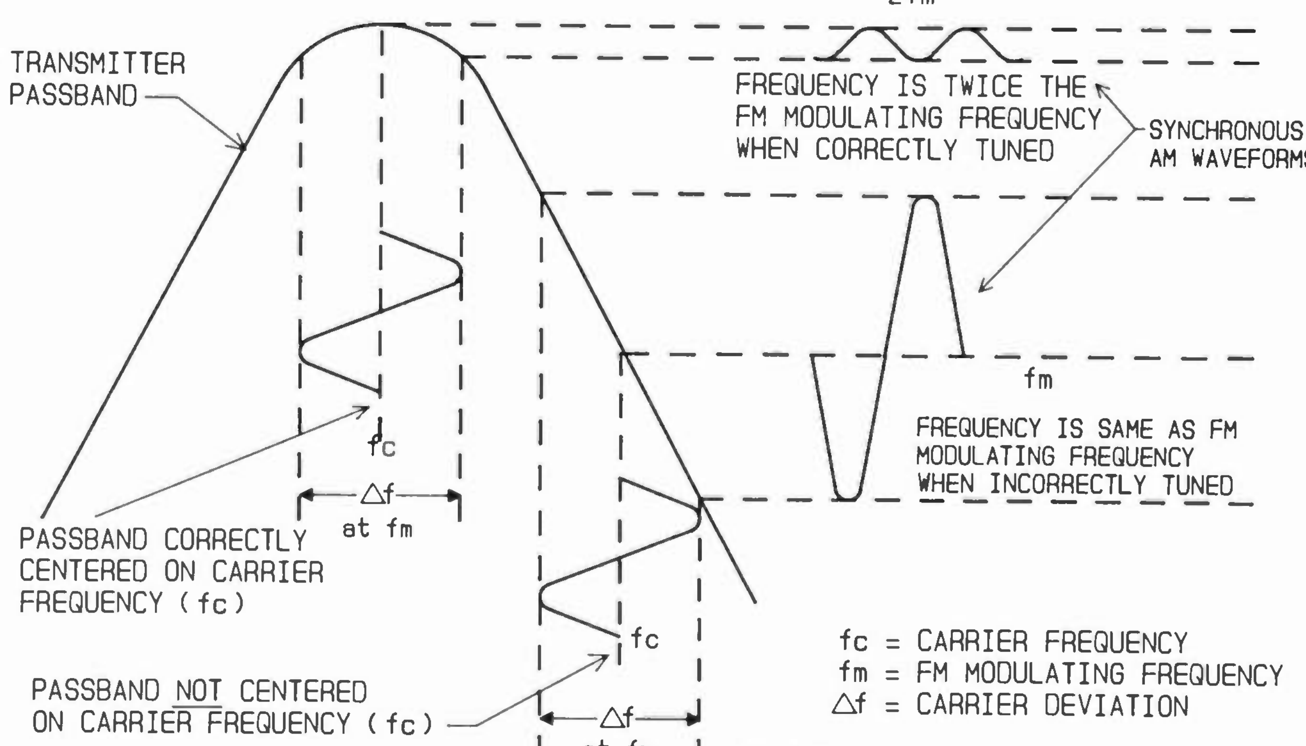


FIGURE - 1

The advantages of observing the demodulated AM waveform versus time, is that the frequency doubling effect is a sensitive, clear, indication of the symmetrical tuning point and the actual level of the AM noise below equivalent 100% AM modulation can be calculated from the waveform's AC and DC components. The disadvantage of this measurement technique, is that it cannot be performed with normal program audio present.

If it is necessary to touch-up the transmitter tuning with normal program audio present, an X - Y display of demodulated AM on the vertical axis (Y input) versus the audio input to the FM exciter on the horizontal (X input) axis, will provide a representation of the transmitter's passband as shown in Figures-2A and 2B. This is called the "PASSBAND" measurement. Figure-2A shows the relative amplitude of the transmitter's output power versus deviation from the center frequency with single tone 400 Hz modulation. Figure-2B shows the same information except that complex program modulation is present.

When making the "PASSBAND" measurement on stereo multiplex transmissions, best results will be obtained if the horizontal input of the scope is driven by a sample of the composite baseband being fed to the FM modulator rather than L+R program audio. A sample of the composite baseband being fed to the FM modulator can be conveniently obtained from the front panel composite test jack provided on some FM exciters.

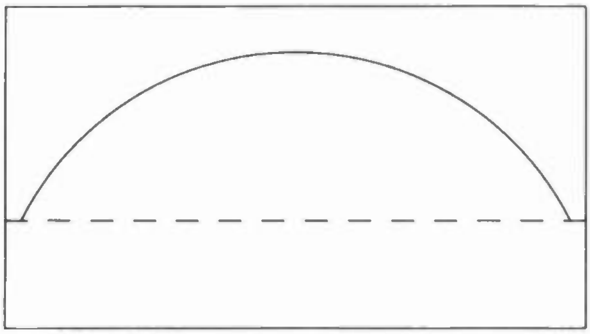
TEST EQUIPMENT SETUP

Figure-3 illustrates a typical test equipment setup and shows a block diagram of the required test equipment for making synchronous AM waveform measurements. A precision envelope detector with high return loss (low input VSWR) is used so that accurate synchronous AM waveforms can be observed while tuning the FM transmitter. Both the "AM WAVEFORM" and "PASSBAND" measurements can be made depending on whether the scope is in the triggered sweep mode or the X-Y mode. Composite baseband can also be routed into the test setup so that fine tuning can be done with normal programming being broadcast. It should be possible to minimize synchronous AM while maintaining output power and sacrificing little efficiency in a properly designed power amplifier.

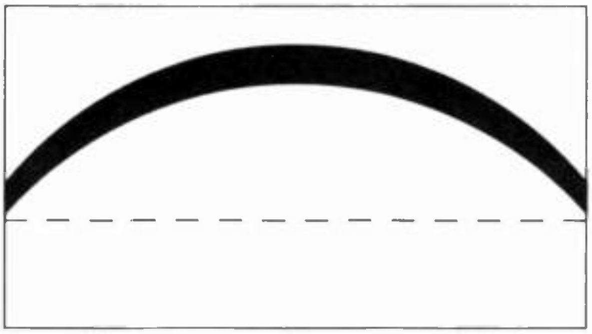
CALCULATING AM NOISE DIRECTLY FROM THE DEMODULATED AM WAVEFORM

Most FM demodulators cannot be relied upon to make accurate asynchronous AM noise measurements so it is a good idea to cross check the demodulator reading directly against the demodulated output of a precision envelope detector. This can be done by first measuring the DC component of the waveform with a voltmeter or by DC coupling the scope input. The scope is then AC coupled and the input sensitivity is increased until an accurate peak to peak measurement of the AC modulation component can be made. The peak to peak AC voltage is then divided by twice the DC component to obtain the "VOLTAGE RATIO".

X(HORIZONTAL) VERSUS Y(VERTICAL) "PASSBAND" WAVE FORMS SHOWING SYNCHRONOUS AM



400Hz TEST TONE
FIGURE - 2A



WITH TYPICAL PROGRAM MODULATION
FIGURE - 2B

BLOCK DIAGRAM FOR SYNCHRONOUS AM MEASUREMENTS

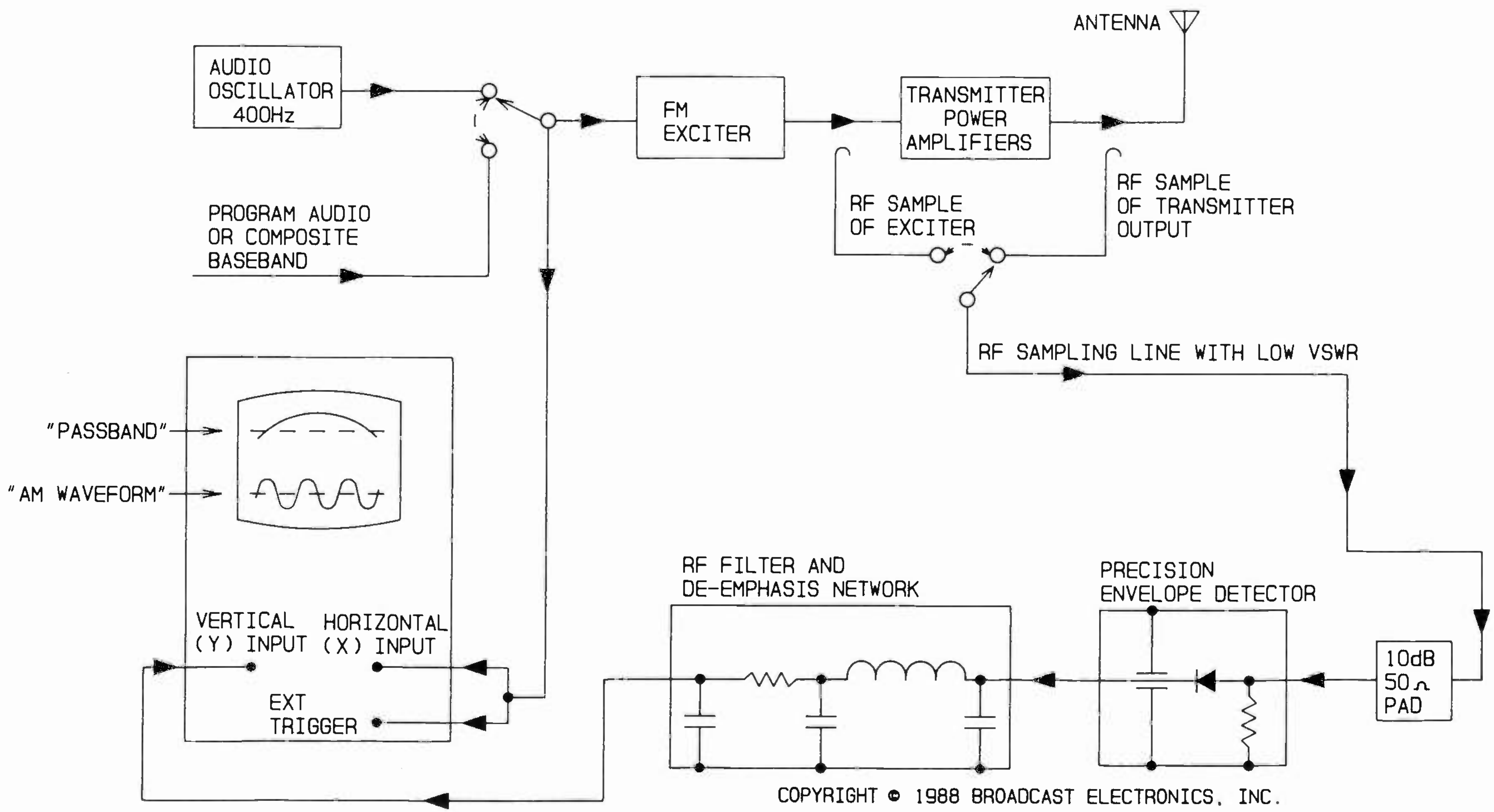


FIGURE - 3

Twenty times the LOG (base 10) of the "VOLTAGE RATIO" is the actual AM noise level in dB below equivalent 100% AM modulation. Multiplying the voltage ratio by one hundred yields the percent of AM modulation. Figure-1 illustrates these calculations.

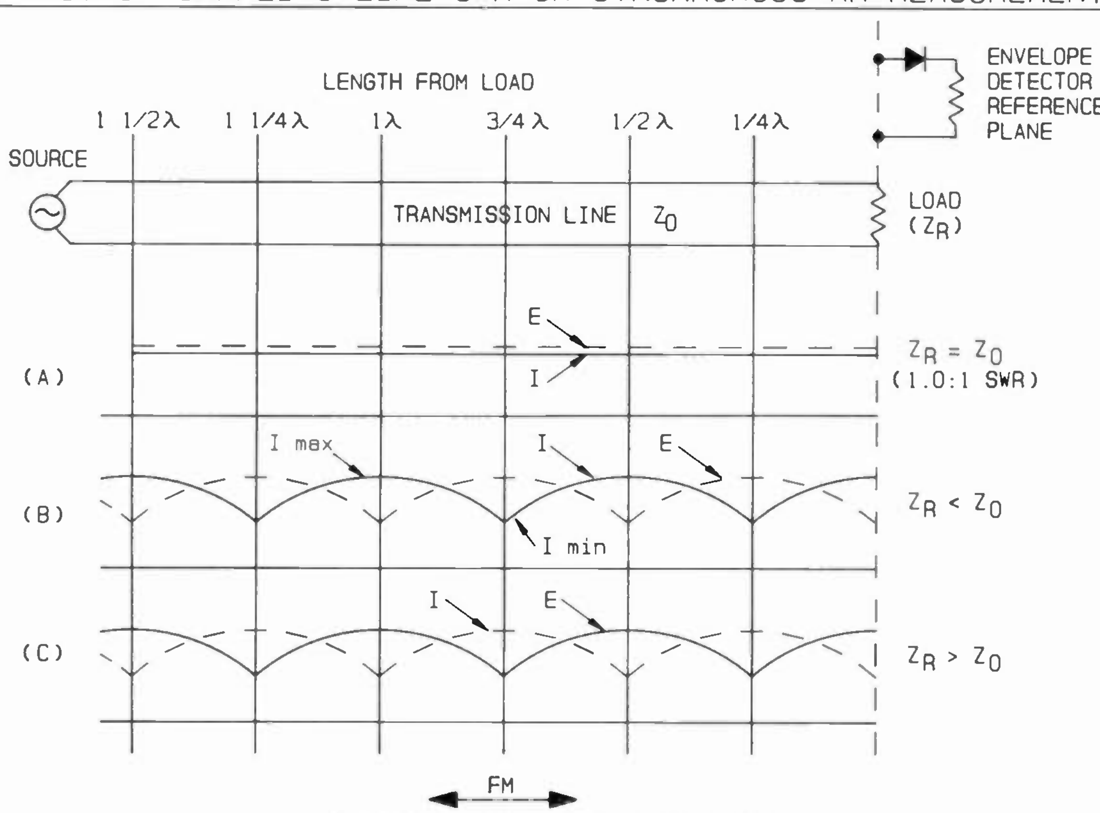
The Need For A Precision Envelope Detector

Care must be taken when making these measurements that the test set-up does not introduce synchronous AM and give erroneous readings which would cause the operator to mistune the transmitter to compensate for errors in the measuring equipment.

The input impedance of the envelope detector must provide a nearly perfect match so that there is a very low VSWR on the sampling line. Any significant VSWR on the sampling line will produce synchronous AM at the detector because the position of the voltage peak caused by the standing wave moves along this line with FM modulation.

Figure-4 illustrates the effect of the standing wave ratio on the RF voltage presented to the envelope detector. As the sampling line length is increased, the amount of erroneous AM caused by a given standing wave ratio also increases because each additional quarter wavelength causes more movement of the standing wave with FM modulation. Unfortunately, the AM detectors supplied with most modulation monitors do not provide a good enough match to be useful for this measurement. Precision envelope detectors are available from Wide Band Engineering Inc. (model A33) and Hewlett Packard (model 8471A option 004) that provide a 30 dB return loss (1.06:1 VSWR) to the sampling line when combined with a 10 dB, 50 ohm resistive pad.

EFFECT OF SAMPLING LINE SWR ON SYNCHRONOUS AM MEASUREMENTS



STANDING WAVES ON A TRANSMISSION LINE TERMINATED IN A RESISTIVE LOAD. FREQUENCY MODULATION MOVES STANDING WAVE BACK AND FORTH ALONG LINE CAUSING SYNCHRONOUS AM TO APPEAR AT ENVELOPE DETECTOR.

FIGURE - 4

Thru-Line Alternative To Precision Envelope Detector

A thru-line type of directional coupler normally used to drive the wattmeter movement, has the envelope detector diode built into the sampling element and provides a DC component that the meter movement responds to plus the demodulated AM noise component that the meter movement does not respond to. If the thru-line element output is fed to an oscilloscope instead of the wattmeter movement, the synchronous waveform can be accurately measured. This approach eliminates the errors due to VSWR on the sampling line, since the detector is located at the sampling point. Figure-5 shows how to use a thru-line coupler for making synchronous AM measurements. The manufacturer of the thru-line coupler can supply the special connectors and/or cables to connect its output to the oscilloscope. Care must be taken to avoid hum pick-up from AC ground loops while making these low level measurements.

RF Filter and De-Emphasis Network

Both the thru-line element detector and the precision envelope detectors have some residual RF on their DC output, so a combination RF filter and 75 microsecond de-emphasis network should be placed between the detector and the input the oscilloscope. Figure-5 shows a suggested configuration for this filter which can be easily constructed in a small shielded enclosure.

Built-In AM Noise Measurement Capability

Broadcast Electronics has recently developed a built-in precision envelope detector and de-emphasis network that will be added to the Automatic Power Control (APC) system used in the "A" series of FM broadcast transmitters. A calibrated front panel AM noise test jack will allow observation of the synchronous AM waveforms or direct measurement of the synchronous AM noise level on a standard audio voltmeter.

MINIMUM SYNCHRONOUS AM 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 for a while 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 elliptical 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.

The point of minimum synchronous AM occurs closer to the minimum plate current than peak efficiency, so there is a compromise between good synchronous AM and best efficiency. A properly designed and neutralized transmitter should be able to achieve minimum synchronous AM without giving up more than about 3% in efficiency.

TUNING SENSITIVITIES

In any of these tests, the input tuning is frequently more critical than the output tuning. This is because the impedance match into the input capacitance of the final tube's grid becomes the bandwidth limiting factor. Even though the amplitude response appears flattened when the grid is heavily driven, the phase response still has a serious effect on the higher order FM sidebands.

SOURCES OF PRECISION ENVELOPE DETECTORS

Wide Band Engineering Inc.
P.O. Box 21652
1838 East University Drive
Phoenix, AZ. 85036
(Model A33)

Hewlett Packard Inc.
1820 Embarcadero Road
Palo Alto, CA. 94303-3308
(Model 8471A option 004)

THRULINE® is a registered trademark of:

Bird Electronic Corporation
30303 Aurora Road
Solon, OH. 44139

DETECTOR SETUP FOR SYNCHRONOUS AM MEASUREMENT

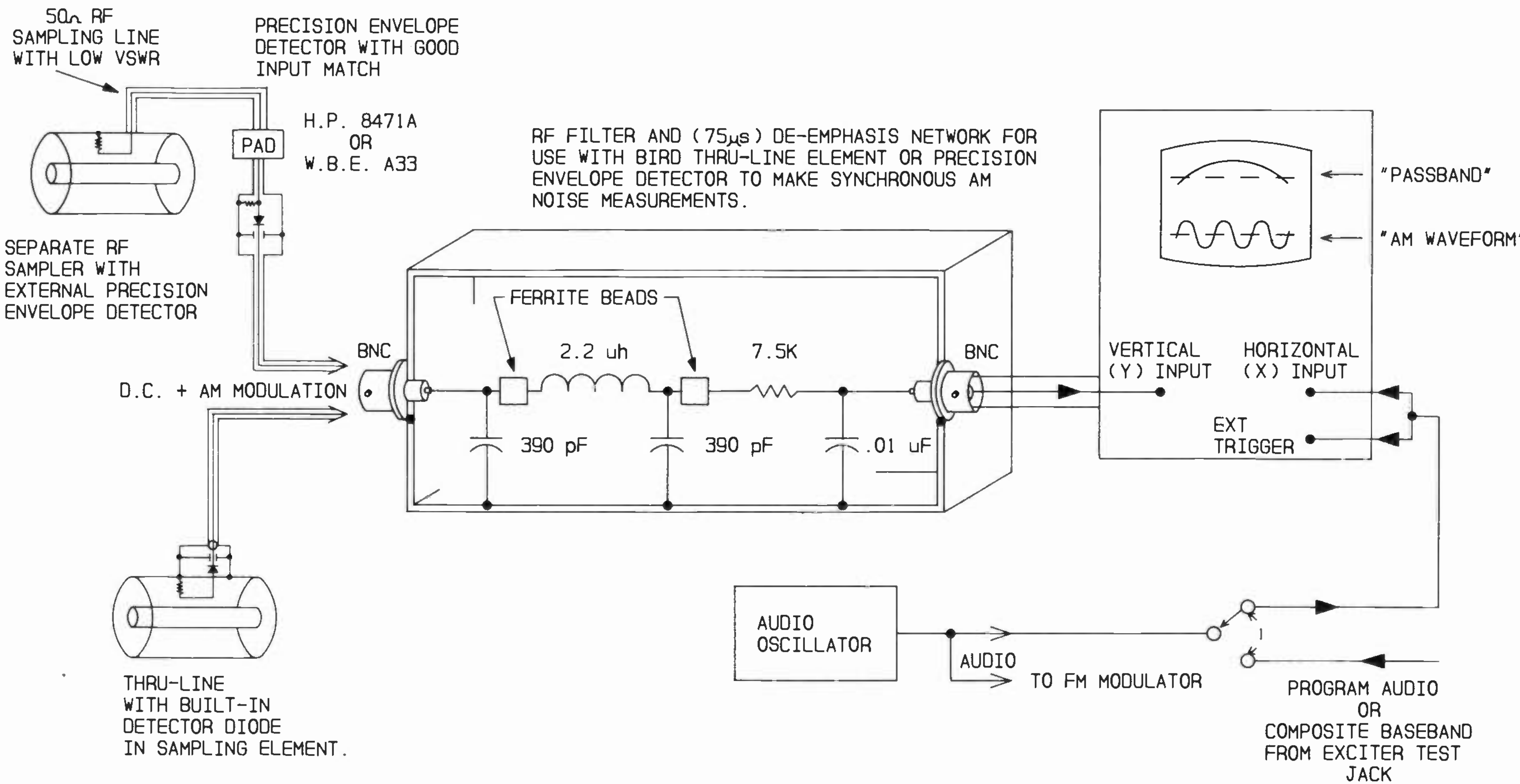


FIGURE - 5

ACKNOWLEDGMENTS

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Mr. Mendenhall is presently serving as Vice President of Engineering for Broadcast Electronics, Inc. in Quincy, Illinois.

The author holds three U.S. Patents for electronic designs utilized in broadcast equipment and is a registered professional engineer in the State of Illinois. He has authored numerous technical papers, is an associate member of the AFCCE, and a senior member of the IEEE.

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SMART AUDIO SWITCHER SOLVES PROGRAMMING AND ROUTING PROBLEMS

Charles Kelly, Jr.
International Tapetronics
Bloomington, Illinois

Radio is changing. In the past 5 years it has become more competititive, more dynamic, and more responsive. Stations are being sold and moved, with formats and calls changing almost daily. A multitude of new programming sources are finding their way into our facilities, from satellite channels to remotes and call in shows. These changes are resulting in changes in the way we as engineers work as well.

In the past, when a studio was remodeled, or built from the ground up, the engineer developed the configuration of the new studios by analyzing the station or AM/FM combination and its format and projecting future needs. The new station may not change appreciably for 5 to 10 years. This process often fails today as the format may be changed before the construction project is complete. What is needed is a method which allows fast and easy reconfiguration of a station.

When the television industry faced a similar problem of mushrooming sources and changing requirements some years ago, they turned to a technology pioneered by the phone company, routing switchers. These electronic marvels can switch our telephone calls quickly and accurately, from millions of phones in use worldwide. This development revolutionized the telephone business, and the operator with her rows of plugs and jackfields are gone forever. In television, the switcher also changed the business, and allowed tremendous flexibility and complexity with a minimum of confusion and error. Hundreds of input sources like cameras and consoles are switched to hundreds of output destinations like monitors and VTR's.

WHAT IS A ROUTING SWITCHER ?

Basically it is a group of switches arranged such that any input may be connected to any output or group of outputs. The manner in which the switches are arranged is often known as a matrix. A matrix can be visualized as a rectangle, with the inputs connected to the vertical side and the outputs along the bottom. Where the lines cross is known as a crosspoint, the actual switch. This crosspoint may either be open or closed, and thus any horizontal line (input) may connect to any vertical line (output). The number of switches required by any switcher is equal to the number of inputs multiplied by the number of outputs. As shown in Figure 1, the rest of a routing switcher merely augments this basic matrix.

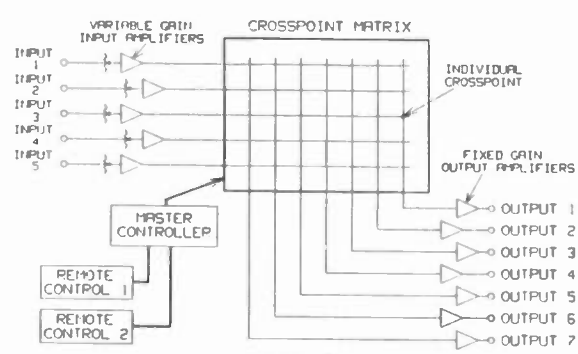

FIGURE 1.

In stereo systems, the mono matrix is doubled, so that a left and right matrix exist, and they are switched in parallel by the controller. This configuration, shown in Figure 2 is ideal for all stereo systems, but radio stations seldom are totally stereo. Even in FM stand alone operations a considerable number of mono inputs often exist such as phone lines and remote news RPU's. When it is necessary to connect a mono input to a stereo dual matrix, the input must be connected in parallel to both the left and the right matrices. This problem gets worse in AM/FM combination facilities.

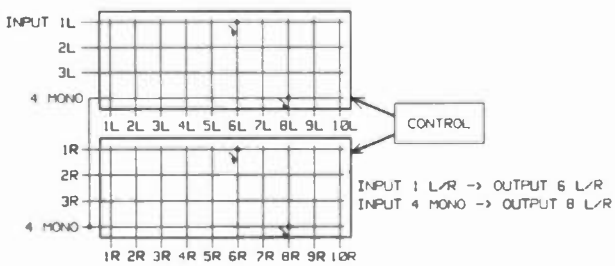

FIGURE 2. STANDARD STEREO DUAL MATRIX

A more efficient and flexible system is shown in Figure 3. Known as "Wild Audio" it is a combined configuration of the left and right matrices into a single matrix. With this matrix, mono inputs need only be connected once, and the controller is programmed to route the mono input to both left and right outputs when connection to a stereo output is desired.

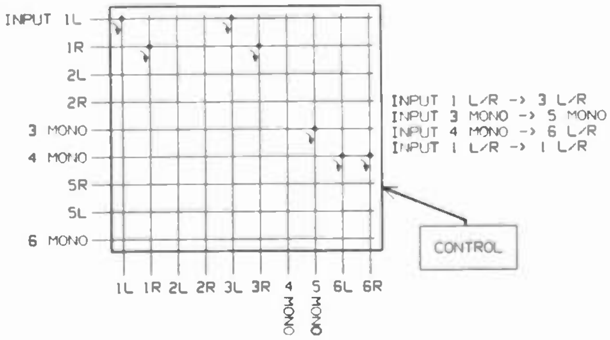

FIGURE 3. WILD AUDIO STEREO/MONO MATRIX

The technology of the actual crosspoint switch is very diverse. Many smaller switchers employ relay switches, and some of those are magnetically latching. The advantage of this approach is that if a power failure occurs, and the system is passive (without input or output amplifiers), the audio will still be routed to the proper outputs without interruption. Other active switchers employ various semiconductor switches in the crosspoint. A key feature to look for with active crosspoint switchers is non-volatile memory, assuring that, on power-up, the crosspoints will be in precisely the same configuration as they were when the power went off. In the past few years, Field Effect Transistor switches have become popular due to their high reliability and good audio performance. Some manufacturers have integrated the crosspoint circuitry into large scale integration or hybrid circuits to reduce size.

In many switchers, input amplifiers before the matrix isolate the inputs and provide gain adjustment. Maximum dynamic range is assured in a system where all inputs are adjusted to a standard level. Most modern switchers provide a balanced, RF protected, variable gain input. If the variable gain input can accommodate a wide enough range, most of the distribution amplifiers in the facility may be eliminated.

Following the matrix an output amplifier is required to prevent the loading of any output from changing the level of any other output being fed from the same input. Generally these output amplifiers also provide balanced outputs capable of driving the levels and loads common in radio today.

The most complex and potentially the most beneficial component in the switcher is the Master Controller. This module, as its name implies, is responsible for controlling all of the matrix switches, and thus the signal flow in the station. The benefits of the Master Controller come from the intelligence and remote control the unit provides.

The Remote Controls are the operator interface to the switcher. They allow the operator to select the inputs which will be routed to the outputs in his studio. There are many types of remote controls available, ranging from simple thumbwheel switch assemblies to intelligent alpha-numeric displays with plain English readout of the selected inputs. The selection of the remote controls is an important one, as operators generally have a difficult time understanding traditional patch bays, and the remote control becomes an ideal opportunity to select an alternative which significantly reduces the confusion and errors on air.

There are many methods of accomplishing the remote control connection to the master control. Some units require a coàxial cable or a multipair cable which carries a BCD logic signal, while others utilize existing twisted pair audio cable.

Most switchers offer redundant power supplies which automatically back each other up in the case of a failure. A good idea is to have these supplies fed from different phases of station power, assuring the switcher will remain operational if a phase is lost.

How do I use a Switcher in my station ?

A routing switcher is most effective when it is connected to all inputs which are used in more than one studio or by than more than one output. Typical inputs and outputs from a switcher are shown in Table 1-1. In this way, the switcher replaces all distribution amplifiers and recorder input selectors, most patch bays, and reduces considerably the need for large and complex consoles as well as minimizing the amount of station wiring needed. Not all patch bays are normally replaced, as the most critical circuits usually are backed up via patch bays in order to reduce the chance of a system failure.

TYPICAL SWITCHER INPUTS

NETWORKS
LOOPS
REMOTE PICKUP UNITS
TELEPHONE INTERFACES
CONSOLE OUTPUTS (BOTH PROGRAM AND AUDITION)
AIR MONITORS

TYPICAL SWITCHER OUTPUTS

CONSOLE INPUTS
REEL TO REEL AND CARTRIDGE RECORDERS
TELEPHONE FEEDS
PROGRAM DIRECTOR/ NEW DIRECTOR OFFICES
TECHNICAL SHOP
TRANSMITTERS (PRIMARY AND BACKUP)

TABLE 1-1.

As opposed to conventional patchbay and distribution amplifier wiring methods, the switcher is easily expandable. If a new satellite dish appears in the back yard, the switcher grows with the addition of input amplifier and matrix elements, instantly providing the new source to all outputs in the station.

Generally, the switcher is located in the main wire room, where all the cables are run from the various studios and where the telephone demarcation blocks are. This minimizes the wiring needed to add new inputs to the system. Methods of connection to switchers range from simple connector blocks to various connectors and even umbilical cables to punch blocks or "Christmas trees". It is advantageous to employ a connection scheme which facilitates quick additions of inputs and outputs while providing identification of the wires and terminals.

WHAT CAN AN INTELLIGENT SWITCHER DO FOR ME ?

Many of the new generation of switchers are intelligent, in that they are capable of doing more than simply making a crosspoint change in response to a BCD remote control. These features allow a tremendous advance in the control, flexibility and efficiency in today's radio stations.

One of the most powerful functions is the Salvo. A salvo is a list of instructions to the matrix which can be executed automatically, and nearly simultaneously. These instructions allow a reconfiguration of a studio, for example, from a music show to a sports program. These types of changes can be quite extensive, and a single errant patch cord can often ruin a program. With the salvo function, the reconfiguration can be memorized by the switcher, and may be invoked at any time.

Many switchers include a clock / calendar which, when used with pre-programmed salvos may cause changes to occur at certain times and / or days. For instance, if a particular program always occurs at a particular time on Saturday afternoon, the switcher can memorize the configuration and time and automatically reconfigure the studio at the proper time, as well as memorizing the configuration for the program which follows to allow a return to normal programming when the ball game is finished. Usually, many salvos may be stored, the total number depending on the amount of storage each requires. This preprogramming may be done at any time, reducing the need for the engineering department to be on hand during a particular program.

Salvos may also, in some systems allow automatic control of a function called "Machine Control". This function consists of a number of relay or opto-isolator contacts under control of the switcher. These may be used to start a recorder to record a particular program at a certain time, or cause any other action which is remotely controllable by a contact closure.

Conclusion

The Audio Routing Switcher represents a technology which can be used to significant advantage when new design and construction is planned. It often is more cost effective than the traditional patch-bay and distribution amplifier method, and allows considerable flexibility in both day to day operations as well as in the changes and growth normal in today's radio.

A STATE-OF-THE-ART MICROPROCESSOR CONTROLLED, ANALOG AUDIO ROUTING SWITCHER WITH ADVANCED FEATURES

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Vienna, Austria

ABSTRACT

The redesign of our routing system CROSSMATIC™ already announced at last year´s NAB-Convention in DALLAS, has been completed. The result is a modern, up-to-date product, whose volume density could be increased by a factor of about 2.2, while at the same time system prices could be cut down to some 75 %. This considerable improvement could be made possible by extensive use of SMD-technology in connection with a completely new control system. It is based on MOTOROLA´s 68k-line of processors, on VME-bus concept, on a powerful operating system package around pSOS and on using the high level language C.

Our second uP-controlled router generation contains even more than just all useful features of its elder brother, such as measuring/test bus at matrix inputs, monitoring bus at matrix outputs, integrated patch panels in two different versions and a reliable emergency and maintenance concept including a matrix selftest and power supply stand-bys with controlled change-over. Different types of system offer control possibilities from a rather simple PC-control onwards via multi-user versions with mixed terminal and keyboard operation up to highly sophisticated host computer systems for e.g. process computer or video control. A schedule system serves to minimize operator´s activities from all pre-scheduled, regular routing down to instant and unforeseen actions.

Although there are certain solutions available for one or the other intricacy concerning fully digitized switching procedures, for the next future we are still relying conservatively on the analog routing system. This, because the problems neither around a fully digital router, nor around the existing muddle of analog/digital studio environment have been solved to a degree, which already today would justify to scrap the well approved analog device.

1 INTRODUCTION

A description of our last generation audio routing system as well as an outlook on present and future developments based on experienced plus envisaged problems for switching digitally, has been the content of a preceding paper last year in DALLAS (1). There already, we have indicated the expedience of a complete redesign of the analog system as an intermediate and transition phase to the digital router and now, we would like te report on this next step. The three main points governing the new approach were

- modernizing all components by application of latest SMD-technology
- reduction of volume, power consumption and cost and
- selection of a powerful control concept capable of upgrading the system lateron for digital audio.

And all that while not just maintaining, but even increasing the useful features of its predecessor, which means outstanding technical data including short switching and reaction times, userfriendly manual as well as automatic operation and the possibility to configure hardware (HW), software (SW) and system functions, being then tailored to individual user requirements. A first prototype with, for the present, only reduced features, is a small 32:32 sample system, simply controlled via a SIEMENS personal computer PCD-2. It has been launched at last year´s TV-Symposium in Montreux/Switzerland and will be exhibited at the NEVE/SIEMENS stand during the forthcoming NAB-Convention.

The complete digital routing system, of which at the time being only the analog switching part is in the endphase of development, has purposely been devided into the three functional blocks

- switching parts (SP)
- control system (CS) and
- peripneral equipment (PE)

indicated in the blockdiagram of Fig. 1 and described (analog SP only) in the following sections.

Fig 1 Functional blocks of audio routing system

2 SWITCHING PARTS

The SP consist mainly of the HW for the different signal paths

- audio (e.g. mono, stereo, second program, feedback monitoring etc)
- intercom (e.g. 2/4 wire with or without conference features, command signals etc.)
- facilities (e.g. on-air and other signalization, remote control, time code and other data etc.)

which are normally handled by separate matrix levels and whose different types of modules (amplifiers, electronic or relay crosspoint cards etc.) are chosen to fit the signals´ parameters and to meet the requested specifications. Most interesting here is the audio chain given as an example in Fig. 2

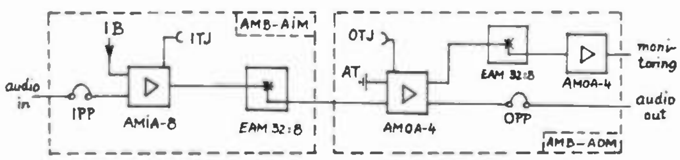


Fig 2 Audio chain

The three main modules here are

o an 8-fold audio matrix input amplifier AMIA-8,
o an electronic (crosspoint) audio matrix EAM 32:8 (in:out) and
o a 4-fold audio matrix output amplifier AMOA-4.

Input amplifiers and matrix modules are housed in 6 rack units (RU) high card frames with motherboards AMB-AIM at the rear, forming building blocks up to 32in:96out without output amplifiers. These are situated in similar 6RU-frames with motherboards AMB-AOM, yielding blocks for 32 output amplifiers. Advanced features of the audio chain are the following:

- integrated input (IPP) and output patch panels (OPP) for emergency by-passing
- integrated output monitoring via up to 8 buses
- SW-controlled audio termination (AT) at the input of each output amplifier by short-circuiting unused channels, thus yielding much better noise figures
- integrated, remote controlled insertion bus (IB) at the input of all input amplifiers for feeding e.g. a test or identification signal into the audio path
- input test jacks (ITJ) at the AMIA-8 and output jacks (OTJ) at the AMOA-4 for each channel on the frontplates, for easy signal tracing with the system being in operation
- BUSY-LEDs for all matrix outputs
- ERROR-LEDs signalling fuse defects and SELECT-LEDs for addressing control on all modules
- matrix inputs and outputs safe against even continuous ringing voltages up to 100 V/20 mA.

The system has excellent technical data (in conformance with the German broadcast standard IRT-3/2) and shows for example switch-over-times between two sources of less than 3 ms.

3 CONTROL SYSTEM

3.1 Hardware

We have chosen MOTOROLA´s 68k-family of processors in connection with VMEbus technology, because we believe that for our control applications, today this is the most capable approach. The basic concept of the control system linking the peripheral equipment (PE) on the left to the switching parts (SP) on the right side is shown in Fig. 3. The intelligent communication controller MVME 333

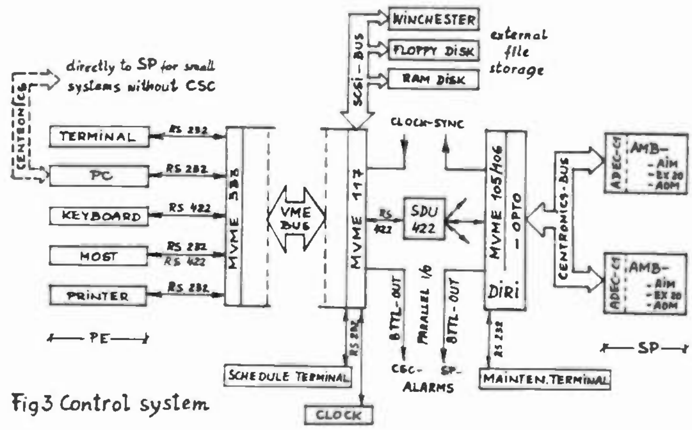


Fig3 Control system

operates as pre-processor for the PE, the monoboard uC-module MVME 117 serves as central processor within the Central System Control (CSC) and the single board computer MVME 105 (or 106 with floppy controller) works as the so-called Digital Intelligent Rack Interface (DIRI), one of which is in each matrix rack. And finally, this DIRI controls the SP (consisting of the amplifier and matrix frames with one of the mother boards AMB-AIM/EX20/AOM) via separated opto-couplers and bus-drivers, a parallel CENTRONICS-bus and an address decoder module ADEC-C1, one per matrix frame. Fig. 3 also indicates the type of data connections between the peripheral equipment and the pre-processor (MVME 333), which turned out to be necessary because of the stringent requirements for the reaction speed in multi-user systems. The CSC, being that functional block of the control system, which at any time has the total information about the system configuration, further incorporates:

- a real-time clock, synchronized by the station´s master clock (for documentation purposes and for fully automatic, time-driven operation following a pre-programmed schedule)
- a separate terminal port for input/output of schedule data (see section 6.2)
- up to 8 TTL-output lines for alarmes and
- external file storage media controlled via an SCSI-bus e.g. a floppy disk containing the boot program and pre-programmed schedule data, and a hard disk with code and configuration files representing a complete replica of the total system including the DIRI´s, keyboard processors and if necessary, a command/-action/error history for later use.

3.2 Software

The HW is supported by pSOS as a sophisticated, real-time operating system kernel, which allows an application to be logically and efficiently organized by multitasking. That means, that an application can be constructed as a group of processes, interrupt handlers and device drivers. And there is a set of powerful system utilities: pRISM as a real-time multiprocessor communications manager, pHILE as a file manager component for under pSOS runnng systems and pROBE as a comprehensive system debugger and analyzer for the whole components family. In general, programs are written in the high level language C, only with time-critical routines in assembler in such a way, that they can be called by C-programs. As one example for the modularity and block structure of the SW, the essential relation between the CSC and the DIRI (controlling a single matrix rack) shall be outlined by the functional diagram of Fig. 4.

two sides. Normally, commands and responses of the DIRI´s are ASCII-strings, which enable the use of standard terminals for test purposes. Correct strings are acknowledged by "ACK", not correct strings by "NAK" and are not further processed. In case of DIRI-responses not being acknowledged within a certain timeout, the string is repeated 3 times at maximum. Repetition and receipt-acknowledgment can be switched off. Standard functions are the following:
- matrix control commands plus conversion from logical coordinates (source, destination and matrix level) into physical ones (number of rack, MB, card, input and output)
- administration of facility levels, tables, authorizations and name lists
- clock functions (i.e. display of clock-time as support of instant manual switching, release of time-driven switching activities within the schedule system and documentation of switching times e.g. for error-protocols etc.

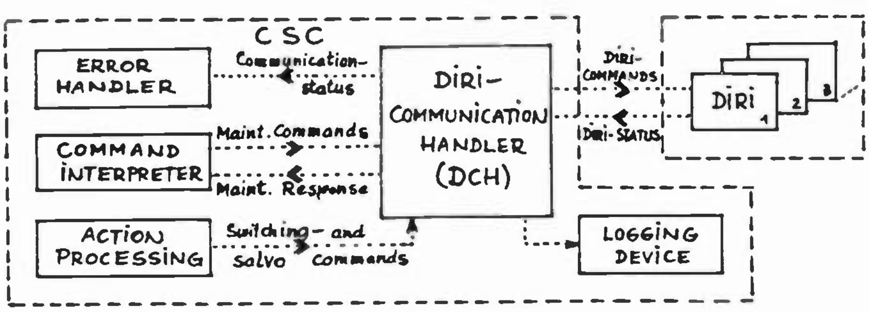


Fig 4 Functions of the DCH

The central SW-segment here is the socalled DIRI-Communication-Handler (DCH), which basically handles the following functions:
- communication between CSC and DIRI via an RS422-IF (plus a serial distribution unit SDU 422, as indicated in Fig. 3, in case of more than one matrix rack)
- receipt of switching, salvo and maintenance commands
- assembly and delivery of commands to the DIRI
- analysis of the DIRI-status and
- assembly and delivery of maintenance-response and communication-status.

Within the CSC, the vis-a-vis for the DCH are the Error-Handler, the Command-Interpreter and the Action-Processing and the communication between pSOS-processes is done via so-called Message Exchanges.

- protocolling (documentation of system events through a printer e.g. error and alarm messages, test results and status reports
- status query of the cards and test of the audio paths (e.g. via the insertion bus at a certain source through the matrix to a certain destination via output-monitoring)
- switching of groups (i.e. connecting and disconnecting of any correlated crosspoints e.g stereo, several matrix levels, preparations and salvos to be executed by a single stroke of a push-button)

Examples for special functions are:
- modul, memory and communication test
- watchdog and supervision of powersupply
- action/command/error history (for documentation and reconstruction of operative actions as help in case of trouble-shooting)

3.3 Functions of the CSC

The Central System Control CSC is configured in such a way, that on the operator´s side, terminals with appropriate dialog, keyboards and host-computers can be controlled and on the matrix side existing analog as well as future digital matrices. Hereby, the CSC checks the semantic of the data coming from either of the

4 PERIPHERAL EQUIPMENT

The various possibilities for the operational periphery are shown on the left hand side of Fig. 3. Two of them shall be discussed more detailed in the following. One is the conventional CRT-terminal (black white/color/graphics) which we still tend to primarily recommend as matrix control considering all the advantages of a CRT with specific user dialog plus all feedback information such as matrix status display and error message. The other one is a keyboard version particularly designed for fastest possible operation of small to middle sized routers or destination oriented parts of it. In addition to those two, the use of a personal computer PC is a very elegant control method for small systems and operationally is very similar to a terminal. And finally control from a host computer (e.g. process computer or in an "audio-follows-video-"control from the video side) depends on the type of the host system, the interface and the transmission procedure, with which we have quite an experience, but must be handled individually according to the peculiarities of the controlling system.

4.1 Terminal

Terminal operation discriminates on one hand between
- manual instant control
- crosspoint schedule control
- menu driven control with a "help"-function as online-tutorial and
- communication (electronic mail/message),

and on the other hand between input of
- primary commands (e.g. connect, disconnect, monitoring etc) and
- socondary commands (e.g. access through password login, clear matrix, set clock or printer control)

via function-, cursor- and softkeys. The display on the CRT-screen is an important part of the man/machine-interface, because its capabilities contribute to ease and quicken the in/out-operation and the continuous information of the operator about his actions and the system status at any time. There are several optimized formats available with special symbols and masks (e.g. for instant operation, manual or automatic monitoring, definition of salvos or the crosspoint control schedule) in connection with color and attention functions (high intensity, blinking with normal intensity etc) for the different situations of system operation including a special concept for maintenance and diagnosis. The operational capabilities of the CRT-terminal may be further expanded by additionally using a "mouse", with its possibilities e.g. to select commands, move record lines or skip to the help-symbol by positioning the mouse-cursor and clicking.

4.2 Keyboard

In single and multi-user systems, keyboards should primarily be used for rather small matrices, destination oriented segments of larger ones or where speedy operation is imparative. Keyboards are connected to either a OIRI (MVME 105) or a central pre-processor (MVME 333) via a peripheral controller PERICO and a serial RS 422. A standard version has purposely been devided into the two parts Basic (BKB) and Extension Keyboard (EKB) shown in Fig. 5. The BKB on the left shows (from top to bottom) 8 push buttons for 8 destinations or integrated monitoring buses together with the corresponding LCDs, a 2x40 digits LCD-strip as input- and information display, a 40-fold alpha-numeric key pad for mnemonic and/or numeric 'source select'

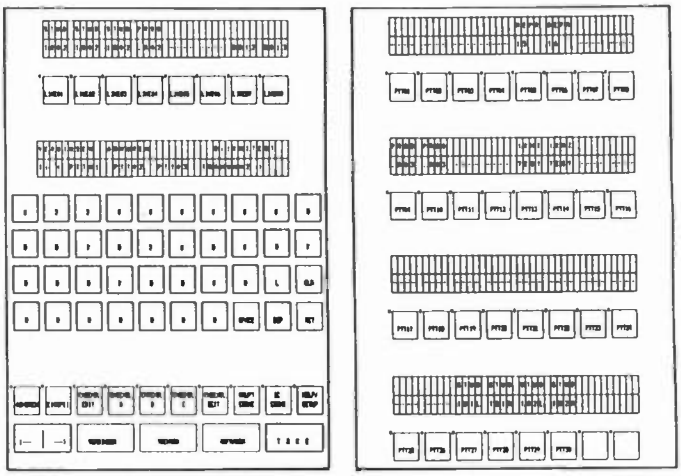


Fig5 Basic (left) and Extension Keyboard (right)

and two rows of function buttons with 1 LED each, 6 for fixed functions (last row) and 10 selectable ones through softkeys. Among those are e.g.: matrix facility levels married or splitted, several salvos, monitoring, time, insertion signal and help/set up-function. The EKB on the right side of Fig. 5 shows 4 rows of 8 destination keys plus 1 LED each, together with LCDs (8 digits each) for displaying the source switched to the selected output. Up to 8 of those Extension Keyboards can be connected to one BKB, thus enabling the control of a matrix with an unlimited number of sources, 8x32=256 destinations and up to 8 monitoring buses.

5 BASIC SYSTEM TYPES

In spite of the useful capability to configure a certain system due to the modularity of the HW and SW in connection with certain operative functions (see section 6.1), some kind of a standardization has been done by defining a number of basic system types having different sizes and operational possibilities. This leads to reduced prices and delivery times in comparison to a specifically tailored, customized version.

Type 1:
small systems up to a size of 96:96 (one level only) and control through an IBM/AT-compatible PC via its CENTRONICS interface as shown in Fig. 6 and represented by the DEMO-unit at the exhibition. Manual and scheduled control are standard with color-screen and printer being options.

Type 2:
matrices as for type 1 controlled through a PC or a CRT-terminal plus up to two keyboards by providing a rack interface module DIRI and control via serial RS 232 and RS 422 (Fig. 7). PC plus keyboard can also be substituted by a single host-computer having the same transmission procedure as the PC.

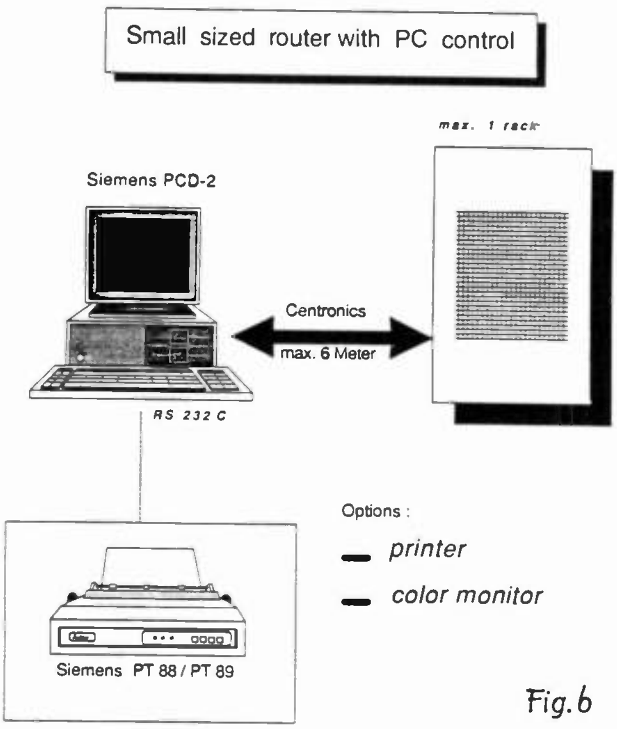
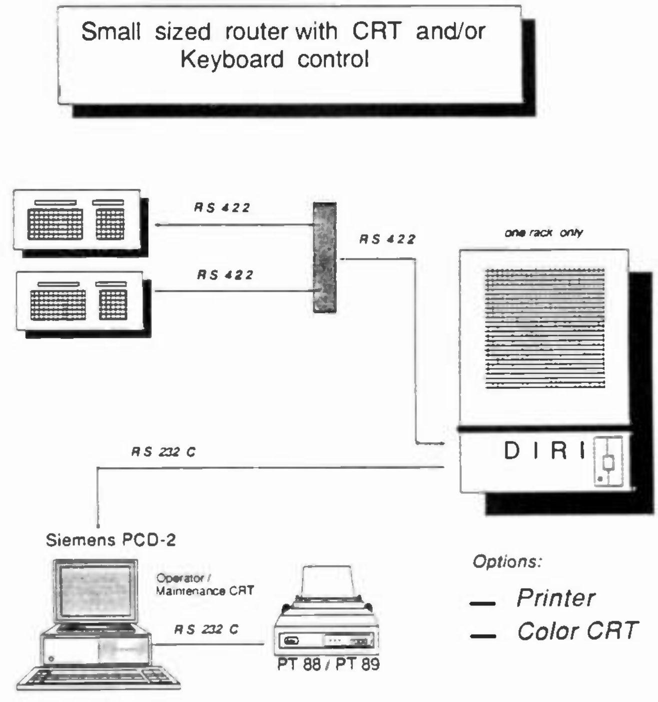


Fig.6



Floppy drive (optional) for loading not DIRI specific software, e.g. Keyboardinterface, etc.

Fig.7

Type 3:
middle sized, multi-user system indicated in Fig. 8, which already allows a matrix size (plus different signal levels) of up to 8 racks controlled by a CRT-terminal and up to 8 keyboards. This quite extensive system now requests a CSC consisting of a central processor with a preprocessor connected via the VMEbus. As for type 2, this periphery can also be replaced by a host-computer. The use of a "mouse" is here an additional option.

Type 4:
this big sized routing system is an expansion of type 3 to a mixed (terminal + keyboard + host) multi-user system on the operational side and

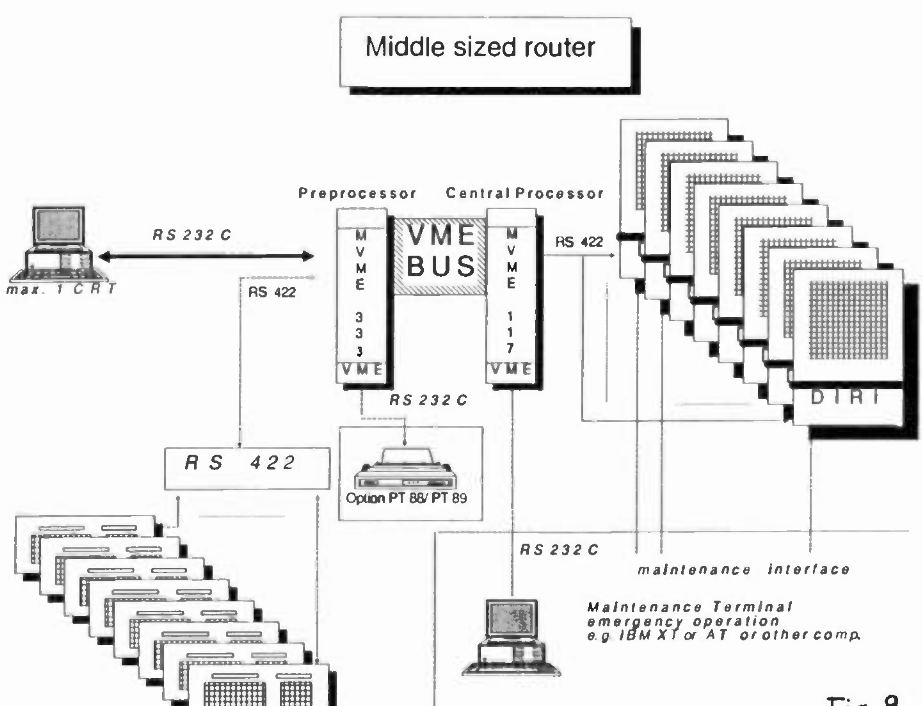


Fig.8

enables the control of a matrix plus facility levels of up to 16 racks on the switching side of the router. It is our most sophisticated type of system, which has already been discussed and shown as Fig. 6 in the prospective section on page 145 of(1).

Type5:
finally is designed for a very special host-control primarily through a VIDEO-system in an "audio-follows-video"-application. There, the host directly controls the individual rack interfaces DIRI via a Serial Distribution Unit SDU 422 and a certain basic transmission protocol without the need of a central system control CSC.

6 ADVANCED FEATURES

Most of the points of the CROSSMATIC D system have already been mentioned, but just by short remarks in the context. Some of the meaningful features shall now be discussed a bit more in detail.

6.1 Configuration run

The modularity of all HW-components allow the implementation of systems having different HW-related characteristics e.g. type of crosspoints and amplifiers, size of matrix plus different levels, availability and number of additional signal paths for (manual or) automatic by-passing in case of a matrix failure, and of monitoring facilities and patch panels etc on one hand. On the other hand number and type of peripheral equipment, both of it working together via the HW-components of the control system, i.e. number and type of distribution units, interfaces and controllers, of (pre/central-) processors and memories plus special modules such as a clock. This to be completed by the necessary switched mode power supply units with partial/full redundancy and automatic switching over to the stand-by block in case of an equipment/power failure. But what´s with all inherent system capabilities due to a block-structured software modularity? Now given a certain HW-structure, if we understand in this context configuration to be the compendium of selected system functions, we can perform and define a configuration run as being the assembling of the system specific parts of the SW. And there are a lot of system functions, which have to be and can be fixed in the course of an initial or renewal configuration run, just a few examples of which shall be listed in the following.

Functions of the PE:
- establishment of the physical interfaces for the communication with the operating elements
- assignment of PE-functions to the operational facilities (e.g. push-buttons, LEDs on keyboards etc.)
- choice of display formats e.g. selection of start-mask at the CRT-screen (for instant manual as well as for schedule operation)

Functions of the SP:
- interactive adjustment of system tables
- establishment of alpha-numeric name and prohibition lists

Functions of the CSC:
- rankordering of failures and errors
- choice of defaults valued as being alarms and their assignment to the 8 available TTL-outputs
- choice of alarm-processing (e.g. output via a printer, via a separate serial or parallel IF, transmission to an indicating element or to a controlling host etc.)
- time-rate of synchronizing the system´s real time clock by the station´s master clock and choice of positive or negative flank of sync-pulse.

These possibilities to functionally configure the system yields an unparalleled flexibility without the need of later, customized SW-development.

6.2 Schedule operation

A very useful feature to drastically reduce the amount of switching manually daily recurring events is a schedule operation, i.e. a fully automatic, real-time driven operation following pre-programmed and stored routing information. This can be considered to be an extension to the instant manual operation, which, overriding the schedule operation, is always possible. The routing information is loaded onto a floppy disk as preparatory storage medium via a standard terminal in socalled "records", which contain the whole switching information for one event, i.e. one source, up to 6 destinations, time of connection and disconnection, type of switching e.g. salvos, mono or stereo, cycle information and so on. They have a special format on the screen with various display and editing functions (positioning of window, scrolling of records, display of preceding and next day etc). The records of one day are downloaded into a non-volatile memory, can be checked, changed and completed as long as the corresponding paths are not yet connected and a one week schedule usually covers 1+7+1=9 days.

6.3 Selftest

Testing all critical parts of the system as preventive and diagnostic measure is of primary importance to ensure a high degree of availability. In addition to certain kind of tests for memories (RAMs, EPROMs), display elements (LEDs, LCDs) and communication paths (PE↔CS↔SP), the rack interface DIRI performs a socalled "Selftest" of the crosspoints in form of a cyclic background routine during normal system operation. By using a separate test-IC and feeding-in a test signal for checking the unused crosspoints, the actual audio paths and not e.g. parallel ones are tested. In case of a crosspoint failure, an error message is issued in full text (type of failure and number of rack, card frame and crosspoint card), which enables its quick localization and removal.

6.4 Maintenance/emergency strategy

Regular maintenance shall help to increase the availability of the system and a particular "maintenance-handler-shell" of commands has been provided, via which the operator can communicate with the system using a separate diagnostic terminal connected to one of the system interfaces at the central processor or at the

DIRIs. And beyond maintenance, there must be a far-reaching emergency strategy comprising the sum of all measures to minimize the system´s down-time by e.g. a quick failure identification, by-passing and putting the system back into full, partial or at least some emergency operation.

For the topics maintenance and emergency a prototype concept called ARTEX (Automated Routing Test Expert System) has been developed, which applies the methods of the artificial intelligence i.e. regulation-based systems and object-oriented programming. Practically it means, that starting from a certain symptom one tries to identify the cause of failure and the best counter-measure, thus arriving at a knowledge-based test and maintenance support for the routing system.

7 CONCLUSION

It has been already indicated in (1), that in parallel we are working at a digital router, but that a couple of unsolved problems still prevent us from producing a fully digital device with technical data and prices at least equating those of a good analog system. No question about having made some progress in the digital area too, but basically, our statement above is still valid. As an intermediate step therefore, we completely redesigned our existing CROSSMATIC system and now we think too that (quotation): "Analog has a lot more life left in it, than people might realize" and that the result is worthwhile to be discussed.

REFERENCE

URBANEK, G.E. "New Generation Audio Routing Switcher Performs Many Functions", 1987 NAB Engr. Conf. Proceedings, pp. 140-146

DESIGNING AND MODELING HIGH POWER FM BOOSTERS

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ABSTRACT

The FCC has recently deregulated the use of FM boosters to allow a power of up to twenty percent of the licensed ERP. This new restructuring of the booster rules has the potential of creating a two edged sword for the broadcaster who may wind up causing as much interference to himself as he eliminates with the booster.

This paper will go through some of the steps Shamrock Broadcasting is going through to ensure that a high power booster we are proposing for our station KABL to cover Pleasanton, California both meets FCC guidelines and provides the best coverage possible.

INTRODUCTION

The new potential of increased power coupled with the ability to carry audio to the booster by most any appropriate means are the two modifications of the booster rules that will likely carry the most impact on booster implementation.Extreme care must be used, however in implementation.

In the past, since high isolation between the booster receive antenna and transmit antenna was a requisite, booster sites were only possible in a handfull of places within the service area.Main to booster interference was generally not a problem since if one could achieve the necessary isolation, it was unlikely that the main transmitter signal would substantially exceed the booster signal over most of the service area. Now this has been changed, and the broadcaster can locate a booster virtually anywhere within his protected contour, so long as he can convince the FCC that he is not extending that contour.

CHOOSING A SITE

In choosing a site for the KABL booster, the very terrain that is blocking the main signal will serve to trap the booster signal within the booster service area. Since usually in mountainous terrain, most population is centered near the lowest point in the valley, this is where the highest booster-to-main signal would be desired. Since there will always be some interference, hopefully, by using the natural terrain features, interference will occur primarily at the top ridge of the blocking terrain.

When locating a site, one should consider which areas are necessary to be best covered.

It may help to list areas in order of importance, realizing that due to inherent limitations some areas may need to suffer somewhat at the expense of others. Also, don,t forget it must be proved to the FCC that your main existing contour will not be extended.

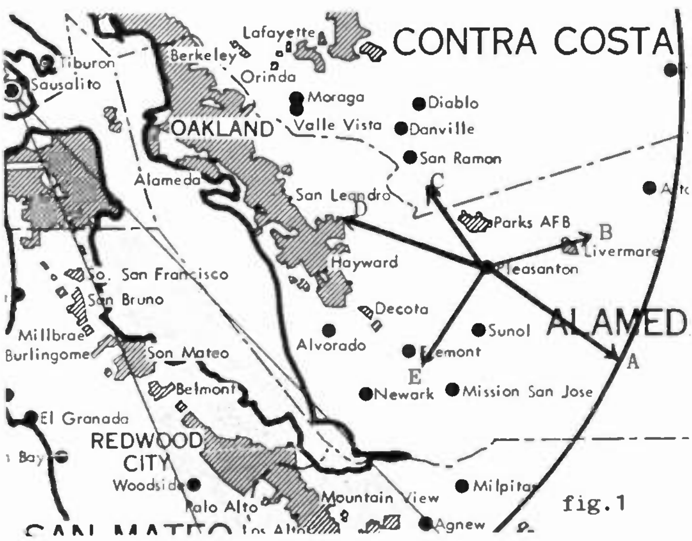


fig.1

Figure 1 shows the considerations taken by Shamrock in locating a booster in Pleasanton, California across the Berkely Hills from Oakland, and San Francisco. As can be seen, several different problems must be overcome:

A- Cannot extend protected contour

B- Maximize signal over a major growing area

C- Maximize signal in heavily populated valley

D- Protect main signal from interference

E- Minimize signal escaping through back of valley and potentially interfering with main

Achieving the results in A, B and C above are relatively easy to obtain from a standard antenna. The deep nulls in D and E are a bit more difficult. The change from null to full power over an arc of 10 degrees is impossible.

Through work and cooperation with Shively Labs, it was determined that the best antenna compromise was obtained by utilizing a dual, stacked, circularly polarized, Yagi array.The pattern is shown in Figure 2. The pattern looks similar to a standard cardioid, but the nulls to the rear are deeper and allow better protection and control to prevent any unnecessary interference to the main signal.

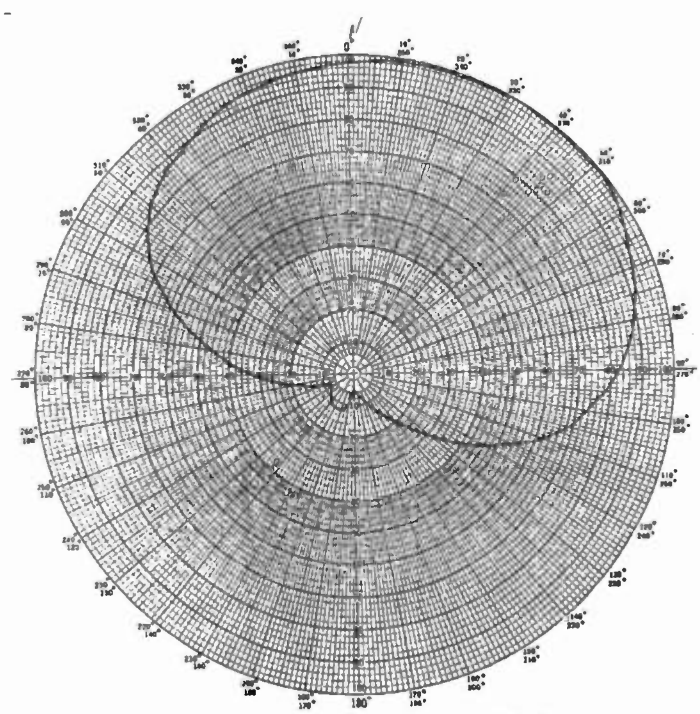


FIGURE 2

Shively Labs

Because the pattern obtained couldn't match the theoretical conditions, a computer study is indicated to model the probable coverage and interference potential of the chosen parameters.

DETERMINING COVERAGE BY COMPUTER MODELING

Obviously, in any booster application, terrain is a very important consideration in determining coverage and interference. Clearly, the FCC 50/50 curve does little to take terrain into consideration when predicting coverage. In our application at Shamrock, the US Commerce Departments' NTIA TAService Coverage and CSPM plots were run to come up with an accurate estimate of what could be expected.

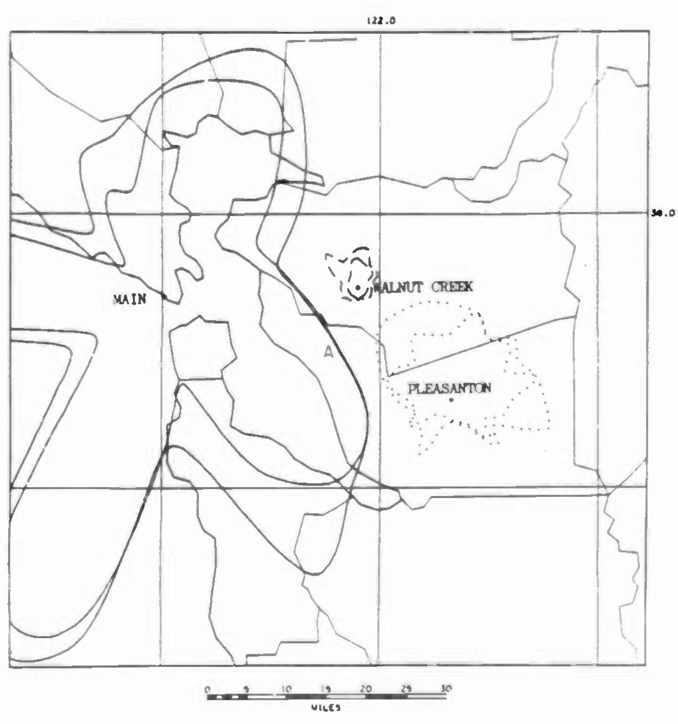
NTIA provides several forms of their database.Due to space limitations of this paper, only the programs applicable to this study will be discussed. Figure 3,4 and 5 show three representations of KABL's coverage in the San Francisco area. There is a substantial difference in the three representations.In order to better understand exactly what the coverage is likely to be, one must understand the basis of the 3 plots.

In figure 3, the FCC 50/50 curve takes very little terrain information into consideration. Signal deficiencies are virtually undetectable.

In figure 4 the NTIA Coverage program was run in the point-to-point mode. In this program, radials are plotted every 15 degrees around the transmit point and a calculation measured every one kilometer is based on the instantaneous height of that point along with any attenuation from previous points. In figure 4, approximately 4,000 points were used to depict coverage. Although data is recorded in chart form for quite some distance (user defined), when plotted in the minimum mode, (inner boundary) the first time that the signal drops below a designated value (in this case 60dbu) the contour is marked and a smooth line is drawn between the sampled radials. In the maximum mode, (outer boundary) the contour indicates the last time the signal remains above the designated value.

This information is helpful in determining in coarse terms, whether or not there are likely to be any widespread signal overlap or other problems in the initial design of the booster. This plot is considerably less expensive to run, costing about 25% of the more detailed CSPM program. The data obtained in chart form can also be used by manually plotting radials and comparing two sets of data to determine where the two signals approach equality and thus interference. In preparing for the CSPM plot, parameters can be modified slightly to compensate for this.

The example below shows the Minimum Coverage plot run for both the high power booster for Pleasanton and the standard booster in Walnut Creek. Although no overlap is shown in the coverage plot, CSPM will likely show some overlap. This plot, hovever, shows that our basic location and antenna design should work.



Since samples taken on Coverage are only every 15 degrees, and 1km apart, much terrain can be lost and precise locations of interference are not really possible. The CSPM program will pinpoint precise problem areas. CSPM plots the calculated signal about every 320 meters and in both a north-south and east-west direction. In th example shown in Figure 5, over 32,000 induvidual points were calculated to acheve the information for just one transmit point. This is repeated for each booster separately.

The user can then ask NTIA to calculate and produce on an overlay, where interference is likely to occur by sampling where 2 plots of the same area are close to each other. This service will be available in the very near future.By examining the data provided, the booster potential can be precisely determined and steps taken at the design stage to eliminate as much interference as possible.

SYNCHRONIZING CARRIERS TO MINIMIZE INTERFERENCE

Since a certain amount of interference is unavoidable, one solution to minimize the problem is to borrow an emerging technology from the AM dial. Synchronous Transmission. The thought behind both methods is essentially the same. There are two useable ways to attempt to sync transmitters for this use. One is to obtain receivers tuned to WWV and control the main transmitter reference oscillator as well as the various booster oscillators to WWV. At this point, we could then time the two transmitters and depend on the accuracy of WWV to maintain the tolerance of 2 hertz that would be necessary for our needs.The main problem with this is the required modification of the main transmitter exciter (which we are not comfortable with and probably wouldn't be accepted well by the FCC).

A second method of reaching this goal would be to provide a subcarrier from the main transmitter that would provide the information needed to lock the boosters' reference oscillator to the original signal. We are looking at a device called a "trigger coherant oscillator" for this purpose. This device will serve as the time base for the high power booster transmitter that we intend to use The oscillator will take an input pulse at any frequency and output the necessary frequency to drive the exciter. In this case we will obtain a 10 MHZ sample from the main exciter, then divide it down to become the actual subcarrier frequency for the main channel or a tone that could modulate the subcarrier. At the microwave relay point, the main subcarrier will be relayed to the microwave STL as a similar subcarrier transposed in frequency.

To finally sync the transmitters, we will use two yagi antennas on a ridge visible to both sites. One antenna will "look" at the main transmitter and the other at the booster. Waveforms can then be matched by a high speed dual trace oscilloscope.

CONCLUSION

With the new booster potential, the broadcaster has several new and exciting challenges open to him. Now, for the first time, stations in extremely mountainous terrain can realize coverage potential more in keeping with their flatland counterparts. As shown in this paper, care must be taken, however, not to move too quickly or without adequate preparation.

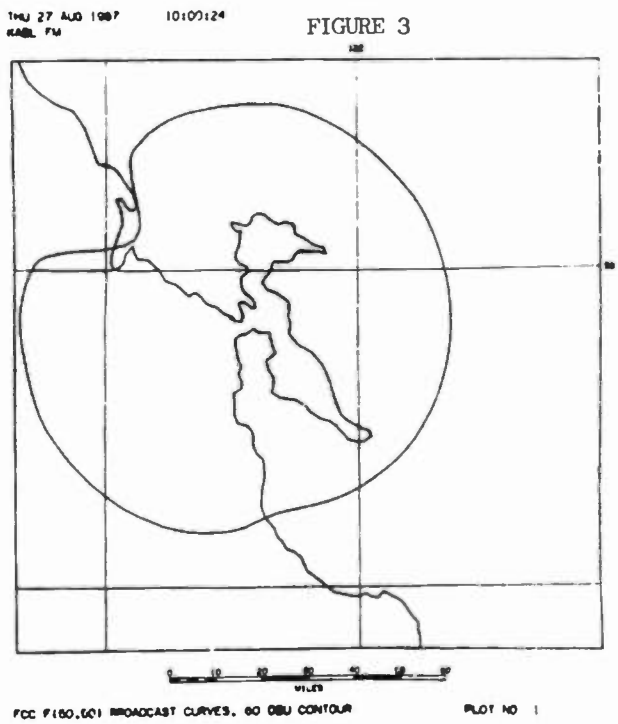


FIGURE 3

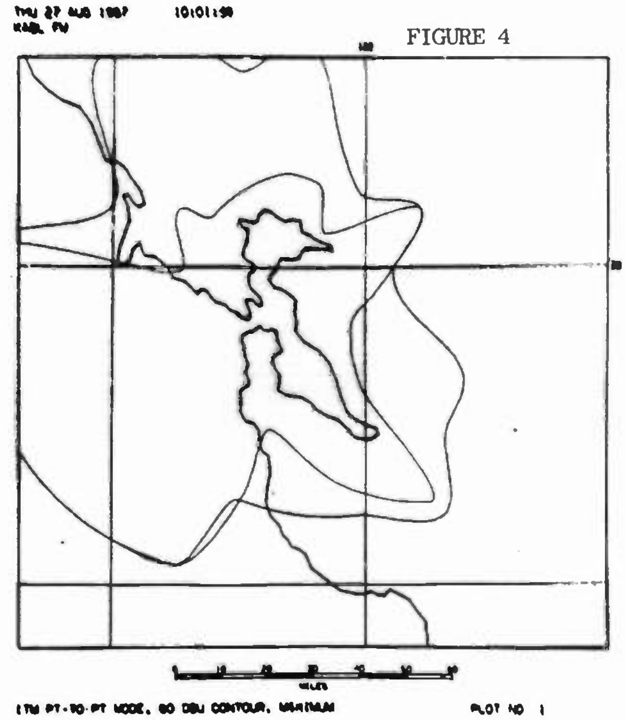


FIGURE 4

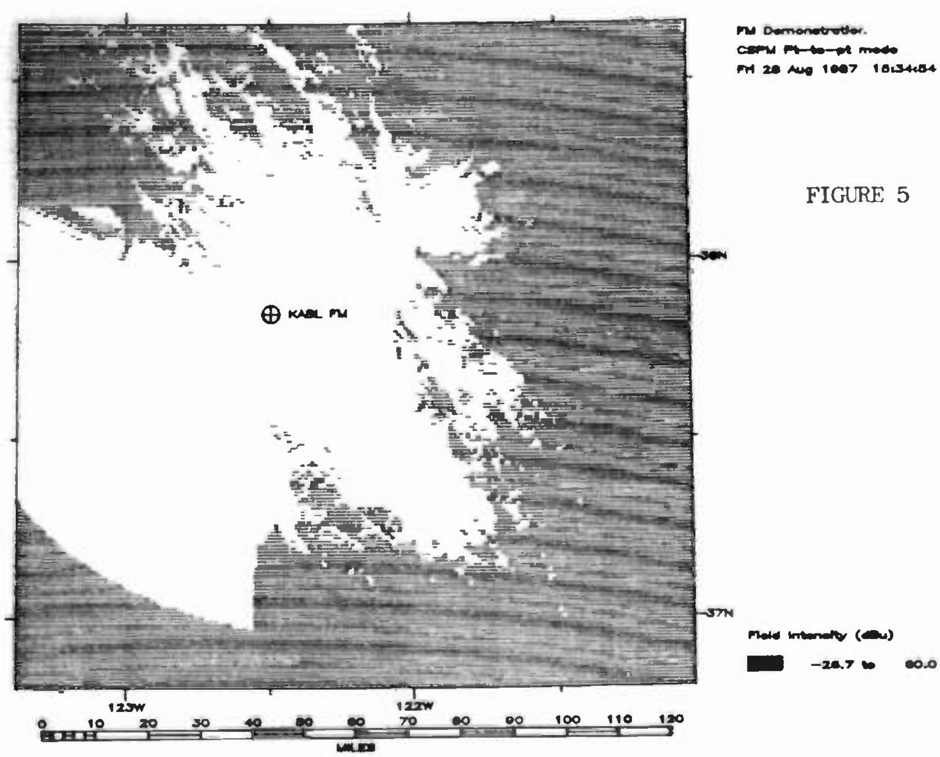


FIGURE 5

GROUNDED-GUY ANTENNA REDUCES STATIC ARCING AND IMPROVES BANDWIDTH

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The grounded-guy antenna is a class of non-directional vertical antenna which has no insulators in its top-level guy wires (Figure 1). The hot guys carry a substantial amount of RF current, and provide a DC path from the tower to ground. This configuration tends to produce an umbrella of protection from induced electrical storm voltages on the remaining guy insulators.

The tower is insulated and series-driven at its base, similar to a conventional tower. However, the impedance characteristic of the grounded-guy tower is unusual (Figure 2). The impedance vs height curve is compressed compared to that of a non-guyed tower of the same dimensions (Figure 3). Parallel resonance occurs near 65 degrees of grounded-guy tower height, which is less than half of the height one would expect in a conventionally guyed tower. This results from the longer electrical length of the RF path on the grounded-guy tower.

If the natural parallel-resonance cannot be avoided, it can be shifted by inserting coils at the bases of the hot guy wires. Consider the 60 degree tall grounded-guy tower: by inserting 200 ohms of inductive reactance at the base of each hot guy, one can drop the tower base impedance from 730 + j1700 to 102 - j19 ohms. The latter value is much easier to deal with. There is no significant effect on radiated field (Table 1).

TABLE 1

GUY BASE CURRENT	GUY BASE COIL	TOWER BASE Z	FIELD AT 1 MI, 1 KW
4.2 amps	+j0 ohms	730 + j1700	189
3.7	50	1850 – j 770	189
3.2	100	380 – j 440	189
2.9	150	170 – j 160	188
2.6	200	102 – j 19	188
2.4	250	73 + j 58	188

The impedance bandwidth of the grounded-guy tower is best for tower heights ranging between 70 and 120 degrees. Bandwidth is always best where the slope of the resistance and reactance curves are least, and where the resistance is greater than the reactance; in this case, near 90 degrees of grounded-guy tower height. In many cases, the channel bandwidth of the grounded-guy tower is better than that of a conventional tower (Table 2).

The hot guys of the grounded-guy tower do provide some top-loading, but this configuration should not be confused with a simple toploaded tower. The hot guys of a normally top-loaded tower are not grounded. In fact, if the guys are hot all the way to the ground, but insulated from ground, the antenna aperture will be closed, radiation resistance will vanish, and little field will be radiated. For this reason, simple toploaded towers rarely have more than two-thirds of the tower shadowed by the hot guy wires. But since the current distribution is different on the grounded-guy tower, this problem of reduced aperture is avoided.

The field from a grounded-guy tower is essentially the same as that of a normally guyed tower of equal height. For heights above 150 degrees or so, circularity begins to degrade as the guys and the tower appear as separate radiators (Figure 4). So long as the hot guys are symmetrical, no significant horizontally-polarized component of radiation will exist for tower heights less than 150 degrees.

Can the grounded-guy tower be used in a directional array? Yes, but only if careful placement of the hot guys and consideration of guy-wire coupling are appreciated. This is best done with general moment-method analysis, and does not lend itself to quick and easy pattern generation.

TABLE 2

			GUYED TOWER			NON-GUYED TOWER		
G	KHZ	GUY COIL	BASE IMPEDANCE	RESONATED	VSWR	BASE IMPEDANCE	RESONATED	VSWR
60°	1000	+j200	102 – j19 ohms	102 + j0	1.00	12.7 – j139	12.7 + j0	1.00
	1010	+j202	101 – j3	101 + j16	1.17	13.0 – j.35	13.0 + j5	1.47
90°	1500	0	448 – j197	448 + j0	1.00	41.1 + j18.5	41.1 + j0	1.00
	1510	0	452 – j195	452 + j4	1.01	42.1 + j21.5	42.1 + j3	1.08

Consider two conventional 60 degree tall towers spaced 60 degrees, and phased 120 degrees, producing the pattern of Figure 5. Compare this to Figure 6, where the hot guys pass quite close to the opposite tower and its guys. There is also a large difference in base driving-point impedances. As mentioned, the area of parallel-resonance is to be avoided when selecting a grounded-guy tower.

RF current stress on the hot guys needs to be considered at higher power levels. The guy current can be higher than the tower base current in some instances. Near parallel-resonance, for example, a tower base input current of one amp may yield four amps at the guy base. Note that Figures 7 through 10 show the sum of the three guy currents. The individual guy currents divide evenly, and are one third of the value shown on the graph.

Keep in mind that the field from the guys tends to oppose that from the tower, because in-phase guy currents are directed down while the tower current is directed up from its feed point. If a phase reversal occurs because of loading, or simply because of a long electrical length, this is denoted by (+) and (-) labels on the graph.

Figure 7 shows a lot of area underneath the current distribution curve, but since the guy field opposes that of the tower, the resulting field is the same as that of a 60 degree tall monopole. The near fields are somewhat different, of course, but the far fields are essentially the same for the lossless case.

If we load the base of each guy wire with an inductor to obtain more convenient base impedance, a phase reversal occurs on the guy (Figure 8). Again, the field is the same as that of a normal 60 degree tall tower.

A 90 degree tall grounded-guy tower without loading also has a phase reversal on its guys, and is beginning to show a phase reversal near the tower base. A 120 degree tall grounded guy tower shows this tower base current phase reversal more clearly.

In all cases (Figures 7 through 10) the far field is not significantly different from that of a conventional monopole, even though the current distribution is different.

All analysis was conducted with a moment-method model over perfect earth. The efficiency of the grounded-guy tower is not radically different from that of a normally guyed tower. However, care must be exercised in bonding the hot guy bases to the ground system. One approach to this detail is to use a separate set of radials around each guy base, bonding these to the tower ground radials in the same manner one would treat a multi-tower array.

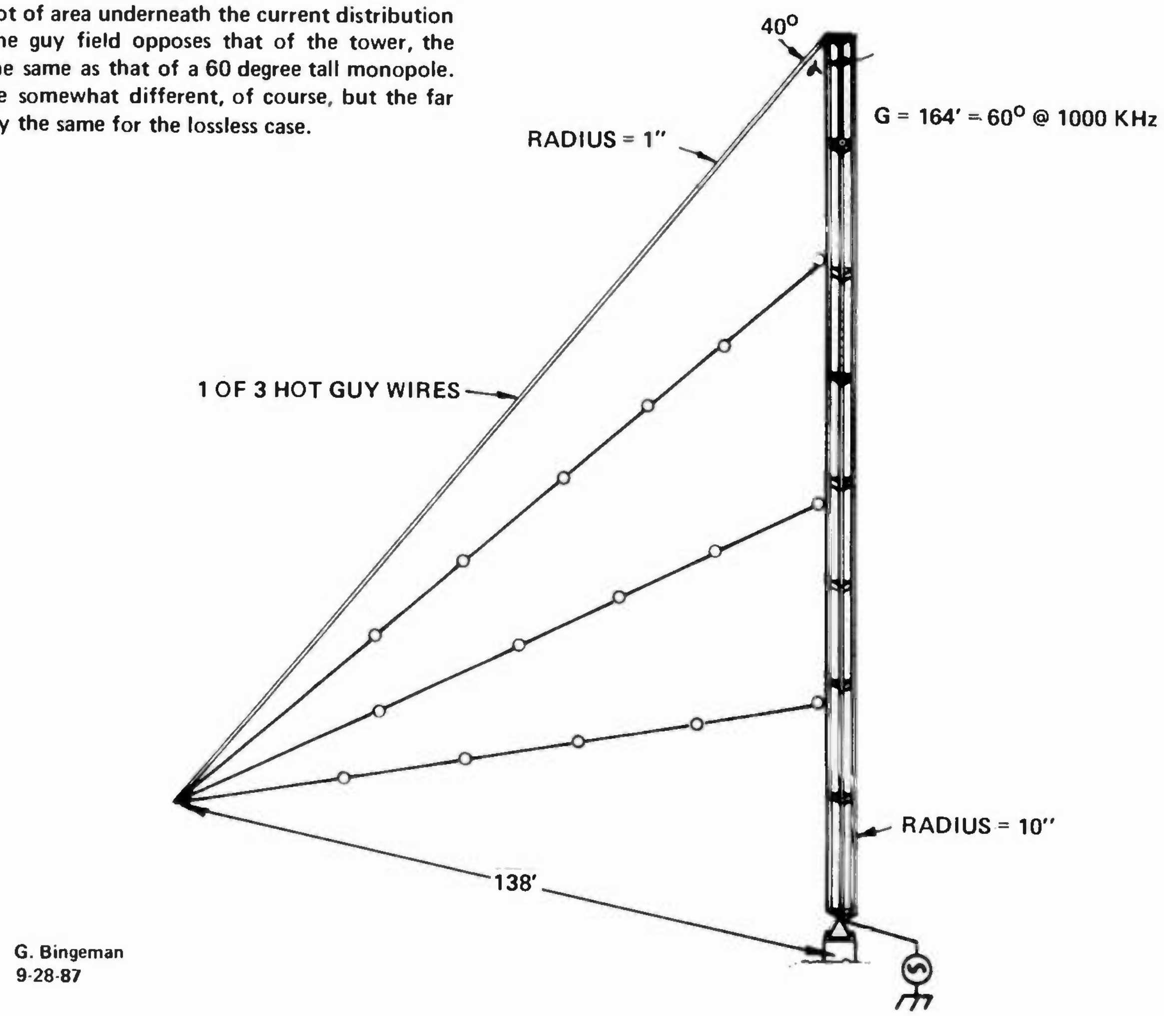
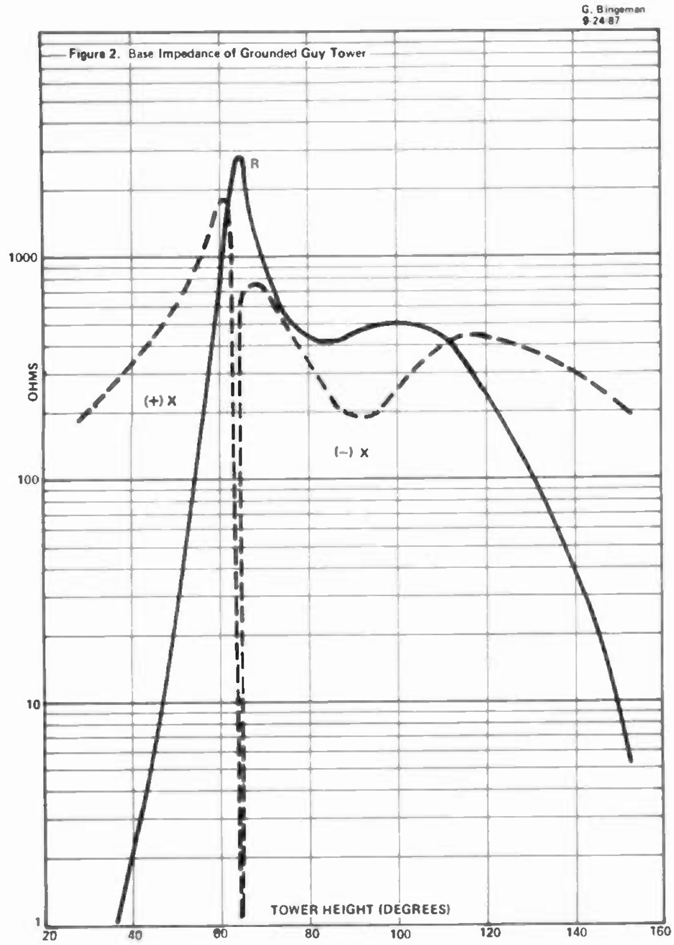
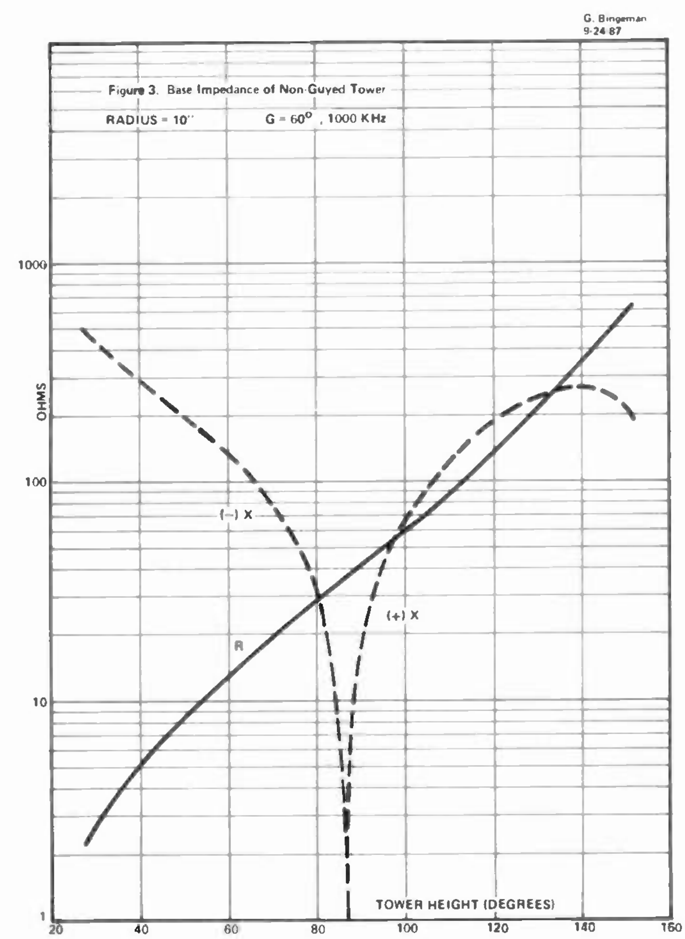


Figure 1. Grounded - Guy Tower

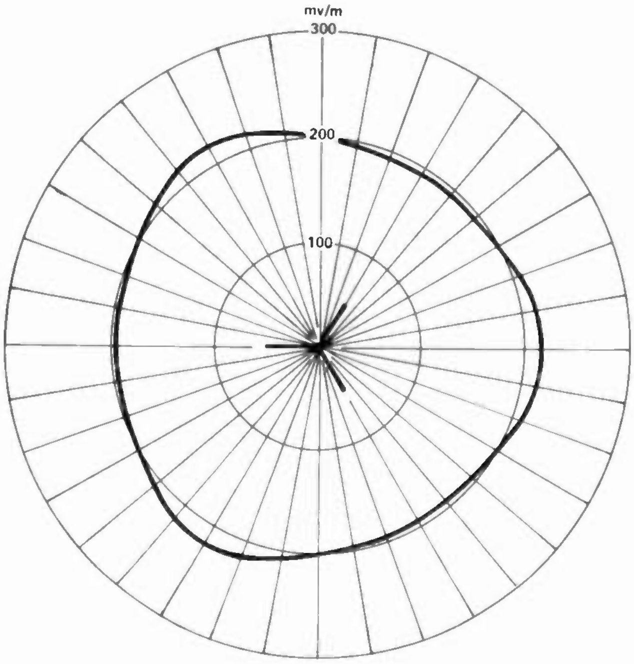

G. Bingeman
9-24-87
Figure 2. Base Impedance of Grounded Guy Tower
OHMS
1000
100
10
1
R
(+) X
(−) X
TOWER HEIGHT (DEGREES)
20
40
60
80
100
120
140
160


G. Bingeman
9-24-87
Figure 3. Base Impedance of Non-Guyed Tower
RADIUS = 10''
G = 60°, 1000 KHz
OHMS
1000
100
10
1
(−) X
(+) X
R
TOWER HEIGHT (DEGREES)
20
40
60
80
100
120
140
160

G. Bingeman
9-28-87

Figure 4. 1 KW Field at 1 Mile, Degradation of Circularity Caused By Hot Guy Wires

G = 150°



ELEV = 0°

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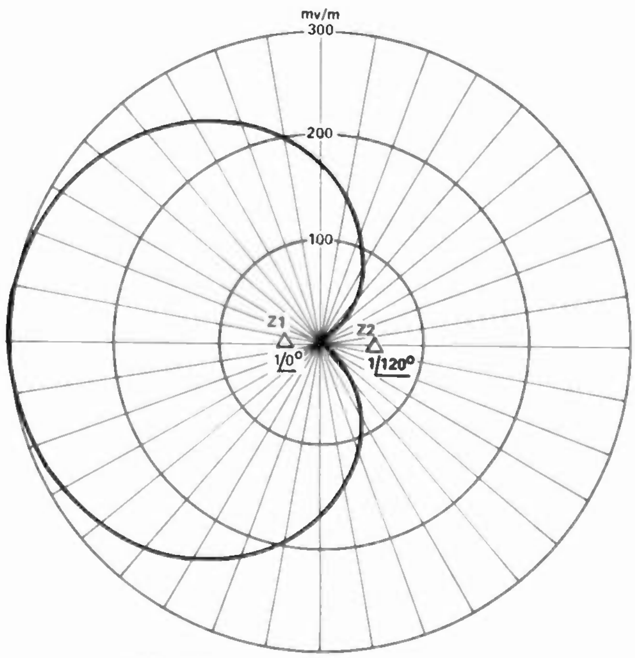
Figure 5. 1 KW Field at 1 Mile

NO GUY WIRES S = 60° G = 60°

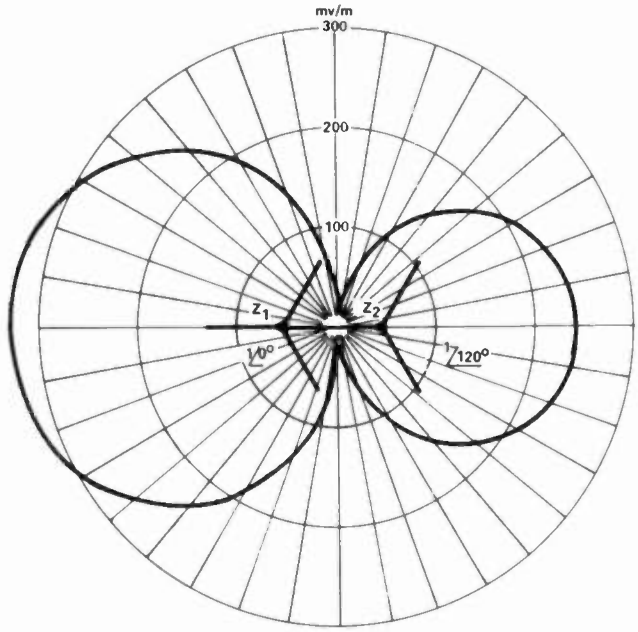


Z1 = 12 - j128 Ω
Z2 = 3 - j145 Ω

ELEV = 0°

G. Bingeman
9-28-87

Figure 6. 1 KW Field at 1 Mile S = 60° G = 60°



Z1 = 3370 + j3790 Ω
Z2 = -290 + j4670 Ω

ELEV - 0°

Figure 7. Grounded - Guy Tower Current Distribution

G. Bingeman
10-2-87

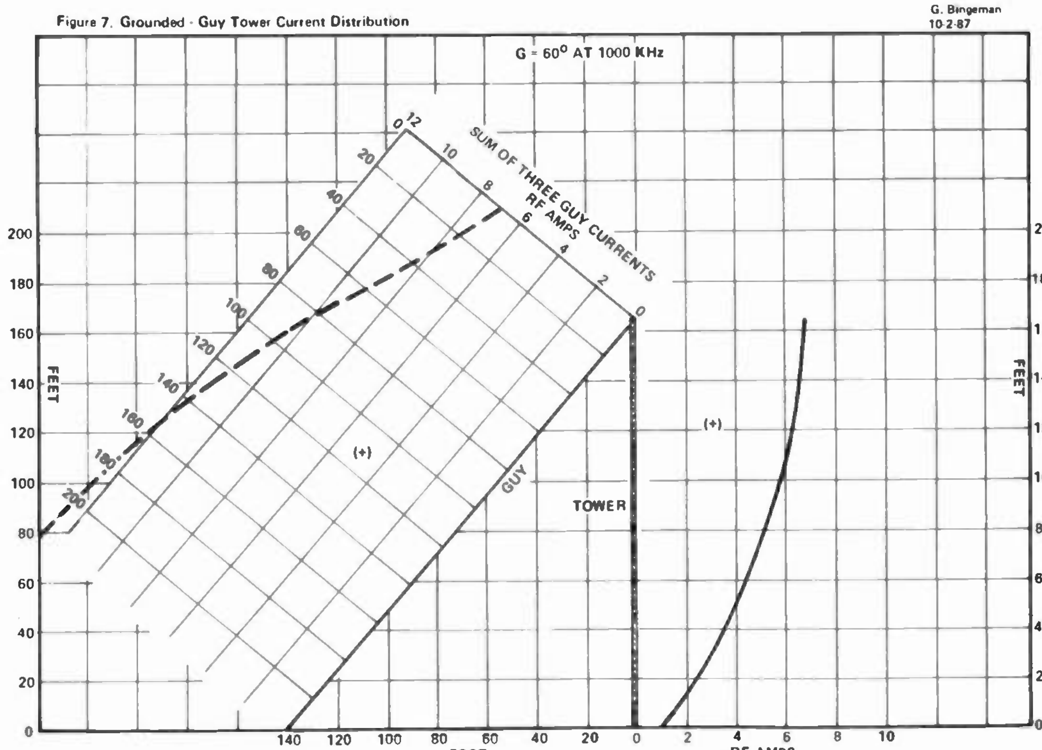


Figure 8. Grounded - Guy Tower Current Distribution

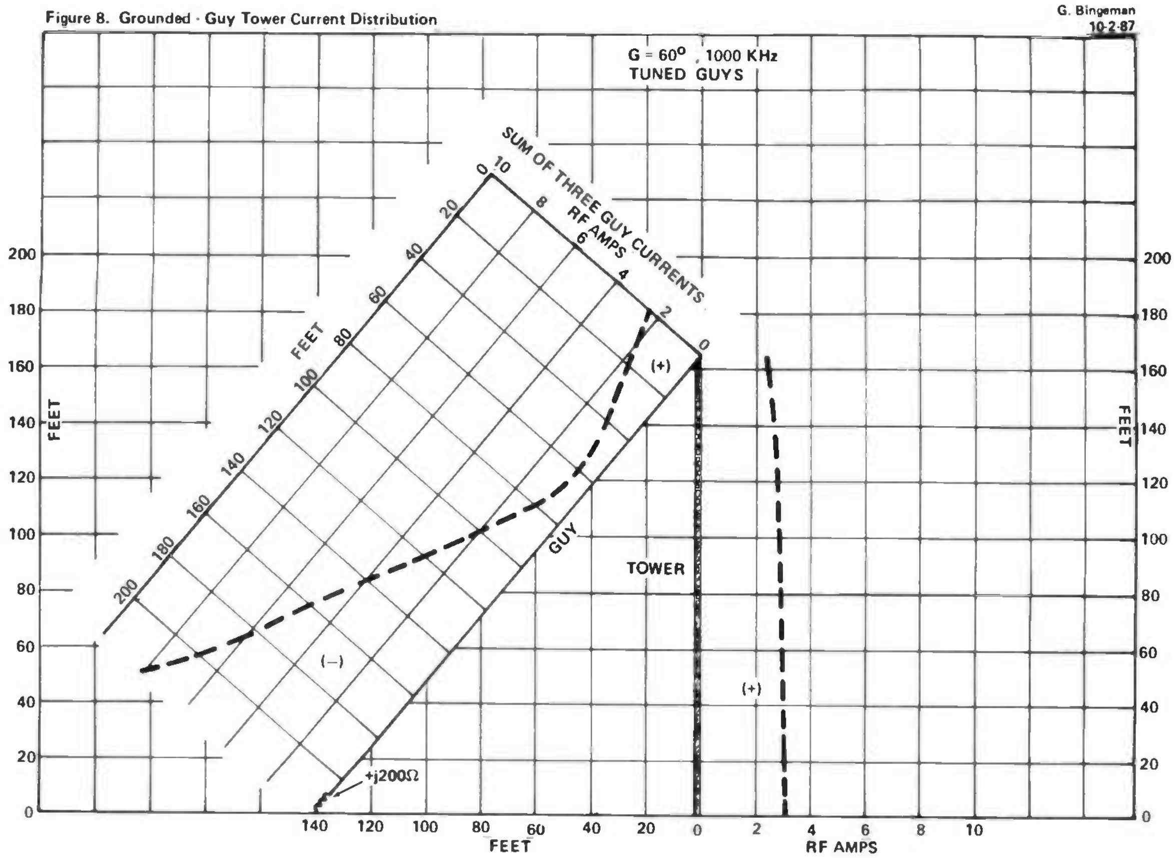


Figure 9. Grounded - Guy Tower Current Distribution

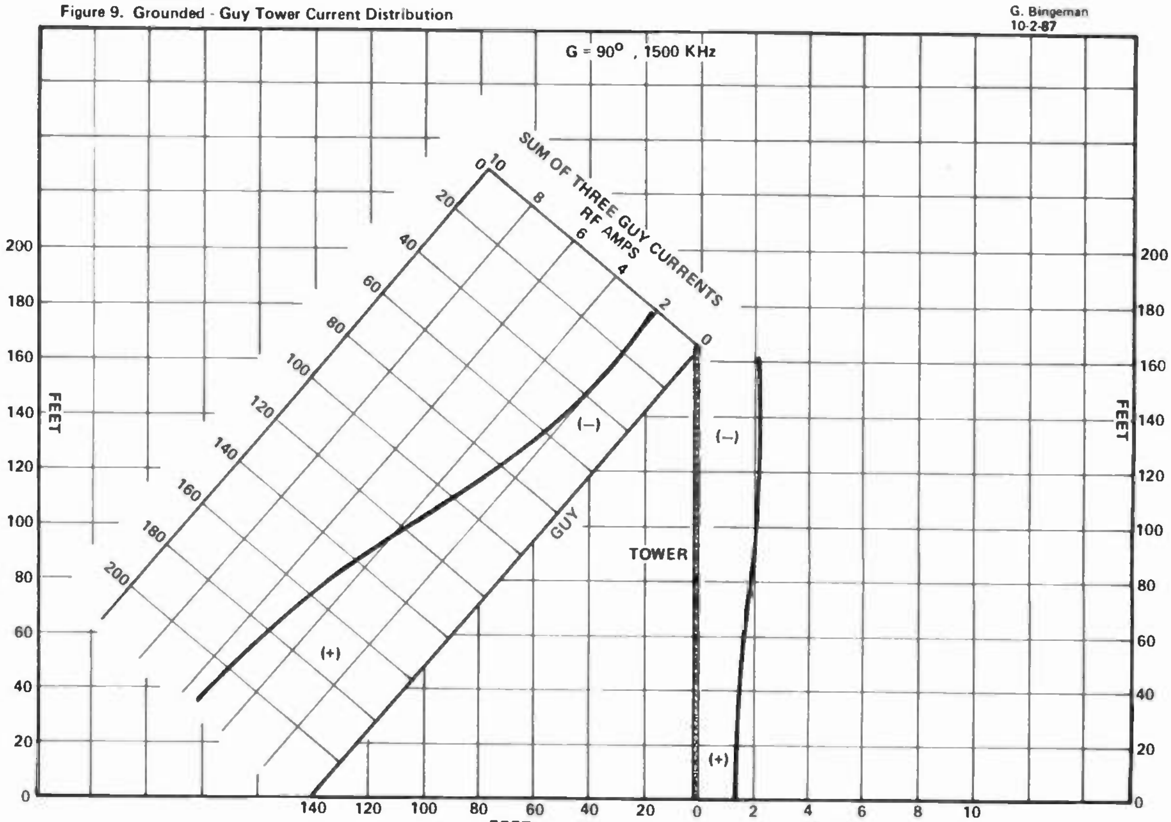
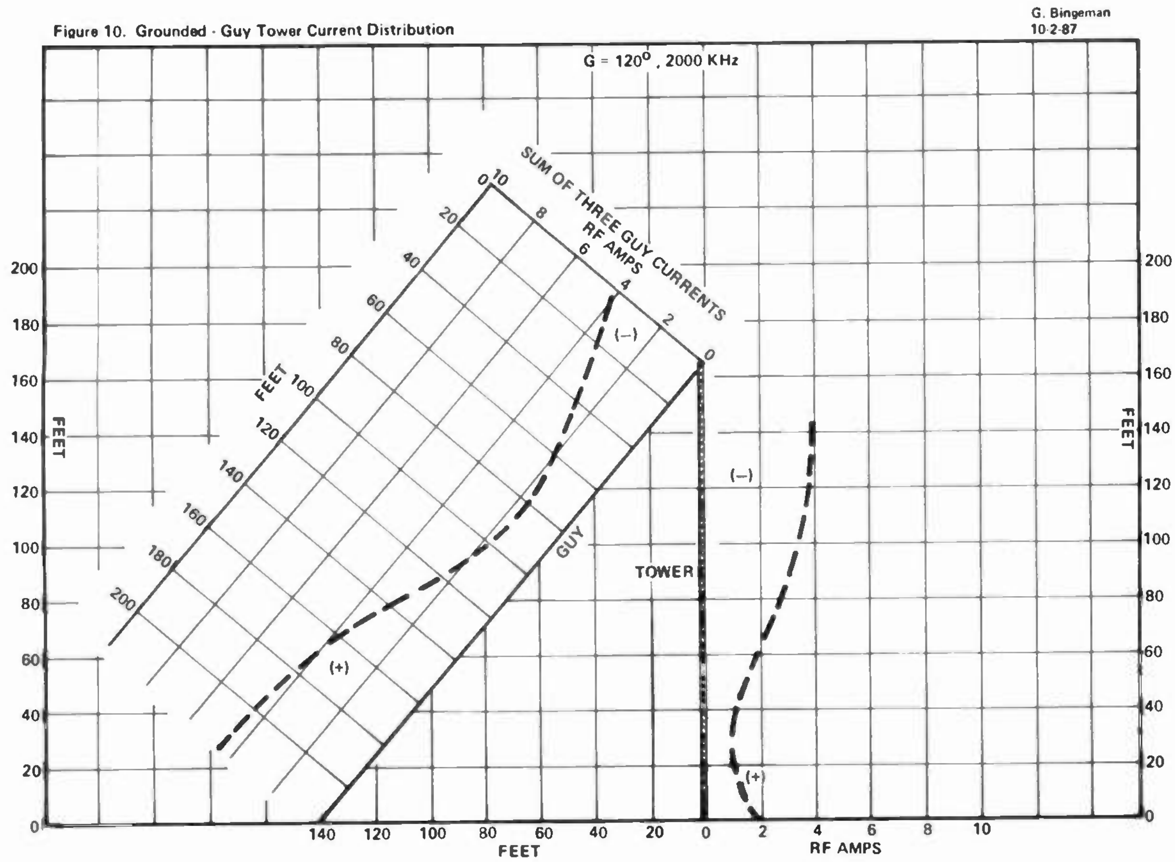


Figure 10. Grounded - Guy Tower Current Distribution



USING FM VERTICAL DIVERSITY TRANSMISSION TO OVERCOME TEMPERATURE INVERSIONS

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Merriam, Kansas

ABSTRACT

In the temperate climate of the Southern United States, a phenominon occurs frequently in the spring and fall months known as a temperature inversion. This layering of different temperature air acts as a partial diffraction radio waves. If the broadcaster is transmitting above these layers, the diffraction can cause a reflection of signal to the extent that virtually all coverage is lost over the desired city for from a few minutes to several hours.

This paper will outline what Shamrock Broadcasting has done to research effects and makeup of the temperature inversion, how we intend to determine when an inversion is in progress, and how we will switch antennas to overcome inversion losses.

INTRODUCTION

Since Shamrock Broadcasting purchased KZFX in Houston, Texas, temperature inversions have been the subject of much research. KZFX is a class C FM station broadcasting from a 2,000 foot tower 35 miles south of downtown Houston.

Being located just eight miles from Chocolate Bay on the Gulf of Mexico, the KZFX signal is subjected to some very severe disturbances due to the temperature inversion layers that form in the area especially during the spring.

These severe inversions can at times make even coverage within the city grade contour of the station virtually useless or worsen multipath conditions (which are normally horrendous) in the shadow of the three major building clusters in Houston. The problem is indeed an important one, since there are certain periods (usually during drive times) that station listenability is very poor.

DEFINING THE PROBLEM

In order to solve the problem of heat inversions, one must first know what causes the problems encountered. During normal transmission of signals, radio waves travel in a relatively straight line from the transmit antenna at 2,000 feet to the surface of the earth. Usually the temperature also varies smoothly from a high point at the surface to gradually lower temperatures as altitude increases.

During a temperature inversion, layers of cooler air are trapped underneath layers of warm air. These inversions are of several types, but the type of most interest to broadcasters takes place usually under 2,500 feet and can last for several hours.During this period, radio waves are deflected upward by the temperature differential between the warm air layer and the cool air layer. If the broadcaster is to overcome this, he must broadcast from a layer of air most like that which is at surface levels, underneath the layer of temperature differential. If one were to switch to an auxiliary antenna under the temperature differential level, one could expect to avoid many of the effects of the inversion. When this is done, it could be expected that by transmitting from a lower height, coverage would suffer. This is not the case. The same temperature differential that bends radio waves upward when transmitted from above, also bends them downward toward the surface when transmitted from below. This tunneling effect can actually increase signal levels on the surface over what one might expect even on days of normal atmospheric conditions.

DETERIMINATION OF INVERSION LEVELS

In order to utilize an auxiliary antenna to transmit underneath the temperature differential level, one must know where that level is occuring. To locate this level, Michael Pass, Meteorologist in Charge of the Houston Area National Weather Service was contacted. With his help we were able to analyze radiosonde observation data supplied by the National Climactic Data Center. Although no radiosonde measurements are made in the immediate vicinity of the KZFX transmit antenna, they are taken in Lake Charles, Loisiana and Victoria, Texas. By using these two sets of data, one can approximate conditions in the Houston area. Two radiosonde readings a day are taken, one at 6:00am and the other at 6:00pm.

The data obtained by the radiosonde balloons is presented in chart form.A reading is transmitted from the unit each time there is a change in the rate or direction of temperature. From this information, inversion levels can be very precisely pinpointed as to height and extent of temperature differential.

Figure 1 illustrates an ideal radiosonde observation. As can be seen by the plot,the temperature decreases smoothly as a function of height. Figure 2 shows quite a different story. The strong shift to the right of temperature indicates a severe temperature inversion. At the time of this reading, KZFX and most staions transmitting from over 1,000 feet in Houston were experiencing severe signal problems.

From radiosonde observations made during April and May, figure 3 graphically shows the levels of temperature differential levels for inversions as plotted against a depiction of the 2,000 foot KZFX tower.By analyzing this information, some tradeoffs become apparent. If one were to transmit from a very low level, it could be expected that virtually all inversions could be overcome, however, even with the tunneling effect coverage would probably be inadequate. therefore, a percentage game needs to be played. Based on the data acquired in figure 3, one could expect to overcome approximately 75% of the problem inversions when transmitting from a level of 250 meters. A height of 275 meters would help in about 62% of the cases and at a height of 300 meters 54% of the inversion problems could be improved upon.

INVERSION DETECTION

In order to take advantage of an auxiliary antenna at a height that will overcome inversion difficultys, one must know when an inversion is taking place and if the lower antenna will be a better choice than the main. To accomplish this, we intend to sample the signal from a station transmitting under 1,000 feet from the city of Houston.

Properly filtered receive antennas will be placed at the 1,000 and 2,000 foot levels on the KZFX transmit tower and benchmark signal levels will be logged. Under ideal conditions, the antenna at the 2,000 foot level will receive more relative signal than the antenna at the 1,000 foot level.During inversion caused interference, the opposite will be true.If both antennas lose signal,(station off air) the alarm will be locked out.With experimentation a level of signal degradation at the upper antenna coupled with an increase in signal at the lower antenna will trigger an alarm at the studio telling the operator of the inversion. The operator then has the option of switching to the lower auxiliary antenna until the inversion condition corrects itself.

SWITCHING ANTENNAS

To affect a quick and nearly unnoticeable antenna switch presents its own problems. Since it is possible that during inversion season antenna transfers can take place several times a day, it is imperative that the switch be extremely quick and for the most part unnoticeable.

The heart of this system is the Continental Electronics 5517,
6 1/8 inch air driven coaxial switch. This unit has the ability to to transfer the antennas in approximately 0.25 second at 30 psi pressure.

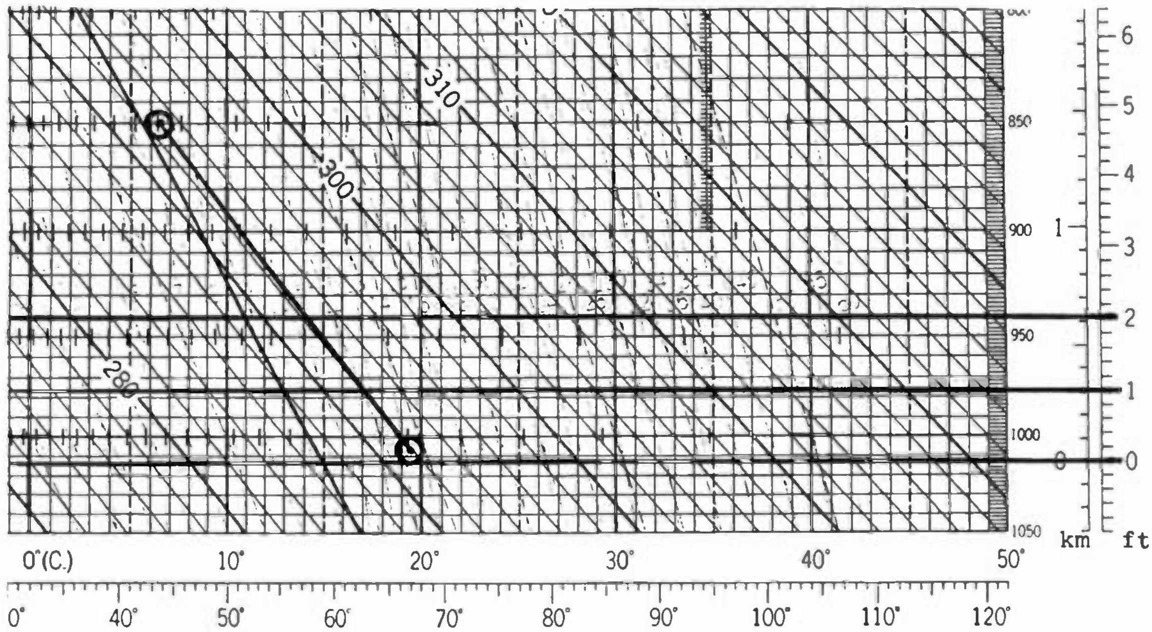
The biggest problem after that is how to quickly switch and interlock two 30 KW transmitters. Although we have not tested the following system, we have every indication that it will work.The system is based on sequential .3second pulses in the following order:

1) Transmitter A, Plate off
2) Transmitter B, Plate off (sense for RF loss before allowing further steps)
3) Switch antennas (sense for proper interlock before allowing further steps)
4) Transmitter A, Plate on
5) Transmitter B, Plate on

CONCLUSION

Microwave users have employed space diversity to overcome the problems encountered by temperature inversions for years, in fact, KZFX uses vertical diversity antennas to provide uninterrupted 950mhz signal for its own STL.

Although, like the weather, we can't change temperature inversions or where they occur, perhaps by changing where it is we broadcast from, the effects of this phenomenon can be minimized.



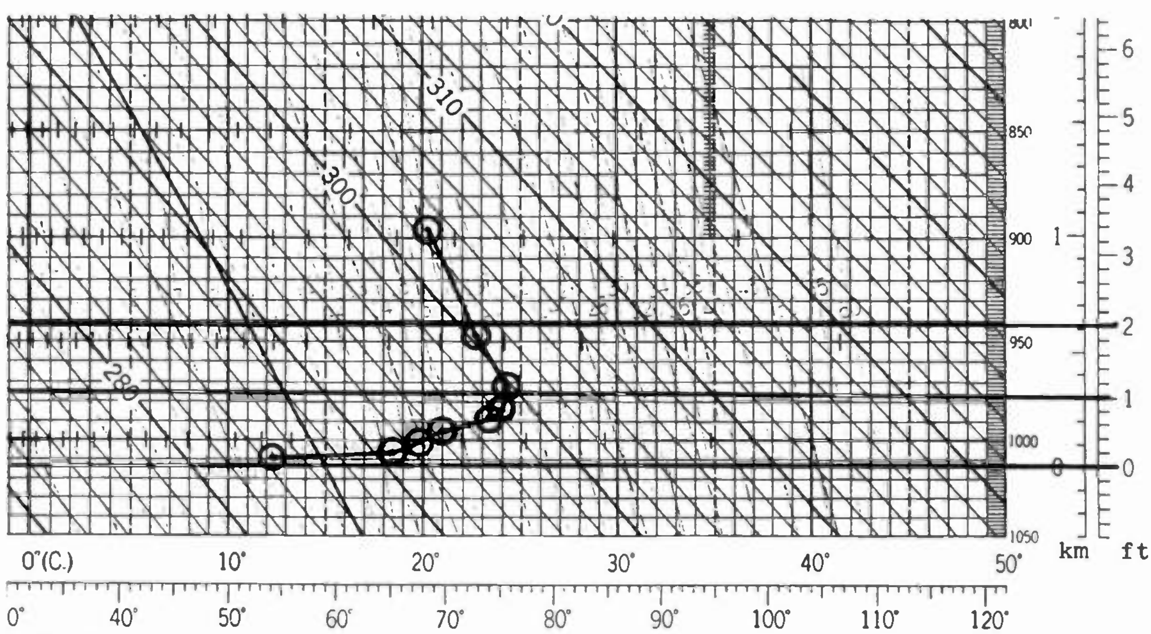
-FIGURE 1-

Illustration of height vs.
Temperature curve on a normal day

Station Lake Charles, Louisiana

Date (G.C.T.) February 28, 1987

Hour (G.C.T.) 6pm CST Drawn by



-Figure 2-

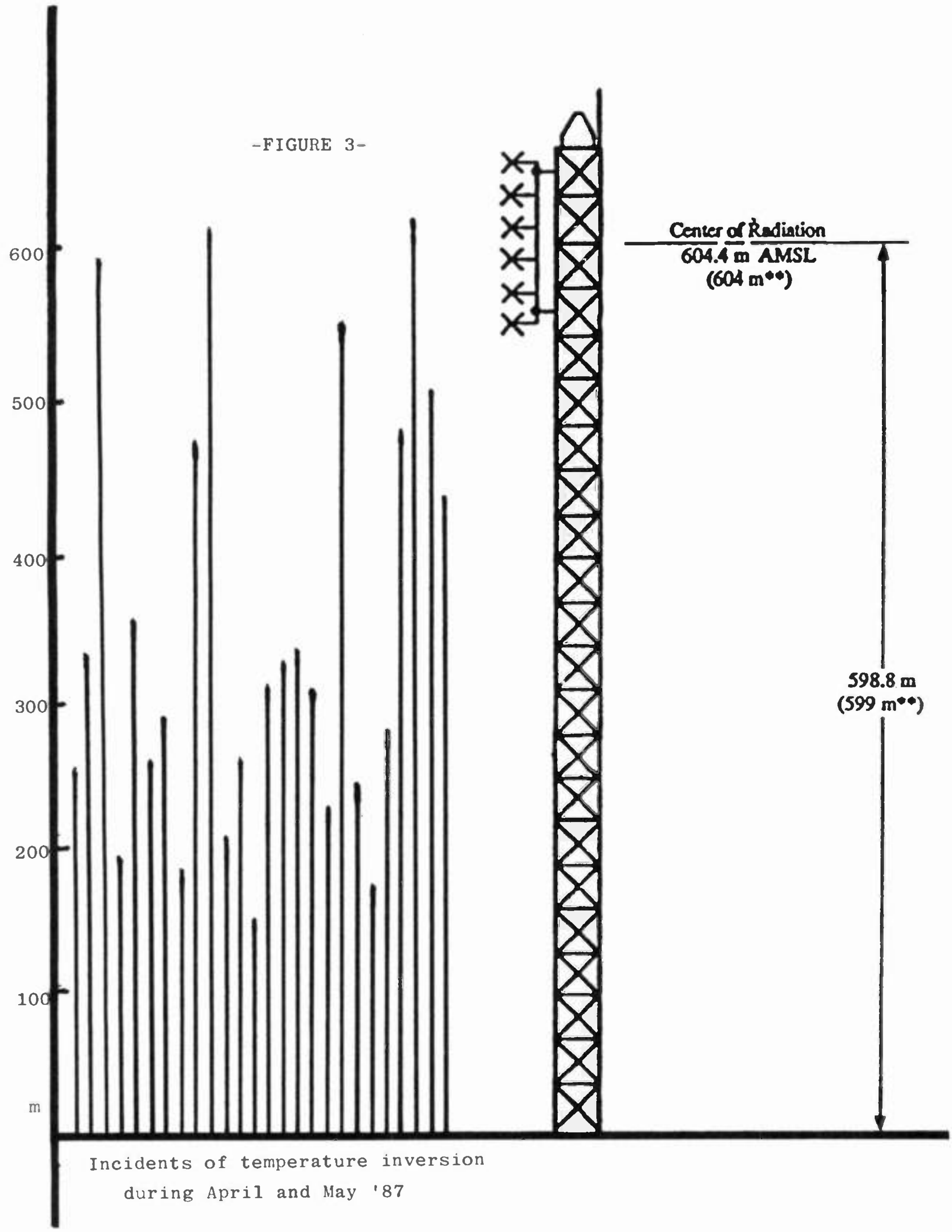
Illustration of severe inversion
layer under 1,000ft.

Station Lake Charles, Louisiana

Date (G.C.T.) April 20, 1987

Hour (G.C.T.) 6am CST Drawn by

-FIGURE 3-


600
500
400
300
200
100
m
Center of Radiation
604.4 m AMSL
(604 m**)
598.8 m
(599 m**)
Incidents of temperature inversion
during April and May '87

ANALYSIS OF AM DIRECTIONAL ARRAYS USING METHOD OF MOMENTS

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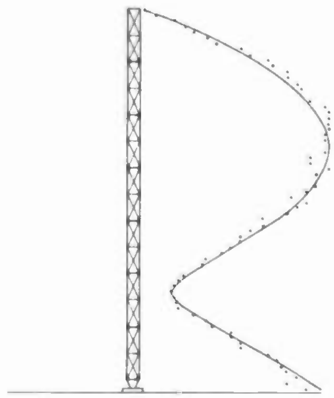
The patterns of AM directional arrays are determined by the "Field Parameters". The Field Parameters are the relative magnitudes and phases of the fields leaving the towers of the array. Directional arrays are adjusted and maintained by monitoring the "Antenna Parameters". The Antenna Parameters are the relative phases and magnitudes of the currents flowing at either the bases or the "loops" of the towers. It is well known that Antenna Parameters and Field Parameters will be different for a particular AM directiona pattern. Method of Moments analysis techiques highlight this difference and provide insight into how to adjust the antenna parameters to achieve the licensed field parameters.

TRADITIONAL ANALYSIS

For over fifty years AM directional antennas have been designed using the same set of assumptions about the relationships between antenna currents and radiated fields. It has generally been assumed that, for arrays which have equal height towers, the ratios and phases, as indicated by the antenna monitor, of the currents at the bases or loops (point of maximum current) of the towers are the same as the field ratios and phases. For this to be true the change in the relative ratios and phases of the currents as a function of height must be the same for every tower in the array. In other words the current must vary the same way along the lengths of all the towers.

NEW INSIGHTS FROM METHOD OF MOMENTS

Computer codes are now available that can be used to determine the details of the current flow in antennas. These programs use an analysis technique called the Method of Moments. Figures One and Two show good agreement between the currents computed by a method of moments program called MININEC III and actual measured currents for a single tall tower and the towers in a three tower directional array. The towers shown in these illustrations are about 5/8 wavelengths tall. Method of Moments analysis of AM directional arrays indicates that the ratios and phases of the tower currents measured by the antenna monitor and sample system are not the same as the field ratios and phases.



COURTESY RON RACKLEY

FIG. 1.
CURRENT DISTRIBUTION ON 5/8 WAVE TOWER

TOWER #1 TOWER#2 TOWER#3

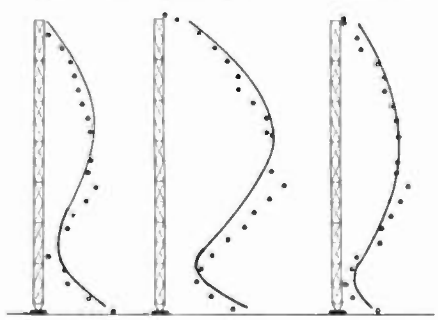


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FIG. 2.
CURRENT DISTRIBUTION FOR 5/8 WAVE 3-TOWER DIRECTIONAL ANTENNA

Figure Three illustrates these differences and shows the variation of the magnitude and phase of the current along the length of the tower. Field ratios and phases are the parameters specified for AM directional antennas by construction permits and FCC records. These "field parameters", for any given tower in an AM array, are the relative magnitude and phase of the electric field component of the radiation leaving that tower.
(One tower is chosen as the reference tower and the ratios and phases of the fields of the other towers are shown relative to that tower.)

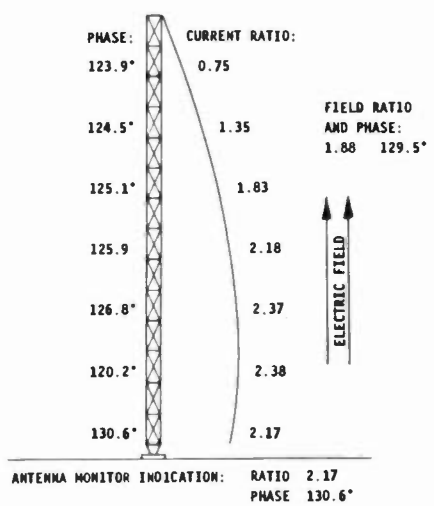

FIG. 3.
CURRENT IN ONE TOWER OF A FOUR TOWER ARRAY

The fields from the towers combine in space to form the directional pattern. The current flowing in each of the towers of an AM array will have different amounts of phase shift and magnitude change along the length of the tower. This effect is due to the energy flowing between towers through mutual coupling.

THEORETICAL PARAMETES DO NOT GIVE THE RIGHT PATTERN

Since the fields leaving the antenna are formed by the sum of all the currents flowing in the tower the field parameters will always be at least slightly different from the antenna monitor ratios and phases. This is not surprising given that the sample system can only provide measurements taken from one particular location on the tower. The field ratios and phases are different from the ratios and phases shown on the antenna monitor.
When a DA pattern has deep nulls this difference can be crucial.

These days most new installations of AM directional antennas employ sample systems that monitor the tower base currents with sample transformers. If there are differences in the heights of towers used in the array large differences can be expected between the antenna monitor indications and the field parameters when the array is properly adjusted. This is particularly true when base currents are sampled even if the situation is not complicated by the presence of re-radiating objects. Figures Four and Five show the standard patterns for the two modes of operation of a four tower directional array. Three of the towers are 72 electrical degrees tall while the fourth tower is 101 electrical degrees tall. The dashed line shows the pattern obtained when the array is adjusted so that the tower base current ratios and phases indicated on the antenna monitor are the same as the theoretical field ratios and phases shown on the construction permit. The fields in the nulls of both patterns exceed the limits of the standard patterns when the array is adjusted to the theoretical field parameters. To achieve the correct pattern the current ratios for the night pattern must be varied by as much as 60% from the CP ratios and the phase angles must be changed by as much as six degrees.

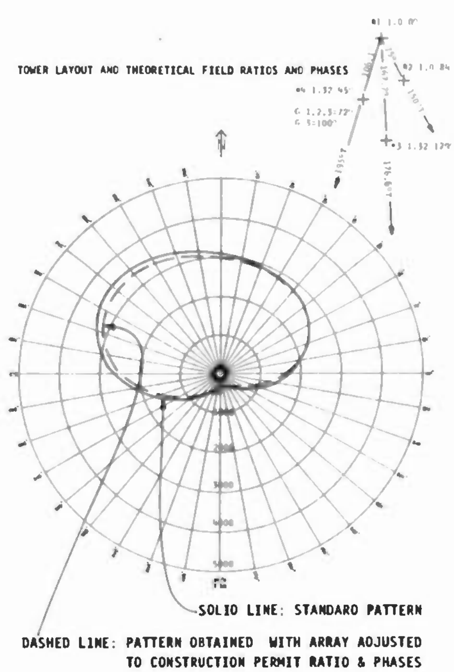

FIG. 4.
DA-D PATTERN

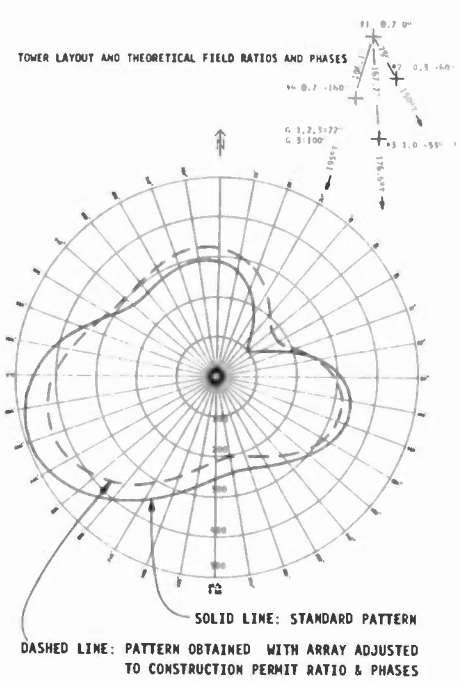

FIG. 5.
DA-N PATTERN

PITFALLS OF TOWER PAIR CALIBRATIONS

Radio engineers have long observed the difference between the theoretical field parameters shown on a construction permit and what the antenna monitor indicates after an AM directional array is properly adjusted. In adjusting arrays where there are more than two towers it has been standard practice to try to overcome this discrepancy by driving the towers in pairs. The base drives to these tower pairs are then adjusted to produce a null in some direction. The field parameters that produce the null are calculated and the differences between the calculated values and the ratio and phase indicated by the antenna monitor are used as correction factors. These factors are then used to correct the ratio and phase indicated by the antenna monitor so that the desired pattern is produced. The problem with this procedure is that the difference between the monitored antenna currents and the field parameters changes when the mutual coupling between the towers changes. When the towers are driven as pairs to determine the correction factors there is mutual coupling only between the two driven towers. When the towers are driven in the full array mutual coupling occurs between the tower in question and every other tower. The correction factors are different for each case. The difference between the base current parameters and the field parameters also changes as the base current ratios and phases change.

An example of this effect would be a four tower array with spacings of 90 degrees between adjacent towers. The first and third towers are 180 degrees apart and can be used to produce an inline null if they are driven identically. For this particular pair of towers the antenna monitor and field parameters are the same. This would indicate that no corrections are necessary for the antenna monitor ratio and phase for tower three. When all four towers are driven to produce the desired pattern the difference between the base current ratio and the field ratio of tower three is 30% and the phase angle difference between the base current and the field is 11 degrees. This may be an extreme example, but it certainly illustrates why antenna monitor readings cannot always be corrected by nulling tower pairs.

MOMENT METHOD YIELDS FIELD PARAMETERS
FROM BASE CURRENT PARAMETERS
AND VICE VERSA

If the field parameters are known for a given directional antenna the antenna monitor ratios and phases that are needed to produce the correct pattern can be found using a method of moments computer code. The field ratios and phases for a particular tower in an AM directional array are proportional to the sum of the current moments for that tower. For this reason the currents for each tower that are calculated by the method of moments code can be used to calculate the field ratios and phases. Matrix inversion techniques can be used to relate the field parameters to the antenna current ratios and phases at a given location on the tower.[1] Conversely, field parameters can be calculated by method of moments codes for a given set of antenna current ratios and phases by using appropriate circuit techniques to convert the current parameters to equivalent constant current sources.

CHANGE THE CURVES AND NOT THE DATA
FOR PROXIMITY EFFECT

The patterns for directional antennas, which are so familiar to most of us, are strictly accurate only at great distances from the array. The approximations that are used to calculate these far field FCC patterns are not accurate close to the towers of an AM directional array. This is called the proximity effect. Proximity effect corrections are sometimes applied to measured field intensity data that is taken close to the towers of the directional array.

Correct field parameters for the operating array must be used to calculate accurate proximity effect corrections. To determine the actual field parameters for an operating array the sample system must give correct indications of the relative tower current parameters. The method of moments computer code can then be used to calculate the operating field parameters for the array.

Since the proximity effect results from an error in the method used to calculate the fields close to the towers it seems more appropriate to correct the calculations rather than the data. In practice this would mean applying the proximity corrections to curves #1 to #19 of 73.184 of the Commission's Rules and Regulations. Measured DA field intensity data would then be fitted to the corrected curve for the particular conductivity resulting from the analysis of the non-directional measured data.

Figure six shows an example of how the conductivity curve would be weighted in a specific case. The 2 MS groundwave conductivity curve has been weighted by the calculated unattenuated fields found close to a hypothetical array. The calculations were based upon the correct near field geometry. The calculated unattenuated fields do not vary inversly as the distance and are generally above the inverse distance line.

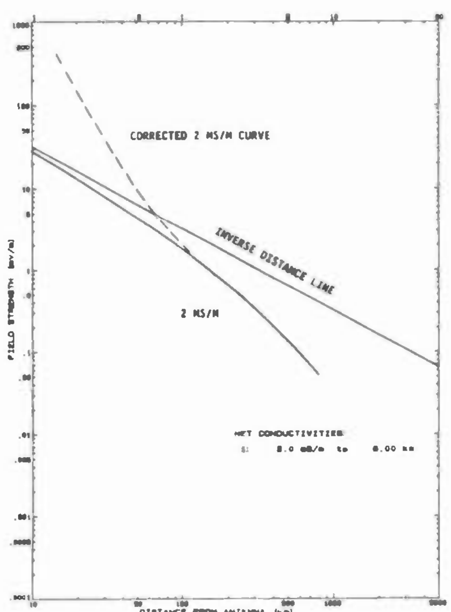

FIG. 6.
GROUNDWAVE CONDUCTIVITY CURVE
CORRECTED FOR PROXIMITY EFFECT

DETUNING RE-RADIATORS

The moment method techniques are applicable to detuning re-radiating poles and towers. The re-radiating object is considered to be a part of the array and the program is used to calculate a drive voltage between the base of the object and ground that will cause its field ratio to be very small. The object can then be de-tuned by using a base resistance and reactance opposite from that which is calculated by the method of moments program. The de-tuning resistance and reactance are connected between the object and ground. The object must be insulated above ground and if the resistance of the object is positive the object must be driven with a voltage at a specific phase angle if it is to be completely de-tuned. In most practical cases the resistive component of the object is so small in comparison to the reactance that the resistive component of the de-tuning impedance can be neglected.

ACKNOWLEDGEMENTS

I wish to thank Ron Rackely for providing the measured tower current distributions illustrated in this paper. He also wrote the lines of program code that had to be added to Mininec III to sum the current moments for field parameter calculations. The use of the matrix to relate field parameters to base voltage and current drives originated with the Harris Corporation where it has been used for a number of years. I would like to thank Jerry Westberg for telling me about it. A paper on this subject by Dr. C. W. Trueman of Concordia University, Montreal, Canada, (see reference #1) will be published in the near future. Our complex matrix inversion program was written by Maria Beth Silkey and Paul Leonard. Dr. R.W. Adler of the Naval Post-Graduate School told me about the technique of converting voltage sources to current sources so that the Numerical Electromagnetic Code could be used to derive field ratios and phases from base current ratios and phases.

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A REALISTIC ASSESSMENT OF AM PATTERN STABILITY

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ABSTRACT

Many technicians and engineers of AM stations with directional antenna systems have observed that, although antenna monitor sample current ratios and phases are well within standard tolerances prescribed by the Federal Communications Commission (FCC), monitor point field strengths may be unacceptably high. For so-called "critical" arrays, the monitor point field strengths may never approach limits, regardless of within-tolerance deviations of antenna monitor readings. The reasons for these phenomena are (1) the FCC's standard antenna monitor indication tolerance is insufficient to assure within-limits radiation in all directions for most patterns and (2) the tolerances usually specified for critical arrays are overprotective. Through the use of a new, random deviation stability study algorithm, a more realistic assessment of the operating parameter tolerance necessary to maintain the pattern within limits may be obtained.

INTRODUCTION

The operating performance of an AM directional antenna system is supervised by observing the antenna monitor indications and taking field strength readings at specified locations in directions of radiation minima and maxima. The antenna monitor indications are, for most cases, directly proportional to the magnitude and relative time phase of the signals radiated by each tower.[1] The array is considered to be operating properly when the antenna monitor indications are within the prescribed operating tolerance and monitor point field strengths are below maximum specified values.

Monitor point field strength limits are established by the FCC on a case-by-case basis, taking into account the field strength at most or all measurement locations along the radial bearing under study, as submitted to that agency in the most recent complete or partial proof of performance. The FCC does not apply any generalized monitor point limit. Each direction, for each station, is assigned maximum values individually.

Such is not the case for sample current ratio and phase tolerances. Unless specified otherwise in the station's license, tolerances of five percent for ratio and three degrees for phase apply,[2] regardless of the nature of the directional pattern. More restrictive tolerances are specified where the pattern has been determined to be "critical". Very few patterns have been so classified by the FCC.

SIMULATION OF ARRAY PERFORMANCE

A complete model of a directional antenna system would take into account the phasing system design and as-adjusted component values, component thermal sensitivities, ground system dimensions and materials, soil characteristics near the array, and local weather conditions. Such an extensive model, while accurate, is not considered practical for the simulation of performance of many patterns. Gathering of the needed data and construction of the model are beyond the economics of AM broadcasting today. Furthermore, it is impractical to apply such a model to the study of all patterns on a given frequency.

A more practical approach is to randomly vary the field magnitude and phase of each tower in the array (other than the reference tower) and compute radiation in pertinent directions. A statistical characterization of the results obtained can be used to determine the relative stability of the pattern.

Subroutine RANDSTUDY

A computer algorithm has been developed to implement a random parameter deviation study. The specific subroutine, RANDSTUDY, was written in compiled, structured BASIC, although it could just as easily have been written in FORTRAN or PASCAL.

RANDSTUDY starts by seeding BASIC's random number generator with the present clock/calendar time so as to achieve a different random number sequence for each study run. Next, all ratio and phase parameters are deviated randomly, except for the reference tower, so as to establish a random set of initial operating conditions for the array. For each ratio or phase parameter, the magnitude of deviation is individually and randomly established between user-specified outer bounds. The pattern multiplying constant ("k-factor") is recomputed based upon the deviated parameters. Theoretical far-field radiation, normalized to one kilometer, is computed for pertinent directions.

The subroutine next randomly selects a single tower, randomly chooses ratio or phase, and then randomly determines a deviation of that selected parameter between the specified limits, such as from -5 to +5 percent for ratio or -3 to +3 degrees for phase. The pattern multiplying constant is again recomputed and theoretical radiation values again found.

One parameter is selected and deviated at a time in order to more closely simulate actual array operation. It is recognized that array type, intertower spacing, and phasing equipment design and adjustment will have a significant effect on the independence of parameter deviations. That is, a significant phase change at one tower induced by the thermal sensitivity of a certain component may "drag along" certain ratio changes at other towers. Even so, this approach is considered reasonable for the purpose of overall assessment.

RANDSTUDY sorts the radiation values computed in each particular direction so as to determine the 90th percentile radiation threshold for that direction, i.e., the radiation value that is exceeded by 10 percent of the trials. Preliminary work with RANDSTUDY suggests that, for a given pattern, the 90th percentile values converge within 0.5 percent from run-to-run for 10,000 random parameter deviations.

The 90th percentile radiation value for each direction of interest can then be compared with the corresponding maximum radiation limit defined by the authorized standard or modified standard radiation pattern to determine whether substantial compliance with radiation limits exists for the ratio and phase tolerances specified.

960 kHz Application of RANDSTUDY[3]

RANDSTUDY is part of a stability analysis program called STABSTUDY, which facilitates a number of pattern stability analyses, including determination of worst-case field strength given operating parameter tolerance, worst-case parameter tolerance given field strength constraints, and the random deviation field strength analysis of RANDSTUDY. The program allows entry of a table of study points defined by azimuthal bearing and elevation angle or automatic determination of bearings corresponding to horizontal plane radiation maxima and minima.

For this study, the directions corresponding to horizontal plane pattern minima and maxima in excess of the pattern RMS were found for each pattern. These directions are those where the FCC usually requires the establishment of monitor points and recommends periodic checks of field strength. The 90th percentile field strength was evaluated at each such bearing.

Thirty-eight directional patterns authorized for use on the 960 kilohertz regional channel were analyzed using RANDSTUDY. For the standard operating tolerance of 5 percent for sample current (field) ratio and 3 degrees for sample current (field) phase, *only 11 patterns (29 percent)* remained below or within one percent of their radiation limits. This total is comprised of 8 of 27 night patterns and 3 of 13 day patterns.[4]

This simulation confirms what station technicians and engineers have known for many years; that maintenance of operating parameters within the standard tolerances of 5 percent for ratio and 3 degrees for phase does not always keep monitor points within limits. It being obvious that five percent and three degrees is an appropriate tolerance for only a minority of patterns, the question of how to determine a more appropriate operating tolerance arises.

CLASSIFICATION OF AUTHORIZED PATTERNS

An extensive, computerized study of array stability has been undertaken using the STABSTUDY program. Nine AM band frequencies were chosen for study, four "regional" frequencies to which Class III stations are assigned, two "I-A" clear channel frequencies to which Class II stations have only been recently assigned, two "I-B" clear channel frequencies to which Class II stations have long been assigned, and one Canadian clear channel which has had sporadic development of usage. The actual frequencies within each group were merely chosen so as to achieve an even spacing across the band within the group.

FCC Classification Procedures

The FCC ordinarily does not determine on its own whether or not a licensed or proposed pattern is "critical". It only investigates the matter if a party opposing the application alleges that the array is "unstable". While the opposing party is often a clear channel station trying to maintain its dominance of the frequency, it may be a competing station trying to delay competition or otherwise harass the applicant station. Regional channel station proposals are seldom subject to opposition based on pattern stability.

In assessing array stability, the FCC makes computer studies accessing the subroutines MRV and/or SMKVAR, described below. The output of these subroutines and the supervisory program, RADIAT, is a table of deviated radiation values in horizontal and vertical intervals of five degrees each. This two-dimensional matrix is compared to the standard or modified standard radiation matrix computed at similar intervals for non-deviated parameters. If the deviated radiation exceeds that permitted at any horizontal bearing or vertical elevation, the specified deviation tolerance is considered insufficient to assure compliance with radiation limits.

FCC Stability Subroutines

The FCC subroutines MRV and SMKVAR, found in its pattern analysis program, RADIAT, are used in assessing stability. Neither MRV nor SMKVAR apply a random study algorithm. Both attempt to determine the worst possible radiation in each pertinent direction for the specified ratio or phase tolerances. There is no assessment of how likely it is that such a combination of parameter variances will occur. It is merely assumed that the worst case scenario can occur and, when it does, whether tomorrow or in the next century, horrific interference will undoubtedly result. MRV and SMKVAR are only useful for determining stability at an extreme.

Neither MRV nor SMKVAR does an absolutely accurate job of assessing the worst case, since (1) neither recomputes the pattern multiplying constant ("k-factor") for deviated parameters, (2) the worst case deviation direction is chosen for each tower and parameter serially, by tower number rather than by a hierarchy of sensitivities, and (3) the unity field ratio and zero phase tower of the pattern studied, usually selected as the reference tower, is unlikely to be the reference tower in actual operation if it is not the highest field tower.

FCC Stability Categories

Under current FCC policy, patterns are categorized as "generally stable", "highly unstable", or evaluated on a case-by-case basis.[5] If worst-case deviations of one percent for ratio and one degree for phase do not result in excessive radiation in any direction, the pattern is considered "generally stable" and specific ratio and phase tolerances need not be assigned. If worst-case deviations of 0.1 percent and 0.1 degree result in excessive radiation, the pattern is considered "highly unstable" and the proposal may be denied or designated for hearing on stability issues.[6] Patterns between these tolerance extremes are evaluated on a case-by-case basis.

If worst-case deviations of 0.5 percent and 0.5 degree result in excessive radiation, the pattern is generally considered "critical" and specific tolerances are required. Patterns which do not exceed limits at 0.5 percent and 0.5 degree but do exceed at one percent and one degree are generally considered "unstable", with the ultimate decision on whether tolerances will be specified and what those tolerances will be based on related factors such as the characteristics of the transmitter site, RSS/RMS ratio, agreement of the opposing parties, etc.

There are several fundamental problems with current FCC policy. First, it is highly discriminatory in that only stations subjected to opposition are evaluated. Second, there need be no connection between excessive radiation and interference in order for tight tolerances to be applied. The bearings where radiation exceeds limits for one percent and one degree worst-case deviations could all be far away from any protected cochannel or adjacent channel station, yet the pattern would still be classified as "unstable". Third, the bearings where radiation is excessive may not be those assigned to be measured during the proof of performance or monitored during the operation of the station.

Subroutine WORSTUDY

A worst-case radiation evaluation subroutine, WORSTUDY, has been written in compiled and structured BASIC to find worst-case tolerance or radiation values without the deficiencies apparent in MRV or SMKVAR. WORSTUDY performs, for each direction, a vector sensitivity analysis on the ratio and phase parameters of each tower other than the highest ratio tower, which is made the reference. Based on that analysis, the magnitude of ratio and phase deviation for all non-reference towers and the sign of deviation for each individual parameter are established. The parameter deviations so determined are implemented and the pattern multiplying constant recomputed. WORSTUDY then reevaluates the deviated radiation and sensitivities. The process repeats until there is no change in the sign of deviation sensitivities from one trial to the next and the deviated radiation just exceeds that permitted, assuring that the worst possible case has been found.

Worst-Case Pattern Classification

A worst-case tolerance study was conducted for the nine frequencies described above, using the subroutine WORSTUDY. For each pattern, the directions corresponding to horizontal plane pattern minima and maxima in excess of the pattern RMS were found. These directions are those where the FCC usually requires the establishment of monitor points and periodic checks of field strength. The worst-case tolerance for each set of directions, i.e., pattern, was determined. The results for all 264 patterns were classified as "stable", "unstable", or "critical", as set forth in Table I. No "highly unstable" patterns were found.

Table I

Worst-Case Pattern Classification

Type	Number	Stable	Unstable	Critical
Total	264	47%	31%	22%
Day	127	50%	35%	15%
Night	144	44%	28%	28%
Clear	100	51%	32%	17%
Regional	164	45%	30%	25%
I-A	22	41%	41%	18%
I-B	46	52%	35%	13%
960 kHz	38	74%	8%	18%

To the station technician or engineer, the results should be surprising, since less than 50 percent of the patterns examined were found to be "stable", yet so few stations have specified tolerances or have been determined to be critical, a fate seemingly reserved for major market Class II stations having six or more towers.

Even more surprising is the small difference between the percentage of day and night patterns considered "stable". Since night radiation is usually suppressed more than day radiation and, often, more towers are used at night, station personnel naturally assume that it is the night pattern that is less stable.

The proportion of authorized patterns actually having specified operating tolerances below 5 percent for ratio and 3 degrees for phase is well below the 22 percent shown above. It is doubtful that more than 50 patterns, on all frequencies nationwide, are so designated.

Random Study Pattern Classification

The 960 kHz patterns were analyzed using RANDSTUDY for field ratio deviations of up to 1.75 percent and field phase deviations of up to 1 degree. In this case, *33 of 38 patterns (87 percent)* remained below or within one percent of their radiation limits.[7] This total includes three patterns whose worst-case tolerance was below 0.5 percent/degree and all patterns but one whose worst-case tolerance was above that threshold.

CONCLUSION

It is readily apparent that the FCC's present pattern stability policy is discriminatory and produces highly anomalous results, i.e., a pattern which is relatively stable under the random study algorithm or in actual operation may be assigned overly restrictive tolerances based on the results of worst-case analysis urged by an opponent, yet a pattern which is not particularly stable under the random study algorithm is never even checked if no opponent steps forward.

If FCC pattern classification were based upon results of the RANDSTUDY algorithm, most 960 kHz patterns would be classified as "generally stable". Under current policy, license-specified operating parameter tolerances would not be required. While these patterns are not "critical", this study shows that the standard tolerances are not appropriate, either. What the results of this study suggest is that maintenance of pattern shape and protection from interference requires antenna monitor parameter operating tolerances that are unique to each pattern, just like monitor point limits which have existed for years.

As stated at the beginning of this paper, maintenance of patterns is regulated by the dual scheme of uniform antenna monitor parameter tolerances and custom monitor point limits, the latter usually exceeded before the former. Present operating requirements specify no interval between monitor point readings. Since monitor point readings were deregulated, many stations have abandoned reading monitor points on a weekly, or even monthly, basis. As a practical matter, the empirical cross-check on the 5%/3° tolerance has largely been removed.

Antenna monitor operating parameter indications are generally more reliable than monitor point readings, since they are relatively unaffected by seasonal ground conductivity changes and construction and/or demolition near the monitor points. Monitor points are sometimes chosen to give similar readings regardless of the overall field strength along the bearing they are on. Even if the monitor points are read regularly, there is no guarantee that within-limits readings consistently result in proper radiation. Canadian stations have no monitor points, yet interference from those stations is not known to be a major problem.

Given this situation, the need for assignment of antenna monitor parameter tolerances related to the stability of the individual pattern is apparent. The assignment of rational, specific operating parameter tolerances to each pattern and the maintenance of them could eventually lead to the elimination of monitor points and all the environmental hassles associated therewith.

FOOTNOTES

1. The mathematical relationship between field ratio, phase, and radiated field strength is given by Equation (1) of Section 73.150(b)(i) of the Commission's Rules, 47 CFR 73.150.

2. See Section 73.62(a) of the FCC Rules, 47 CFR 73.62

3. As of the publication deadline for this paper, only patterns on this frequency have been studied. The results for several other frequencies will be presented orally and are available from this author after 30 April 1988.

4. Unlimited-time patterns were considered twice.

5. Letter from Richard J. Shiben, Chief of the Broadcast Bureau, to the Association of Federal Communications Consulting Engineers, ref. 8800-DW, 9 August 1979.

   See also Memorandum Opinion and Order, *WKKQ, Inc.*; BC Docket 82-227, FCC 82-195, released 29 April 1982

6. See Memorandum Opinion and Order, *Radio Nevada Corp. (KDWN)*, BC Docket 79-313, FCC 79-748, released 28 November 1979

7. The publication deadline foreclosed the possibility of including evaluation of tolerances for the five patterns exceeding limits at this tolerance. Such results will be presented orally.

OPTIMUM USE OF TALL TOWERS FOR AM

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ABSTRACT

The success of FM broadcasting and the FCC's action in Docket 80-90 has led to an increase in the construction of tall towers for FM use . Many of these towers are built on sites which are jointly used by an AM station. For the first time more vertical aperture than needed for AM antenna efficiency may be available. Methods of controlling current distribution are reviewed. The relation between some current distributions and vertical radiation patterns is described. The analysis leads to the optimum current distribution for AM efficiency and minimum high angle radiation from the tower and makes provision for the grounding of the tower base in many cases.

I. INTRODUCTION

FM broadcasting has become the dominant aural radio medium. The efforts of many broadcasters to improve their facilities, partially because of the FCC action in Docket 80-90, has resulted in the construction of many tall towers on or adjacent to the transmitter sites of existing AM stations. These towers if properly located can be useful to the AM facility. However for the first time we find that we may have available more vertical aperture than we need for the normal approach to AM antenna systems.

These towers are in many instances too tall to be used as conventional base fed radiators. Figure 1 shows the current distributions and vertical radiation patterns for towers ranging in height from 0.2 to 0.8 wavelengths. It is obvious that for tower heights above 0.6 wavelengths they can not be used as is for AM.

Most of the work on this project had been completed before the author had a satisfactory method of moments computer program in operating condition. A computer program was written which calculated the theta radiation component from a section of antenna with sinusoidal current distribution, summed up such components from several such sections and provided outputs of current distribution and vertical radiation pattern. The current

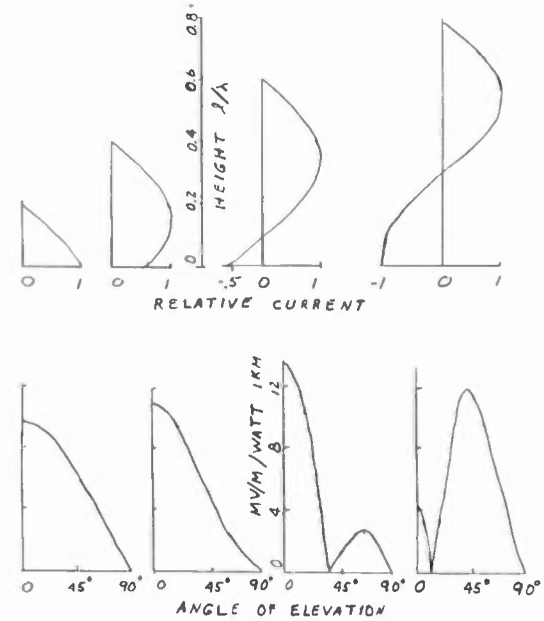

FIGURE 1

availability of MININEC3 has made it possible to add some analysis using that code. The illustrations herein come from both codes.

II. USEFUL CONTROL OF CURRENT DISTRIBUTION

Let us first discuss the tools that we have available to control the current distribution and vertical radiation pattern from the tower. We will examine some array pattern concepts, sectionalizing of the tower, guy cable top loading,and the choice of feed points on the tower.

A. USEFUL ARRAY PATTERN CONCEPTS

In the design of linear arrays we find different requirements for maximum gain, minimum side lobes, slewing (beam tilt), and other desirable or undesirable side effects. Let us first examine the conditions which yield minimum or no side lobes.

Figure 2 shows the geometry and vertical radiation patterns for a series of antennas in a linear array. For simplicity the antenna element is considered to an isotropic source. For any other element the final pattern is merely the product of this array pattern and the element pattern. The array pattern for a pair of elements is readily found from the geometry of Figure 2a. The phase advance for upper element is S/2xCOSθ where theta is the angle from the line of the array and the phase delay for the lower element is the same. Adding the real components we get 2x(S/2)xCOSθ and the imaginary components cancel.

If we now consider that two element array as an element itself and space two such systems again at spacing of S between the centers we obtain the array of Figure 2b. Since the total pattern is the product of the array pattern and the element pattern the array pattern of the three element pattern in Figure 2b is the square of the array pattern for the two element array. This process can be continued and the array pattern is the (n-1)'th power of the two element array for an n element array.

Now let us make the spacing S equal to one half wavelength. Note that for the two element array we have a major lobe normal to the line of antennas and a far field zero in the line of the array. Note further that there is still a zero in the line of the array for all larger numbers of elements in the array and there are no side lobes. Too be sure the gain is lower than it would be for uniform current in all elements but we may be interested in control of the current distribution to obtain a specific vertical pattern from an array.

Consider now an array of six elements as shown in Figure 3. Assume that the elements are all fed in phase and with equal currents. As shown in Figure 3a this antenna has a major lobe normal to the line of elements and a first null at an angle from the line of the array whose cosine is the wavelength divided by n times the spacing. Examine now the illustration of Figure 3b. Here we have a vector diagram of the summation of the element vectors from the six elements in the direction normal to the array line. We have shown the vectors paired that are at equal distances from the array center. The total radiation is proportion to twice the length of vector summation shown.

Now consider Figure 3c, here we have changed the angle to 3.2 degrees from the normal to the array (theta is 86.8 degrees). Note that the symmetrical pairing results in a linear combination of the projections of the vectors along the chosen angle and the net field is reduced.

Next we will look at Figure 3d. Here we have further changed the angle from the normal to 6.4 degrees. We now have a significant reduction in field.

Figure 3e shows the vector diagram for 9.6 degrees from the normal which is that for the first null as described above. Note now that the resultant from the 1-6 pair exactly cancels the resultant

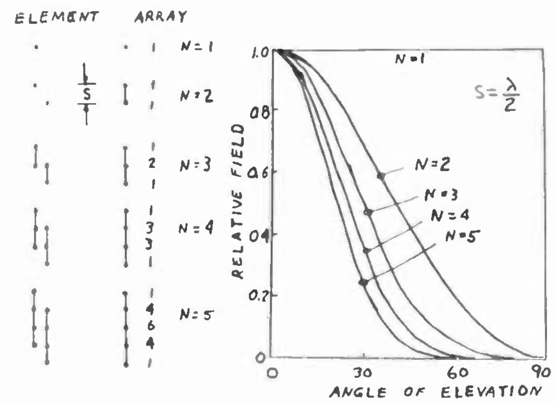

FIGURE 2

from the 3-4 pair and that the 2-5 pair are out of phase.

Now consider what would happen at this angle if we made the 1-6 vector pair have more radiated field and the 3-4 pair have less. Under that circumstance the net field is no longer zero but has changed sign and is negative, we have gone past the null. Therefore for some angle less than that which we calculated earlier for

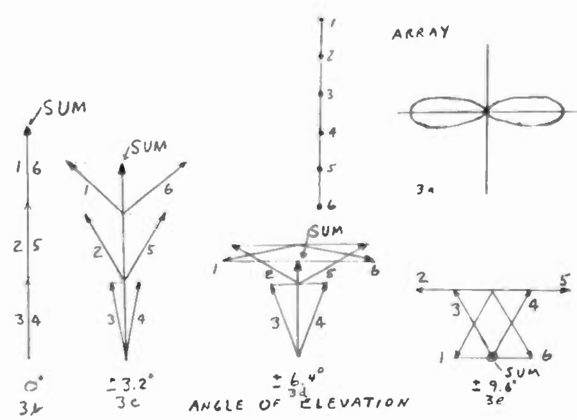

FIGURE 3

the first null we find the null, the beam width has been reduced, consequently the gain has been increased. The effect is not large but it is real. A consequence of this change is that the first minor lobe increases in size.

The two concepts described have opposing effects, one broadens the major lobe and removes minor lobes while the other narrows the major lobe and increases the first minor lobe. These concepts will be useful in working with stacked AM antenna arrays.

B. SECTIONALIZED VERTICAL ANTENNAS

Let us now look at Figure 4. Here we have depicted a base driven vertical antenna 3/8 ths wavelengths in height with a variety of sectionalizing reactances. The power is 1 kW and the currents are in amperes. These examples were derived with the aid of MININEC3 and both the quadrature (solid line) and in phase (dashed line) components have been shown.

Figure 4a shows the current distribution for the antenna with no sectionalalizing loading.

In Figure 4b we have added a sectionalizing inductive reactance (+J 150 ohms). Note that the current distribution is now in two parts, the part above the loading reactance which has similar sinusoidal shape as it did for the unloaded antenna and a second section below the loading reactance which is of a similar sinusoidal distribution but the current loop and the following current minimum have moved up on the tower.

Figure 4c shows the antenna with a larger sectionalizing reactance (+J 300 ohms), here the current minimum has clearly moved up onto the tower. Figure 4d shows a further increase in sectionalizing reactance (+J 1000 ohms). Here the current minimum has moved up almost to the sectionalizing point.

Figure 4e illustrates the condition for an essentially an open circuit at the section point. Note the significant change in phase of the principal current for 4e and the move of the minimum to the sectionalizing point.

Figure 4f shows the current distribution for a capacitive (-J 300 ohms) loading reactance and Figure 4g is for a loading of -J 3000 ohms. It is seen that for the case of capacitive loading the current loop and minimum move down on the tower in contrast to the upward move for inductive loading.

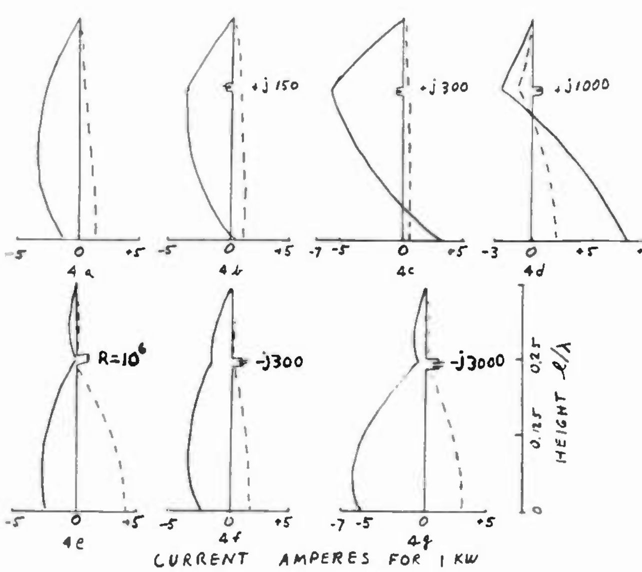

FIGURE 4

C. GUY CABLE TOP LOADING

Current distribution may also be altered by the addition of a capacitive hat or sections of top loading guy cables. This process has the effect of moving the current loop and minimum up on the tower.

Figure 5 illustrates this effect on the current distribution of a 3/8 ths wavelength tower. The guy angle was selected to be 30 degrees from the vertical and four different lengths are shown.

The current loop and the following minimum move up on the tower with increasing loading. This is not of much use to us here but it is included for completeness.

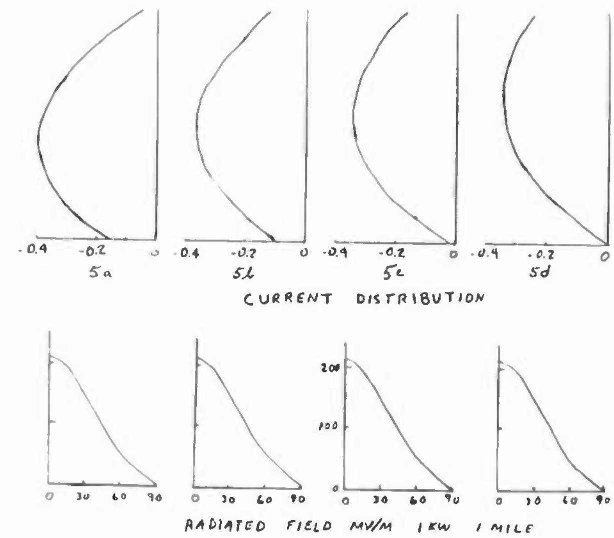

FIGURE 5

D. NON BASE FED ANTENNAS

When the simple vertical antenna is base fed its current distribution is to a first approximation that of a sine wave from the top of the tower downward. However when the feed point is moved up the tower a change in the distribution occurs for the region below the feed point. The current below the feed point behaves like a cosine function from the base up to the feed point.

Figure 6 shows a half wavelength antenna under various feed point conditions. Figure 6a shows the current distribution for base fed antenna. Figure 6b shows the current distribution when the antenna is fed at a point 1/4 of the way up the tower.

Figure 6c shows the current distribution when the antenna is fed at the mid point. Note that when the lower portion of the antenna exceeds a quarter wavelength there is a phase reversal between the current loops in the upper and lower portions of the antenna.

Figure 6d shows the current distribution when the antenna is fed at the midpoint and with a reactance of +J 100 ohms in series with the base. Figure 6e shows the conditions when fed at the midpoint and with -J 100 ohms at the base. Here the radiation resistance is low and the reactance high accounting for the high currents. Also the large current areas have substantial cancellation of radiation in the horizontal direction, consequently this would not be a good arrangement.

These then are our basic current distribution control tools.

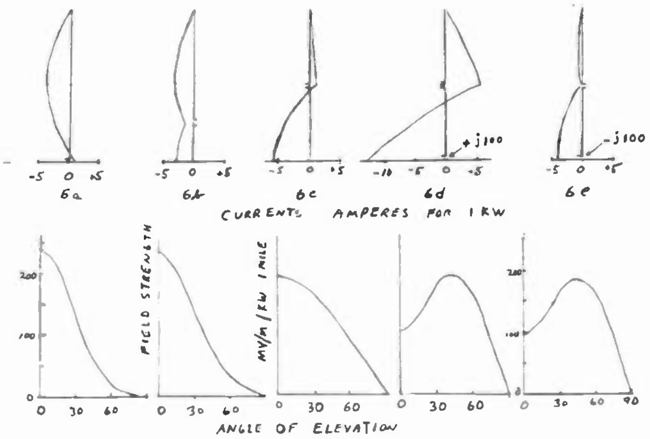

FIGURE 6

III. DESIRED CURRENT DISTRIBUTIONS

In the past we have generally been looking for more antenna height to give us higher gain and to control fading on high power stations. Now for the first time we appear to have systems where we have an excess of aperture and can use it to obtain our best performance. In many instances we can have high gain and still have the luxury of minimum or no high angle lobes.

Our preferred vertical pattern for most uses would be the highest gain that we can get without any high angle radiation which would cause night time fading or excessive nearby interference. It may be that in some instances an existing directional antenna would have to be redesigned to accommodate a different vertical pattern from one element. Should that be difficult then the optimum current distribution and vertical pattern would have to contoured to closely match that for the element it is replacing in the array.

We then have two targets for our best fit. One to match a quarter wavelength or similar antenna and the second to obtain the maximum gain with essentially no minor lobes.

A. METHODS OF ANALYSIS

In the discussion above we referred to a computer code which would enable us to analyze directly the behavior of an antenna with a specified current distribution. Figure 7 shows the geometry of an antenna with a arbitrary current distribution which consists of segments of sinusoidal current distributions. Note that three terms M(n),S(n),and L(n) adequately describe the geometry of the n'th section of current. Further if I(1) is specified then I(2) can be calculated directly from I(1),M(1),and L(1). Then I(3) can be calculated from I(2), M(2), S(2) and L(2), etc.

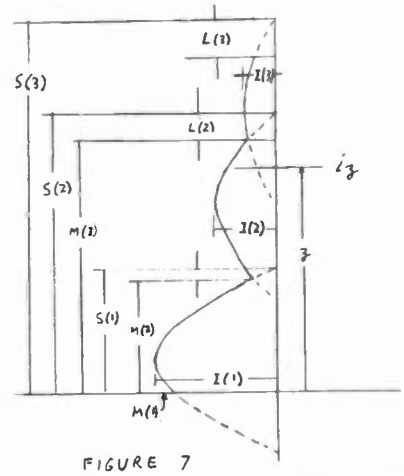

FIGURE 7

The equation for the current distribution for the n'th section can be written directly and since it uses sinusoidal currents it can also be integrated directly to obtain the radiated field strength and its phase at the angle theta from the vertical. These contributions can then be summed up in a computer program to obtain the field from the entire tower at that angle.

Further by summing up the entire power flow we can determine the drive point resistance and correct the radiated fields for the actual power radiated.

This program which is called AMSTAK permits us to directly determine the vertical pattern from a specified current distribution. Then by the use of a method of moments program such as MININEC3 we can by trial determine the exact nature of the drive point impedance and the values of sectionalizing reactance needed to generate that current distribution.

AMSTAK assumes that the phase of the current for the individual sections is uniform throughout their length. This a reasonably good assumption if the sectionalizing points are close to equidistant from the current loops on either side of the feed junction and the continuing section is not being feed near a current minimum. The use of MININEC3 as a follow up corrects these errors and permits a second and more accurate trial for the desired distribution.

Note also that the drive point impedance is controlled by the current magnitude with respect to the current loop at the feed point. High drive point impedances are to be expected when the feed point is not near a current loop. However, as in any antenna, the voltages at these low current points are going to be high whether or not it is a feed point.

B. HIGH GAIN WITHOUT MINOR LOBES

The array patterns discussed above are the clew to this current distribution. Note that the spacing between elements in that array had to be a half wavelength or less to eliminate the minor lobes. This requires sections that are less than a half wavelength long and if the base is to grounded the bottom section must be less that a quarter wavelength. (The bottom section can be modified from that requirement if we put a reactance to ground.)

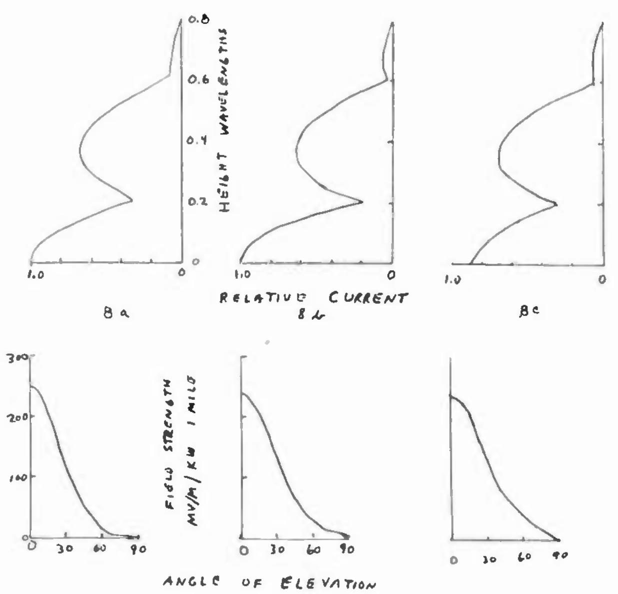

FIGURE 8

Consider a tower 0.8 wavelengths in height at the AM frequency. Figure 8 shows current distributions and vertical patterns for this tower. Figure 8a shows the assumed distribution for a first try with AMSTAK for the 0.8 wavelength antenna. The 6/4/1 distribution for the top half of the 5 element array of Figure 2 was chosen for the three section antenna. Its vertical pattern as calculated by AMSTAK is shown below the current distribution.

Figure 8b is the current distribution and vertical pattern as calculated by MININEC3 using the same drive and loading points. It took three choices of sectionalizing reactance to get this close to the simple AMSTAK analysis.

Even with further attempts at choosing the sectionalizing point and the value of the reactance we don't get as close to the objective of no minor lobes and very low high angle radiation as it seems that we should.

Why? A series of calculations with MININEC3 provides data that show lack of independent control of the amplitude and phase of the current loop for sections one and two. We are unable to obtain the desired combination of spacing, amplitude and phase for the two most important sections of current moment.

Lets take another look at Figure 2. In the array example we see that there are three coordinates necessary to calculate the pattern: spacing, amplitude and phase. In order to match that pattern or any pattern we need three variables which control the sum of the radiated fields from the array. We are using only two, reactance and its position on the tower.

We need another parameter to improve the situation. We can move the drive point or add a reactance between the antenna and ground which moves the location of the current loop in the lower section of the tower. The moving of the drive point as a tuning parameter is difficult and it would be a poor choice. therefore for optimum performance we must add a reactance between the base and ground.

If our goal is not maximum distance to the fading zone as desired for a clear channel station we can leave the base grounded and still make a very acceptable antenna.

Figure 8c shows the current distribution for the same antenna when base fed and with loading reactances at the former feed point as well as the former loading point. This is also a viable pattern but we have lost he advantage of a grounded base for the FM and other uses.

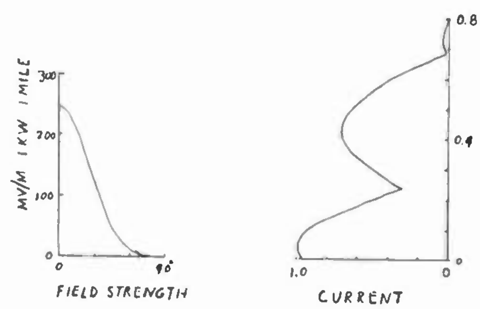

FIGURE 9

Figure 9 shows the current distribution and vertical pattern for the optimum use of the tower height. This was derived from repeated calculations to reach the current amplitude and phases which give the maximum suppression of the high angle radiation. The control of current distribution this way results in an antenna whose antifading characteristics are better than the typical Franklin antenna.

C. COMPROMISE VERTICAL PATTERNS

In the foregoing section we indicated that some arbitrary choices (grounding the base and fixing the sectionalizing point) could be made while still obtaining very good vertical patterns. These choices can be made since the in pattern control only the minima require precision whereas the major lobes are much less demanding of detail.

From the discussion of the effects of loading reactance we see that large capacitive reactances keep the feed through currents low and essentially in phase with the currents below. As we have seen short sections with tapered in phase currents lead to patterns with small or no minor lobes. Inductive loading causes a phase reversal on the tower and is more difficult to control or accommodate.

The current distribution shown in Figure 4g suggests a quarter wavelength antenna and the vertical pattern confirms this. Further the small current above the sectionalizing point can not significantly effect the major lobe. It is only effective at angles of low radiation.

A base fed antenna with appropriate sectionalizing and reactances would then appear to be the' way to approximate a specific current distribution. As a first approximation the bottom section should be slightly shorter that the target antenna and large capacitive reactances should be used for sectionalizing.

Lets then take our same 0.8 wavelength tower and try to match a quarter wavelength antenna.

Figure 10 shows the current distribution and vertical pattern for several approximation to possible desired tower heights. Figure 10a is a model for a 90 degree tower. 10b for a 135 degree tower and 10c for a 190 degree tower. Inspection or calculation will show that these are good approximations for the specified towers and are readily obtainable.

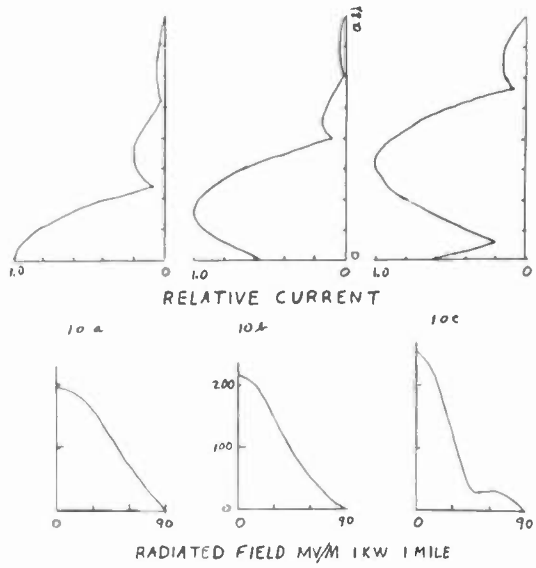

FIGURE 10

IV. SUMMARY AND CONCLUSIONS

We have shown the available tools to control current distribution and thereby obtain desired vertical radiation patterns. Optimum patterns have been identified along with means of generating them. The procedure for obtaining special current distributions and their radiation patterns have been demonstrated.

Methods of skirting towers for such sectionalizing as well as methods of detuning the towers when not in use are beyond the scope of this paper.

ADVANCED EDITING FEATURES FROM DIRECT ACCESS MEDIA

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0. Abstract

Direct access, disc-based media make possible efficient and reliable editing features. By combining this media with digital signal processing, new techniques can be introduced. This paper describes how different data formats, variable length cross-fades, reel-rocking and variable speed operation are carried out in a multi-channel environment, with complete freedom to handle each channel independently. The human interface for such complex operations has also been examined in great detail and an overview is given of an editing console which combines an interactive touch-screen with hard keys for frequently used operations.

1. Introduction

Direct access, disc based media offer tremendous potential for the storage, editing and processing of digital audio signals. However, there are a number of requirements to be satisfied before a disc-based system can be used as an ultimately flexible audio recorder and editor.

* Data throughput - the digital audio data rate must be sustained under all conditions, including editing which corresponds to increased numbers of seek movements in the drive.

* Data format - all operations should be applicable to audio wordlengths of 16, 20 or more bits.

* Signal processing - variable speed replay direct from disc should be provided so that familiar features such as 'reel rocking' and pitch adjustments are available.

* Edit processing - variable length crossfades should be available at edit points to minimise residual transient switching.

* Human interface - disc system operation should be presented to a user in a form that is found pleasant and productive to use.

This paper describes the implementation of the above features in SoundStation II, a disc-based system for professional audio use. Up to eight channels of audio can be transferred from a single drive though for high quality applications, more than one drive would be required. Each channel is controlled completely independently for editing, speed, gain, direction, etc. and the level of features and performance can be readily balanced making it suitable for a wide range of applications.

2. Direct Access Media

2.1 Disc vs. Tape

One of the earliest recording techniques, used in the early 1920s, was acoustic recording direct to disc. In this case the sound waves actuated a diaphragm to which the stylus was attached, mechanically engraving a sound track. A single spiral track on an easily manufactured disc yielded a simple an effective recording technique.

Until recently, the mechanically replayed disc has been the universal medium for commercial distribution of sound ptogrammes. There have even been notable attempts to extend the technique to multi-channel (greater than two) reproduction. However, for professional applications, longitudinally recorded tape has been used for its versatility in providing simultaneous access to many audio channels.

One of the key issues addressed in this paper is new methods for recording audio on disc which retain all the existing virtues of working with tape, but add new dimensions of operational flexibility by including the inherent features of discs.

These features can be exploited by the use of digital techniques whereby the data representing one or more audio channels can be distributed over the disc in a manner geared specifically to ease the replay process, freeing the technology from the need for a single, contiguous audio track. In contrast to tape, this gives:

* direct access to any part of a recording ("random" access).
* constant data reading speed independent of audio speed.
* no head/media contact, no degradation after heavy use.
* 'virtual' channels, independent of 'logical' channels.
* simple transport mechanics with minimal servicing needs and very high proven reliability.
* high speed, zero error rate copying.
* compatible read/write and archival media.
* no error concealments.

2.2 Factors related to efficient data transfer

The performance of the system depends critically on the dynamic characteristics of the disc, on its controller, on the method of data buffering and on the sampling rate and number of channels of audio to be handled.

A disc drive may have several data surfaces, each of which may have more than one head. Data is formatted on the surfaces into cylinders, tracks and sectors. The disc drive parameters relate to the time it takes for the heads to 'seek' a new sector and include the latency time (period of one revolution), track to track access time and full sweep time (maximum seek time for a new track).

The disc controller also plays a key role. Important features include its ability to transfer a track of corrected data at a single pass and to minimise the gap in data transfer at track boundaries by 'spiralling'.

Spiralling represents the degree to which data can be transferred without interruption and several modes can be identified. The most rapid mode occurs when the sector seek at a track boundary is sufficiently fast to permit continuous data transfer across the boundary without incurring a track latency delay. This is possible for multi-head drives when the heads can be electrically switched within cylinders.

However, under normal circumstances, when a seek to an adjacent cylinder is initiated, the access time caused by head movement (typically 2–10ms) will incur a latency. Skew sectoring is a technique by which the first sector in each cylinder is offset to anticipate this delay, and can be used to great advantage in disc systems with many tracks of low capacity.

The factors determining the net transfer rates are therefore numerous and complex. An appreciation of the problem may be gained by considering the following simplified example for a typical Winchester drive:

If, number of data surfaces	=	8
usable bytes/track	=	16384
worst case seek time	=	50ms
rotational latency (max)	=	16.67ms

An allocation unit may be defined as a convenient unit of storage which can be transferred without incurring latencies, i.e. in the above case 144KB., (KB = 1024b), (kB = 1000b).

Four channels of audio at 48kHz sampling rate and 20 bits/sample = 480KB/s.

For independent transfer of audio channels, a minimum transfer would then entail four allocation units = 576KB.

The transfer of 576KB takes approx:

36 revolutions at 16.67ms	=	600ms
+ 4 seeks (allow worst case)	=	200ms
+ 4 latencies (worst case)	=	68ms

Since the audio data can be delivered to the memory buffers in this total time of 868ms but represents a play time of 1200ms, the disc system can keep up with the real-time requirement. The excess in transfer rate provides time to deal with edits.

2.3 Relative performance of optical media

The use of optical media is extremely attractive because it permits interchange and is compact. With error specifications matching that of Winchester drives, it is safe to use, not only for audio data but also edit decision lists and other computer related files. Thus the audio and its control information may be easily passed from one system to another. Most WORM (Write Once, Read Many) optical drives that are currently available have low data transfer rates and significantly longer seek times than magnetic drives.

For audio applications, the data reading rate is probably the most important factor. High transfer rates provide the excess data for buffer memory which covers the seek periods when no data can be read.

TABLE 1. - TYPICAL COMPARATIVE SPECS FOR 5 1/4" MEDIA

	magnetic	CD-ROM	WORM	high perf WORM
data reading rate	10Mb/s	1.25Mb/s	2.5Mb/s	7.5Mb/s
latency	16.67ms(CAV)	110-300ms(CLV)	33ms(CAV)	25ms(CAV)
access time (av/wc)	23ms/50ms	250/800ms	100/300ms	85/195ms
capacity	to 760MB	to 680MB	to 400MB	to 750MB
number of tracks	1600/surface	20,000	15,000/side	16,000/side

CLV - Constant Linear Velocity
CAV - Constant Angular Velocity

Comparative specifications for currently available media are given in Table 1. The high performance WORM has specifications approaching those of the best small Winchesters. The reduced data reading rate and longer access times still provide a level of performance which can support full facilities for two-channel operation. Additionally, disc-to-disc copying at the maximum rate will support two-channel back up at four times real-time rates.

3. DATA FORMATTING

3.1 Wordlength conversion

Discs are invariably byte-organised devices. Audio is commonly 16 bits (i.e. 2 bytes) but there is a growing body of opinion that more bits are necessary for high quality, professional applications. At the other extreme, data compression techniques may be acceptable in fields such as video post-production. This indicates the need for a flexible approach to formatting the audio data prior to recording on disc.

The data buffers which serve to smooth the data flow from disc can be used as processing buffers. Data may be modified 'in-place' by providing access to a signal processor. By this means q, N-bit audio samples may be converted to p, 16-bit samples as the data is transferred. Each 16-bit sample is then stored as two sequential bytes on the disc. As will be shown later, the tight coupling of a high speed DSP and the processing buffer provide the mechanism for crossfades, gain adjustment and variable speed operations.

3.2 Channel independence

An important objective in the SoundStation II project was to maintain facilities on each audio channel independently. The most obvious reason for this is to permit 'slipping' of one track relative to another. However, other features are also useful when applied to individual channels:

* signal level.
* record or replay.
* forward or reverse.
* edit control parameters (e.g. crossfade length).
* data format (number of bit/sample, coding technique)exchangeable.
* speed.
* pitch, etc.

Each processing buffer is therefore provided with its own control structure which contains all the data relevant to the recording or replay of that channel.

3.3 Demand driven operation

Each channel delivers data only on demand. This is essential for variable speed operation and where data format changes are imposed. In order that the supervising computer can be aware of which data is actually being monitored at the loudspeakers, the processing buffer and associated DSP maintains independent time-line information for each channel. Thus, the host can compare the current 'now-line' with its own record/replay information and maintain the buffers accordingly.

4. SIGNAL PROCESSING

4.1 Crossfades and gain adjustment

There are three methods which can be considered for producing crossfades in disc-based systems:

1. Precalculate the crossfaded audio and store it on disc as an additional segment to be incorporated in the playback sequence. This is a poor approach because it eliminates the possibility of real-time manual control.

2. Double up on replay channels from the disc and

calculate the crossfade as the data is output. This is wasteful of resources when there are only a few crossfades to be made, and halves the number of channels that could be available to the user.

3. Initiate a temporary additional replay channel for the duration of the crossfade. This maximises the number of channels that can be guaranteed and uses the processing buffer which has already been shown to be desirable.

4.2 Variable speed operation

As has been well demonstrated by the use of 'sampling' electronic musical instruments, pitch changes can be readily achieved by replaying audio data at above or below the normal sampling rate. However, in the context of professional digital audio studio operations, it is essential that such facilities are provided with a -constant- audio sampling rate, i.e. the rate common to all digital audio equipment in that studio. This is then a problem to which a Sampling Frequency Convertor (SFC) must be applied.

The general SFC problem can be stated as the need to calculate output samples with the timing of the output clock from a knowledge of the input samples with the timing of the input clock. This can be achieved by notionally increasing the sampling rate by a sufficiently large factor so that selecting the nearest sample at the output incurs an error of less than 1/2LSB. The detailed techniques for achieving this have been described elsewhere and here we will concentrate on the specification for such conversion in an editing environment.

The lowpass digital filter used has the following specification:

passband edge (-3dB)	18.5kHz
stopband edge (-100dB)	36.0kHz
in-band ripple	<0.1dB

This is readily achieved with modest signal processing hardware in conjunction with with the processing buffer already mentioned. This specification represents a compromise between maintaining a flat passband while rejecting the many alias components so that they are inaudible. At low speeds this is particularly difficult because the audio signals are by definition at low frequencies and will have no significant masking effect over aliases which may occur at the most sensitive region of hearing at around 2 kHz.

Nevertheless in static performance tests, the above filter provides total distortion better than 0.1%, even at 1/1000 normal speed. Subjectively, the use of this filter may be acceptable in a number of applications where the minimal quality loss can be tolerated. Dynamic effects are more difficult to quantify, since residual alias components are subjectively more easily identified when they move. However, for normal programme (rather than high level test tones), there are no discernible additional distortions. Note that the filter design is such that the passband tracks the replay speed, and optimises the replay bandwidth, e.g. at half speed the passband is reduced to 9.25kHz.

4.3 Off-line non real-time capabilities

It is an unfortunate fact that algorithm developments always tend to require more processing then is currently available. This is particularly true in digital audio, where for example the performance achievable in artificial reverberation, and now sampling frequency conversion or pitch change, has always outstripped what can be commercially provided in hardware. Additionally, as special purpose DSP circuits are introduced, the programming skills become more specialised and devious! There have been courageous attempts to sidestep this problem by the writing of automated code generators. Such approaches will always to some extent depend on the DSP implementation.

An alternative approach, used in SoundStation II, is to safeguard the investment in algorithm development by keeping it at the high level language. This ensures early and successful implementation because of the enormous number of software tools and debug utilities that are available and a degree of future-proofness. Rather than make the time-consuming and expensive shift to an implementation on specialised hardware, this software can be made immediately available to run on standard, commercially available in quantity, high performance single board computer cards. (SBCs).

In SoundStation II, up to four 32 bit SBCs can operate in parallel using the Multibus II architecture. This architecture and operating system support the concurrent running of tasks on multiple processors with high speed (40MB/s) message passing between tasks. These processors have full access to the audio data stored on disc and permit even complex algorithms to be implemented at near real-time rates. The use of this technique to implement TimeWarp, a stereo signal pitch shifting software package will be described in a future paper.

5. THE DATA PATH, FROM DISC TO STUDIO EQUIPMENT

The high performance 5 1/4" drives that are suited to audio data applications transfer data at 10Mb/s over an Enhanced Small Disc Interface (ESDI) to the disc controller. In SoundStation II, the disc controller is compatible with the Small Computer System Interface (SCSI) which currently supports transfers at a maximum of 1.25Mb/s (i.e. the same net rate). The controller contains buffering for one track which means that the SCSI bus need not be busy while the drive is seeking. This arrangement can readily support up to 7 controllers, each of which has 4 drives. The use of SCSI is highly desirable because it insulates the rest of the system from the rapid changes which are occurring in storage technology. For example, SCSI II, currently under discussion, promises an upgrade to 16 bit data, 8Mb/s, with a Common Command Set (CCS) for all devices including WORM, tape and random devices.

In every SCSI system there must be at least one Initiator, the name given to the unit which controls the data transfers. SoundStation II includes a module which combines an Initiator with 2Mb of data buffering, a DSP and a supervisory microcomputer. Additional modules can be added as required for large systems, limited only by the backplane capacity. Each module can support up to eight audio channels which are output to a Multiplexed Audio Bus (MAB) which performs internal signal routing for up to 128 audio paths. Currently, both analog and digital I/O is supported by the following modules:

* 4 channel, 18 bit A/D convertor
* 4 channel, 18 bit D/A convertor
* 4 channel, AES/EBU input (with isolating transformers)
* 4 channel, AES/EBU output (with isolating transformers)

5.1 Three buffer flow

The Initiator module has the job of smoothing the data flow from disc so that real-time audio is presented to the MAB (which operates locked to the studio sampling rate). This is done in three levels of buffer management (fig ....).

1. Bulk timing correction (2MB)
2. Processing buffer (small cache memory)
3. Fine timing buffer.

Within the bulk memory, storage is dynamically allocated for each active channel. Areas are reserved for commands to control DSP activity and also to monitor the 'now-lines' so that buffers can be loaded as required. Audio data is maintained both ahead and behind the now-line so that instant reversal of direction can be achieved.

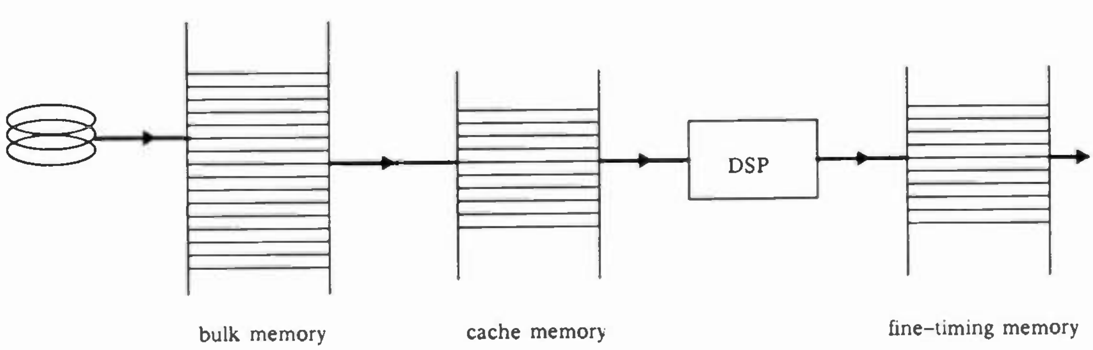
The processing buffer is based on very high speed memory which is accessed by the DSP chip. Crossfades are executed on the data in this buffer and digital filtering, for example for varispeed, uses this area as the filter data stores.

The fine-timing buffer serves to decouple the DSP operations from tight real time calculations. The output of the buffer is at precisely the studio sampling rate. The DSP can be viewed as both manager and processor for data between the processing buffer and the fine-timing buffer. For example, with data format changes, the net data rate out of the processing buffer can be different to the net data rate into the fine-timing buffer.

6. HUMAN INTERFACE

6.1 Data Organisation

Regardless of how well designed the system hardware is, considerable emphasis must similarly be placed on presenting to the system user, both vast amounts of



information and mechanisms for reviewing and changing this information in an efficient and, if possible, familiar manner.

In creating the basic unit of data which will represent a "unit of audio", consideration must be given to the fact that the data structures must contain data elements which are relevant to the host computer operating system, to the signal processing hardware and to the user. Furthermore, any complexity which is not of relevance to the user should be hidden.

In SoundStation-II, the basic units of audio are called Segments. The definitions for this and related structures are given in Table 2.

Organisation of segments should be natural to the context in which they are being used. For audio applications, we have chosen a "tree structure" utilising the concepts of audio Reels, Segments and Groups of Segments. This hierarchy allows data to be hidden by the user as it becomes less relevant, by simply giving a name to a group of consecutive Segments. This Group name then "replaces" the names of individual segments in any display unless the user chooses to examine the hidden detail, which remains readily accessible.

This mechanism also permits Segments, which deliberately appear simple to a user, to hide complex sequences of edited audio data as we shall see in the next Section.

Table 2. Definitions from the users point of view

Segment A piece of audio with both variable duration in time and width in tracks.
Group A named Segment comprising other consecutive segments.
Reel A virtual location in the system where Segments are grouped for efficient access and identification.
Set A user defined unnamed collection of Segments which will be operated upon by a subsequent command.

The attributes of a Segment are kept by the system in a Segment Information Block which typically contains the following information.

Table 3. Segment Information Block Contents

Full Name (Reel, Groups)
Playback Start Time Code
Duration
Comments
Number of Tracks

Additional Information - for Optional Display

Date/Time Of Recording
Session Name
Source Tape Id
Source Time Code Start
Source Time Code End
Sampling Frequency
Emphasis/Eq
Gain (fixed, variable=list)
Data Format
Date/Time Last Mod

System Information - Invisible to User

Pointer To Audio Data
Parent Segment Pointers
Display Attributes

6.2 Segment Origination and Attributes

6.2.1. Origination

Segments are initially created during recording, which will not be described here in detail. Suffice it to say that audio data may be written to the disc using the techniques previously described, and given a name either before or after the recording. Furthermore, original recorded Segments initially have a system attribute indicating they are "parents" of any subsequent Segments which refer to the same audio data. During the recording process, the user may press a key to "rough cut" the recording on the fly, which electronically creates a new parent Segment starting at the time of the key press, with Segment names automatically derived from the original name.

6.2.2 Segment attributes

Segments have two general types of attributes, those which the user may display and manipulate such as Segment Name or Length, and those which are fixed by the system, such as pointers to the position of data on the disc and original sampling frequency. System-related Segment attributes may be kept hidden from the user. Attributes which the user can manipulate can be made visible or hidden at the user's request. Our concern in the this paper is with displaying Segments and viewing and changing certain user manipulated attributes.

6.3 Segment Manipulation

By ensuring Segments have the appropriate system related attributes, they become data objects which may be manipulated by efficient commands, whose syntax can be constructed to follow natural language patterns as closely as possible. Done properly, this can encourage intuitive exploration of the operation of the system.

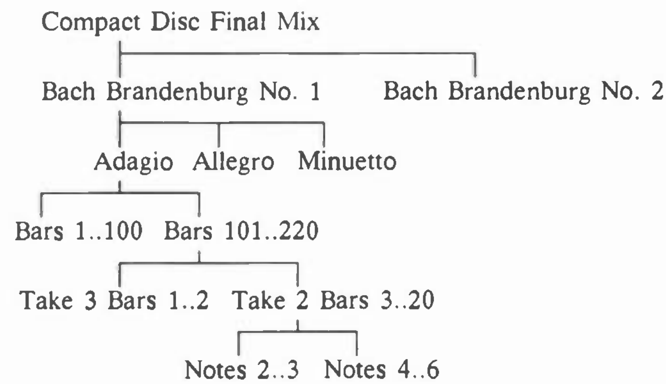
A wide repertoire of commands may be implemented to perform the creation, restructuring, or movement of audio sequences. Typical commands which can be applied equally to Segments, Groups of Segments or even Reels are: {NAME}, {COPY}, {MOVE}, {EXCHANGE}, {FIND}, or {SORT}. (Note that the "{ }" used to enclose commands indicates that the key is a "touch key" on the control console. Commands enclosed with "[]" are implemented on a hard console key and "< >" signifies any object of the type described by the enclosed name, e.g <Segments>.

6.3.1 Name syntax

Perhaps the most useful command for maintaining an organised and efficiently accessed segment data base is the {NAME} command. The syntax for this command is as follows:

This command applies a user generated name (e.g. "Take7 Bar 5-20") to a single segment or to a Group of consecutive Segments, the latter use being highly recommended to simultaneously identify the Group's contents and hide unwanted detail. Once a Group is named, if the displayed Group name is selected by touch, the command {REVEAL} will redisplay the detailed contents of that Group. To restore the Group name to the display and replace the detail, a complementary command, {HIDE} is available. Note that these commands only affect the information displayed and the sequence structure remains constant.

This set of commands coupled with a user defined data organisation, allows the user to move from displaying Segments representing several hours of audio data, to Segments displaying fractions of seconds, with a very



small number of keystrokes. The example below presents an imaginary but representative example whereby only five consecutive presses of the {REVEAL} key could take a user from an hour programme down to a few notes of interest in the recording.

6.3.2 Copy Syntax

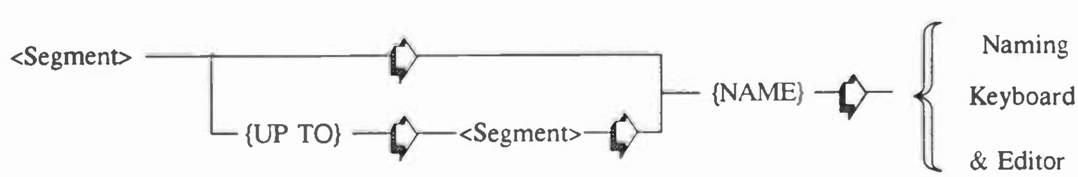
Using the {COPY} command, a Segment, a set of optionally non-consecutively displayed Segments (using the {AND} and {UP TO} commands) or Segment Group is first selected (by touch). After this, the {COPY} key is pressed followed by an optional number (of copies to make) followed by touching the screen destination position. Thus, with as few as three key presses, a Segment can be selected and copied.

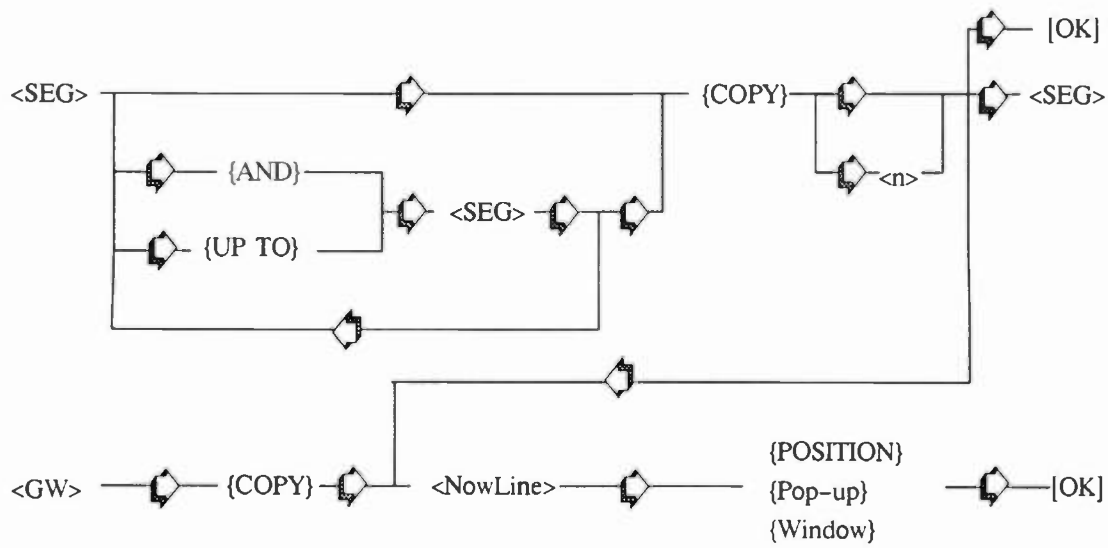
Parameters and descriptors:

n Specifies the number of copies (>= 1).
<GW> A Group Window, which contains a single Segment name which is a Group of Segments.
<Now-Line> A Current Position Indicator in the PLAYBACK SEQUENCE.

6.4 Display of Segments

The task in editing audio is to build, from an unstructured collection of Segments, a sequence of Segments with audio events organised to occur in the desired channels at the desired times. One method of supporting this activity on a display screen, whereby audio data objects are





efficiently manipulated, is to create two areas on the display. In one of these areas, the Segment List, the relative timing and channel information of the original information is hidden and the display is organised as follows.

6.4.1 Segment Lists and Group Windows

At the top of the display screen, lists of consecutive Segments are displayed vertically with the appropriate Group or Reel Name containing the corresponding Segment list indicated separately in "Group Windows" at the left side of the display. By having several lists in view (say, three to five) the user can examine the audio data base in a completely flexible, and yet controlled and efficient manner. For example, different "ends" of the same reels can be viewed in two lists simultaneously at any level of detail.

Perhaps the most important feature to add to such a display used for audio editing is to support immediate playback (with only one keypress) of any displayed item. This strategy ensures no time is lost when the user wishes to audition the content of a Segment. Thereby, different levels of detail of the same segment may be displayed in different lists to permit say, alternatively playing an edited fragment for detailed inspection and then immediately playing the segment in context of a larger group of Segments for continuity assessment.

In displaying Segments and Group names, icons may be used to indicate effectively whether a Segment in the list is itself a Group or Reel, but this has not been done in the above Figure for clarity.

6.4.2 Playback Sequence

A lower area of the screen is used for creating a multichannel output sequence with Segments displayed according to their width in tracks and relative durations. In effect, this display is analagous to a final multitrack audio tape with names and markers indicating where Segments start and stop. In the figure above, time is shown increasing from left to right, corresponding to the direction of increasing time in musical scores and text for dialogue, making labelling readable and logical and the notional "playback head" is represented by a vertical line known as the "NowLine".

The major differences between the representation of digital audio shown on a display and audio on tape is that each Segment may be individually manipulated in time, gain, playback speed etc. and the implied "splices" between Segments may be individually manipulated. As the audio is played, the Segments scroll past the NowLine at a rate proportional to the time scale on the Playback Sequence display which is made user adjustable. The time code (which may be elapsed time from the Segment start or a System time) is displayed with the NowLine. This combination of display and control flexiblity provides complete control over all the editable parameters of audio.

6.5 Segment Selection

In the specific task of audio editing, the traditional approach is to manually move audio elements on tape into the desired sequence either physically or electronically. Apart from the joining process required, the manual selection of elements, if sensibly organised for rapid access, can be extremely efficient.

SoundStation-II supports touch-based selection of Segments on a Touch Screen, as this appears to be the most natural, direct and efficient selection mechanism, in contrast to moving cursors through long unorganised lists. Information on the full name, duration, time code start, etc. of the selected Segment may also be displayed in a dedicated area of the display screen, with options for accessing further information in the Segment Information Block. Additional commands for accessing specific Reels, Finding and Sorting Segments are also available to the user.

6.6 Construction of Playback Sequences

Although considerable arranging of Segments into a desired running order can be acheived in the Segment Lists, precise editing work is acheived by moving Segments from the Lists to the desired channels and accurate time scale of the Playback Sequence. The user can select a Segment, Group or Reel by touch and {MOVE} or {COPY} it to any point in the Playback sequence.

Once a Segment is in this display area, the lead-ins and lead-outs of contiguous Segments create "virtual" splices which are completely adjustable. By the mechanism of "zooming" the displayed Time Scale to represent smaller and smaller time increments, the user can make micro-adjustments to either retract or extend the positions of the lead-in or lead-out points at each virtual splice. Similarly, time slipping of Segments on a track by track basis is also a straightforward operation. Such modifications would be inconceivable using conventional methods.

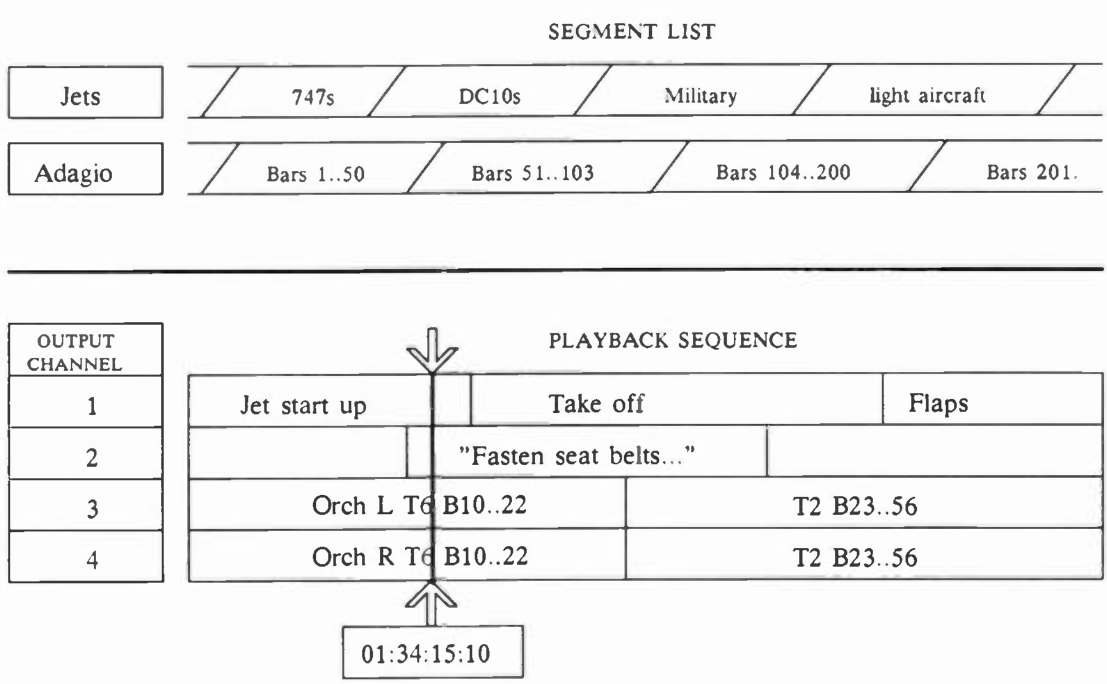
Further adjustments to parameters governing each Segment, such as gain, playback speed, duration and pitch may independently be assigned to single Segments or Groups as desired.

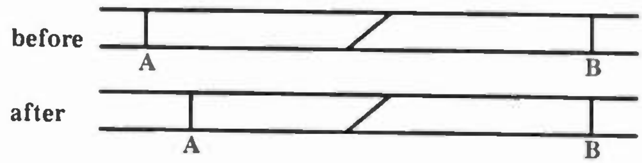
6.7 Sliding and Trimming

There are some operations available to the users of a disc based audio system which have no real equivalent in traditional audio editing methods. Sliding and trimming are two examples of making adjustments to an edit -after- the initial edit decision has been made.

6.7.1 Trimming

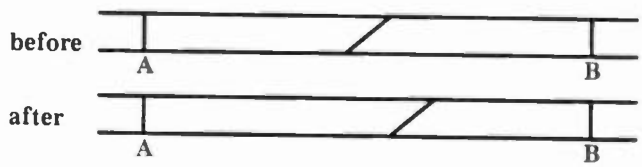
Note that the start of the lead-out segments remains fixed and the duration of the lead-in segment is varied. This is called "Trim-in". In the alternative case, the duration of the lead-in segment remains constant while the start of the lead-out segment is varied. This is called "Trim-out". The overall duration of the segment pair varies during trimming.





6.7.2 Sliding

In this case the splice 'slides' across the two segments by, in this example, extending the duration of the lead-in segment with a corresponding shortening of the lead-out segment. The overall duration of the segment pair remains constant during sliding.



7. ACKNOWLEDGEMENTS

The authors wish to acknowledge Kevin Dauphinee, Mike Parker, Paul Corcoran, Ian Wigmore and Brian Moscrop for their contributions to the SoundStation II project.

APPLICATIONS OF THE TAPELESS STUDIO IN BROADCAST

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If one were to consider the evolution of the word processor in the office environment, one could say that a similar evolution is occurring in the field of audio production. Just as the benefits of the word processor eventually led to its universal acceptance, the "tapeless" studio is well on its way to becoming a standard tool in any high-end audio facility.

The purpose of this paper is to review recent developments of the tapeless studio, and to outline some of the immediate applications in the field of broadcast.

THE TAPELESS STUDIO

The major tasks in audio production can be broken down roughly into three major areas:

1) Sound Generation
2) Archival/Retrieval
3) Signal Processing

Traditionally, sound generation has been achieved either by playing sounds live or by triggering cart machines, compact disc players, and analog tape machines. The recording process, or the archival and retrieval of sounds, has been accomplished largely on multitrack analog tape, which is a linear, serial access medium. The shortcomings of tape are evident mostly in the limitations of access and control, requiring retakes or laborious editing procedures in order to make changes once the recording is finished. Signal processing and mixing has been achieved largely through analog consoles, which although highly evolved, have not as a whole addressed the concept of being able to edit program material and have the mix follow automatically.

The "tapeless studio" is an environment in which all of the traditional functions of audio production--producing, recording, and processing sound--can be performed digitally and in an integrated random access system. "Random access" is the cornerstone of the tapeless studio and refers to the ability of advanced computer systems to instantly play back, locate, process, and edit any sound without having to rewind tape. The advantages of this technology are many but can be summarized as follows:

1) Speed

Due to the instant access to large sound databases, the tapeless studio suffers no down time for sound effects retrieval or tape rewind time. Sounds can be previewed and edited a dozen times in the time it would take to retrieve even one sound from a cart or tape library.

2) Quality

The tapeless studio also enjoys the sonic benefits of high quality digital audio, with no sound degradation due the bouncing of tracks or the accumulation of analog tape noise. In addition, the tapeless studio does not require elaborate error-correction schemes to compensate for media errors, as is necessary in digital tape systems.

3) Flexibility

Most importantly, the tapeless studio is a "non-destructive" editing environment. That is, all edit, production, and mixing decisions can be reviewed and altered any number of times without destroying the original audio material and without having to make safety copies. Other areas of flexibility include the ability to edit sound on a "micro" level, down to as fine as 1/100,000 of a second, with no damage to the audio surrounding the edit point.

New England Digital Synclavier and Direct-to-Disk Systems

New England Digital has currently provided two of the three functions of the Tapeless Studio, namely, the sound generation, and the archival and retrieval, or recording function. The sound generation function is provided by the Synclavier and the archival/retrieval, or recording function is provided by the Direct-to-Disk Multitrack Recorder. It is significant to realize that total capacity, that is, number of tracks, expandability, recording time, and the total number of audio channels is an important factor in assessing the tapeless studio. For a system to be truly viable, it should offer no less capacity than existing systems.

The Synclavier

The Synclavier provides sound generation through the use of 16-bit, 100 kHz digital sampling, FM synthesis, additive synthesis, and resynthesis. The Synclavier also provides up to 96 independent digital audio channels, or sound generation channels. These channels, or "voices", are controlled through software edit decision lists which are subdivided into 200 separate tracks of information. These edit decision lists can be edited on the computer terminal and contain information regarding volume, stereo pan, SMPTE start and end times, and other real-time effects information. The tracks can record either musical performances, sound effects, or dialogue.

Optical Disk Storage

Musical instruments and sound effects material for the Synclavier are stored on a 2 gigabyte Optical WORM drive. The optical media is a write-once, read-many removable format and can contain 5 and 1/2 hours of sound recorded at a 50 kHz sampling rate, or 20,000 one-second sound effects. The optical disk can locate any sound on the disk in less than a second, and it can load the sound into the system in 1/2 of real-time. Thus, a ten second sound would take less than a second to find and 5 seconds to load.

200-Track Sequencer/Recorder

Both the Synclavier and the Direct-to-Disk systems utilize a 200-track sequencer/recorder. A sequencer is a software-based recording system which does not record audio. Instead, the sequencer records cue information fed to it either by a piano-like keyboard, a sound effects triggering interface, or from other software interfaces on the computer terminal. The cue events then trigger audio segments which are held in either solid-state RAM, in the case of the Synclavier, or winchester disks in the case of the Direct-to-Disk system.

These cued events can be edited on the computer terminal or from the remote control unit in case of the Direct-to-Disk system. The cues can be moved in time, they can have volume or stereo pan information attached to them, or they can be exchanged for other cues while retaining the same trigger times. In effect, the audio can be edited like a paragraph on a word processor and moved, copied, or deleted.

The audio events are organized into "tracks" for convenience. The Synclavier and Direct-to-Disk share a common sequencer and can have up to 200 tracks of information. Since the sequencer format between the two systems is common, projects can move about between different systems and be totally compatible. For example, a voiceover session could be recorded and edited on the Direct-to-Disk, and then sent to a Synclavier in another room to add music and sound effects, all within the same sequencer and editing software.

Multichannel Outputs

The Synclavier can also support up to 64 multichannel outputs, which can be routed directly into any standard mixing console. Any of the 200 tracks of edit decision list information can be assigned in the software to any multichannel output. That is, the system allocates the voices as needed to the different sequencer tracks, but then routes the audio to whatever multichannel output the user has specified. The voices that play the events on any particular sequencer track need not be the same voice; this allows the voices to be used as needed depending on the load on different tracks at different times.

The system also allows software controlled submixing and summing of tracks, which can be sent to the multichannel outputs sub-mixed in mono, stereo, or any other cofiguration of outputs. In addition, dynamic volume fades, dynamic stereo panning, and overall track balancing can also be recorded and edited in the system.

The Direct-to-Disk Multitrack Recording System

The Direct-to-Disk Multitrack Recording System is a hard-disk based mulitrack recording device that records audio digitally and stores the data onto a system of winchester disks. The winchester disk format allows instant access to any portion of the audio program without having to wait for any rewind time associated with tape-based recording. In addition, the hard-disk system is free of data errors which are inherent in tape-based digital recording, which improves the overall audio quality of the program material.

The system is available in 4, 8, 12, 16, or 32 track configurations and has a recording capacity of 26 to 125 minutes per track and can run at any sampling rate between 0 and 100 kHz. Digital-to-Digital transfer to either the AES/EBU, Sony, or Mitsubishi digital formats is available.

The Stand-Alone Direct-to-Disk

In addition to configurations that are integrated with the Synclavier, the Direct-to-Disk is available as a stand-alone unit. That is, it has its own controlling computer, system storage devices, and specialized control interfaces including a terminal and a remote control unit for editing, synchronization, and track status control.

The "Post-Pro" 8 Track Stand-Alone

A special configuration for the post-production and broadcast markets has been designed called the "Post-Pro." This unit is self-contained ,complete with digital real-time backup, MIDI and SMPTE interfaces, and a complete audio editing software package. The unit will interface to an Optical Disk to allow direct digtal transfer of sound effects, program I.D.'s, and jingles within the system.

APPLICATIONS

The applications base for the Synclavier and Direct-to-Disk systems is extremely broad. In the following pages, I will outline some applications in television, radio, and video post-production.

Video Post-Production

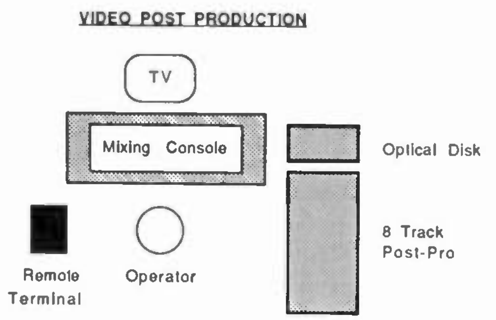
Video post-production is an ideal application for the tapeless studio (see diagram #1). The hard-disk system allows ample recording time (up to several hours) of audio material. Editing is achieved easily and does not require any physical cutting of tape, and does not damage the audio in any way. Sound effects, music layback material, and dialogue are stored as "cues" in a directory which can be called upon instantly to recall and audition any sound element. The cues are then assembled by "dragging" them with a mouse device onto an electronic cue sheet . SMPTE time code start and end times for the cues can then be entered onto the cue sheet.

Equipment and time savings are accrued here by not having to have multiple tape machines which are either manually triggered or controlled by an external synchronizer. Retrieval time for sound effects is virtually eliminated.

If large numbers of sound effects or music composition is required, then the Synclavier is appropriate due to its extremely large (up to 96) number of discrete audio channels. A maximally configured Synclavier is capable of playing back 96 channels of sound effects at one time, fed from an edit decision list of 200 independent tracks.

Television

Television applications range from full post-production audio for network television programs and features, to high-speed news editing and live sound effects performance for game shows. The tapeless studio has a special significance for taped television production. Unlike post-production studios which rely on multitrack tape machines and synchronizers to edit sound to video tape, the Synclavier and Direct-to-Disk systems are able to quickly and easily slide tracks and edit events within tracks. Since everything within the tapeless studio is in a random access medium, it is never necessary to record from one tape machine to another merely to slide a track. In addition, the tapeless studio is quite capable of quickly and inexpensively creating music and effects tracks for television features that have or would like to have foreign audiences. Essentially, the tapeless studio brings the flexiblity of film sound editing to taped television soundtracks, without the expense or sound degradation of using magnetic film stock.



- o Optical Disk: instant access to sound effects, backgrounds, backgrounds, station I.D.
- o Post-Pro: on-line storage of 3.5 hours audio, full editing and sync lock of events to picture via electronic cue sheet, post sync of dialogue and music elements. Non-destructive edits.
- o Project back-up to high speed data cartridge tapes

(Diagram #1)

Sound Effects

Sound effects are stored on the Optical disk and can be fed to either the Direct-to-Disk multitrack system or the Synclavier. For heavy sound effects applications such as animation or action shows, the Synclavier is more powerful due to the large number of channels available and the ability to manipulate the sound extensively. In addition, up to 76 sounds can be accessed from the keyboard at one time and "played" from the keyboard. These events are recorded into the Synclavier's sequencer as edit decision lists which can be further edited. The timing of the events can be edited by changing the SMPTE time code numbers automatically assigned to the events.

Thus, for game shows and other formats where sound effects need to be played "live" to tape, the Synclavier can easily handle an entire show with one operator. Or, when sound effects need to be post-synced to picture, the Synclavier allows instant recall of effects which can be assigned SMPTE start and end times from the terminal.

Laughs and Applause

Audience reaction sounds can be easily stored on the Optical Disk and recalled to the keyboard. These sounds can be stereo and can be played and mixed in real-time using the keyboard's pressure sensitivity to control volume. In addition, a much more varied selection of laughs can be stored on the Optical disk than is currently available in other devices without evening beginning to tap the capacity of the disk. Thus, a "showpack" of sound effects and laughs can easily be assembled on the system to be accessed in one post session without having to have any additional specialized equipment.

High-Speed News Editing

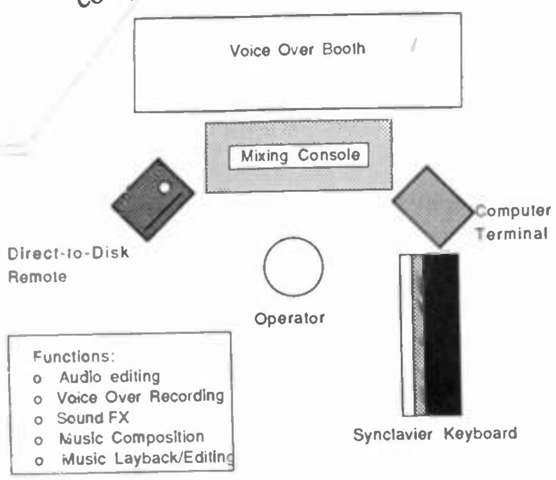
Where time is especially critical, the speed of the Direct-to-Disk system can be extremely beneficial. For example, using the Direct-to-Disk and Optical systems, sound effects and voiceovers to news stories can be accessed instantly. Voiceovers can be broken down into cues as the recording is taking place and then synced to picture simply by adjusting the timing of the cues on the electronic cue sheet. Final mixing need not be slowed by excessive retakes, editing, or by waiting for tape machines to rewind.

Dialogue Editing and ADR

Dialogue editing and Automatic Dialogue Replacement is also extremely efficient on the Direct-to-Disk system. During ADR recording, for example, any number of takes can be recorded onto the system and the system will automatically record the sync information and store it with each take. The takes are automatically labeled and stored in a cue directory, at which point the editor or director can simply "click " on the take that he wishes to hear with picture. Sections from individual takes can be quickly edited together and stored as new cues which can then be instantly heard and compared, all without ever physically cutting the original recordings in any way.

Dialogue editing tasks such as ...oothing cuts from
one scene to another are faci...d by having all of
the "dailies" (sync sound ...segments) on the
system at one time. ...l dailies can be
accessed instantly ...ng tasks such as
copying ambi... ...kgrounds from any of the
audio segr... ...done extemely quickly and
compl... ...digital domain.

Radio Jingle Workstation



(Diagram #2)

Radio

One of the primary advantages of the tapeless studio in radio is the instant access to the thousands of sound effects, station i.d.'s, jingles, and voice clips. The reliance on cart machines, turntables, and tape machines as sound sources is limited by the number of such devices an operator can be reasonably expected to control. With the Synclavier, for example, up to 76 sounds can be accessed on a single keyboard settting. The optical disk allows an operator access to thousands of sounds or jingles that he can select in less time than it takes to pick up a cart. Or, on the Direct-to-Disk system, music, interview material, and pre-assembled news material can be automatically reviewed, edited, and programmed in advance by a program director without having to rely on the operator at all to execute the format.

More impact on the air: Due to the efficiencies in speed and control, radio personnel can accomplish more impressive sounding program material in less time. For example, it would not be unreasonable to expect a single production room to be able to produce more and better sounding material than three or four conventionally equipped rooms. In addition, manipulating sounds after they are selected is extremely fast on the Synclavier, allowing the operator not only a much larger selection of sounds, but more control over the sounds.

News editing

Editing of news stories, where time is critical, can be accomplished easily on the Direct-to-Disk system and extremely quickly. The ability to instantly access any portion of the audio for review and editing is extremely important here. In addition, background sounds and library music that may be needed can be accessed quickly from the Optical disk system and instantly placed and mixed on other tracks.

Jingle Production

The tapeless studio is a complete production environment for jingle production (see diagram #2). First, all of the music for the spot can be composed on the Synclavier. Second, all of the sound effects can be recalled from the Optical Disk and placed either live or through the computer terminal. Third, all of the voiceover recording can be recorded on the Direct-to-Disk system, edited, and assembled to the music and effects tracks completely within the sequencer structure of the system. Fourth, if the spot requires library music, the music can be recorded and edited on the Direct-to-Disk system and placed within the spot.

Voiceovers

Allocate Mode: One of the most important features of the Direct-to-Disk system for voiceover recording is its ability to record any number of takes on the same track and still maintain any sync time desired. Analogue tape requires that takes be recorded on seperate tracks if sync is to be maintained. Because the Direct-to-Disk system is random access, the various takes can be stacked out on the track in any position but played back at any time. In addition, the takes are automatically labeled and numbered during recording and placed in a cue directory. Thus, voiceover takes can be reviewed by number instantly, simply by viewing the directory and clicking the mouse on the desired take.

Block Mode: "Block Mode" refers to a unique ability of the Direct-to-Disk system to establish edit points in an audio track in real-time. This feat is accomplished by playing the audio back and clicking the mouse at the desired edit points. The system will then number the edits and list them in a cue directory for review. This mode is extremely useful when editing voiceovers, or breaking-up repeated takes of a line for subsequent review and editing.

Automated Playlists

Due to the capacity of the Direct-to-Disk system to play back many hours of audio information in a pre-described event list, the step into automated radio station systems is a natural one. In a multiple system facility, program material can be created on the Synclavier and Direct-to-Disk systems, stored as files, and then sent to a master playback system. Radio management can then formulate a program list simply by entering the program decisions into the Direct-to-Disk computer.

Networking

It is in the future of the ta[illegible] into a complete system net[illegible] storage systems, backup, an[illegible]studio to evolve Individual workstations will be [illegible] shared mass today, but will be able to communi[illegible]ftware. effectively with other workstations an[illegible] master controller system that will coordi[illegible]are entire facility. One of the key factors in ho[illegible] tapeless studio will be and should be judged is how modular the system is in its approach, and also in how well integrated these modules are to each other.

CONCLUSION

Although it is impossible to detail these applications thoroughly in this paper, the hardware and software tools are in place today to accomplish these tasks within the tapeless studio, with the benefits of speed, random access, and non-destructive editing. These capabilities, although extraordinary, are no longer theoretical and are in fact fundamental techniques in audio production that will be standard procedure in the very near future.

THE DIGITAL AUDIO CARTRIDGE DISK RECORDER, REPRODUCER AND EDITOR FOR BROADCAST USE

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Abstract

Today's broadcast audio tape cartridge machines and reel to reel decks have advanced significantly from their humble beginnings over 40 years ago. They are generally reliable on-air and offer straightforward razorblade and splicing tape editing. However, even the most advanced analog and digital tape machines are inherently prone to high levels of wear, jamming, and require frequent maintenance to provide peak performance. Manual audio editing with a razorblade requires considerable experience, dexterity and time. Magnetic tape media itself wears out in less than 2000 plays and loses high audio fidelity long before that.

Cartridge magnetic disk digital audio recorders, reproducers and electronic editors achieve Compact Disc levels of audio fidelity for thousands of plays per disk. In the one year that cartridge disk equipment has been commercially available, users have reported that the system is reliable, requires little maintenance and minimal training for electronic editing proficiency. Media and equipment costs are competitive with conventional tape equipment.

This paper describes the technology and the operational characteristics of the CompuSonics DSP 1500e digital cartridge disk machine in audio recording, editing and playback modes. Comparative test data for the cart disk machine, a high quality NAB tape cart machine and a digital audio tape (DAT) deck is presented.

Background

Beginning in 1982, an R & D effort to develop a microcomputer replacement for audio tape recorders and hand-editing was initiated by the author.[1] In 1984, the first commercial result of this work, a digital audio workstation based on magnetic hard disk drives, was shipped to customers.[2] A number of other manufacturers subsequently introduced similar equipment, and the digital audio workstation is now a recognized necessity for high quality-high speed audio production for film and TV. Total worldwide installations of this genre of computer are estimated to be approaching 200 units. However, radio broadcast installations of hard disk-based digital audio workstations are few and far between due to several factors: the high cost of these systems which typically start at about $30,000 and frequently exceed $100,000, and the "fixed" nature of the storage medium which does not allow removal or easy transfer of audio.

The cost barrier was broken in 1986 by the first desktop audio computer, the CompuSonics DSP 1000, [3] which cost less than $7,000. This machine has limited applications in broadcasting due to its storage medium, the removable optical disk cartridge which is write-once and non-erasable. Prototypes of floppy disk-based broadcast recorder/reproducers [4] had previously been demonstrated at the NAB convention in 1985 by both EMT of Germany and CompuSonics. These machines featured low-cost erasable media and somewhat complex operation.

Field tests of CompuSonics' prototype beginning in 1985 proved that a more rugged recording medium and simpler operation were needed for the broadcast environment. These and the other specific demands made on tape cartridge machines and reel to reel decks by broadcast production and on-air operations were directly addressed by the author and his colleagues during the 3 year development period of the audio cart disk system. In April, 1987 at the NAB convention in Dallas the production versions of the CompuSonics DSP 1500 cart disk recorder and DSP 1200 reproducer were introduced, and shipped to radio stations later the same month.

Since April, 1987 DSP 1500s and 1200s have been utilized in radio broadcasting both on the air and in production studios. Units are now also in service in foreign countries including England, France, Italy, Sweden, and Germany. European production of DSP 1500s and 1200s has recently begun to service the European Economic Community. The manufacturer in England, AVM Ferrograph, Ltd., produces the machines under license from CompuSonics. Their Model 9500 corresponds to the DSP 1500, and their Model 9200 corresponds to the DSP 1200.

The Hardware

Chassis

Both the recorder/reproducer and the playback decks are 19 inch rack-mount units 5 1/4" tall and 17 inches deep weighing 28 pounds. (Fig. 1) The rack-mount expansion module with 4 disk drives is 4 rack units tall and weighs 40 pounds. For editing functions a personal computer or serial data terminal is required. These range in size from an IBM PC down to hand-held units.

Inside the chassis are 2 subassemblies: the disk drive & power tray, and the single board audio computer (SBAC), (Fig. 2). The drive & power tray contains a linear power supply for the analog portion of the SBAC, a switching power supply for the digital portion of the SBAC, the magnetic cartridge disk drive, disk drive controller circuit board and wiring harness.

The SBAC is an 11 million instructions per second multi-processor computer designed from the ground up for high reliability audio data processing. With the exception of the microprocessors and the analog converters, virtually the entire board is populated with low power consumption CMOS process integrated circuits. Every component is an off-the-shelf part. For purposes of discussion in this paper, the SBAC will be considered in 4 distinct functional parts: the converter section, the digital signal processing (DSP) section, the central processing unit with its memory (CPU), and input-output (I/O) section.

Converter Section

The converter section is composed of active 13 pole elliptical filters, 16 bit linear analog to digital and digital to analog converters, and a low-noise actively balanced amplifier circuit for +4 dB level signals. The stereo audio signal path is maintained as two separate monophonic channels, each with their own converters, filters and amplifiers, locked to the same sample clock. The effective dynamic range and signal to noise ratio is about 90 dB with almost zero third harmonic distortion, wow, and flutter. Total harmonic distortion plus noise (THD+N) is less than 0.03%. Associated with the converter section is the VU metering circuit which displays average signal levels for both input and output. Between the converter section and the DSP section is a first in-first out (FIFO) buffer that stores up to 512 digital audio samples so they can be transmitted in bursts by the DSP section.

DSP Section

The DSP section of the audio computer is based on the Texas Instruments TMS32010 signal processing chip. This special purpose microprocessor is clocked at 20 megaHertz, features 200 nanosecond instruction cycle time, on-chip 16 bit multiplier, two 32 bit accumulators, and 144 words of internal data memory. The signal processing section of the SBAC is actually composed of two TMS320-based subsections; one for each digital audio channel.

Each DSP subsection contains a TMS320, 8 kilobytes of high speed static random access memory (SRAM), a port to the CPU section, and both data and control ports to the analog section of the audio computer. The SRAM can be addressed by either the TMS320s or the CPU itself.

Audio data is moved by the TMS320s in 8 millisecond blocks of audio samples to and from the analog section via the FIFO buffers. Processed audio blocks are "stolen" from the static RAM by the CPU while the DSP chips are "asleep" in the reset state. This means of transferring data from one microprocessor to another is made possible by mapping the addresses of a particular range of RAM into the memory space of both the DSP chips and the CPU.

CPU Section

The CPU section of the SBAC is based on a Motorola MC68000 microprocessor clocked at 8 megaHertz, 512 kilobytes of dynamic RAM running without wait-states, 128 kilobytes of EPROM, and a 4 channel direct memory access controller. The CPU supports the EPROM-based control program of the audio computer which is a proprietary real-time disk operating and editing system (RTDOES, also called "the opsys", which is discussed below under "Software"). The CPU receives commands from the user and manages the internal operations of the cart disk deck to achieve the user's desired results.

The data path starts with audio entering the A to Ds of the converter section, then moving through the DSP section where the CPU accesses the data and moves it to its own RAM. The CPU then transfers the data to the disk drive controller which passes the data to the drive. For playback, the path is the same but reversed with D to A converters at the output. The half-megabyte of RAM is used by the CPU to buffer audio data while the disk drive seeks and acquires the data. This is especially critical during on-line editing when the audio data from two different takes stored far apart on the cart disk is being spliced together on the fly to playback a finished edited production.

I/O Section

The Input/Output section is closely coupled to the CPU section of the audio computer. I/O of data, status information and user commands is supported by 6 separate channels to the CPU:

* Small Computer Standard Interface (SCSI) data bus
* Remote control port
* RS232c serial data port, 9600 baud
* Parallel data port (not presently utilized)
* Liquid crystal display (LCD) output-only port
* Front panel switch decoder input-only port

The SCSI bus can have as many as seven disk drives attached to it. As supported in the present operating system software, the SCSI port accomodates one expansion chassis which either contains 2 or 4 disk drives. Thus, one audio computer may have up to 5 disk drives. The LCD port provides machine status information to the user on a 32 character backlit display that shows the number and name of the cut cued up to play, counts the time up and down as necessary, and labels the soft function keys.

The front panel switch status is decoded at an input-only port that is read by the CPU more than 1000 times per second to provide fast response to user commands. The remote control port is bi-directional so both remote switch closures and user-acknowlegement signals are supported. DSP 1500s have been connected to the remote start and stop switches of a variety of broadcast consoles. Typically, solid state relays are used to make the dry switch closures needed by the 1500. The remote port directly powers LED indicator lamps at the console to indicate ready, playing, paused, and cueing status of the 1500.

The RS232c serial port allows the CPU to communicate with other computers, "dumb" serial data terminals, and custom microprocessor-based control panels. The RTDOES contains a complete RS232c-based command language that supports both data terminals and personal computers running compatible control and editing software. The data terminal support software in the 1500's EPROMs contains commands for simple record and playback as well as powerful "cut and paste" style electronic editing. Help screens are available whenever the machine is idle that explain the built-in commands.

Disk Drive

The magnetic disk drive that stores the digital audio data is a Bernoulli cartridge unit of the type manufactured by Iomega Corporation of Roy, Utah. The cartridges are sealed units 5 1/4" square and 5/16" thick. Over 300,000 of these drives and millions of cartridges are in use in computers of all types worldwide. The drive is in the half-height 5 1/4" form factor common in personal computers. Some PC makers, like Tandy and Leading Edge utilize the Iomega drive as one of the "standard" drives offered in their equipment. Many other peripheral data storage device manufacturers offer the Iomega as an add-in or add-on device for Apple and IBM PCs.

The Bernoulli cartridge contains two floppy disks mounted back to back on a common hub. The basic aerodynamic principles discovered by Daniel Bernoulli are employed in the disk cartridge design to stabilize the recording head to disk interface by taking advantage of disk rotation induced mass airflow. This arrangement allows the magnetic heads on each side of the twin disk to maintain good contact with the media at very low contact pressures. The heads actually "surf" in the air-lubricant boundaries on the disk surfaces.

The Bernoulli drive data recording technique is similar to that of many high performance magnetic hard disk drives. Digital data bits are represented by magnetic flux changes in the recording media. When audio is recorded, every bit of data is verified to be written correctly to the disk by reading the data after it has been written to the disk. In addition, the disk controller circuitry computes a 56 bit polynomial error correction code for each 256 byte sector. The error correction codes (ECC) are recorded as additional data within each sector on the disk.

To maintain the continued ability of the drive to store error-free data, the disk drive controller has the ability to reformat the disk cartridges. Formatting is the process by which the disk controller tests each sector of the disk for its recording and playback performance by writing and reading back difficult to handle binary data patterns. Two independant patterns are used during formatting. Any sector that does not meet specs for either pattern is removed from the cartridge's own map of good sectors. More than 100 kilobytes of spare data space is reserved on new cartridges to allow for the gradual loss of sectors as the cartridges wear over their more than 5000 play lifetime.

The disk drive will record and playback data at a continuous data transfer rate of over 1,000,000 bits per second. Although the data burst transfer rate for a single sector of information is more than 3 times this rate, digital audio recording requires continuous high bandwidth data transfers for extended periods of time. In the case of the DSP 1500, CSX digital audio encoding (discussed below in "Software") enables two channels of 16 bit digital audio samples to be recorded in real-time within the performance limits of the disk drive.

Disk Cartridge

Iomega and CompuSonics jointly developed a special version of the Bernoulli cartridge for digital audio use that is available in three formatted capacities: 20 megabyte, 14.5 megabyte, and 7 megabyte which correspond to about 7, 4.5 and 2 minutes of stereo recording respectively. The disks are priced from $10 to $30 each. Although standard Bernoulli carts as purchased at Radio Shack for $95 will work properly in the audio deck, digital audio cart disks will not work in a Tandy computer. This is a planned consequence of the development of the digital audio cart disk that protects the retailers of ordinary Bernoulli carts from possible leakage of lower priced audio disks into the personal computer marketplace.

At first glance the cart disks would seem to be far more expensive than tape carts. However, the disk carts appear to offer substantially longer life than tapes so the additional cost is offset. Preliminary test data collected over the past 14 months indicates that the disks will last for over 5000 use cycles. In continuous playback operation, 20,000 plays is not uncommon and some test disks have gone over 100,000 plays without audio degradation. The primary wear factors appear to be related to insertion-removal cycles in dusty environments. Excessive heat build-up in the equipment rack where the units are mounted may also shorten disk life. Additional data is being gathered related to these issues. Also, the manufacturers of the disk media are constantly improving their magnetic coating technology which is expected to further increase heat and wear resistance.

Other Types of Audio Computer Disk Drives

Although beyond the scope of this paper, it is worthwhile noting in passing that other models of audio computers, based on the identical SBAC, are also available for broadcast use. The primary difference among the models is their disk drives. In the "Background" section above, the optical disk cartridge version was mentioned. The DSP 1000XLR, based on optical disk carts, stores 2 hours of stereo on each write-once cart. The DSP 1800e stores 3 hours of stereo on a fixed hard disk drive, also referred to as a "winchester" drive. Both the 1000XLR and the 1800e are about the same size as the 1500e. Expansion chassis with up to 4 additional disk drives are available for the 1500e and the 1800e, but not the 1000XLR.

The Software

Being a computer, the cart disk machine would be useless without software. To achieve full functionality, two sets of software programs have been developed for the machine: an operating system and a digital audio signal processing system. The operating system is the control program for the computer that contains the instructions for audio recording, playback and editing. The second set of programs, for signal processing, is named CSX, a trademarked abbreviation of CompuSonics.

CSX Software

The DSP section of the computer runs CSX software to perform two tasks: encoding the digital audio samples so that they require the minimum amount of disk storage space and insuring the integrity of the data. CSX encoding [5,6,7] is a patented process whereby the audio signal is analyzed in real-time with respect to both its amplitude history and bandwidth characteristics on a block by block basis; each block containing about 8 milliseconds of audio samples. In effect, the recording time on the cart disk is at least doubled with respect to raw samples. Table 2 shows the audio performance and storage capacities of the DSP 1500 for its 3 modes of CSX recording. As can be seen from Tables 1 and 2, the audio quality is excellent. This result corrects the misinformation published in the recent SBE Conference Proceedings by the technical manager of another company in the cart business [8].

Without CSX encoding, not only is recording time severely restricted, but the probability of disk drive read errors effecting the audio data transfer rate increases substantially as well. Read errors are normally encountered while accessing data from digital storage media of all types, and are either corrected by the disk drive's controller circuit before the SBAC gets the data, automatic retries or the SBAC requests the disk drive to try again. The combination of these three strategies reduces errors to negligible levels; on the order of 10 to the -12 power (1 error every million million bits). CSX encoding further increases the reliability of the data by minimizing the absolute number of bits that must be transferred every second to playback continuous audio. This gives the error correction systems fewer bits to inspect and more time to do correction or recovery when needed.

An additional measure of audio playback reliability is insured by a checksum word that is computed for each CSX encoded block of audio samples. If the signal processors encounter a checksum error, that audio block is muted. This method of error avoidance is very robust. Even severe physical damage to the disk will not interrupt audio playback.

RTDOES Software

The real-time disk operating and editing system, RTDOES, was developed specifically for the audio computer. It supports recording and playback operations directly from the unit's front panel or remote console switches, and editing operations from a serial data terminal or personal computer. Controlling the machine from the front panel is similar in operation to a conventional tape cartridge recorder/reproducer. There are buttons for start, stop, cue, record, secondary and tertiary tone placement. The last three buttons are displayed on the LCD, and are "soft" function keys. That is, they are redefined by the RTDOES as needed to provide additional controls without additional discrete switches.

When a cart disk is inserted, the DSP 1500 displays the first cut number available for playback, or the name of the cut if one has been entered during the recording session. If the disk is blank, the display reads "disk is blank" instead of a name or number. Under the LCD, the 3 function keys are titled "rec", "sec" and "ter". Descriptions of the basic procedures for the most common broadcast audio operations follow.

Recording

With a blank cart disk in the drive, the user presses the "rec" button, the green "start" button begins flashing, and the DSP 1500 begins monitoring audio. The level of the input signal may then be adjusted with the gain control pots on the front panel. To begin recording the user presses the "start" button. The LCD display shows the number 1, indicating that the first cut is being recorded, and the timer counts up in minutes and seconds. To pause during recording and continue to monitor audio, the "stop" button is pressed once. The red "stop" button begins flashing, the LCD shows the word "paused" and the count-up timer is paused. To resume recording, the user presses the "start" button. A clean, glitch-free splice is created in the audio where it was paused. To stop recording, the user presses the "stop" button twice. The directory information about the recording is written to the disk and the cut is automatically cued for playback.

Recordings can be made in mono or stereo and at 3 different "speeds". They can also be recorded as endless loops. Up to 180 cuts, some or all of which may be endless loops can be recorded on one disk. The recording "speeds" are actually different CSX encoding modes. The audio quality, bandwidth and recording times for these modes are listed in Table 2.

During recording, "cue tones" can be set at any time by pressing the "sec" or "ter" buttons on the front panel. These are not tones, but electronic markers that will light the front panel "sec" and "ter" buttons whenever they are encountered in a recording. The "tones" also cause output pins on the rear panel remote control port to go to a 5 volt level for the duration of the "tone". As many as 14 markers can be placed anywhere in any cut. These markers may be deleted as a group from the cut at any time, and re-recorded without disturbing the audio tracks. The markers also serve as edit decision points for slicing up cuts while editing.

Playback

To playback the cut just recorded, the user presses "start". Punch-in is virtually instant, with no wow or flutter since there is no tape to accelerate and the disk is always spinning. During playback, the LCD shows the timer counting down the time remaining on the cut. Playback may be paused at any time by pressing "stop" once. The LCD shows the word "paused", and the timer is paused. Pressing "stop" twice stops playback and cues the same cut ready for playback. If the cut is allowed to play to the end, the next cut on the disk is cued up and ready to punch-in about 2 seconds after the first cut ends. Seamless playback, without cueing delays, of up to 30 cuts is accomplished by creating a "Chain" with the built-in editing software.

The DSP 1200 is the on-air playback-only version of the DSP 1500e. It contains specific software for random access playback from multiple disk drives in conjunction with an optional Apple Macintosh or serial data terminal. On the model 1200 the directory screen can display up to 40 cuts at once. This makes it easier for the on-air talent to see and cue-up the next cut. In a fast-paced audio effects drop-in situation this eliminates tape cart shuffling and helps reduce miscues.

Cueing

A cart disk can hold up to 180 cuts. Cueing the cuts from the front panel is accomplished by pressing the "Cue" button. The button can be pressed as fast as humanly possible to skip ahead on the disk as many cuts as the number of times the button was pressed. The LCD shows the number or name of every cut as it is addressed prior to cueing it up. This allows rapid paging through the cuts without waiting for the disk to cue up the audio. Once the user stops pressing the "cue" button, the selected cut is ready to play in about 2 seconds. True random access based on cut number is available via the RS232 serial data port.

Editing

Electronic editing allows the user to create new cuts which reference selected segments of the originally recorded audio. This is called "non-destructive" editing because the original audio material remains intact; it is only the playback instructions which change. Editing is useful for eliminating unwanted audio material from a cut and for creating audio segments to be sequenced later using the chaining capability. The software that makes editing possible is a set of functions built into the DSP 1500e's RTDOES in non-volatile memory so they are always available instantly.

The RTDOES contains a complete command language for RS232c serial data communications with terminals and other computers. This feature supports editing and extended functions that would be difficult if not impossible to operate from a simple switch-based front panel. Serial data terminals of many types and brands have been used successfully with the DSP 1500e. Personal computers running terminal emulation software have also been used. CompuSonics has also developed specific software for IBM PCs called "PC/Sonics" and for the Apple Macintosh called "MacSonics" to take advantage of their ability to make the user interface more "friendly" than a plain terminal. However, both software packages rely on the same fundamental RS232c command language that drives the "dumb" terminals. This command language supports a variety of powerful functions, a few of which are discussed below.

These functions are presented in approximately the order they would be utilized by an audio production person editing on a data terminal connected to a DSP 1500e. To access a function, the user types the character indicated, and sometimes is required to answer "y" for yes and "n" for no to prevent accidental use of functions such as "e"; erase the disk.

h show the help screens

Typing "h" begins a series of 3 screens of help which lists all commands and what they do. Typing "q" aborts the help screens.

d show the disk directory

Displays the directory of the cart disk which includes cut number, name, comment, recording speed, channels, running time, and cut status: deleted, looped, or chained. Figure 4 shows a typical directory.

c change to a new cut

When "c" is typed the machine prompts the user for a cut number. After typing the cut number and pressing the carriage return key, the new cut is cued up for playback or editing.

n name and comment about a cut

When "n" is typed the user is prompted for a cut title. After typing the title and pressing the carriage return key, the user can enter a comment as well. These will appear in the disk directory and up to 16 characters of the name will also appear on the unit's front panel LCD.

a enter editing mode

This enables the audio "rocking" functions that are useful for finding edit points.

p play the cued cut

This punches in the audio playback.

4 rock audio forward 320 milliseconds and stop

Pressing the number "4" plays 320 milliseconds of audio and stops. The editor can proceed forward stepwise until the point of interest is reached. The actual time location of the "playback head" in the audio track is displayed on the terminal screen in seconds and milliseconds. Pressing "4" during playback automatically pauses playback.

1 rock audio backwards 320 milliseconds and stop

Pressing the number "1" backs up the audio track 320 milliseconds, inaudibly. The actual time location of the "playback head" in the audio track is displayed on the terminal screen in seconds and milliseconds. Pressing "1" during playback automatically pauses playback.

3 rock audio forward 40 milliseconds and stop

Pressing the number "3" plays 40 milliseconds of audio and stops. The editor can proceed forward audibly in a stepwise fashion until the point of interest is reached. The actual time location of the "playback head" in the audio track is displayed on the terminal screen in seconds and milliseconds. Pressing "3" during playback automatically pauses playback.

2 rock audio backwards 40 milliseconds and stop

Pressing the number "2" backs up the audio track 40 milliseconds, inaudibly. The actual time location of the "playback head" in the audio track is displayed on the terminal screen in seconds and milliseconds. Pressing "2" during playback automatically pauses playback.

Note that the 1, 2, 3, and 4 keys are arranged in a row on the top of all typewriter-style keyboards. These keys can easily be pressed by four fingers on the left hand to control the "audio transport" while searching for an edit point. This differs substantially from physical tape rocking since there are no reels of tape to rock. Furthermore, the audio is not heard backwards or at any slower speed. However, the benefits of millisecond precision, true-pitch audio playback, and no audio amplitude loss are usually appreciated and found more than adequate by editors who have used the equipment.

s mark audio track with secondary "cue tone"

Typing "s" places a "secondary" electronic marker alongside the audio data. This key can be pressed while audio is playing back or while paused.

t mark audio track with a tertiary "cue tone"

Typing "t" places a "tertiary" electronic marker alongside the audio data. This key can be pressed while audio is playing back or while paused.

S slice a segment out of a cut

Typing "S" produces a list of the "cue tones" presently set in the cut being edited. A typical list is shown in Figure 5 below. Note that the tones are numbered sequentially, and the time remaining to play in the cut from each cue tone is shown. The DSP 1500 prompts the user to select the tone which corresponds to the desired punch-in point. If the punch-in is to be the beginning of the cut, a zero is entered. Then the user is prompted to select the cue tone corresponding to the punch-out point. As soon as that tone is selected, the machine creates a new cut which contains only the audio between the punch-in and punch-out points selected by the user. To delete a segment of audio from the middle of a cut, the slice function must be used twice to create two new cuts: the beginning and the ending. These new cuts are then spliced together seamlessly by the chain function, "C".

C create a chain of cuts

Typing "C" results in a prompt to the user to select the first cut from the directory of the disk that is desired in a seamless chain of cuts. As soon as a cut number is entered, the user is prompted for the second entry in the chain, and so on until the user types "999" to tell the DSP 1500 that no more cuts are needed for the chain. Up to 30 cuts can be entered into a chain. To create an edited version of a cut shortened by deleting some segment of audio in the middle of it, a chain composed of two cuts would be created. The first cut in the chain would be the beginning of the material and the second entry in the chain would be the cut containing the end of the original audio.

L enter a new chain of cuts into the directory as a single cut

Typing "L" adds the freshly created chain to the list of cuts already in the directory. The new directory entry will cue up and play just like an originally recorded cut, even though it may contain up to 30 other cuts.

Frequently, after a chain has been created and rehearsed it is apparent to the editor that one of the cuts in the chain must be trimmed to create a better sounding splice, or that the audio was up-cut. Allowing for this, the RS232c command language also contains commands for trimming and extending the audio at splice points. Although they are beyond the scope of this paper, the DSP 1500e's operating system presently supports over 50 commands for audio recording, playback and editing functions.

Comparative Test Data

Equipment Under Test

Three pieces of audio recording equipment were tested for audio quality specifications with new media and then retested after many plays as a measure of system performance in typical broadcast duty cycles. The equipment tested was a CompuSonics DSP 1500 cartridge disk recorder, a Pacific Recorders Maxtrax Tomcat cartridge tape recorder, and a JVC digital audio tape (DAT) cassette recorder. Each piece of equipment was fully calibrated and in like-new condition with less than 500 hours of use. The recording media for the CompuSonics was a Bernoulli magnetic flexible disk cartridge. For the Tomcat the media was a Capitol Audiopak AA-4, SGS-4 70 second tape cartridge. The JVC DAT recorder media was a Maxell R-60DM tape cassette.

Test Rig

The test bench set-up is shown in Figure 3. The key equipment was a Sound Technology 1510A tape recorder/audio tester, a Tektronix TM 515 chassis with an AA 501 distortion analyzer and SG 505 oscillator, and a Tektronix 2236 100 mHz scope, oscillator and counter-timer. Shielded cables and proper grounding procedures were utilized.

Test Procedures and Results

Seven types of tests were performed on the subject units:

* Signal to noise with new media
* Frequency response 20Hz to 15kHz with new media
* Total harmonic distortion + noise with new media
* Third harmonic distortion with 1 kHz input with new media
* Signal loss after 2000 plays of same media
* THD + N after 2000 plays of same media
* Third harmonic distortion with 1 kHz input after 2000 plays

Each recorder was provided with precision sine wave test signals at the optimum levels for that recorder in order to maximize sonic performance. The results are summarized in Table 1.

In the case of the DAT machine, the wear test must be considered preliminary and not conclusive due to the difficulties encountered in achieving a reasonable number of trials with that unit. The problem appears to be premature tape wear which first manifested itself as the inability of the DAT machine to recue its selected audio segment after 84 plays. Rerecording the cue markers in a slightly different location on the tape after the failure allowed continuation of the test. However, rerecording the cues caused false cues and noise spikes in the audio at the starting cue point on every subsequent playback. Rerecording of the cues was done a total of 4 times. This problem notwithstanding, the test was continued for a total of 624 plays at which point the error correction system of the DAT was muting (no output signal at all) 50 out of the 65 seconds in the test. The 15 seconds that remained playable was close to the original recording's specifications. Further testing is underway with another brand of DAT recording media to determine if the problem is the particular unit being tested or the tape itself.

The test results indicate that, for the small sample tested, the cart disk machine and the DAT recorder were approximately equal in audio perfomance at the beginning of the test after only a few playbacks of new media. Both the DAT unit and the cart disk machine clearly outperformed the NAB tape cart machine, in audio quality, under the initial conditions. After less than 100 plays of all machines, changes became apparent. The DAT unit began to miscue, and at about 600 plays was unusable, as discussed above. After 2000 plays the NAB tape cart was still playing, but showed signs of serious tape wear; its output signal had dropped by 36% and THD+N increased 15%. The extent of tape wear is made even more evident by the measurement of third harmonic distortion with 1 kHz input after 2000 plays. The almost total loss of frequency response above 8 kHz appears as a reduction in measured third harmonic distortion due to the lack of signal on the tape.

The cart disk recording unit suffered no audio losses for the duration of the test. Its audio quality was the same on the first play and the 2000th play.

Operational Comparisons

Audio quality testing and media durability testing do not disclose the complete picture of how the cart disk, tape cart, and DAT machines compare with one another. Table 3 provides another comparison among the three machines that includes most operations required by broadcasters of audio cartridge recording equipment.

Summary

This paper has described the DSP 1500e, one of the first machines in a new generation of digital audio computers that are well suited to replacing stereo tape decks of all types in the broadcast industry.

These machines offer high audio quality, reliability, low maintenance, ease of use, and advanced electronic editing features. The cartridge disks have proved reliable and long-lived. The performance of the machine and its media have compared favorably with other equipment of similar functionality.

The CompuSonics machine can record, edit and air the exact same piece of removable media without damaging the original recording or dubbing it off to any other format. This capability is entirely new for radio broadcast facilities and can result in significant efficiency gains in production and on-air operations.

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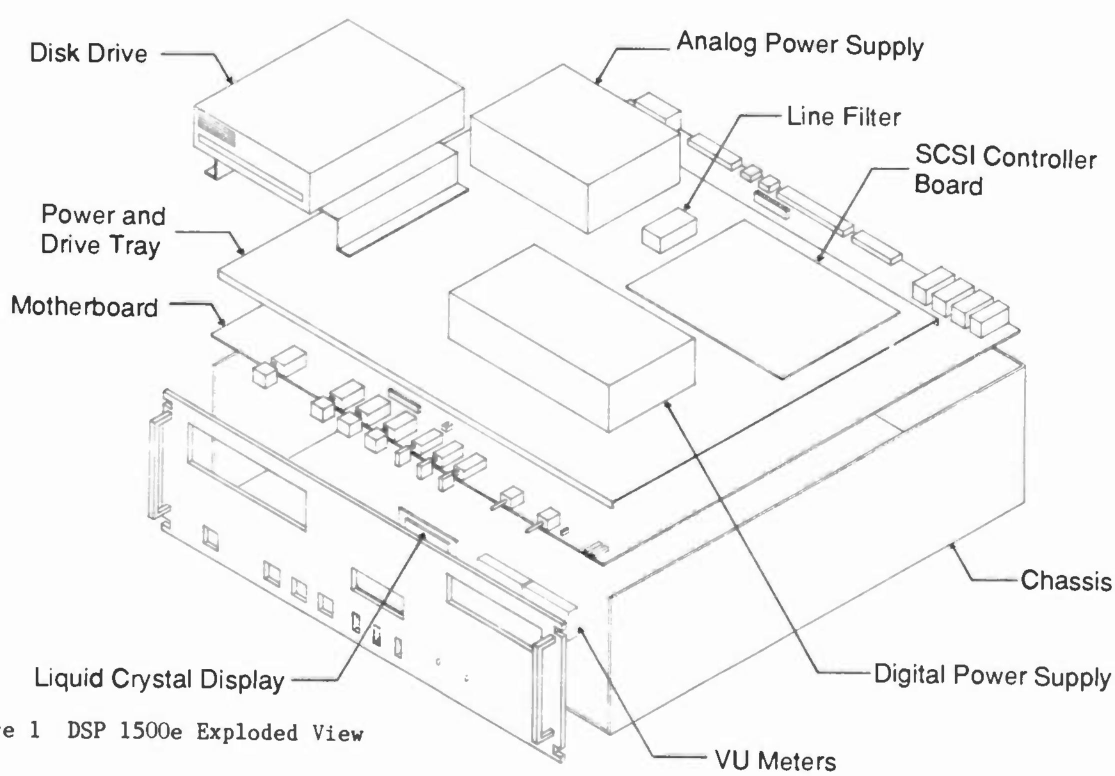

Figure 1 DSP 1500e Exploded View

Broadcast Audio Recorders Test Summary			8 February 1988
	Compusonics DSP 1500	PR Tomcat	JVC DAT
Signal to Noise Ratio	88 dB	67 dB	89 dB
Frequency Response at: 20 Hz	-0.5 dB	0.1 dB	0.0 dB
50 Hz	-0.1	0.1	0.0
100 Hz	-0.1	0.2	0.0
200 Hz	0.0	0.2	0.0
500 Hz	0.0	0.2	0.0
1 kHz	0.1	0.2	0.0
2 kHz	0.1	0.2	0.0
4 kHz	-0.1	0.3	0.1
5 kHz	-0.2	0.3	0.2
6 kHz	-0.2	0.3	0.1
7.3 kHz	0.0	0.2	0.2
8.3 kHz	0.3	0.2	0.2
9.4 kHz	0.6	0.2	0.2
10.5 kHz	0.4	0.2	0.2
11.5 kHz	0.2	0.1	0.1
12 kHz	0.3	0.1	0.0
13.5 kHz	0.6	0.1	0.1
14.5 kHz	0.3	0.1	0.2
15 kHZ	-0.6	0.1	0.3
THD +N at 1 kHZ input	0.026%	0.818%	0.0165%
3rd harmonic distortion at 1 kHz input	0.004%	0.26%	0.004%
Signal Loss after 2000 plays	0.0 dBm	0.6 dBm	Failed at 624 plays
THD + N after 2000 plays	0.026%	0.960%	
3rd harmonic distortion @ 1kHz after 2000 plays	0.004%	0.10%	

Table 1

CompuSonics DSP 1500: 5 1/4" magnetic cartridge disk recorder/reproducer		
Sampling Rate: 32 kHz A/D/A Resolution: 16 bits		Disk Data Capacity: 20.2 Megabytes (formatted) Audio Data Storage Allotment: 20.1 Megabytes
Recording Speed	Applications	Performance
High (CSX II)	Live music recording; Studio recording of speech and music for broadcast; Sound effects; Audio archiving	Typical bit rate*: 532 kbits/sec (stereo) Avg. recording time: 5 min. 2 sec. (stereo) Dynamic range: 88 dB Freq. response: 15 kHz THD+N: .026 % @ 1 kHz input 3rd Harmonic distortion: Undetectable
Medium (CSX II)	Broadcast production recorder; Recording and playback of pre-recorded music, commercials, and jingles for broadcast; Studio sound effects	Typical bit rate*: 378 kbits/sec (stereo) Avg. recording time: 7 min. 5 sec. (stereo) Dynamic range: 88 dB Freq. response: 15 kHz THD+N: .175 % @ 1 kHz input 3rd Harmonic distortion: Undetectable
Low (CSX I)	Broadcast recording and playback of speech and announcements; background music	Avg. bit rate: 140 kbits/sec (stereo) Recording time: 19 min. 2 sec. (stereo) Dynamic range: 88 dB Freq. response: 4 kHz THD+N: .431 % @ 1 kHz input 3rd Harmonic distortion: .052 % @ 1 kHz input

Table 2. Performance and Applications of the CompuSonics DSP 1500.

* Bit rates for CSX II vary depending on the type of music recorded. Both the High and Medium speed bit rates in this table were determined by recording a wide range of selections of rock and roll and classical music from compact discs.

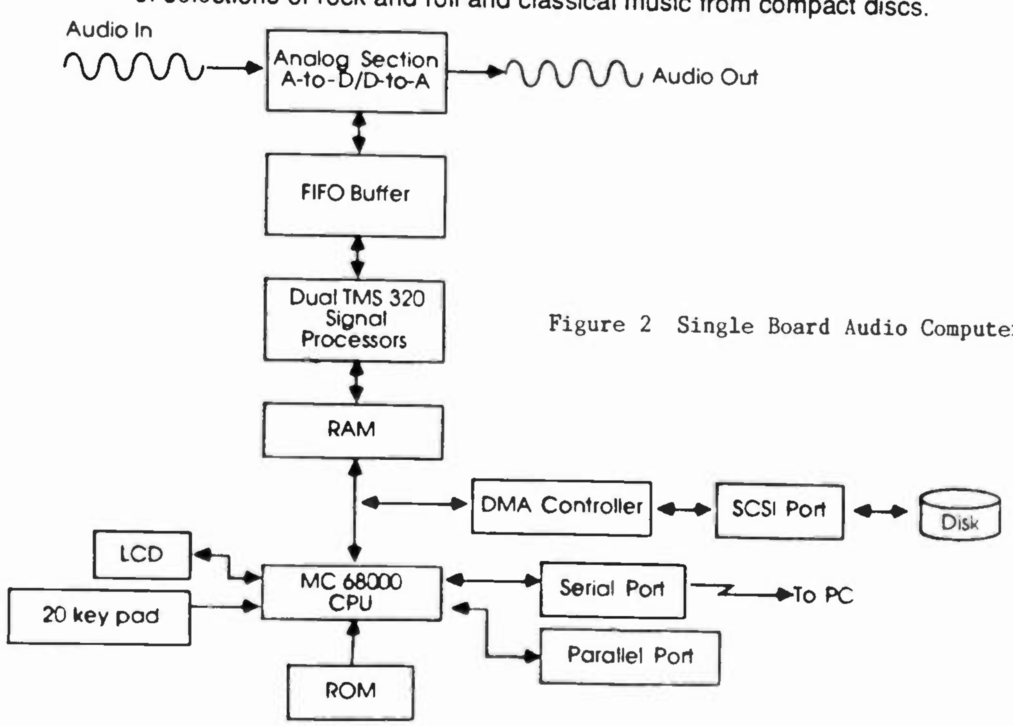

Figure 2 Single Board Audio Computer Diagram

Broadcast Audio Recorders Operational Comparison Table 3 8 February 1988			
	CompuSonics	Tomcat	JVC DAT
Time to load and cue	6 seconds	1 second	10 seconds
Time to unload	4 sec	1/3 sec	4 sec
Time to access any cut	2 sec	depends on location	15 sec
Looping ability	yes	yes	no
Audio trimming	yes	no	no
Cut and paste editing	yes	no	no
RS232c command port	yes	no	no
Bi-directional remote port	yes	yes	no
Alphanumeric display	yes	no	no
Cue tones	yes	yes	no
Cut Indexing	yes	no	yes
+4 to +8dB balanced audio	yes	yes	no
Digital to digital capability	yes	N.A.	yes
Erase function	yes	no	N.A.
Rack-mountable	yes	yes	no
Number of Recording speeds	3	2	2
Double record time in mono	yes	no	no

Figure 3

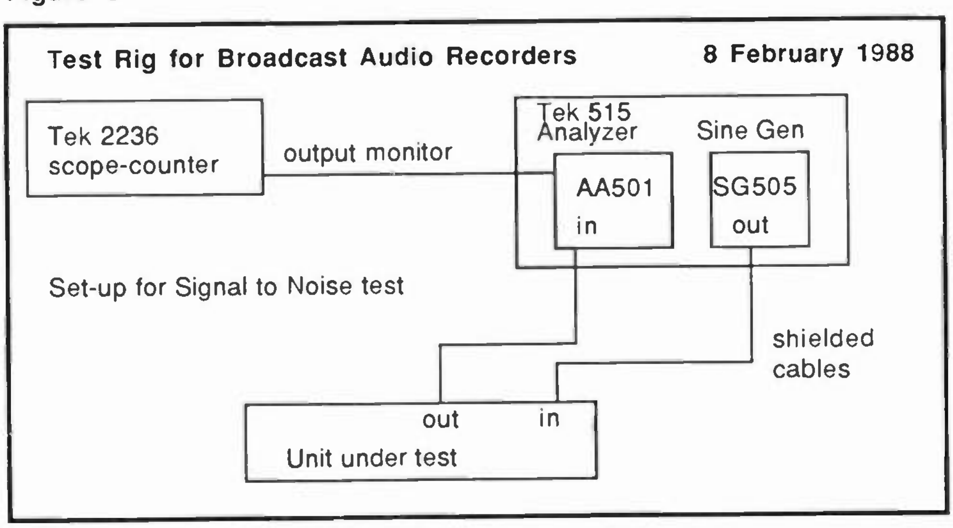

Figure 4 Example of a Cartridge Disk Directory

disk title: WXYZ Sampler

cut	name	comments	speed	chnls	length	
1	station jingle 1		med	2	:10	
2	station jingle 2		med	2	:20	DEL
3	typewriters	background	med	2	1:00	(L)
4	hard days night	Beatles	fst	1	2:36	
5	congress news	actuality	slw	1	:45	

number of formats: 1

Figure 5 Example of a list of cue tones on cart disk

The Current Tones are:

number	tone	duration(seconds)	location(remaining)
1	SEC	12	5:08
2	TER	5	5:05
3	SEC	7	4:53
4	TER	9	4:53
5	SEC	2	:10

DIGITAL STORAGE AND RANDOM ACCESS OF BROADCAST AUDIO

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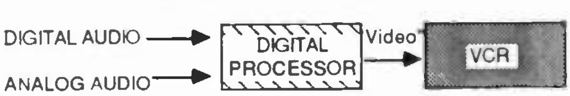
The Schafer Digital System incorporates a number of VCR's on which are played program material recorded with the use of digital processors. Commercials, ID's and other short events are stored on VCR's and automatically down loaded to a hard disk to allow instant access in any order desired. The system is computer controlled. Playlists and commercial schedules may be inputted from other computers, from music syndicators and traffic and accounting systems. Random access is accomplished and controlled with the use of an IBM compatible AT computer. Sufficient storage is obtained to allow the storage of a complete music library and complete commercial and other short event library. The system is designed for use in a totally automatic, semi-automatic or live assist mode. Walk away time is measured in days or weeks. Quality is equal to the best compact disk.

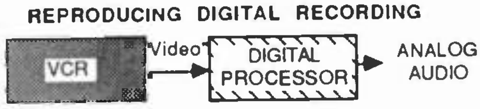
The Schafer Digital System utilizes the EIAJ standard. Digital or analog audio is fed to a digital processor which converts the information to a digital form which can be recorded (like video) on a Beta video cassette recorder. The format is 16 bit quantization with a 44.1 KHz sampling frequency, providing excellent quality, equal to the best compact disk.

DIGITAL RECORDING ON VCR



In playback, material digitally recorded on the video track of the video cassette is converted to analog by a digital processor.

REPRODUCING DIGITAL RECORDING

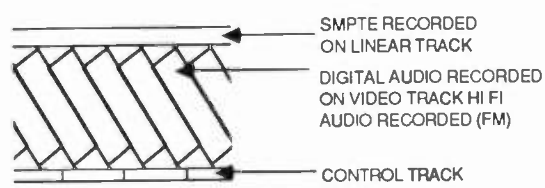


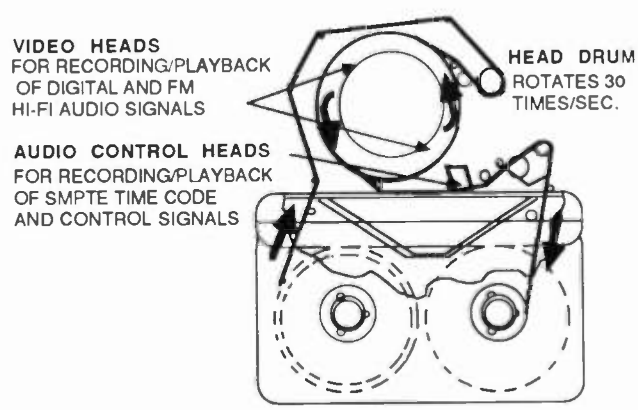
Recording in Beta III gives excellent results and permit recording of up to 5 hours digitally on an L-830 cassette. In addition, up to 5 hours may be recorded on the FM (hi-fi) track, a total of 10 hours per cassette. (9 hours on an L-750).

Specifications, record through playback of digital and FM:

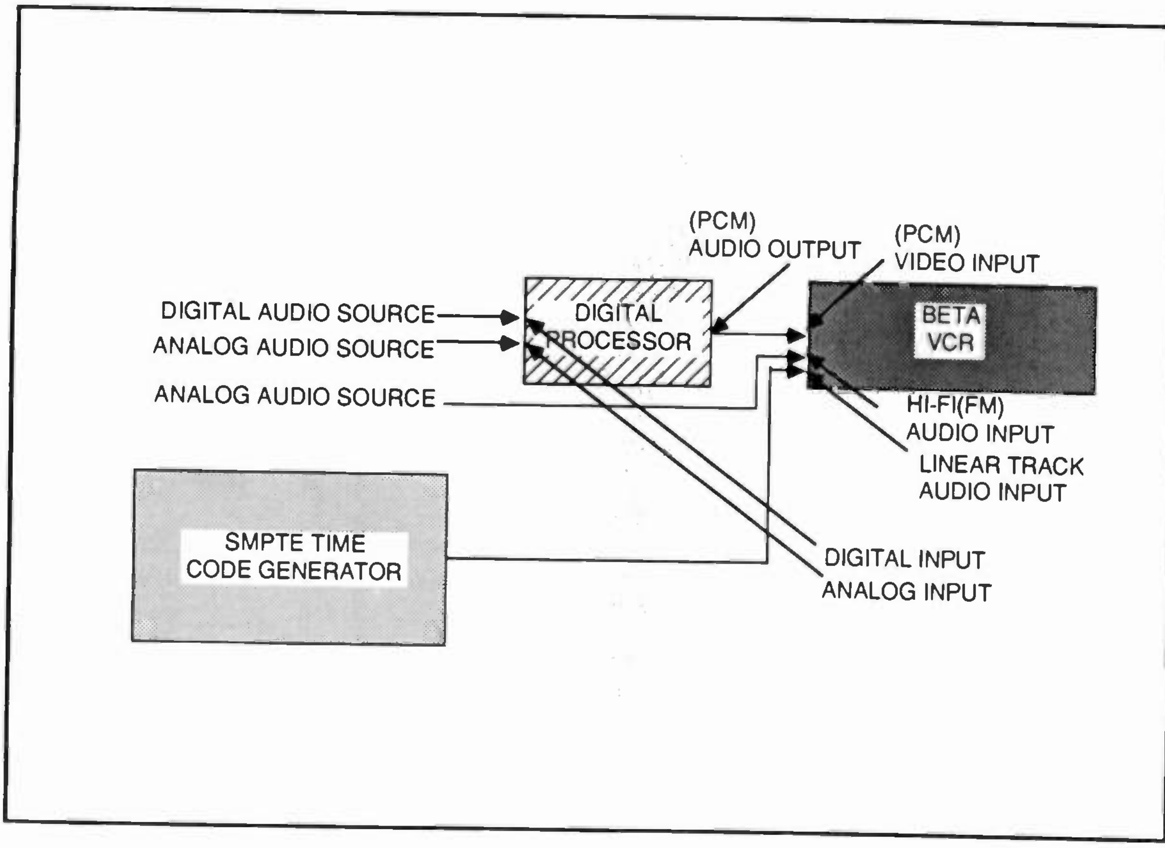
Frequency response	20 - 20,000 Hz + - .5 db
Harmonic distortion	Less than .005%
Dynamic range	More than 90 db digital; 80 db FM.
Wow and flutter	Below measurable limit.

Recording up to 10 hours on each cassette allows the storage of up to 80 hours in a system with 8 VCR's. The key to flexibility is the ability to search and cue to any song or recorded event automatically. This is accomplished by recording the SMPTE time code on the linear audio track located on one edge of the tape. A control track is located on the other edge of the tape. The video track and the hi-fi track is located in the center of the tape. Search is accomplished without the video heads coming into contact with the tape.





As a cassette is recorded, a (SMPTE) time code is striped (recorded). This provides the ability to cue to within one frame, that is to 1/30 of a second. (There are 30 frames to the second.) A typical address could be: 04:19:55:20.
(4 hours, 19 minutes, 55 seconds, 20 frames).
A user bit number is interlaced in the time code. Each time the cassette is cued or played, the cassette number and the address on the cassette are identified and confirmed. This assures that the correct event is being played from the correct cassette.



Block diagram above of Schafer Digital recording set-up shows a system to allow recording from analog or digital sources. A digital source such as a compact disc may be recorded (cloned) by connecting the digital output of a CD player to the digital input of the digital processor. Analog audio may be recorded by connecting to the analog input of the digital processor. In either case, the signal will be converted to digital form to be recorded on the video track of the cassette.

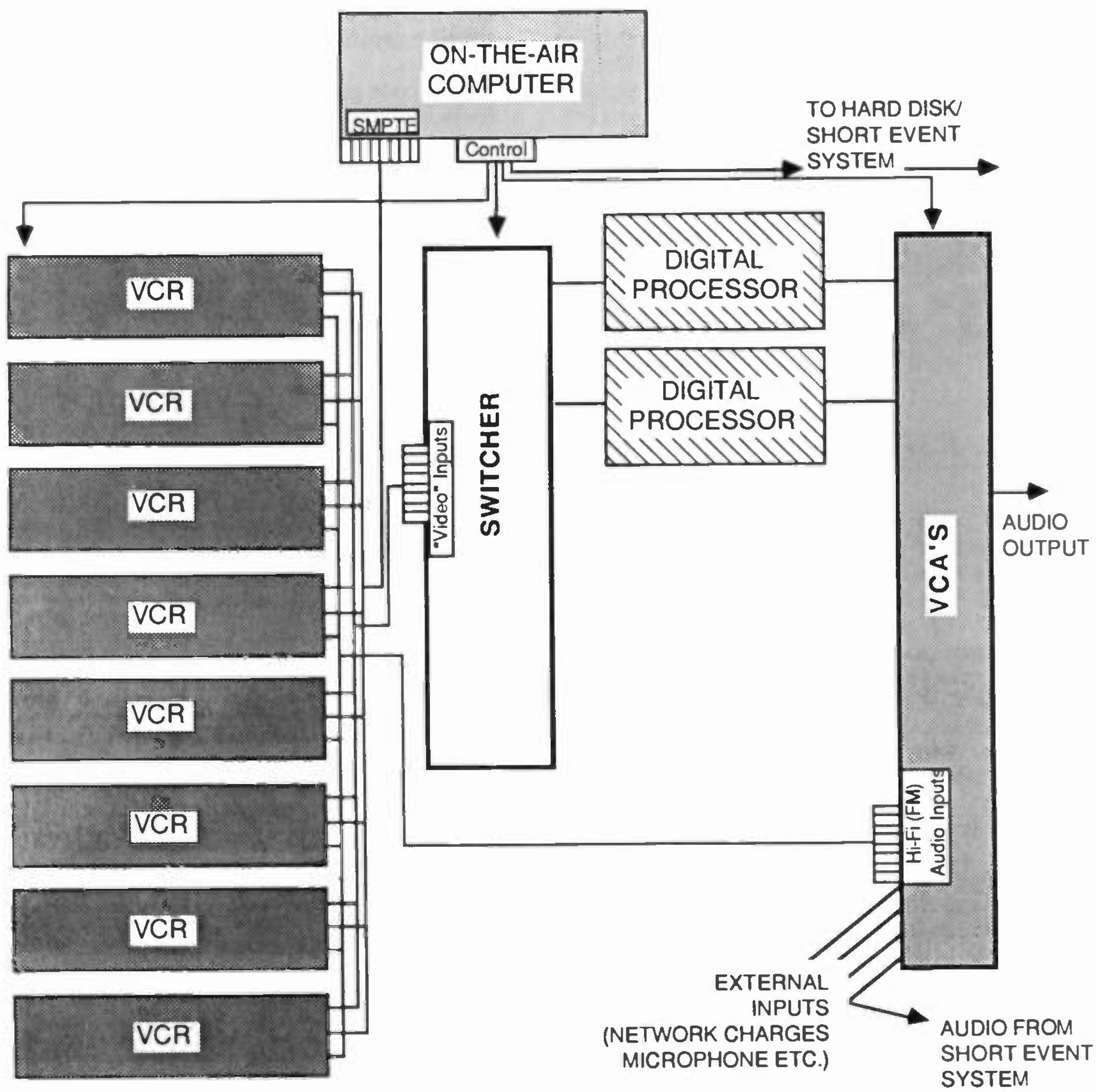
Analog audio may also be recorded on the FM (hi-fi) track of the video cassette by connecting to the hi-fi input of the VCR. This doubles the amount of time which can be recorded on a video cassette to 10 hours on an L-830 cassette, 9 hours on an L-750 cassette.The L-750 cassette has standard thickness tape.

The SMPTE time code may be recorded at the same time both the digital and FM tracks are being recorded or a cassette may be "striped" with SMPTE time code and the digital and FM tracks may be recorded and/or edited later. It is not possible to edit the digital (video) track and the FM (hi-fi) track separately. A VCR with editing capability, such as the Sony SLHF-1000 is used for recording and editing to allow recording and editing on the digital and FM tracks without affecting the SMPTE time code track. Special inputs and outputs are installed on the VCR to allow simultaneous recording and reproduction of all tracks.

After recordings are made, the information including the song title, artist, length and any other pertinent data including the cassette number and the address must be entered into the data base in the computer. The address information (SMPTE time code) and length may be developed and entered manually or an automatic system is available to record beginning and end times. It is important to note that the end time of an event which is calculated in the computer from the SMPTE code at which the event begins plus the length of the song or the event, is the point at which the computer will begin the fade of the on-the-air event and the start of the next event. The length of time of an event is therefore calculated to the time at which the fade will begin. A song which ends in a crescendo should have the length indicated, therefore, as the point in time at the precise end of the crescendo. This allows programming to be as tight or loose as the program director wishes, a function of entering length of time of events and/or a function of software. Length of fade is hardware adjustable, as well. Program services supplied by music syndicators will include all data information supplied on floppy disks to correspond to the appropriate cassette. Playlists are available from music suppliers and can be modified by the station program director. Whether a particular radio station requires a complete production station or not depends on the extent to which it is desired to record material an enter and change data.

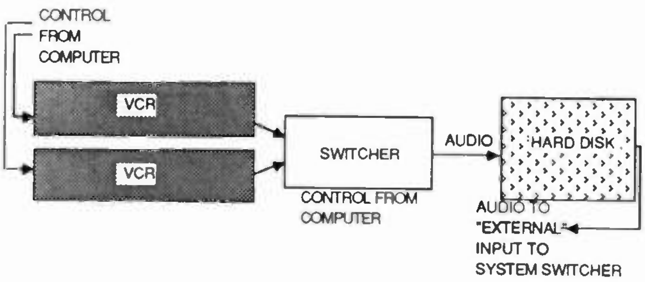
The block diagram below shows the computer in a typical 8 VCR system. Two digital processors are employed to allow a cross-fade and overlap from one on-the-air event to another. External inputs are shown. The computer and switching system will accommodate external sources such as the hard disk system for instant access of commercials, ID's and other short events which is described in this paper, as well as other sources, network, CD players, multiple or single cartridge players; any audio source including a simple microphone.

An IBM compatible AT computer is employed to control the VCR's and switching, as well as the scheduling of the order in which events will be aired. Controller boards and software are installed in the computer and connected to interface with each VCR and to the switcher and VCA controller. Interfacing is available to allow control of external sources, including selection of individual cartridges or CD's in an external multiple player. Communication is accomplished in RS-232 or other data stream to the interface circuitry.



INSTANT ACCESS HARD DISK SYSTEM

The block diagram shows the instant access hard disk system utilized for storage of short events which must be available for play in any order. A video cassette can store hours of audio, ideal for music, but the access time can be minutes, not ideal for short events such as commercials, ID's, jingles and the like which must be available for instant access. The hard disk is perfect for storage of events for instant access, but the cost is generally considered prohibitive and the fear of losing material in the event of a hard disk crash is another consideration. The system we describe here incorporates the advantages of both with the disadvantages of neither.



The short events are recorded on a video cassette together with the SMPTE time code as in the case of the music cassettes described earlier. The data describing the location of each event is stored in the computer, as well as the schedule indicating when each item will be aired. An L-750 cassette can store up to 4.5 hours of such events in either digital or FM form. These can be edited without disturbing the SMPTE time code on the editing model VCR. The hard disk with 300 megabytes reserved for the recording of 30 minutes of stereo in 16 bit format (48 KHz sampling frequency) assures fine quality. The computer down loads from the VCR or VCR's to the hard disk in advance of air time. Typically, the system will be an hour or more ahead, allowing time for editing during the broadcast day. The on-the-air computer controls the down-loading of events as well as the airing of events as they are scheduled. Frequently broadcast events will automatically remain on the hard disk. In any event, all events are ready for broadcast when scheduled. A single VCR system is adequate for a light broadcast schedule. A dual VCR system is recommended for heavier broadcast schedules. New material may be recorded at any time by removing the cassette from service. This represents no problem in that material required for broadcast in the near term has already been down-loaded to the hard disk.

PRODUCTION STATION

The production station will require a reproducer of some kind to play material to be recorded into the digital system. This can range from a simple analog production control room to a special facility including digital recorders and reproducers to permit copying (cloning) keeping the material in digital form until it leaves the Schafer Digital System, ready to enter the station transmitter.

Production station requirements will vary from station to station. A production station which includes its own computer is ideal for most stations and is a necessity for a radio station in which playlists and/or commercial schedules are created and edited extensively during the broadcast day.

A time code generator is required if a radio station requirement includes recording music cassettes, unless the station wishes to purchase cassettes pre-striped with SMPTE time code.

A digital processor is required in the production station, if a radio station requirement includes recording music cassettes and/or short events digitally. It is considered acceptable to most to record and edit commercials and other short events on the FM track, eliminating the need for a digital processor in the production station.

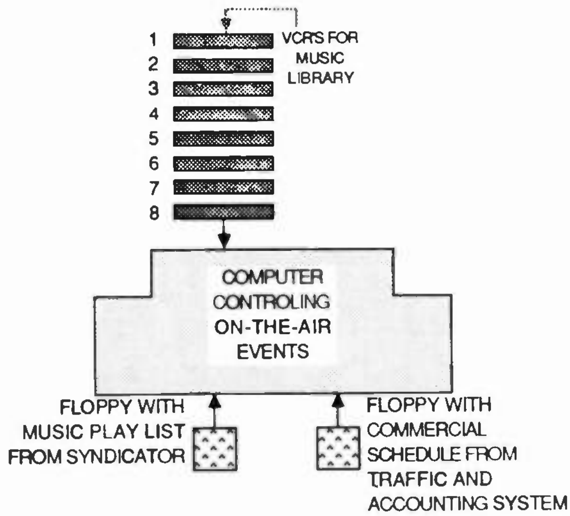
A compact disk player with a digital output is required to copy (clone) compact disks, digital to digital. A special input is provided on the special digital processor to allow digital to digital cloning.

An editing VCR such as a Sony SLHF-1000 is essential for editing. It allows recording and editing without disturbing the SMPTE time code track.

No production station would be required by a station receiving music cassettes from a music syndicator, assuming the station did not require the ability to edit playlists extensively. It is possible to make changes in the play list from the on-the-air computer prior to air time, but more extensive changes would indicate the need for a production station including a computer.

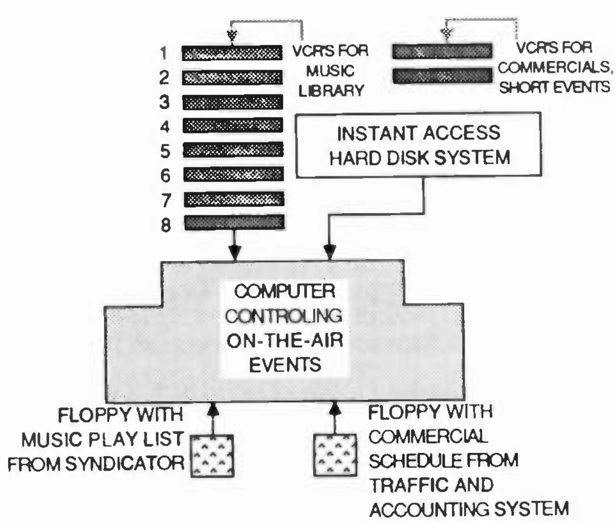
The production station computer may be used to create music play lists and commercial schedules and/or to merge and edit them. Traffic schedules and play lists from external sources may be introduced in more than one way as shown next.

ON THE AIR SYSTEM

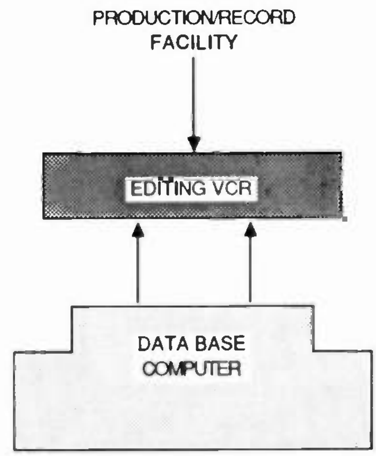


The simplest system shown above, a music only system would require external source(s) for commercials, ID's and other short events, if they are to be controlled by the system. The music cassettes and the music playlist could be provided by a music syndicator.

Below is shown the simplest system incorporating the instant access commercial and short event system. The commercial schedule could be provided by a traffic and accounting system and introduced to the on-the-air computer via a floppy disk. The music playlist could be provided by a music syndicator and introduced to the on-the-air computer via a floppy disk.

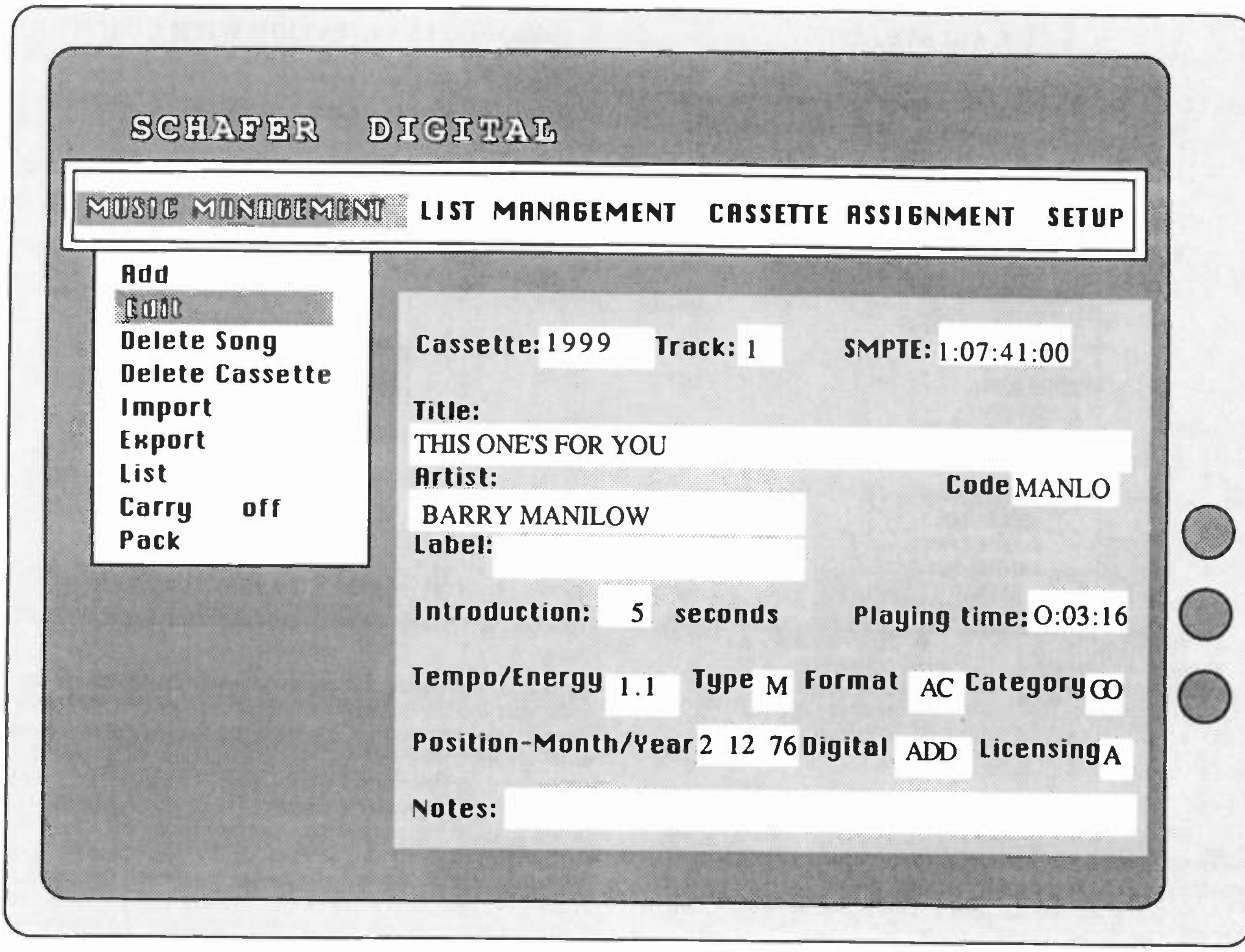


PRODUCTION STATION WITH COMPUTER



Typically, the on-the-air computer and the production station computer would be identical, an ideal situation for back-up should a failure occur. Currently, the Schafer Digital System employs an IBM compatible AT computer made by NEC. It includes a parallel and 2 serial ports and 640K of RAM. It employs a 1.2 megabyte floppy disk drive. Additional disk drives may be installed to facilitate accepting data from other computers in the radio station. A 20 megabyte hard disk provides more than adequate storage for the required data base and operating software provided with the system. The computer has an operating speed of 10 MHz and has eight full size expansion slots. Controller boards are installed to work with the software provided. These boards include SMPTE time code reading circuits and circuitry to interface with all external interfacing. A color monitor is recommended to facilitate viewing. Friendly software provides a window type format to make entry and changing easier.

The production station computer includes the interface for the editing VCR. The data base information is stored in this computer. The commercial schedule may be generated in this computer. The music playlist may be generated in this computer. The music playlist may be merged with the commercial schedule in this computer, whether they are generated in this computer or introduced via floppy disks from other computers.



COMMUNICATING WITH THE COMPUTER

The illustration of the computer screen above indicates the ease with which data may be entered into the data base. In color, it is even more indicative. By clicking on any of the headers, one selects the task to perform.

Music management provides the ability to:
Add a new song or recorded event to the library.
Edit a record previously entered.
Delete Song completely from the data base.
Delete Cassette completely from the data base.
Import data from another computer via floppy disk.
Export data from this computer to a floppy disk.
List to screen or printer.
Carry off or on
Pack, that is eliminate from the hard disk records deleted.

Cassette number is the number assigned to the particular cassette and identified in the user bit number recorded as part of the SMPTE code. The entire user bit code may be recorded to include 8 digits. The 4 digits not shown may be used to identify a particular syndicator or user.

Track number. 1 = digital; 2 FM (hi-fi). All "tracks" are stereo but for special application may be doubled in monaural.

SMPTE time code. Hours 1-9; Minutes 0-59; seconds 0-59; frames 0-29. Cassettes normally begin at one hour.

Title, Artist, Label self explanatory

Code may be used to categorize artist to program separation by artists in a broadcast format.

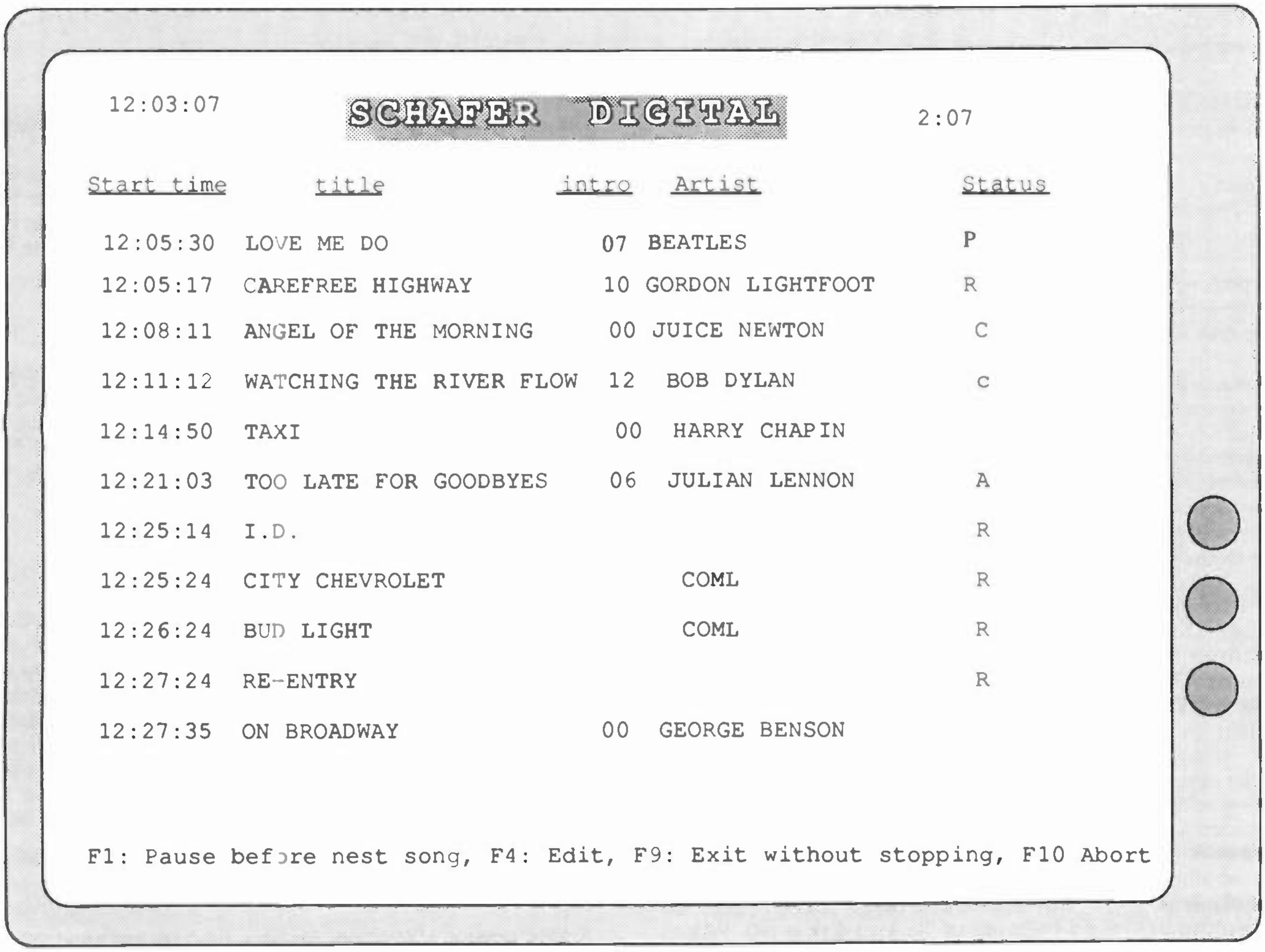
Introduction __ seconds indicates point at which vocal begins for use of disk jockey in live assist operation.

Tempo/Energy, Type, Format, Category, Position and/or other categories of descriptions of type of music used to define songs for particular positions in a broadcast format.

Digital indicates how a song was recorded, mastered and finally copied.

Licensing ASCAP, BMI, SESAC

Notes: Space for any additional information desired.



LIST MANAGEMENT

Under **List Management**, either **Playlists** or **Formats** may be selected.

Formats

Add a new format, that is a template describing the characteristics of each song in a given hour, for the purpose of preparing a playlist.
Edit an existing format.
Delete an existing format.
Pack, that is permanently remove deletions.

A radio station subscribing to a music service which provides a playlist would not need to design a format nor create music playlists. Those desiring to create their own will be assisted with the software provided.

Playlists

Edit play list or create a new playlist.
Copy a playlist to repeat or edit for later broadcast.
Import a playlist from a floppy disk.
Export a playlist to a floppy disk.
Merge a playlist with a commercial schedule.
Delete a playlist.

Creating a new playlist is accomplished by selecting a format. As each song is to be selected, pressing the F2 key will display the list of all songs which fit the criteria of the song selected for that position in the format. Variations of software are available for this function so we will not go into detail about these at this time.

Whether the playlist is created at the radio station or supplied by an outside source, it may be modified in the production computer. The playlist may be modified with the on-the-air computer prior to the playing of a song. Depressing the F4 key will cause to be displayed the on-the air playlist ready to be edited. The screen shown above illustrates the typical screen. Songs, except those already cued, may be cancelled (aborted), or re-arranged. Note the letters opposite the songs in the playlist above.
P= Playing; **R**= Ready; **C**= Cued; **c**= rough cued.A= Aborted

12:03:07 The time in the upper left hand corner of the screen is the current time.
2:07 The time in the upper right hand corner of the screen is a count-down to the end of the on-the-air event.
The times in the left hand column indicate the time each event will begin. Should the schedule be altered, the computer will immediately re-calculate the times.
Pressing the **F1** key will cause the system to pause following the on-the-air event.
Pressing the **F4** key will put the system in a mode to edit, to allow editing or re-arranging up-coming schedule.
F9 will exit the program without stopping the cassette playing.
F10 will instantly abort and kill the on-the-air program.

COMPUTER / VCR OPERATION

Music, commercials and other events are recorded in digital and FM on video cassettes, together with control pulses and SMPTE time code, to facilitate search and play as desired.

To prolong head and tape life, high speed search functions are performed in true fast forward and re-wind, at which time the tape does not come into contact with the video heads.

When the system is turned on, the first operation is the configuration of each VCR. One at a time, the computer checks the SMPTE time code on each video cassette, to determine that it finds good SMPTE code and that the correct cassette is in each VCR, according to the cassette schedule and the playlist. It then cues the next songs to be played. The pattern is always: rough cue the song after the next song, then cue the next song. The song is cued to 12 seconds before the actual start time of the song. 12 seconds before air time, the VCR is pre-rolled, the SMPTE is double checked, it is "jogged" to assure it is on the exactly correct frame 5 seconds before air time. The video switcher opens up 2 seconds before air time. The VCA fades the audio up at air time. As soon as a song is placed on the air, the next two are cued up in the same pattern.

In the case of the instant access short event hard disk system, the same on-the-air computer, checks ahead and prepares the commercials and other short events to be down-loaded to the hard disk. The computer calculates how much time is required for the down-loading of each commercial, and so long as there is sufficient time to down-load another, it will continue to down-load until the 30 minutes of the hard disk is full. When the hard disk is going to play one or more events on the air, down-loading is suspended until they have been aired and until sufficient time remains to down load more. The hard disk does not require pre-rolling. It is capable of instant start and instant access. Therefore commercials and other short events may be programmed to follow one another in any order desired.

The computer smoothly airs music, commercials and other scheduled events according to the schedule. Playlists are identified by each hour of the day. Events from the instant access hard disk are inserted from the commercial schedule which has been merged into the music playlist. The schedule of music and commercials is displayed on the computer screen. It is possible to change the schedule at the last minute,except for the next 2 songs that are already cued. Ever they can be aborted, if desired.

WHY BETA?

Extensive tests clearly indicated that the tape handling capability of the BETA VCR is far superior to other types of VCR's including VHS and 8 mm. The video cassette was chosen because of the ability to record and reproduce without problems from one VCR to another. The 16 bit, 44.1 KHz format assures quality equal to the best compact disk.

WON'T BETA GO AWAY?

The recent announcement of Sony has raised the question. Sony will make VHS recorders. Sony will continue to make and support Beta VCR's. Sony VHS recorders will be the less expensive ones. Sony has discontinued low end of the line Beta VCR's. They are already adding to the top end of the line Beta VCR's. Sony continues to be dedicated to provide the finest quality VCR's in Beta.

WHEN IS DIGITAL NOT DIGITAL?

There is more than one digital format. To achieve quality equal to the quality of the best compact disk, it is important that the digital format be of 16 bit (There are 8 bit formats, as an example.) and the sampling frequency be more than 44 KHz.) There are 32 KHz formats, as an example. All are digital, but it is generally considered that to achieve "perfect" quality, the 16 bit, 44.1 KHz format must be employed.

WHY NOT R-DAT?

R-DAT is true 16 bit digital and is good quality. The R-DAT cassette is limited to 2 hours, thereby precluding the design of a cost-effective system to store the complete music library in a system, making possible complete flexibility in programming and virtually unlimited walk away time without ever repeating a "program". There is some question as the repeatability of quality from one recorder to another and a question about tape life. I think R-DAT is a viable system and will serve the industry in many ways, not the least of which will be in recording of news.

WHO WILL PROVIDE MUSIC IN DIGITAL?

At this writing, at least one music syndicator has announced a program service designed specifically for the Schafer Digital System, that is EIAJ format, 16 bit, 44.1 KHz, with SMPTE time code. Announcements regarding two other services are expected in the near future.

WHAT ABOUT TAPE AND HEAD WEAR?

In actual operation employing standard grade L-750 cassettes, playing video, no noticeable degradation was apparent after the equivalent of 3 years of playing the same cassette in the same VCR. Heads should be cleaned periodically, of course. The error correction inherent in the EIAJ format, corrects for the occasional dropout. The resultant quality is truly incredible.

IS DIGITAL THE FUTURE IN RADIO BROADCASTING?

Absolutely. I think the Schafer Digital System is the future, but then I'm prejudiced.

Paul Schafer, known as the father of automation founded Schafer Electronics in 1953, Schafer International in 1969 and started the **Schafer Digital** project in 1986.
Schafer Digital/Schafer International is located at
5801 Soledad Mountain Road, La Jolla, CA 92037
Telephone 619 456 8000 FAX 619 456 1350 Telex 697939

COMPARING FM TRANSMISSION SYSTEM PERFORMANCE AND RECEIVER CAPABILITIES: HOW GOOD MUST YOUR STATION BE?

Jerry Whitaker
Broadcast Engineering Magazine
Overland Park, Kansas

AUTHOR'S NOTE:

The author wishes to thank the following individuals and companies for contributing equipment that made preparation of this paper possible.

- Bob Metzler, Audio Precision
- Howard Mullinack, Orban Associates
- Jesse Maxenchs, TFT
- Steve Claterbaugh, Continental Electronics,
- the staff of Brandsmart (a consumer audio/video outlet in Overland Park, KS).

The radio industry has made great strides in recent years toward the realization of a transparent medium through which programming can flow to listeners. And now, more than ever before, the need for excellence in all phases of station operation is becoming painfully obvious to engineers and managers across the country. The public is more discriminating, and competition has never been tougher.

The consumer has demonstrated a strong desire for high quality music programming. Compact disc players are gaining wide acceptance in the marketplace and consumers are beginning to judge the performance of all audio systems against the CD. That's a tough act to follow.

Consumer FM stereo receivers have also achieved an impressive level of sophistication. In fact, some units can out-perform FM radio stations.

In order to compete, broadcasters need to keep current with the state-of-the-art. The first step in this process is to determine what the state-of-the-art is.

In an effort to identify just how good the current crop of FM receivers are, and thereby determine how good FM radio stations need to be, detailed audio performance tests were conducted by the author on a representative sample of 10 consumer receivers in the price range of $299 to $850. The measurements were made under the same conditions for all units tested.

Results of the study indicate that although FM receivers have reached an impressive level of sophistication, a well-maintained radio station using current-technology equipment can still out-perform the receivers used by most consumers today. However, the gap between broadcast quality and receiver quality is narrowing.

TESTING PROCEDURE

The tests were performed using the latest available broadcast hardware. To facilitate rapid and accurate measurements, an Audio Precision System One (AP/S1) automated audio test set was employed and used for all measurements. The stereo signal was generated with an Orban 8100-A FM Optimod (switched to the proof mode) and fed to a Continental Electronics 802-A exciter operating into a dummy load. Monitoring was accomplished with a TFT 844 modulation monitor.

A test procedure was programmed on the AP/S1 to measure the following parameters on each channel of the sample stereo FM receivers:

- Frequency response from 30Hz to 15kHz at 95% modulation.
- Total harmonic distortion (THD) at 95% modulation as a function of frequency (30Hz to 7.5kHz).
- SMPTE intermodulation distortion (IMD) as a function of amplitude (from 95% modulation to 15dB below 95% modulation).
- Separation from 30Hz to 15kHz at 95% modulation.
- Noise referenced to 95% modulation (400Hz reference frequency).

The receivers were fed by an over-the-air RF signal. The FM signal was provided by leakage from a coaxial cable (5-foot in length) that connected the exciter output to the 100W dummy load. The RF output of the exciter was adjusted to provide sufficient field strength to drive the receivers under test.

All receivers used only a wire dipole antenna connected directly to the back of the unit. The physical separation from the receiver antennas to the exciter-to-load cable was about 10-feet. It was found that a power output of 5W to 10W was sufficient to provide full quieting of the receivers.

On units that featured a front-panel signal strength indicator, the FM exciter was adjusted to provide a reading of approximately 75% of full scale. On receivers that did not feature a signal strength indicator, RF power output was adjusted to the point that -- when using the station scan feature on the receiver -- the unit would lock onto the test signal.

The modulation monitor was driven using the same approach. The 844 features a switch-selectable field strength meter, which was driven to about 75% of full scale.

No efforts were made to optimize any of the receivers for the tests. The measurements were taken at a suburban Kansas City stereo dealer showroom using receivers selected at random. The left and right tape-out ports were used to provide the input signal to the AP/S1 audio test set. Because the goal of the study was to characterize the level of performance typically achieved in the consumer's home, no special adjustments or optimization was conducted on any of the test receivers.

Baseline performance measurements were also taken on the broadcast equipment used for the test. The hardware was supplied to the author on a loan basis and aligned prior to shipment from the manufacturer's factory. No additional adjustments were made by the author.

Measurement parameters

The procedures used to make the measurements were simple and straightforward. They were patterned as closely as possible after the old FCC equipment performance measurements (EPM) for FM broadcasting.

The AP/S1 provides a generator amplitude regulation feature that permits EPM-type constant-modulation measurements, however, the procedure requires access to a signal source without de-emphasis. When using a modulation monitor, such a signal is readily available. Measurements on consumer receivers, however, do not permit access to a flat (no de-emphasis) demodulator output. In fact, to bypass the de-emphasis circuit in the receiver would have failed to check an important element of receiver operation.

The AP/S1, however, does permit the generation of test signals at amplitudes determined by pre-selected pre-emphasis or de-emphasis curves. The 75-microsecond de-emphasis curve was switched into the generator output to provide a modulating signal that would maintain approximately 95% modulation. This approach was taken after confirming the accuracy of the AP/S1 de-emphasis curve and Optimod 8100-A pre-emphasis circuit.

A total of 31 test points were made per sweep on each channel of the device under test (DUT). A total of 250 individual measurements were conducted on each DUT.

The load impedance on all DUTs was 100k-ohms, unbalanced. Baseline noise readings taken before the tests began confirmed that RFI or 60Hz hum were not a problem.

Automatic ranging of input signals was used for all measurements. The AP/S1 autoranging control circuitry responds to the peak value of the input signal, rather than the RMS or average value, preventing overload and non-linearity on signals with high crest factors.

Measurements of noise and separation were made with the generator outputs back-terminated in a resistance equal to the selected source impedance (600-ohms). This permitted measurements to be made without the necessity of disconnecting cables or connecting termination resistors to the inputs of the stereo generator.

Measurement of total harmonic distortion (THD) were made using true-RMS detectors, and a 22Hz high pass filter and 30kHz low pass filter switched into the analyzer. THD measurements were conducted up to and including 7.5kHz. Measurements were not made above this frequency because of the inherently invalid results that are produced when testing an FM transmission system for THD at frequencies above 8kHz. Because all stereo generators filter the input audio above 15kHz to 17kHz to protect the 19kHz pilot, even the second harmonics of 8kHz and higher frequencies are also filtered. Under these conditions, the distortion meter is basically reading residual noise and analyzer filter irregularities.

Reference readings

A series of baseline readings were taken before testing began on receivers to establish the fundamental performance levels of the assembled equipment. Figures 1-5 show the results of the tests, which can be assumed to represent the performance possible today from a well-maintained FM station using current-technology equipment.

Frequency response was within 0.2dB from 40Hz to 15kHz. As shown in Figure 1, both channels track well over the range of measurements. The furthest excursion from the 75-microsecond pre-emphasis curve occurs at 30Hz, which is down 0.5dB on both channels.

Baseline distortion was less than 0.3% from 30Hz to 7.5kHz. Figure 2 shows that THD was well below 0.2% at frequencies less than about 3.5kHz.

Intermodulation distortion at 95% modulation measured 0.3%. Figure 3 charts IMD as a function of modulating level. The chart illustrates the effects of noise on the measurement, reaching 0.45% IMD at 15dB below 95% modulation.

Stereo separation tracked well between the left and right channels. Low frequency separation was, at worst, approximately -40db, improving to about -54dB at the high end of the audio passband. Figures 4 and 5 show the baseline stereo separation performance for each channel of the test set-up.

The measured signal-to-noise ratio was -61dB for each channel (with de-emphasis). Because of the types of measurements conducted in this study, de-emphasis was used on all baseline measurements. As mentioned previously, tests on consumer FM receivers -- by definition -- requires the use of de-emphasis.

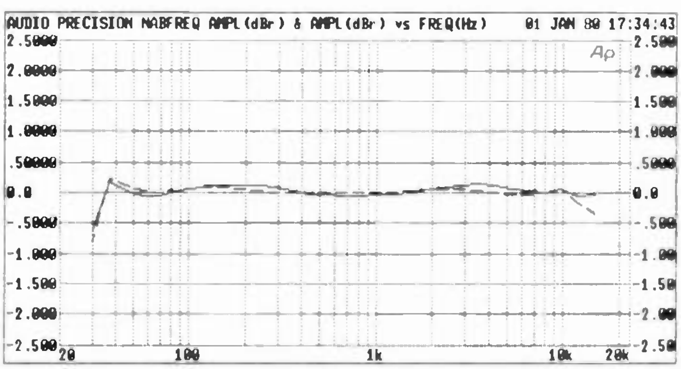


Figure 1. Baseline frequency response for the right and left channels of the test set-up used to examine the receivers. The solid line represents the left channel, and the dashed line represents the right channel. The printout shows net deviation from the 75-microsecond pre-emphasis curve.

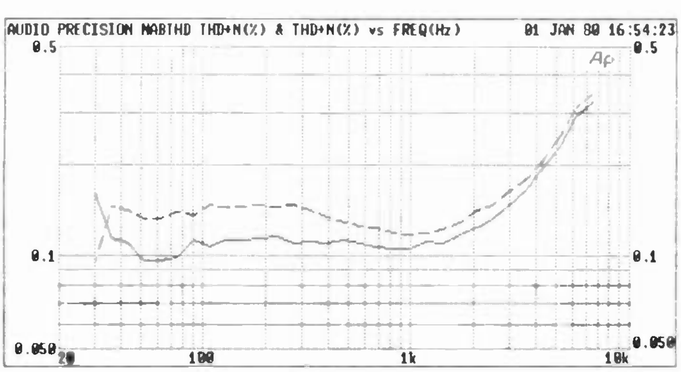


Figure 2. Total harmonic distortion (plus noise) as a function of frequency for the test hardware. Note that all THD components are below 0.3%.

The factory test performance data supplied with each piece of equipment used in the measurements were significantly better than the measurements documented in Figures 1-5. The explanation comes from the method chosen to perform the tests. Because an off-the-air test routine was required to check receiver performance, the same method was used to make the baseline measurements.

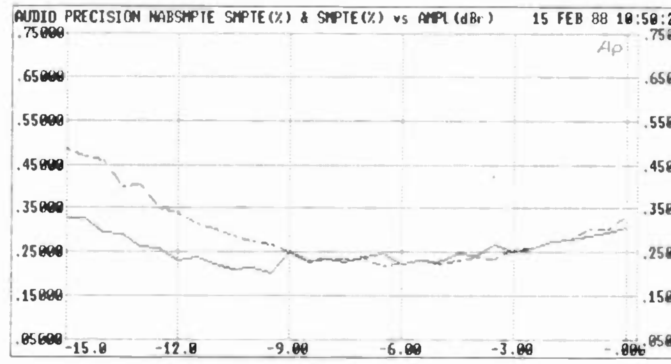


Figure 3. Baseline SMPTE intermodulation distortion as a function of modulation level. 0dB on the chart represents 95% modulation.

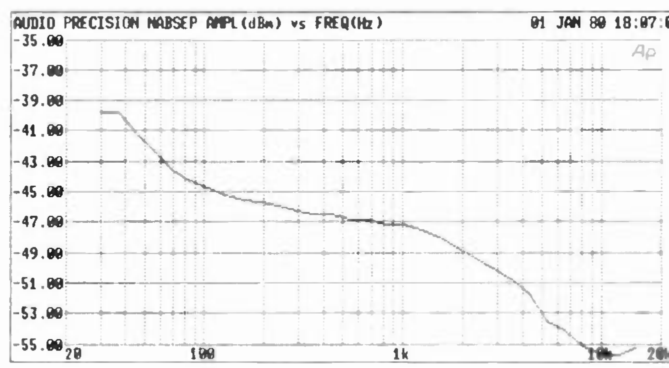


Figure 4. Left channel separation for the test set-up. Tracking between left and right channels was well within 1dB.

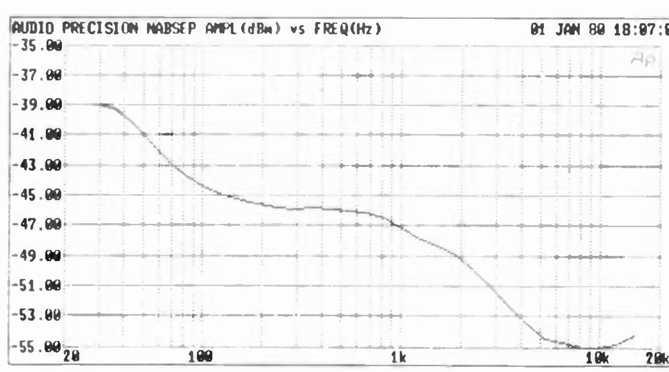


Figure 5. Baseline right channel separation. The baseline readings for the parameters documented in Figures 1-5 were taken just before the FM receiver tests were conducted.

The goal of the study was not to establish the performance level of broadcast equipment or consumer receivers, but rather to assess the relative quality of each. All tests contained in this report should be viewed from that perspective.

Test results

FM broadcasting has certain limitations that prevent it from ever being a completely transparent medium. Top on the list is multipath distortion. In many locations, some degree of multipath is unavoidable. Running a close second are the practical audio bandwidth limitations of the FM stereo multiplex system. The theoretical limit is 19kHz, however, real-world filter designs result in a high-end passband of between 15kHz and 17kHz.

Receiver IF bandwidth is another limitation placed on the FM system. The problem involves adjacent channel interference that is usually hidden by narrowing the receiver IF bandwidth. This effect can be demonstrated using a tuner that features switch-selected IF bandwidth. Many stations are not listenable in the "wide" mode because of interference. But, when the receiver is switched to the "normal" mode, most stations are reasonably clean, and well within the practical limitations of current broadcast technology. Often, a clearly audible reduction in high frequency distortion and noise is noted when switching from "wide" to "narrow."

Some receivers do better than others in dealing with these inherent limitations. The series of tests documented in this report were conducted to establish the current state-of-the-art in consumer receivers, and thereby establish the performance requirements for FM radio stations. Table 1 lists the overall performance the receivers tested.

Because of the large number of readings taken on each piece of equipment (250 for each receiver), averaging of most data was required to provide realistic numbers for meaningful comparison. It should be emphasized that the performance of an individual unit is not the primary concern of this report. The intent, rather, is to identify basic performance levels. The approximate list price of the units tested is given to roughly classify the intended market for the receiver.

Table 1. Measured performance of a representative sample of consumer FM receivers. All units were tested using the same parameters in the same environment.

TABLE 1

DUT	FREQ.RESP.	THD	IMD	SEP/L	SEP/R	S/N
BASELINE	+/- 0.2dB	0.3%	0.3%	-41/-55dB	-39/-53dB	-61/-61dB
RECEIVER 1 (Yamaha T-85 tuner-only, $450.00	+/- 0.2dB	0.3%	0.1%	-37/-52dB	-37/-50dB	-63/-71dB
RECEIVER 2 Onkyo TX-82 receiver, $359.00	+/- 0.3dB	4.0%	0.8%	-17/-28dB	-17/-28dB	-41/-41dB
RECEIVER 3 Carver 150 receiver, $850.00	+/- 0.2dB	1.0%	0.9%	-45/-55dB	-43/-55dB	-58/-57dB
RECEIVER 4 Sony STR AV-780 receiver, $750.00	+/- 0.5dB	1.5%	0.5%	-39/-53dB	-39/-52dB	-50/-50dB
RECEIVER 5 Onkyo TX-82 receiver, $459.00	+/- 0.5dB	2.0%	0.7%	-41/-47dB	-41/-46dB	-53/-53dB
RECEIVER 6 Yamaha RX500U receiver $420.00	+/- 0.3dB	1.9%	1.5%	-35/-41dB	-35/-41dB	-38/-38dB
RECEIVER 7 Onkyo T4150 tuner-only, $299.00	+/- 0.3dB	2.0%	0.7%	-35/-41dB	-35/-41dB	-47/-48dB
RECEIVER 8 Yamaha RX-700U receiver, $589.00	+/- 0.5db	1.8%	0.9%	-35/-53dB	-33/-53dB	-48/-49dB
RECEIVER 9 Carver 900 receiver, $635.00	+/- 1.0dB	1.9%	0.6%	-49/-57dB	-45/-54dB	-58/-58dB
RECEIVER 10 Technics SA-937 receiver, $315.00	+/- 1.5dB	7.5%	4.0%	-24/-42dB	-24/-43dB	-43/-42dB
COMPOSITE	**+/- 0.5dB**	**2.4%**	**1.1%**	**-36/-47dB**	**-35/-46dB**	**-50/-51dB**

NOTES:

1. Measurement categories are defined as follows:

DUT	Device under test.
FREQ RESP.	Typical measured frequency response of both channels from 30Hz to 15kHz, reference to the 75us pre-emphasis curve.
THD	Typical measured total harmonic distortion of both channels over the frequency range of 30Hz to 7.5kHz at 95% modulation.
IMD	Typical measured intermodulation distortion of both channels at 95% modulation
SEP/L	Left channel separation with the right channel modulated at 95% with tones from 30Hz to 15kHz. The first reading is the lowest separation reading recorded, and the second reading is the highest separation reading recorded.
SEP/R	Right channel separation with the left channel modulated at 95% modulation with tones from 30Hz to 15kHz. The first reading is the lowest separation reading recorded, and the second reading is the highest separation reading recorded.
S/N	Measured signal-to-noise reading referenced to a 400Hz tone at 95% modulation. The first reading is the left channel noise figure, and the second reading is the right channel noise figure.
COMPOSITE AVERAGE	The average reading in each category for all 10 receivers tested.

It must be noted that price alone does not characterize the expected performance level of the FM section of a receiver. The features offered to consumers vary widely depending on the manufacturer. Variables include the number of inputs, type and sophistication of equalization provided and -- most importantly -- the power output of the stereo amplifier section. It is, therefore, dangerous to classify a consumer FM receiver based on price alone.

The quality of the power amplifier section of receivers tested was not a factor in the measurements conducted for this report because the output signal from the receivers was taken from the tape-out port, bypassing the power amplifier.

Virtually all receivers tested performed well with regard to frequency response. No significant problems were noted on any units. The greatest deviation measured from the 75-microsecond pre-emphasis curve was +/- 1.5dB.

Total harmonic distortion, however, as another matter. Some of the receivers had serious problems with THD at high modulating frequencies. Many of the units performed well with regard to THD up to about 2.5kHz. Receiver 2, for example, measured about 1% THD from 30Hz to 2.0kHz. Above 2.5kHz, however, distortion rose rapidly and reached 4% at 7.5kHz. All of the receivers experiencing excessive THD did so between 2.5kHz and 7.5kHz.

Intermodulation distortion (SMPTE 4:1) measurements produced some rather impressive figures on the receivers tested, especially when compared to THD performance. Most IMD figures were below 1%. The data given for IMD reports performance with the test signal modulating the exciter to 95%. At lower levels of modulation, IMD rose significantly on most units tested, apparently because of the effects of the noise floor.

There was one exception to the IMD rule, however. Receiver 10 had its best IMD performance (1.5%) at 10dB below the operating level and rose to about 4% at operating level (95% modulation).

Separation figures for both left and right audio channels tracked closely. It was unusual to see more than 1dB difference between channels. Separation performance was dependent on the modulating frequency, with figures on most receivers lower at low frequencies and higher as frequency increased. It was common to see separation numbers between -35dB and -45dB at mid-band.

Signal-to-noise performance ranged from excellent to poor. One receiver's measured S/N reading was only -38dB for each channel. At the other end of the spectrum, however, another unit (receiver 1) provided an outstanding showing of -63dB for left channel noise and -71dB for right channel noise. The author is at a loss to explain the 8dB difference between the two channels, especially in view of how closely the other measurements on the receiver tracked.

The S/N measurements also provided an interesting paradox. In three instances (receivers 4, 6 and 8), the measured S/N was several dB worse than the measured stereo separation for those same units. It is reasonable to assume that the noise floor would place a limit on the separation figures that could be measured. In any event, the numbers shown are those recorded by the test equipment.

Full documentation was taken on all receivers measured. Figures 6-10 show the measured data of the top-performing unit (receiver 1). The unit performed well with respect to frequency response, as shown in Figure 6. Particularly impressive were the tuner's THD and IMD performance, graphed in Figures 7 and 8. No other unit tested exhibited such low distortion and, in the case of IMD, linear distortion with regard to modulating amplitude. Channel separation performance (shown in Figures 9 and 10) was impressive, reaching as much as

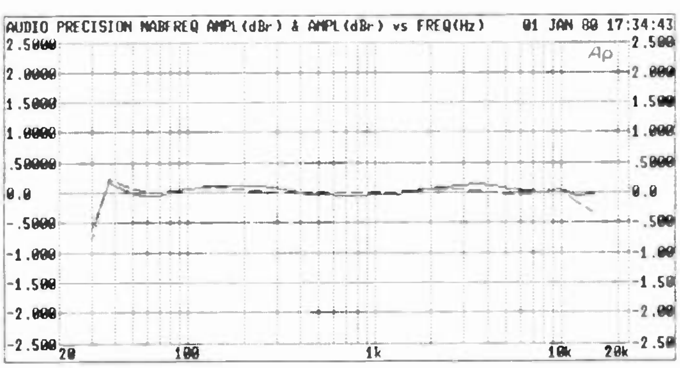


Figure 6. Frequency response relative to the 75-microsecond pre-emphasis curve for test receiver 1. Most receivers measured showed adherence to the pre-emphasis curve of within +/- 0.5dB.

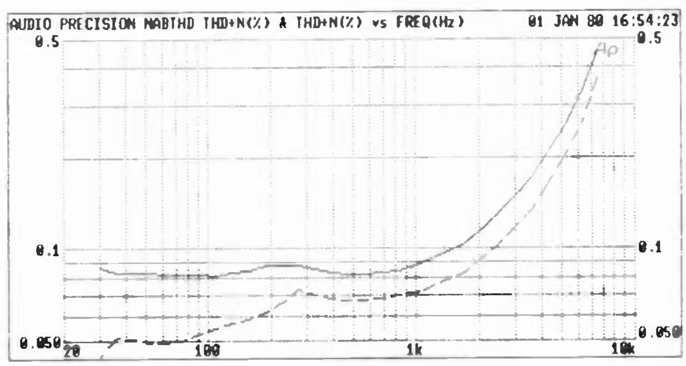


Figure 7. THD performance of test receiver 1. Note the exceptionally low distortion below 2kHz. This receiver, like most tested, exhibited sharply rising distortion at frequencies above 2.5kHz.

-55dB at about 10kHz. As mentioned previously, the S/N performance of the receiver was remarkable (-63dB and -71dB).

Receiver 1 and a couple of others tested featured a front-panel selectable bandwidth control with "wide/normal" positions. If selectable by the consumer, the control was placed in the "wide" mode. Whether or not the consumer could use the receiver in that mode, given the field strength of the desired station, is another matter. It was felt, however, that use of the "wide" mode would provide the data needed on what constitutes the current state-of-the-art in FM receiver development.

So far as the author knows, no dynamic noise reduction (DNR) circuits were active in any of the receivers during the tests. Most receiver designs place the DNR circuits (if used) in the pre-amplifier section of the power amplifier, which was bypassed by using the tape-out ports.

The measured performance of receiver 1 demonstrates the sophistication that is available to consumers in the marketplace. And although few other receivers tested came close to the performance of that unit, most did quite well overall. Broadcasters cannot assume that the receivers being used by consumers are the limiting factor in the delivery of programming to the public. Quality is a moving target, and FM radio stations need to stay ahead in the race.

TRANSMISSION SYSTEM PERFORMANCE TARGETS

The receiver performance tests documented in this report demonstrate that to remain competitive in the marketplace today, a high level of audio quality is required of FM stations. To that end, a set of recommended performance targets is proposed. Those targets are based on three fundamental constraints:

- o Practical fidelity requirements,
- o inherent system limitations, and
- o the need for effective average modulation levels.

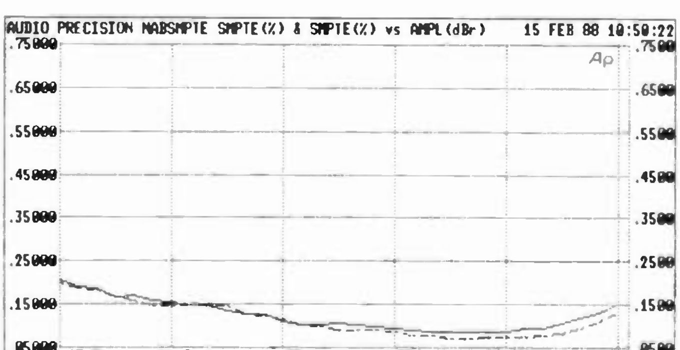


Figure 8. Test receiver 1 SMPTE IMD as a function of modulating level. This receiver exhibited the most uniform IMD performance across the modulating range of any unit tested.

It is much easier to engineer a loud station than it is to fashion a loud and clean one. Cleaning up the signal without losing level is a much more complicated task than simply turning down the processing. It starts with the cleanest, flattest possible transmission system. In the end, a systems design approach, involving everything from the tape heads and microphones to the antenna is required.

The performance targets suggested by the author are tough, but they are achievable. Table 2 lists the recommended performance levels and measurement procedures. Even though some of the targets may be tighter than the manufacturer's specs on some individual links in the system, factory specifications are usually conservative with regard to product performance. If a component of the transmission chain, when tested by itself, does not make the grade, replacement (or at least maintenance) should be considered.

Don't overlook the modulation monitor when aiming for high performance. Although the mod monitor has no effect on how the air signal sounds, it is the reference by which the entire system is measured and adjusted. If the monitor provides less than optimum per-

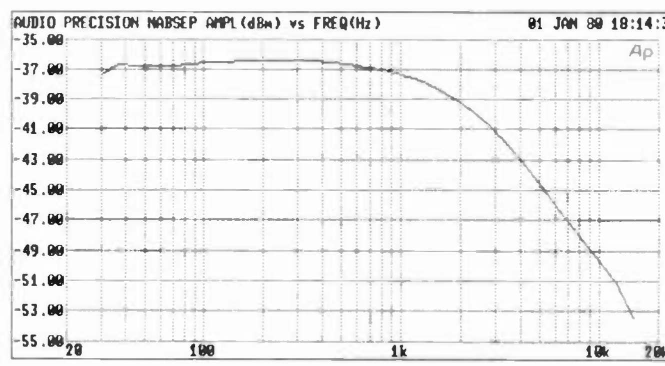


Figure 9. Left channel separation as a function of frequency for test receiver 1. Because of de-emphasis, most receivers exhibited their best separation performance at frequencies above 2.5kHz.

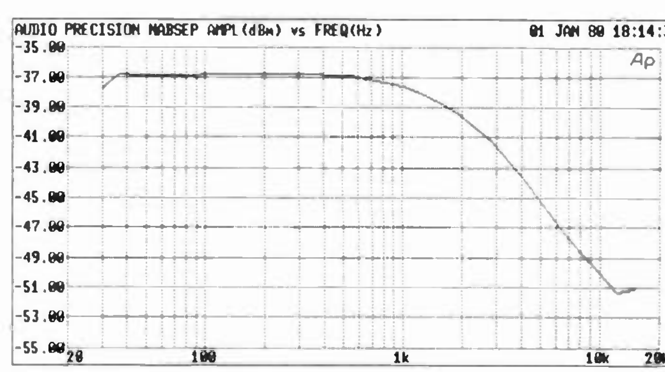


Figure 10. Right channel separation for receiver 1 as a function of frequency. Tracking between channels during separation measurements was within 0.5dB.

TABLE 2

GENERAL TEST CONDITIONS

- Switch system to stereo mode.
- Apply input signals to the console line input used for most program sources.
- Sample and demodulate the system output at the transmitter antenna output.
- Leave all processing and equalization in-line.
- Define the operating level as 0VU or equivalent at the main audio console.

FREQUENCY RESPONSE MEASUREMENTS

Test conditions:

- Switch the AGC voltages off. Do not simply patch around the audio processor. Unfortunately, not all processors provide this feature. In such cases, patch around the unit.
- Set modulation for any convenient level between 75% and 100%. It is suggested that the normal modulation level produced by feeding a 400Hz tone into the system be used as the reference modulation level.
- Adjust the input level as required to maintain the reference modulation at each frequency used.
- Calculate the response error as the input level deviation required to maintain reference modulation, compared to the 75-microsecond pre-emphasis curve.

Recommended performance:

- +/- 1dB 30Hz to 15kHz
- +/- 0.5dB 50Hz to 15kHz
- +/- 0.2dB 100Hz to 10kHz

DISTORTION MEASUREMENTS

Test conditions:

- Switch AGC voltages on, if appropriate (see the text). Otherwise, switch AGC voltages off. Do not simply bypass the unit unless absolutely necessary.
- Set modulation for any convenient level between 75% and 100%. It is suggested that the normal modulation level produced by feeding a 400Hz tone into the system be used as the reference modulation level.
- Adjust the input level as required to maintain the reference modulation level at each frequency measured.
- Switch the monitor de-emphasis in.

Recommended performance:

- THD = 0.3%, 30Hz to 7.5kHz
- IMD = 0.3%, 60Hz and 7kHz, 4:1 (SMPTE IMD)

NOISE MEASUREMENT

Test conditions:

- Measure noise at each stereo audio channel output with all processing equipment in-line and adjusted for normal operation.
- Reference the noise measurement to the output level produced by a 400Hz input signal at 0VU at the console.
- Take the measurements unweighted, with de-emphasis switched in.

Recommended performance (each channel):

- -60dB

SEPARATION MEASUREMENT

Test conditions:

- Set modulation for any convenient level between 75% and 100%. It is suggested that the normal modulation level produced by feeding a 400Hz tone into the system be used as the reference modulation level.
- Adjust the input level as required to maintain reference modulation at each frequency used.
- Switch the AGC voltages off. Do not simply patch around the audio processor. Unfortunately, not all processors provide this feature. In such cases, patch around the unit.
- Terminate the unused input channel with a 600-ohm wirewound resistor (or other appropriate value resistor).
- Measure residual leakage into the other stereo audio channel.

Recommended performance:

- -40dB, 400Hz to 15kHz
- -35dB, 30Hz to 400Hz

Table 2. Recommended test procedures and performance targets for FM radio stations. These measurements are designed to simulate as closely as possible real-world operating conditions.

formance, consider replacement or maintenance to bring it up to spec. The monitor, by itself, must be capable of residual distortion of less than 0.1% and a S/N performance of at least 70dB when using the high level RF (transmitter sample) input.

The recommended equipment performance measurement (REPM) test conditions are designed to simulate as closely as possible the normal operating conditions of an FM radio station. Although sampling of the transmission system at the output of the transmitter is recommended, a high-quality off-air demodulator is preferred, if available. This approach provides the advantage of taking transmitter and antenna bandpass irregularities into account in the measurements. The demod must, however, be very flat to avoid invalid results.

Frequency response targets

Absolute frequency response accuracy over the audio passband does make an audible difference. Researchers exploring subtle differences in audio amplifier designs have found that errors as small as 0.2dB can be heard. Therefore, flat response (strict adherence to the 75-microsecond pre-emphasis curve) is reflected in the performance targets. The recommended frequency response limits are +/- 1dB, 30Hz to 15kHz, +/-0.5dB 50Hz to 15kHz, and +/-0.2dB, 100Hz to 10kHz.

Because most musical content is in the 100Hz to 10kHz range, tighter specifications are recommended for this band. With the equipment available today to broadcasters, there is no reason that an FM system cannot be absolutely flat over this range, and in view of how critical flat response is to overall fidelity, it pays to optimize here.

Somewhat looser tolerances are specified at the frequency extremes in recognition of the practical high-pass and low-pass filter considerations of stereo generator design.

Distortion targets

It is preferable to make the distortion measurements with the AGC (and limiter) voltages switched on. This simulates the real-world operation of a radio station. Excessively fast attack-time constants will produce low-frequency and IM distortion in older limiter designs, and excessive high-frequency clipping will -- obviously -- increase high frequency distortion.

It must be noted that, depending upon the set-up of the audio processor, the distortion targets of 0.3% THD (30Hz to 7.5kHz) and 0.3% IMD are probably impossible to meet with the AGC voltages switched on. Because of the highly competitive nature of broadcasting today, audio processors cannot always be adjusted to provide for the purest reproduction of the incoming signal. While this is regrettable, it represents the real-world that engineers must deal with.

When making the distortion measurements, take a set of readings with the AGC and limiter engaged, and then with the audio processor in the proof mode. Save both sets of readings for reference. This way, at least you will know the effects of your audio processing.

It is fair to point out that listeners do not listen to test tones, but to program material. With this in mind, it is not too difficult to rationalize audio processor settings that produce steady-state distortion in the range of 1%-2.5% at mid-to- high frequencies (a common occurrence).

The recommended THD test frequencies are kept low enough (7.5kHz maximum) that at least the second harmonic of the input signal will fall within the system's 15kHz passband.

Although the IMD tests are relatively impervious to system noise at full modulation, THD tests are limited by the noise floor. It is important, therefore, to get the noise floor as low as possible, so that low levels of THD can be read.

THD and IMD tests alone do not check dynamic instability problems like transient intermodulation distortion (TIM), however, careful selection of high slew rate components in the audio chain and THD/IMD figures in the noise floor will leave an audiophile audience impressed.

Noise targets

In many cases, system noise is the most difficult parameter to bring under control. The recommended performance target of -60dB per channel reflects state-of-the-art exciter/transmitter performance (about -66dB baseband noise) and assumes that the noise contribution of other elements of the audio signal chain is minimal. Referenced to 100% modulation, -66dB noise at the transmitter means -60dB out of each audio channel. If the audio chain noise is kept down to -70dB or lower, the overall S/N reading for the system will be close to -60dB.

Because the recommended noise measurements are performed with the audio processor switched to the operate mode, it is important that all equipment preceeding the compressor have a low noise floor. Any residual noise in the audio console, STL or other components in the chain will be boosted by the amount of compression typically delivered by the processor.

Although 60dB of dynamic range doesn't sound too impressive in this age of compact disks, it is important to keep two facts in mind. First of all, limited dynamic range isn't limited at all unless the program input signal exhibits greater dynamic range. Most program material in most formats stays within a 20dB range most of the time.

Another key factor is that apparent loudness continues to increase as the threshold of limiting is exceeded and compression begins. The limiter may present a peak modulation barrier, but loudness pushes ahead as density increases.

Separation targets

The separation test is performed in the traditional manner by feeding tones into one channel while measuring leakage into the other channel (whose input should be terminated with a 600-ohm wirewound resistor, or other appropriate value). The target readings are -35dB, 30Hz to 400Hz, and -40dB, 400Hz to 15kHz.

The low frequency separation recommendations are looser than targets for mid- and high-frequencies in recognition of the non-directional acoustic properties of long audio wavelengths. In the mid- and high-frequency ranges greater separation is recommended to preserve stereo imaging. Program sources rarely provide greater than 30dB of separation, so an additional 10dB of separation headroom is recommended to be sure that the transmission system is not the limiting element.

Test equipment

For any test measurements to be of value, the test equipment used must be carefully selected and accurately calibrated. The following instruments will be required to correctly run the measurements recommended:

- A low distortion audio signal generator with a metered output and calibrated attenuator. Distortion must be below 0.1% at all frequencies and output levels to be used in the tests.
- A distortion analyzer capable of measuring THD and SMPTE IMD to an accuracy of at least 0.1%.
- An audio voltmeter capable of accurately measuring signals to at least -70dBm. This function is usually provided on distortion analyzers. Frequency response linearity across the audio frequency band must be within 0.1dB (20Hz to 20kHz).
- A properly calibrated FM stereo modulation monitor. Because measurements will be made with de-emphasis, the accuracy of the built-in de-emphasis circuits must be verified and documented.

Before attempting to run the REPM tests, check the audio generator and distortion analyzer frequency response and residual distortion at all frequencies of interest. Confirm that response is flat to within at least 0.1dB from 20Hz to 20kHz, distortion is below 0.1% for all frequencies to be measured, and the noise floor of the distortion analyzer/audio voltmeter is at least -70dBm. If adjustments or repairs are indicated in the closed-loop tests, make them before attempting to take any measurements.

The residual test equipment distortion values may not be subtracted from the total system distortion figures obtained when running the actual REPM. Subtracting test instrument residual distortion is not a valid procedure because distortion components do not necessarily add. In fact, the only time they add is when all of the harmonics are in-phase, a near impossibility when you consider that this would have to be true for every modulating frequency used during the measurements.

How many frequencies?

To accurately evaluate a broadcast transmission system, the performance of the equipment must be checked at a sufficient number of discrete frequencies. It is recommended that 28 separate frequencies be measured between 30Hz and 15kHz. These points are based on 1/3-octave ISO (International Standards Organization) center frequencies, with three minor modifications. The measurement frequencies are shown in Table 3.

The lowest frequency to be measured is 30Hz. The actual ISO frequency is 31Hz. All other frequencies are standard ISO centers, except 7.5kHz (the standard ISO frequency is 8kHz) and 15kHz (the standard ISO frequency is 16kHz). These modifications to the ISO 1/3-octave center frequencies provide compatibility with the key frequencies specified in the old FCC EPM tests. This provides a measure of comparison between data

TABLE 3

FREQUENCY	FREQUENCY
30Hz	800Hz
40Hz	1.0kHz
50Hz	1.25kHz
63Hz	1.6kHz
80Hz	2.0kHz
100Hz	2.5kHz
125Hz	3.15kHz
160Hz	4.0kHz
200Hz	5.0kHz
250Hz	6.3kHz
315Hz	7.5kHz
400Hz (reference)	10.0kHz
500Hz	12.5kHz
630Hz	15.0kHz

Table 3. Recommended frequencies for measurement of frequency response, separation and distortion of FM broadcast systems.

taken now and that taken in previous audio proof-of-performance tests.

If an automated audio test set is available, it is -- obviously -- desirable to use swept frequency measurements. Automated systems provide the user with improved flexibility in conducting audio measurements, and greater information on performance of the overall system.

Audio processing considerations

Every station engineer and program director will have their own opinion as to what the optimum processor input level (and other settings) should be. The author is not so presumptuous as to suggest a set of compression figures that will suit different stations programming different formats. However, some thoughts about high compression ratios are in order.

If 0VU on the console is just at the threshold of limiting (under these conditions, 6dB to 10dB of compression will be indicated with program material), a 7.5kHz input signal will be compressed by nearly 12dB because of pre-emphasis. If the input level is increased 10dB, 22dB of compression will result. Most systems should still provide fairly low distortion at 22dB compression. At levels above this point, however, the signal will likely get into the safety clippers. The resulting distortion will not buy you additional loudness, but listener fatigue instead. In the case of audio processing, sometimes less is more.

It is an unhappy fact of life that audio processing of any type involves a tradeoff between loudness and distortion. In general, you can trade one for the other. The current generation of audio processors make the tradeoffs slight and generally acceptable. However, each station should know the cost -- if any -- of its on-air "sound." The best way to gain this information is to conduct before-and-after measurements of all key audio parameters, with the processing switched-in and then switched-out.

THE FUTURE FOR FM RADIO

The future holds many exciting possibilities for radio broadcasters, including digital-based program source equipment, computer-controlled transmission gear and improved consumer receivers. If broadcasters are to hold their positions in the marketplace, they must move with the times. Those who fall behind will find that new technologies and more aggressive competitors have walked away with their audiences.

The broadcast industry is faced with unprecedented challenges from alternative programming sources and new technologies. Stations can only compete with these services by delivering to their audiences top-quality programs through top-quality transmission systems. Excellence in broadcast audio is an expensive and time consuming enterprise, but one that can pay handsome dividends.

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A DIFFERENT APPROACH TO THE OLD PROBLEM OF AUDIO LEVEL METERING

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ABSTRACT

Years of Hands-On-Experience with both VU and PPM metering have led to the conclusion that neither provides the true representation of the program material.

This has led to the development of a meter which addresses a relationship between average loudness and peak excursions of the composite waveform on one simple easy-to-read scale.

HISTORY

In the development of any new product, it is always advisable to consider history.

Both the VU and PPM meters were conceived in the late 1930's as solutions to the need for standardizing metering.

Various devices had been used for indicating the amplitude of the broadcast signal here in the United States, but it was not until the VU meter became a standard in 1939 that there was any common reference meter.

The development of the VU meter was the result of a joint effort on the part of CBS, NBC and Bell Telephone Labs to provide a common correlation of volume level. The VU meter still functions today in that service.

It is suggested that those interested in the background and the history of program level metering review the IRE paper "A New Standard Volume Indicator and Reference Level" by H.A. Chinn, D.K. Gannett and R.M. Morris in the proceedings of January 1940.

This paper points out that while complex and non-sinusoidal waveforms can be measured in predictable numerical terms using peak, average or RMS techniques, the results, using the examples of speech and music found in average program material, could no longer be expressed in those same simple terms. What was needed was a practical method of measuring and expressing the magnitude of program material with a simple numerical indication.

These considerations led to the conception of a value known as "Volume". This defined value was considered as purely empirical to meet a practical need. The term "Volume" was then defined in terms of a reading on an instrument known as a Volume Indicator.

Simply, the basic requirements to the development of the VU instrument were to indicate suitable operating level, check for transmission gains or losses and provide for reasonable indication of comparative loudness. It was further agreed that the indicator bear some ballistic relationship to the sound of the program material.

Of particular interest is that consideration was given to the preference for peak or rms indication, as at the time of the development of the VU meter, the peak reading instrument was already in favor by European engineers.

Because it was felt that the indicator should bear some relationship to the cadence of the program material, an averaging type metering instrument was favored.

The reference level of +4dBm was specifically chosen to be 10dB below the peak levels of average speech which could be accommodated by program circuits of that time.

This is where design philosophy differed from the European approach. The ballistics chosen for the PPM device favored a faster integration time to display any peak energy which would not pass through the broadcast chain without undue coloration from clipping. The slower release time was chosen to relieve the potential of an oscillating pointer.

Thus, the major difference between the VU and PPM is the time and manner in which the complex wave is integrated.

Both devices consider potential distortion of the circuit by peak amplitudes, but do so in a different manner. While the VU is referenced with respect to an acceptable level of circuit capability, the PPM is referenced with respect to an acceptable level of clipping.

Therefore, it is reasonable to expect that an average level difference between these two meters can be predicted. This is brought forward in the AES paper "VU Meter and Peak Program Meter - - Peaceful Coexistance" by Stephen F. Temmer presented at the 63rd Convention May 1979. The investigations in this paper show a spread of 6 to 8dB between the two meters on significant peaks, with the consensus that 6dB was reasonable to expect as an acceptable differential.

Thus, the usage of the device becomes the major factor. The VU has found favor in applications where the requirement for the integration of various program material prevails, and the PPM has found favor where the requirement is to adjust level with respect to peak modulation.

OBSERVATIONS

Each device has its own shortcomings. The VU meter assumes that peak excursions of composite waveform do not exceed the reference level by more than 10dB. However, there are techniques used in modern studio recording which exhibit peak amplitude excursions of short integration times which often exceed the limitations of the circuitry which pass unnoticed by the VU meter.

Conversely, the PPM meter makes the assumption that the peak to average in program content is a relative constant. While the ballistics of the PPM capture those amplitudes in program material of short duration, the longer release time of the PPM has the attendent tendency to mask the important comparative average levels or volumes. This, of course, presents problems when inserts into a program differ in average volume or listening level with respect to peak amplitude such as found in the relationship of compressed to un-compressed program material. Program gain adjustments made with respect to peak meter indications will vary in listening level by the peak to average ratio in program content.

THE NEED FOR A NEW CONCEPT

Conclusions obviously indicate that each of these two standardized meters is viable in its own area of usage. Each meter has its own advantages and disadvantages.

The need is for an instrument which indicates both the true peak of the waveform as well as its relationship to the quasi-average volume level.

HOW CAN WE ACCOMPLISH THIS?

Clearly, there appears to be a need to capture and display the peak amplitude of the signal. From this display it is possible to ascertain that program operation is within the parameters of the system. However, the development of the VU meter concludes that simple peak metering is not acceptable. It was also learned from that development, that a ballistic relationship to cadence is desirable. Because of the dynamic relationship between the peak and average parameters there becomes a need to display both so that the relationship can be viewed.

The concept of placing VU and PPM meters on a single panel provides some indication of this dynamic condition, but it is also likely that, what I term as, "eyeball wobble" will develop in the attempt to follow two adjacent meters with differing ballistics. Consideration of two pointers with such differing ballistics on a single scale will confirm that the PPM does indicate the greater amplitude, and that the differential in release time between the two instruments is difficult to interpret.

Observation of the complex audio signal in an oscilloscope indicates the peak excursions in program material. The use of an oscilloscope with long or variable persistance CRT exhibits additional information relative to recurrent amplitudes which are displayed as a band of energy concentrated about the center by the persistance of the CRT.

These two pieces of information provide the composition of acoustically related peak to quasi-average information.

The combination of LED and digital technology provides the solution to the display. The use of a dot display for "Peak" information and a bar display for "Persistance" allows a single display for both ballistics. Each lamp in the display is therefore driven by two drivers, one for peak, the other for persistance.

This representation presents a display of a dot riding on top of a bargraph. However, in order to make this display meaningful there has to be a relationship between these two ballistics.

Rather than accept the 10 millisecond integration time of the PPM, it was deemed wiser to allow the peak display to indicate the true peak of the composite signal.

The relationship of the persistance display to the actual peak was next to be determined. Equal energy, properly weighted for program material is discerned as equal in loudness. Since energy can be displayed as a function of amplitude and time, the oscilloscope can be used to confirm that large amplitudes of short tone bursts can be equal to longer tone bursts of lower amplitude. Average Power is defined as equal to the area under the curve divided by the time interval. Since the area is equal to the input energy "w" or watts during the interval, the Average Power = w/t where "w" is in watts and "t" is in seconds. Thus averaging type metering can provide an indication of power. The VU meter is an averaging type voltmeter with its scale calibrated in power.

With the peak integration time as defined, the next step in the development of this meter was to determine a meaningful integration time for the bar display.

Although one would think that the integration time incorporated in the VU meter might be the logical choice, a review of the development of the VU meter tells us that the ballistics were chosen with reasonable compromise to the peak energy content. The characteristics of the PPM were not considered because of its slow decay time.

Investigation into various integration times soon revealed that the proper choice would lead to the logical conclusion that the power in the complex waveform would follow a form of cadence as well as indicate a level of loudness. All of this, of course, would relate to the peak of the excursion.

Results of this investigation showed that with an integration time twice that of the VU meter for the "persistance" bargraph display with its reference at 65% of the full meter scale, and the corresponding dot display for the "Peak" with its reference at full scale, provided a metering device which would indicate both the peak and quasi-average value of the composite waveform in a manner relative to the effective loudness of the program material.

Monitoring program material is now as simple as riding gain in a manner that neither the "Peak" or "Persistance" reference level is exceeded.

This allows all program material to be adjusted for equal perceived loudness while protecting the peak of the waveform.

RELATIVE BALLISTIC CHARACTERISTICS

1) General

a) Integration networks are fed from Low Z drivers.

b) In an integration network, the charge of the capacitor reaches 63.6% of the full value of the peak in one time constant, 86.5% in two and 95% in three.

c) Standard measurement of rise time using an oscilloscope is defined as the time for the waveform to rise from 10% to 90% of full value.

d) Maximum voltage excursion of a sinewave is at 90 and 270 degrees.

2) Peak Display

a) Integration network employed is 47 ohms and.1uf.

b) Time constant - 4.7 usec.

c) Rise time = approximately 2 time constants or approximately 10 usec. This is 1/1000 the integration time or rise time of the PPM.

d) The period of a 15kHz sinewave is 67 usec. It takes 1/4 cycle to reach 90 degrees or 16.67 usec. Therefore the peak display easily detects the peak* of a 15kHz waveform.

*This circuit is followed by a 3 pole 20Hz filter to avoid jitter in the peak display. The delay introduced by this circuit results in a 2.4msec delay in the indication only.

e) The decay time of the peak display is 18msec/dB (180msec).

3) Persistance Display

a) Integration network employed is 2.7megohm and .1uf.

b) Time constant - 270msec

c) Rise time = approximately 2 time constants or approximately 600usec. This is twice the integration time of the VU meter. With both meters set at reference with a steady state tone, the standard test signal of 300msec rise time for a "O" or reference indication on the VU meter will indicate approximately 3dB below the Persistance reference level.

ADDITIONAL FEATURES

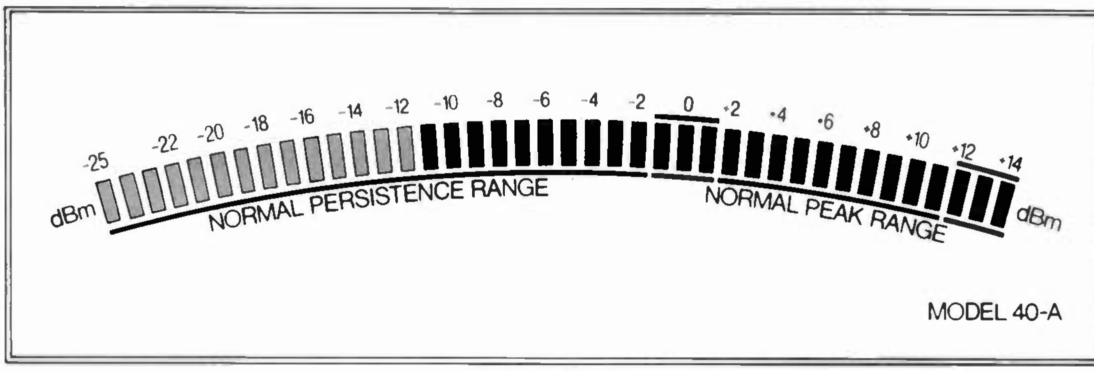
1) The meter is amplifier isolated and bridges the program line.

2) The meter has two inputs. This feature provides for the metering of Sum and Difference channels in stereophonic operation. It is the Sum Channel which sets the relative loudness of stereophonic material. Metering the Difference Channel is an indication of the stereophonic illusion. Monophonic program material should display no difference information.

CONCLUSIONS

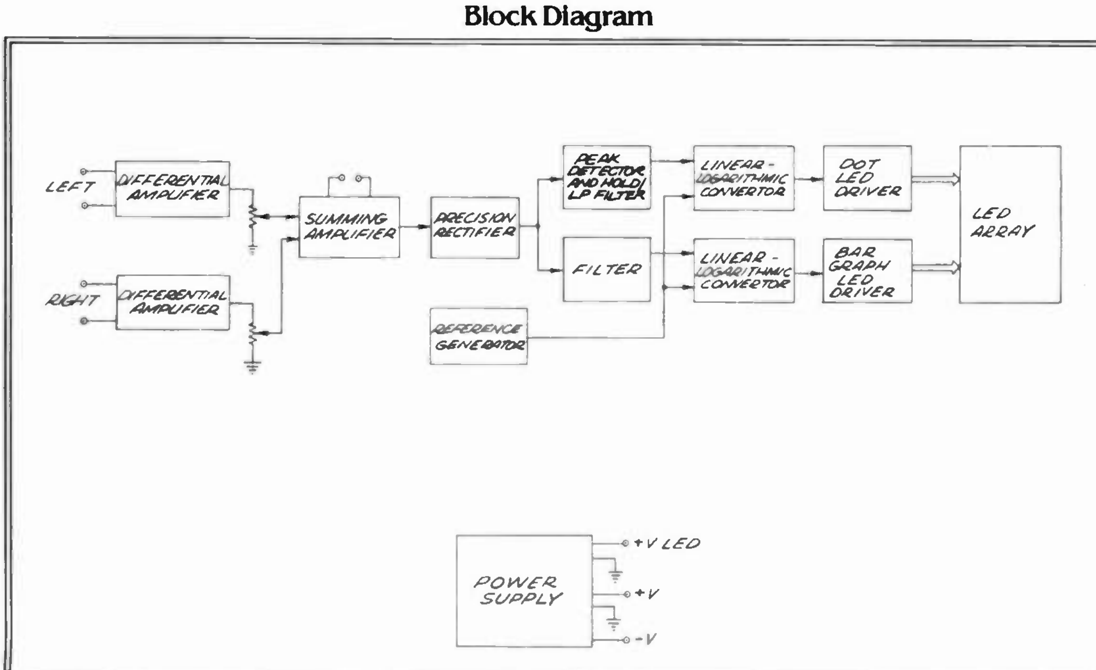
This meter displays the instantaneous peak of the waveform, thus providing the operator with the assurance that available headroom is not exceeded.

Research has established a relationship between integration time of the peak and average of the signal and the display of the two ballistics on a single simple to read scale relating to equal loudness for both compressed and uncompressed program material.



Meter Scale

Block Diagram



RADIO WITH PICTURES

Ralph Beaver
Q105 Radio
Tampa, Florida

Overview

In October 1985, negotiations began between Tampa Cable, later sold to Jones Intercable, Inc., and Q105 radio to put video with the morning QZOO audio and present it as the "QZOO TUBE" to cable subscribers in the city of Tampa as part of their "basic" cable. This experiment started as a marketing effort aimed at "subscriber retention" for the cable company. On May 1, 1987, the first QZOO TUBE program aired, and "Radio with Pictures" was a reality in the Tampa Bay area.

SYNERGY

The QZOO TUBE required the engineering, production, and programming personel of Jones Intercable to work with the engineering, production, and programming personel of Q105. This hybrid of departments, people, and technologies was focused on creating a third product that would be impossible for either separate entity to accomplish alone..."SYNERGY"! One of the beautiful elements of this experiment is the two-way benefit. We are finding new advantages each day, but I will try to outline a few obvious ones.

Benefits for cable company

The term "subscriber retention" was a new one to me. I had not realized this was a problem in the cable industry. I did realize that cable companies may need a good source of "local origination" material, and this could fill that need. We were proposing to allow the local cable people to use our highly-acclaimed morning show as a starting point, add pictures, sell advertising, and market this newest product to other Tampa Bay Area cable companies.

In preparation for this live TV venture, the cable company had to set up a new television production facility at the radio station. This additional facility provides expanded production facilities for cable ventures during non-simulcast hours.

Another shocking achievement is the continued surprises of working with artists with egos. Cable people weren't used to this. A constant explanation was in order to calm them down and reassure them that the end of the world was not right around the corner!

Benefits for radio station

During promotional appearances, our cable TV remote crew documents the radio station activities and uses them as non-synchronous video during periods in need of video "fill". (Record companies have supplied us with a large number of videos, but never 100% of our playlist.) The radio station then puts together a promotional tape of these documentations and uses them as examples of our successful promotions.

Cable system audiences are not measured by Arbitron Radio or Birch Radio. Since the first broadcast, however, the QZOO TUBE has experienced wide audience acceptance. The results of a cable subscriber bill-stuffer survey, indicate that 20% of cable households that are able to watch the QZOO TUBE, do so on a daily basis.

Another un-measured benefit is the national attention we received. While it may or may not reflect in local radio ratings, it does feel good.

Preserving the "radio" environment
One item was immediately agreed upon by both parties: the delicate environment of the radio broadcast must not be disturbed. The cable company wanted to try this experiment because we had the top-rated morning show. If we made compromises that degraded the quality of our radio program, nobody's purpose would be served.

The two permanent, remote-controlled cameras are hidden in the walls and do NOT have tally lights. The on-air radio talent does not know if either camera is on or off until they view the tape after the morning show.

There are NO cable TV monitors anywhere near the control room areas. There are NO special TV lights in the control room. The cameras selected were able to produce a very nice picture in very low light levels.

Equipment needed for TV
With as many experts working on this project as we had, "technically impossible" simply meant that something would be finished in the afternoon instead of the morning. One of the impossible tasks was to convert a very small area (8' by 10') into a video production center for live television. Due to this size constraint, the concept for construction was a "mobil video van" environment with compact equipment space to maximize people space.

Complex interfacing requires teamwork. It also required working late at night to avoid the normal heavy daytime traffic in the radio station.

Picture control room
We had to look at all possible video playback units, since we were really not certain of our video sources. Dang, there are a bunch of video tape formats! The final choices were 1" for professional quality music video with good signal-to-noise and stereo phase, 3/4" for locally produced non-sync video, and 1/2" VHS for those video oppoutunities archived only on VHS. For all other types of video, spigots were located throughout the building that allow connection of any other type of video playback device. This allows us to use "home video" from any consumer machine, and instantly route it through the digital video effects machine to sync it with the program.

The video control room also had to be equipped with the normal complement of video switcher, DVE (digital video effects), graphics computer, paint box, electronic still store, video distribution and the like.

The biggest problem was space. Our radio station did not have any excess space planned, certainly not enough for a fully equipped television production facility. Somehow, using every inch of floor and wall space near our technical operations center, we were able to house all the video equipment, people, and tape storage with no room to spare.

Interfacing Cable with Radio
The Q105 technical crew was responsible for any circuits directly involving our main control room electronics. There had to be a lot of interfacing to insure a smooth, live RADIO-WITH-PICTURES program.

The most ambitious project within this project was SACPI. Studio Audio Cart Protocol Interface (pronounced "sack-pie") is the computer hardware and software which enables the radio station's cart machines to directly access the cable company's electronic still-store (a type of electronic slide projector). It was created to preserve the spontaneity of the radio program. EXAMPLE: A caller wishes to hear a certain pre-recorded bit about Ajax Company. The morning personality receives the phone call, grabs the pre-recorded cart, jams the cart into a cart machine, and starts the comedy bit. Elapsed time for this type of request from telephone mention (live on-air) to actually playing the bit is about five to ten seconds. When the cart is started, the electronic still-store has the correct picture ready for the TV director to "take" in less than one second! This effort is to permit a "pre-produced" look on the TV screen although it is being created and aired live.

Remote starts for the 1" video machines were directly interfaced to the Q105 main control room console. These interfaces allow un-restricted use of these machines by the video production center during non- broadcast times, and fully syncronized playback during broadcast.

Another "must" is a slave of the count-down timer that is reset each time a new audio cart machine is started. We also made this timer reset with the start of the 1" video machines from the radio main control room.

Another group of lights in the video center indicate the real-time status of all radio control room audio channels, auto sequencer, and end- of-message warning.

An INFORMATION CENTER DISPLAY is maintained in the Q105 GO PATROL operations center, and remote displays are located in radio control rooms and in the video production center. The display is updated with weather information, traffic conditions, and cues and orders as necessary. This is an effort to get information to all necessary positions instantly.

Obtaining music videos
It was agreed at the start that the Cable Company has the responsibility of obtaining Videos. The radio station has no use for them otherwise, and has no equipment to play them. The record companies were understandably confused by our requests for video, but played along with us anyhow. I shall always be grateful for that.

When no music video is available...
When we have a remote from a major theme park such as Walt Disney World or Busch Gardens, we try to have live pictures to add sparkle to the program. During live-via-satellite transmissions from New York, London, and other domestic locations, audio interviews are accompanied by either still shots of promotional photos and album covers, or rolling video supplied by the promoters.

Weather updates
Great care is used to create "graphics that match the spoken word" in weather forecasts. Our standard radio format calls for frequent temperature checks during the program. This creates numerous opportunities for mistakes. The cure for these possible errors is the information screen available in all Q105 control and news rooms and in the Jones Intercable video control center. In theory, since everyone is reading the same screen, everyone should come up with the same information. In fact...it works!

News and Sports updates
When news or sports reports are delivered from our newsroom or main control room, the normal QZOO TUBE system is adequate. Action footage is used whenever available.

Traffic Updates
Rolling video shots of the Q105 "GO PATROL" vehicles and aircraft are used during traffic reports along with generic shots of traffic clogs at the normal clogging places. We were testing the idea of LIVE shots from our aircraft, but the cost-vs- benefits ratio was well out of line. Care is taken to match the correct GO PATROL vehicle video with the correct GO PATROL reporter.

Distribution
NOW...how do we get this "Radio-With-Pictures" product to the Tampa Bay cable subscribers? It was agreed by both parties that distribution of the QZOO TUBE is the sole responsibility of the cable company. In distributing the program, it was decided to keep the program "cable exclusive" in order to develop significant locally origanted programming that is available to only those subscribers hooked up to cable TV.

The Jones Intercable system is wired with 5 "hubs" for distribution. "Hub 1" is the primary origination site for the system, and is geographically located near the Q105 studio. The signal from the Q105 video studio is connected by an upstream return path on the cable system, and is then routed to "hub 4", and injected on channel 47, the designated channel on the Jones Intercable system for the QZOO TUBE. In order to distribute the program to other cable systems in the market, other technologies not normally used for interconnection of cable systems were utilized. At the Q105 FM transmitter site, on a downtown Tampa bank building, is also the location of several omni-directional television distribution systems. One of these systems was used to transmit the live signal to other cable systems in the market. The use of this available technology marked the first time several cable systems in the market were interconnected for any reason. This "network" is checked on a regular basis to allow time to scramble technicians into place should problems occur.

The Jones Intercable subscriber base in Tampa is currently 46,000 homes. By distributing the program to the other Tampa Bay Area cable companies, the potential subscriber base expands to approximately a half-million homes. Currently the program is distributed to 7 affiliate headends throughout a tri-county area. Subscribers able to receive the program currently number 190,000. The distribution system necessary to serve these needs was constructed amid Tampa's lightning season, and during other massive cable plant re-construction. Cable crews installed equipment in a few short weeks (with 3 days to spare) that had not been planned until the next year. The teamwork exhibited by these professionals was inspiring!

What a blast!
I have enjoyed the poop out of this! The pains have been worth the pleasure. We have all learned a lot.

Shaky start-up
April Fool's day, 1986. What a perfect day to start this project! A variety of setbacks prevented this launch, but 13 months later, Jones Intercable viewers enjoyed the first of many QZOO TUBE cablecasts.

About 50 people were involved in the early start-up stages involving contracts, executive decisions, technical inventions, inter-company interfacing, sales efforts, and space planning. Any one of those 50 people could have put an end to this experiment, but all 50 had to agree to give it a try...and we did!

During our sensitive negotiations, "Tampa Cable" was purchased by "Jones Intercable". All of our hand-shake agreements were put on hold, and re-started after the sale was completed. Jones Intercable allowed us to continue with this project. I consider that one of the smartest moves in cable television since CNN!

Great opening show
Friday, May 1, 1987 was the official launch day for the QZOO TUBE. 40,000 Jones Intercable households were now able to listen AND watch the Q- Morning Zoo™. The opening day festivities were covered by the local Tampa Bay Area newspapers and television, CNN, Radio & Records, and several other fine media entities with superb taste.

The kick-off party was held the same night at a local hotel. Many TV sets were arranged around the hotel lobby to display that morning's program.

SACPI was still under development, and was not used.

Future plans
The QZOO TUBE is a joint experiment in the mixing of an all-radio environment and a non-radio environment. It is a joint experiment in mixing a new collection of equipment and people. It is a cable experiment in creating new media dollars.

We have frequent joint cable/radio meetings to discuss the last show and plan the next one(s). Some of these are very intense with much fist-pounding, veins popping out of foreheads, screaming...but each person involved in these meetings cares very much about how Tampa feels about what we are doing as a radio station, a cable company, and the combination thereof. After less than a year, the Tampa Bay Area has begun to accept "QZOO TUBE" as a household word. I like that!

While I can't reveal all the plans right now, it looks promising that future expansion for the QZOO TUBE could be "Radio for the Deaf!"

Conclusion
We are not trying to be a slick television show. This is indeed "Radio with Pictures". Some mis-timings must occur due to the live nature of this product. It is our responsibility to keep these to a minimum.

We have to ask ourselves "would we do this again?" Why not! Radio is LIVE with heavy input from our LIVE audience. The TV crew has to really be awake to follow our audio with a picture that makes sense. You may notice this is a little backward from "audio-follow-video" that most TV is based on. Twenty hours of LIVE television each week is a staggering statistic for even a "normal" television station.

Acknowledgements
For me to name ALL the people I should here would take too long and could cause damage to my operation when someone else discovers just how talented this group is. A project of this type does not necessarily take a lot of people to accomplish, but the chosen few players must perform beyond normal bounds. All connected with this project are extraordinary in my view, and have worked to make this project happen with the phenomenal effort that the Tampa Bay has grown to consider "normal" from Q105. Thank you powers-that-be at Edens Broadcasting who allowed "Bevo" to try yet another one of his "great ideas". My life is much richer having been associated with these people.

A MOBILE RADIO PRODUCTION FACILITY FOR THE SPACE AGE

William Ryan
KVIL
Dallas, Texas

ABSTRACT

In July 1987, KVIL's Programming Department asked us to create a new "motorhome" type remote studio to originate Ron Chapman's top rated morning show from anywhere in the Dallas-Fort Worth Metroplex. It must look good, sound as good as the main studio, do everything the main studio can do, and be ready for the first show in two months. It will be used every day for three months from a different location every day to be chosen the day before. This report outlines the project from that idea to the first live broadcast in October.

CONCEPT

In the first meeting with programming, a design criteria was developed defining the goal of this project and detailing the objectives. The stated goal was straightforward...to be able to broadcast the entire morning show in stereo, with all the same facilities and On-Air quality as the main studio provides, from any location within a 45 mile radius of the station, in a "First Class" motorhome studio designed to reflect the quality image of KVIL. Specifically, it must be set up to: Play all music, commercial and drop-in carts or CD's; interview guests inside or talk with the live audience outside; handle telephone calls from listeners live or on reel tape; place calls anywhere live or pre-taped; communicate with the traffic helicopter and van and put them On-Air live; allow fulltime intercom with the main studio producer; provide P.A. for the audience; be free of any "land line" connections for power, telco, or stereo program link; provide full kitchen, bath and lounge facilities for staff and guests; be able to operate live every week-day from 6am-10am for 3 months at a different location each day on short notice; have low initial cost and very low operating expense; and be on the air by mid October.

We succeeded and here's how.

PLAN & DESIGN

Any potentially successful project must be well thought through, carefully planned and designed in detail. A coordinated plan and well informed, enthusiastic participants will make the project work.

Discussions were started immediately within the Engineering group to assess every possible aspect of this project. Each of our three very creative Engineers brought to every discussion a list of ideas, potential problems and solutions. The potential problems were generally based on the timeframe and cost--Anything can be done with enough time and money.

A list of preliminary questions was put together.

1. What configuration should this remote studio take...motorhome, trailer, portable stage?
2. What power source will be needed...generators, shore hook-up, solar, wind?
3. Will it be possible to provide a full studio telephone system that operates reliably under these conditions...cellular, RPU, pack, phone patch?
4. How will the stereo program reach the main studio dependably every day from yet unknown locations?
5. What features would allow the studio to be used in other ways later on?
6. What could be built into this project to handle future expansion easily?

From our discussion sessions, a set of general specs was listed to give the project a starting base.

1. Absolute minimum studio space must be 8'x 10' for equipment and people.
2. On-board generator power must be provided for all facilities and floodlighting since we will be setting up and starting the shows in the dark.
3. A cellular phone provides the instantaneous telco service desired.

4. A stereo link to the main studio from all unknown locations might be configured by PCM encoding the audio and connecting to a minicam van rented from a local TV station set up to cover the area...they're not usually using minicams early in the morning.

Armed with these guidelines, we went to work to investigate the possibilities.

THE STUDIO FORM

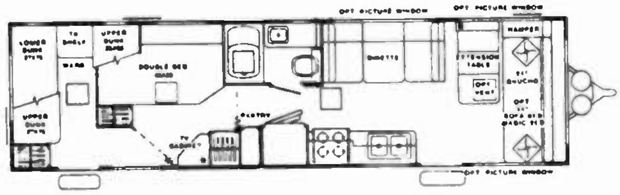
A sketch to scale of our proposed studio layout was drawn up and visits were scheduled to motorhome dealers. We soon realized the cost and space available in a motorhome would be restrictive to the project as envisioned...full studio, kitchen, bath, and lounge.

Trailers were the next choice and after the first trip through some RV trailers, the overall concept of this new studio's form began to change. An RV trailer projects a different general image than that of a motorhome. The lookout angle is much lower and, in our case, would provide a more direct eyeline with the audience...a more intimate feeling. The interior areas are comfortable...homey...again creating a closer feeling with the audience.

On the practical side, the lower floor and higher ceiling clearance provide more "room" inside...ceiling height is 78" and overall interior width is 90". Cost, space, and delivery were also much better.

With another look at this project by programming, the KVIL Rolling Coffeebar was born...a mobile studio to broadcast the morning show and serve coffee to all who stop by.

The selection process was fairly simple given this new direction. A Fleetwood Model 31Z was the choice for a number of reasons. The large rear sleeping area, Figure 1, would provide all the studio space without serious modifications, while bath (with shower), kitchen, dinette, and lounge areas could remain unmodified...already finished and working.



MODEL 31Z

The dealer also happened to have available a one ton, dual, rearwheel drive, diesel pickup truck, fully equipped to tow this trailer, and the name of a qualified remodeler to install windows, reinforce floors, wire and paint the package to our specs.

Fleetwood's factory rep was contacted by the dealer and supplied a full set of structural drawings for us to begin the remodeling evaluation. A floor plan was overdrawn on the print, Figure 2, to make sure everything would fit. Pedestals above the wheel wells allow reinforcing bases for equipment racks and a telephone "room." The area of 7' 6"x 11' 0" was clear for a console pedestal and people. Large windows were drawn into the side and rear walls and the rear door was retained to allow easy installation and operational access.

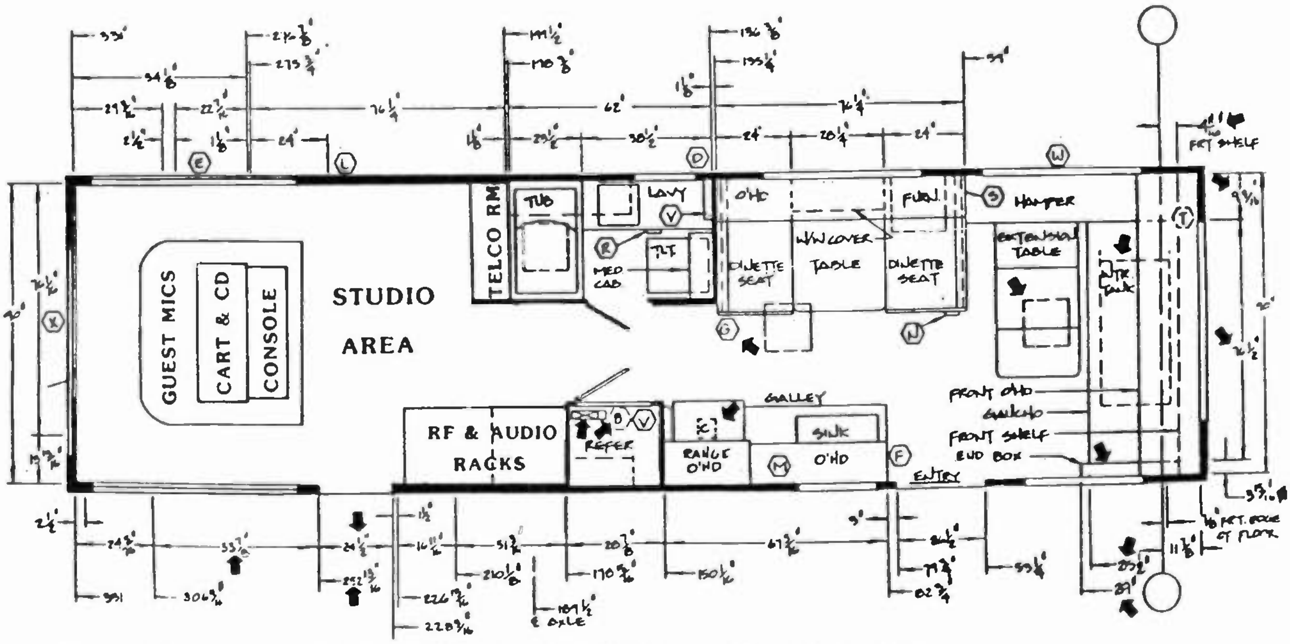


Figure 2 **KVIL REMOTE STUDIO - Preliminary**

A set of drawings was developed for: electrical rewiring, audio and control wiring, weight distribution/reinforcement, sound and noise controls, etc. A logistical plan for towing, setup, teardown and storage was estimated to determine operating expense and flexibility. From these plans, costs were figured, along with time tables, to buy, remodel, and transform this package into our desired studio.

Budgets were reviewed, discussed, modified and ultimately approved. On September 5, the trailer was towed to my house to begin gutting the rear bedroom area and draw up detail plans for the remodeler. The following week, he towed it to his shop and began removing the outer "skin."

THE SYSTEMS

While the trailer was being rebuilt, all final plans and purchase orders were completed for the other components of the project.

The project was broken down into individual systems, and each Engineer was given primary responsibility for one of them. Documentation was maintained through each phase and equipment files set up to keep everyone informed as the project proceeded.

STUDIO

Leslie Triplett handled the studio installation. A final floorplan was drawn up showing wiring runs to an RF rack, audio rack, telephone room, and outside outlets for PA speakers and mics.

The new Autogram R/TV-12 console was chosen for ease of operation, quality, ruggedness and an extensive list of features that fit the project perfectly. For example, it's external "Channel On" control will remotely turn the appropriate channel on at a preset level so that a cart or mic started at the remote location (the guest mic was used by our sports commentator who fires his own commercials) will air at the right level even if the main mic position is unattended. This modular console also offers eight mic pre-amps, independent of any channels and assignable to any channel. We used all eight for main mic, guest, stereo outside crowd mics, stereo outside AKG handheld and stereo mic in the kitchen for sit-down interviews. For maintenance, all components are connectorized....even the front panel mounted switches.

An IFB was designed into the outside PA system to allow the On Air talent to talk to the live audience with one button to dim the PA air monitor and place the main mic on the PA speakers. A similar system was designed for the intercom to kitchen and lounge areas.

Dallas studio cabinetmaker Ron Rentfro took our studio sketch and returned in three days with the console pedestal, Figure 3, that houses equipment racks below and a cart/CD overbridge above, built precisely to our specs. An equipment order was placed through Pat Hurley at the Dallas Allied office and equipment started arriving within a few days. Each piece was checked out and installed in its enclosure with wiring harness built to fit. The entire studio was assembled in a storage room for pre-installation tests. Thanks to proper planning, careful documentation and wiring, everything worked the first time the system was turned on and all stereo was in phase.

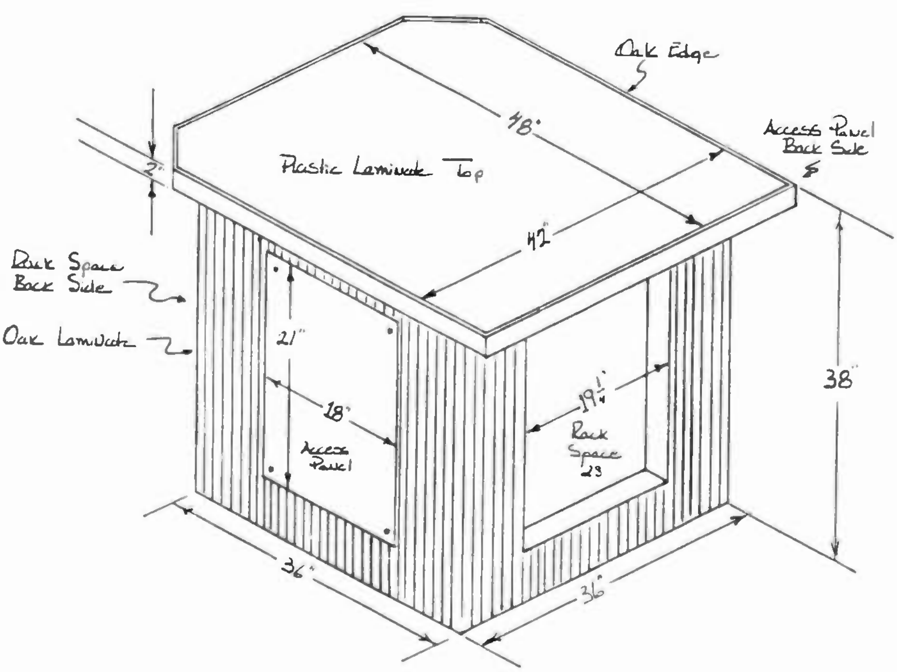


Figure 3

POWER

Jim Fitzgerald designed and installed the extensive power system for this project. Included were two Onan 7.5KVA diesel 230VAC generators mounted on the bed of our tow truck, a custom main power disconnect panel, 100 ft. cables to the trailer, two receptacles for the mains built into access panels on the side of the trailer, and distribution panels inside to provide wiring termination for all A.C. and 12V D.C. systems.

The original trailer system, set up for a 110V RV Park feed, was rebuilt to accept 230 volts with an added mains box and all additional wiring for floodlighting outside. One generator was dedicated to the trailer electrical system for lighting, air conditioning, furnace, refrigerator, microwave, and utility outlets inside, plus the added outside floodlighting since the intended use of this studio would be 6am-10am, or about 2 hours of darkness everyday.

The second generator was fed to a new main distribution panel and provides individual circuits for the RF rack, audio racks, console pedestal, telephone room and outside outlets plus the most important feed...an industrial size 230V five pot coffeemaker to supply a continuous flow of coffee for the audience. We made between 30 and 50 gallons of coffee per four hour show, or 480 to 800 cups.

Hookups between the trailer and generator/tow truck were made by cables with mating Hubble power connectors so that if either generator failed, the other could be plugged in to take it's place. Power distribution and load balance to each generator were carefully planned. A failure of either generator would only cost us A/C, interior utility outlets and some exterior lights. Trailer lighting, including the studio, is all 12 VDC, so in worst case conditions, a trailer plug on the truck can supply interior lights for all areas as well as the furnace power. Microwave and air conditioner are all run from A.C. in the trailer. The stove, refrigerator, water heater and pumps are all operated on 12 volts.

Jim also designed a full range of outlets outside the studio. For example, all floodlight locations have outdoor outlet boxes installed directly under the light fixture with a special duplex outlet...one plug is a twistlock for floodlights and the other is a standard three prong A.C. for utility use. By this design, "a drill" cannot be plugged into a circuit load-designated for flood lights, thereby maintaining the load distribution on each generator.

The generators are mounted on special pads built in-house using Mason mounts to control horizontal and vertical movement and noise transmission. They are covered by a Gem Top with large side doors for complete access to the service side of each generator, a rear service for access to set-up tools, and dual roof vents to exhaust fumes, air, and noise, which is projected up and away from people and microphones.

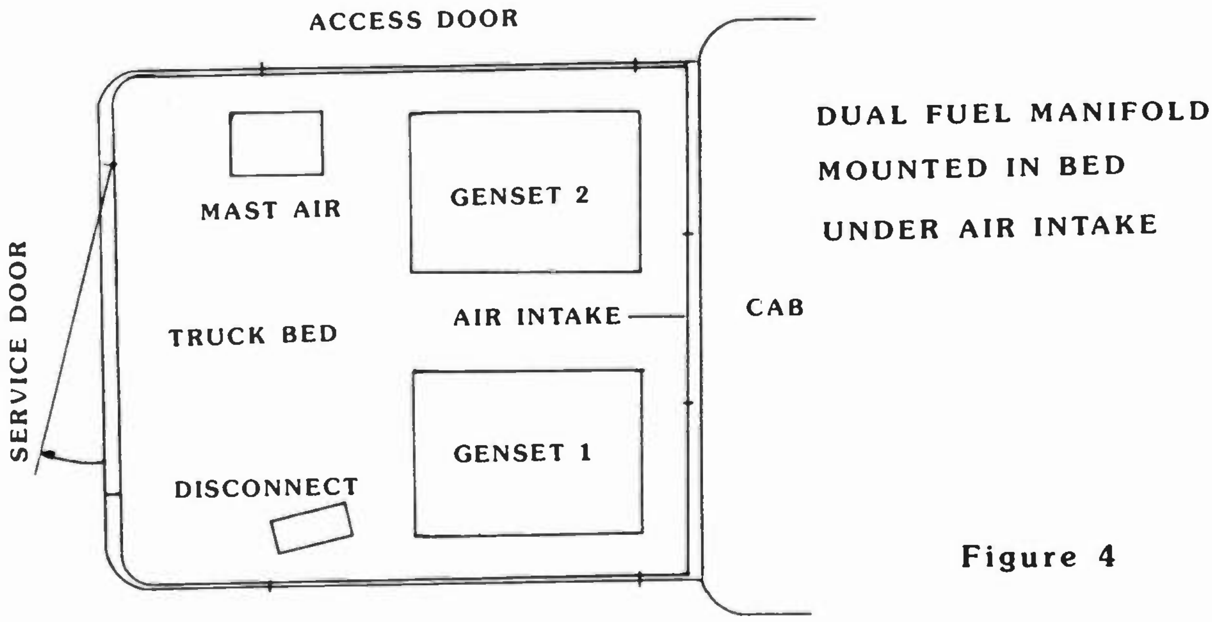


Figure 4

KVIL GENERATOR/TOW TRUCK

A fuel distribution manifold, to feed supply and return lines from the front auxiliary fuel tank of the tow truck, was installed. The entire power system will supply continuous 24 hour operation for two days without refueling. The main tank of the truck was not tapped so there is always fuel to get home.

RF

·Ike Blevins headed up the RF system creation which consisted of five cellular telephone transceivers, two complete two-way radio installations, a dual/stereo RPU link and an FM monitor receiver...all of which must be able to operate simultaneously without interference or RF noise in the audio system. Most of the RF equipment requires 12VDC for power, so three 12V/50A converters were installed and bussed to the RF and phone systems.

The minicam/PCM audio idea in our original plan was not workable, so our primary stereo program link was designed around two Marti RPT-30 transmitters, two power amplifiers, a combiner, and a 4-bay Yagi antenna mounted atop a 30 ft. Wilburt pneumatic mast in a rotor, Figure 5. A truck mounted compressor allows the operator to raise the mast easily while watching for obstructions. The rotor control is inside so that coordination with the main studio by phone or two-way simplifies orientation of the antenna for maximum signal to noise. The main receive site is the tallest building in Dallas, First Republic Bank Plaza, and a 15KHz stereo telco pair carries the audio back to the main studio. A back-up receive site is installed on the roof of our studio building and proved to be as good as the primary site for most of the locations.

One addition to this system enhanced the audio quality remarkably. A C-4 Noise Reduction system from RAM in Chicago was installed to encode audio into the Marti transmitters and decode at the studio end of the telco stereo pair. The results were better than expected. The link is QUIET! You can hear low level record surface noise on music carts dubbed from vinyl records as the music fades out.

TELEPHONE

The studio and lounge phone system was one of the unique parts of this project. Five GE cellular transceivers were installed with Morrison and Dempsey AB1X Cellular Interface Units. The interface plugs into the cellular transceiver with a supplied cable and provides a "land line" style RJ11 modular jack with ring voltage, loop current, dial tone, and tone or pulse dial input for a standard phone instrument. Figure 6 shows a typical installation of the AB1X.

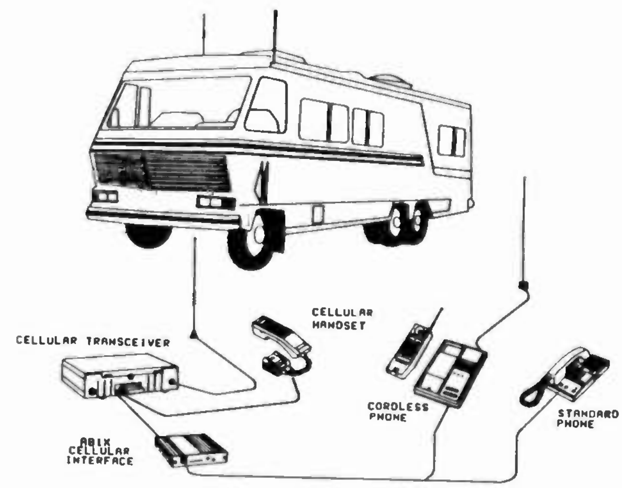


Figure 6 RV INSTALLATION

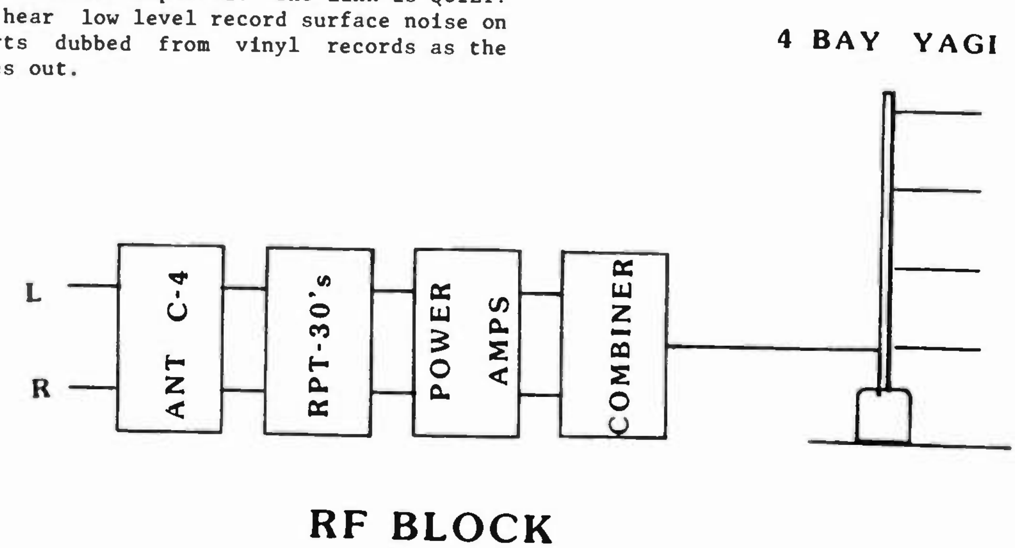


Figure 5

We installed five of these combinations into a shelf unit built by our cabinetmaker, plugged the five modular connectors into a 1A2 key system, plugged in our six button telephone Amphenol, and the mobile studio telephone worked just like the main studio. A keystrip and speakerphone completed the On-Air system and a mixer/limiter provided a reel tape mix of phone and live voice for pre-recording calls.

Listeners now call our regular studio number, calls are forwarded to the cellular system and ring in the lounge and studio. The 1A2 key box flashes the appropriate button on the phone, allows calls to be place on "hold" after screening, and picked up live or on tape, in the studio end of the trailer.

Outgoing calls are placed through this system by picking up the handset, punching a line and dialing. The AB1X interface provides local dial tone and automatically initiates "send" and "end" signals for the cellular system.

ACOUSTICS

Noise control for the studio was not considered a serious problem due to the fact that the presence of the audience on-site would be part of the show. And, as it turned out, the outside door was open and people invited into the studio most of the time.

Reinforcement of roof, walls, and floor structure in the studio area of the trailer, for the extra weight of equipment and people, helped to increase noise transmission loss. A layer of foam board was placed all around under the outerskin when it was reinstalled to keep rain and other small impact noises from directly entering the room. Walls are stuffed with fiberglass batts and sealed. The large windows and thin walls made it difficult to build serious acoustical walls in the timeframe we were given.

Interior acoustic treatment of the room is somewhat limited by large window space, but the usual wall carpeting and a styrofoam "egg carton" style ceiling tile was installed in the studio area. In this size room, everything is virtually near-field and monitor levels are kept low.

RESULTS

After broadcasting 35 consecutive 4-hour live shows, and 2 days of 9-hours each, the project has been heralded as a resounding success in every respect.

From a sales standpoint, the capital investment and all operating expenses of the project were returned several times.

Technically, the studio met all design requirements, sounded and operated better than expected, and ran reliably every day.

In the final analysis, Arbitron indicated a substantial increase in the morning ratings for the Fall Book, convincing everyone that the hard work was "worth it", and clearly demonstrating what can be achieved with a good idea, detailed plan, and excellent people to support both.

A MICROPROCESSOR PERFORMANCE OPTIMIZER FOR ALL TAPE FORMULATIONS

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The proliferation of broadcast tape formulations has required the broadcast engineer to either accept performance degrading compromise settings for bias and equalization, restrict his use to only one tape formulation or equalize independently and frequently for each tape type. Even if the engineer chooses one tape type from one manufacturer, batch to batch variations in performance are inevitable.

This paper describes a system that optimizes performance of various tape formulations rapidly and automatically. The system stores and retrieves the optimum settings for a variety of tape formulations permitting maximum audio tape performance without time consuming manual adjustments for each and every recorded tape.

BIAS, SENSITIVITY, AND EQUALIZATION VARIANCE IN BROADCAST CARTRIDGE TAPE

Table 1 shows the statistical results of bias, equalization and sensitivity testing performed on three different groups of broadcast cartridges on a single tape cartridge machine. The first group was new cartridges of the same length from the same date code and the same manufacturer. The second group was from the same manufacturer, but of different lengths, ages and date codes. The third group was a random selection of new cartridges from different manufacturers.

As shown in Table 1, 99% of the first group of tapes were within ±0.325 dB of the average bias current level of the group. 99% of the second group of tapes covered a range of ±0.96 dB of the average bias current level. 99% of the third group of tapes were within ±2.3 dB of the average bias level. 99% of the first group of cartridges were within ±0.65 dB of the average equalizer level of the group. 99% of the second group of tapes covered a range of ±2.25 dB of the average equalizer level of the group. 99% of the third group were within ±3.3 dB of the average equalizer level of the group. The equalizer level is the amount of high frequency (12 kHz) gain necessary to match the 1kHz output level. 99% of the first group were within ±0.6 dB of the mean sensitivity of the group. 99% of the second group were within ±1.15 dB of the mean sensitivity of the group. 99% of the third group of tapes were within ±2.4 dB of the mean sensitivity of the group.

To demonstrate the audio performance impact of the statistical results, -10 dB record/replay frequency responses were plotted. The bias, equalization and sensitivity were optimized for a tape on the mean of each group. The frequency response of other tapes in the group was then plotted without altering any parameters. These plots are shown in Figures 1, 2, and 3. The first group of new tapes, as would be expected from the statistical data and common sense, was the only group that had consistent response. Plots of the other, non-optimum groups, show large performance variations. With the normal station cart mix, optimum tape machine performance requires frequent and time consuming adjustments for each batch and type of tape cartridges. Automatic record alignment, with easy storage and retrieval of settings for various tape formulations and batches, is a valuable tool for the broadcast engineer.

TABLE 1. VARIANCE IN BROADCAST CARTRIDGE TAPE

	BIAS		EQUALIZATION		SENSITIVITY	
	MEAN	3σ	MEAN	3σ	MEAN	3σ
GROUP1	61.7	0.32 dB	42	0.65 dB	35.4	0.6 dB
GROUP2	62.9	0.96 dB	38.6	2.25 dB	34.9	1.15 dB
GROUP3	59.1	2.30 dB	45.3	3.30 dB	34.5	2.40 dB

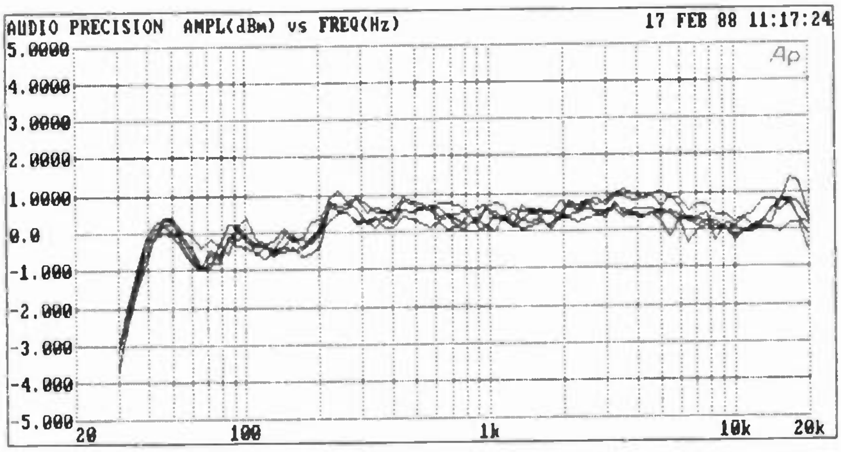

FIGURE 1.

SAMPLE OF FREQUENCY RESPONSE VARIATIONS - GROUP 1 TAPES

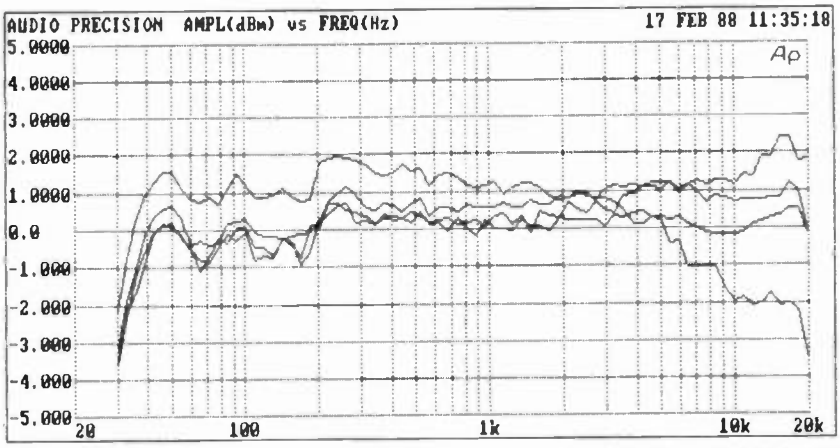

FIGURE 2.

SAMPLE OF FREQUENCY RESPONSE VARIATION - GROUP 2 TAPES

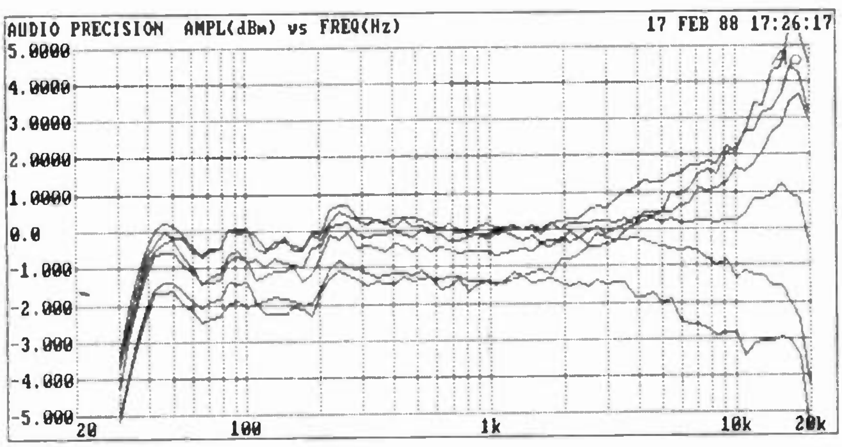

FIGURE 3.

SAMPLE OF FREQUENCY RESPONSE VARIATIONS - GROUP 3 TAPES

AUTOMATIC TAPE PARAMETER ADJUSTMENTS-THE LEARN MODE

The learn mode of operation in the PT90RPS optimizes three critical recording parameters for any tape formulation: bias, equalization, and sensitivity.

Criteria For Bias Adjustment

There are many proposed criteria for defining the optimum bias level for a particular tape formulation and tape machine combination. The five least controversial criteria for optimum bias level settings are:

1. The bias level at which the third harmonic distortion is minimum.
2. The bias level at which the recorded sensitivity of a reference frequency is maximum.
3. The bias level at which the low frequency MOL is maximum.
4. The bias level at which the IM distortion is minimum.
5. The bias level at which the AM modulation distortion is minimum.

Although sensitivity and distortion requirements cannot be satisfied simultaneously at all frequencies, the distortion and sensitivity versus bias of broadcast tape formulations at 7.5 ips form fairly broad curves as shown in Figure 4. In particular, the first two criteria mentioned above are almost equivalent. The LEARN mode was developed using a modified version of criteria 1 and 2 above. This criteria leans toward reduced noise and distortion at the expense of more complex record equalization circuitry.

As seen from Figure 5, tape sensitivity versus bias level plots usually have only one, easy to detect maximum. However, other factors of tape machine design and construction can cause false maximum detection. The two biggest causes of false peak detection are the inter-head time delay and tape dropouts.

The record and play head gaps of a broadcast cartridge machine are separated by a nominal center-to-center distance of 1.125 inches. At the normal 7.5 ips tape speed, this is equivalent to a time delay of 0.15 seconds. Suppression of the dropout error requires that the bias be stepped in small increments (0.05 dB) and the playback output be rectified and filtered before input to the microprocessor. To accommodate the 3.75 ips speed and the time delay in the detection filter, the bias level is incremented at 0.5 second intervals.

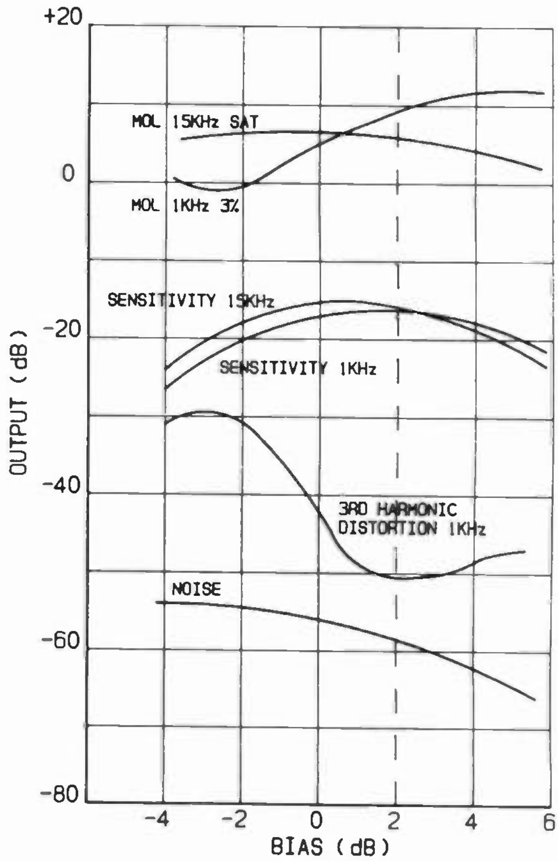

FIGURE 4.
TAPE PARAMETERS VERSUS BIAS CURRENT

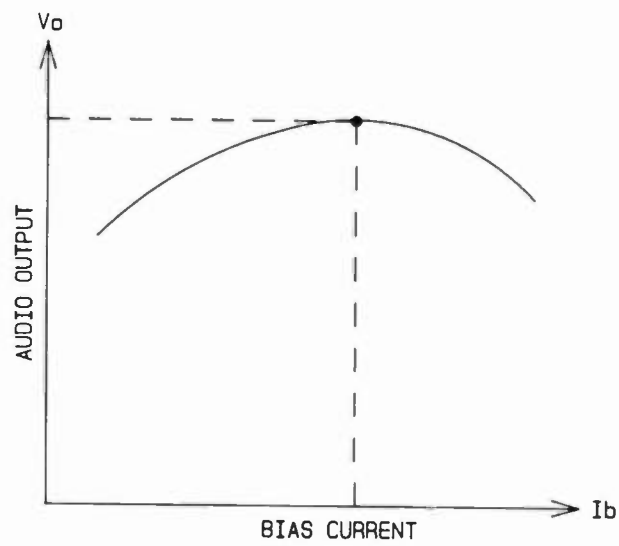

FIGURE 5.
AUDIO OUTPUT MAXIMUM VERSUS BIAS CURRENT

Bias Adjustments

The bias adjustment is performed using the same audio detection and A/D circuitry used to establish the equalization and sensitivity levels. The internal 12 kHz oscillator is selected and the microprocessor increments the bias D/A converters output in 0.05 dB steps. In order to reduce the number of steps and therefore the amount of time needed to make the adjustment, the initial bias level is set to a predetermined, non-zero level. The bias current is increased until the playback audio level peaks. The peak audio level is multiplied by 0.86 (2 dB) and stored. The bias is then decremented until the playback audio level matches the stored -2 dB level. This is called "2 dB overbiasing". The frequency response difference of each tape is then compensated for with an adjustable record equalizer. Figure 6 shows a flow chart of the bias adjustment system.

Tape Sensitivity Adjustment

The tape sensitivity adjustment is performed using the same audio detection and A/D circuitry used to establish the bias level. The internal 1kHz oscillator is selected and the audio gain VCAs in the input audio chain are incremented in 0.1 dB steps until a predetermined, user-defined level is reached. In order to reduce the number of steps and therefore the amount of time needed to make the adjustment, the initial level is set to a predetermined, non-zero level. The input sensitivity control has a range of ±10 dB. The input sensitivity circuitry is factory set to adjust for both the 160 nWb/M level or the 250 nWb/M level. The status of the deck elevated level sensor determines which level is used as the reference. Figure 7 shows a flow chart of the input sensitivity adjustment system.

Record Equalization Adjustment

The record equalization adjustment is performed using the audio detection and A/D circuitry used to set the bias and input sensitivity. The internal 12 kHz oscillator is selected and the VCAs in the input high frequency circuitry are incremented in 0.05 dB steps until the same predetermined, user defined level used for the sensitivity adjustment is reached. In order to reduce the number of steps and therefore the amount of time needed to make the adjustment, the initial equalization level is set to a predetermined, non-zero level. The high frequency record equalization control has an adjustment range of ±6 dB. Figure 8 shows a flow chart of the high frequency equalization adjustment.

The low frequency response of tapes is dominated by the response of the playback head. The low frequency "contour" effect causes response irregularities of as much as ±1 dB in a well designed head and as much as ±6 dB in older designs. Since the low frequency equalization adjustment is not as sensitive to tape formulation, it is controlled by a potentiometer adjustment with enough range to equalize the low frequency response for the most commonly used international equalization standards.

The LEARN mode is designed to permit storage in battery-backed memory of as many as 10 sets of bias, equalization and input sensitivity data for instant recall. The elevated level function permits automatic switching between two sets of data. For example, record settings can be automatically switched between a normal bias tape and elevated bias level tape. If the tape being optimized exceeds the range of the bias, equalization or level adjustments, the machine exits the LEARN mode, stops and the front panel displays the word "FAIL". A simplified flow chart for the entire LEARN mode is presented in Figure 9. The block diagram for the LEARN functions of the cartridge machine is given in Figure 10.

The frequency response sweeps in Figure 11 show the performance of three different tape formulations after each was optimized using the LEARN mode. The sweep marked "A" is a normal bias tape. The sweep marked "B" is a high bias tape. The tape marked "C" is a new tape formulation requiring even higher bias than "B". The time required to optimize these tapes were: A=43 seconds; B=48 seconds; C=55 seconds. The time range to LEARN a tape was 42 seconds to 57 seconds for the tapes tested.

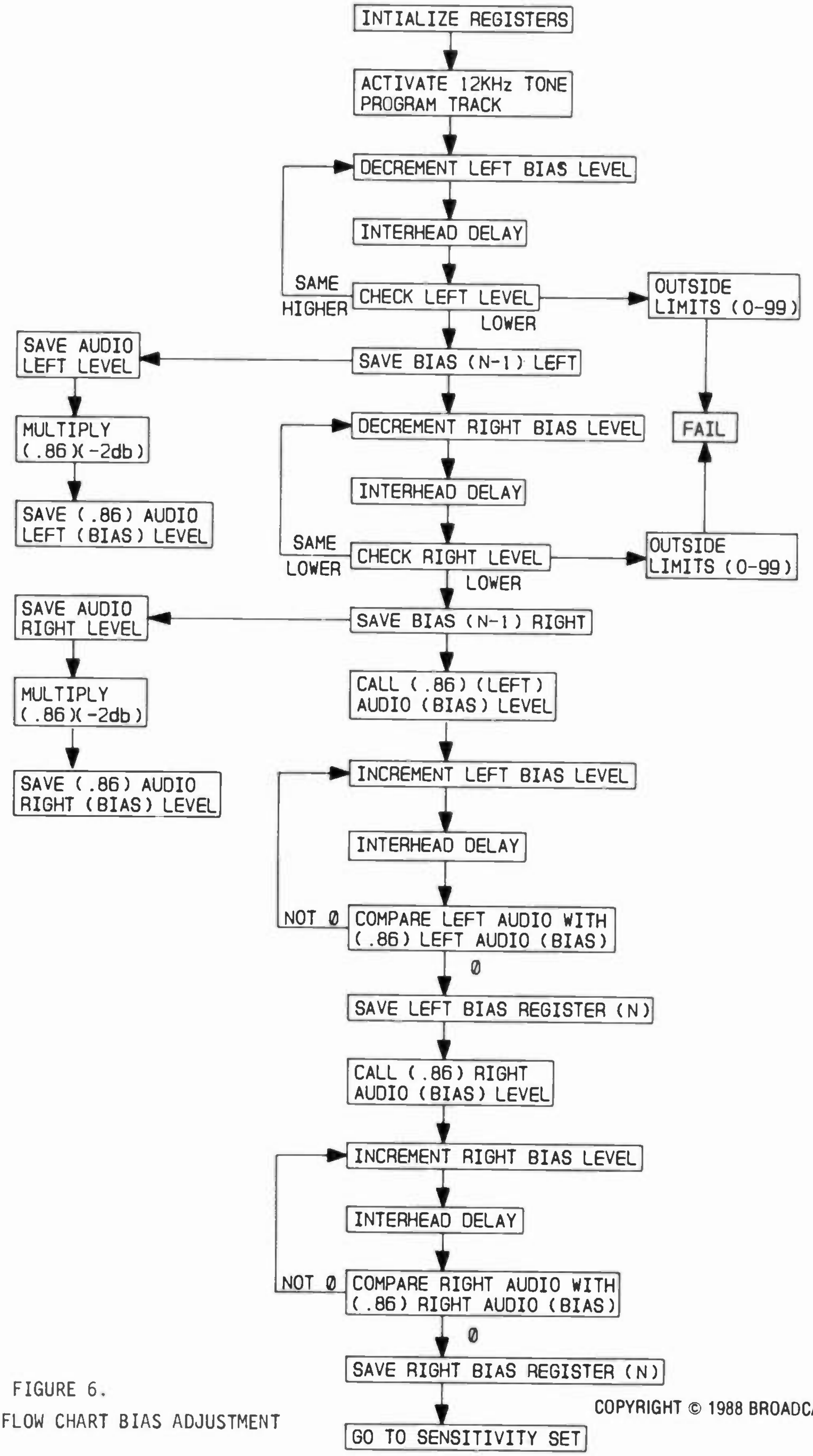

FIGURE 6.

LEARN MODE FLOW CHART BIAS ADJUSTMENT

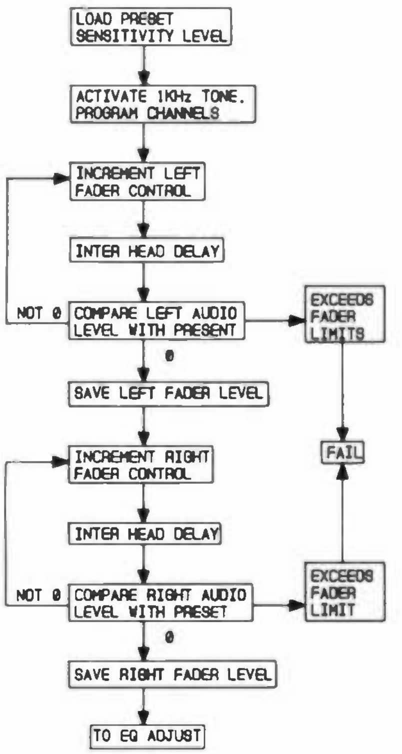

FIGURE 7.

FLOWCHART LEARN MODE SENSITIVITY ADJUSTMENT

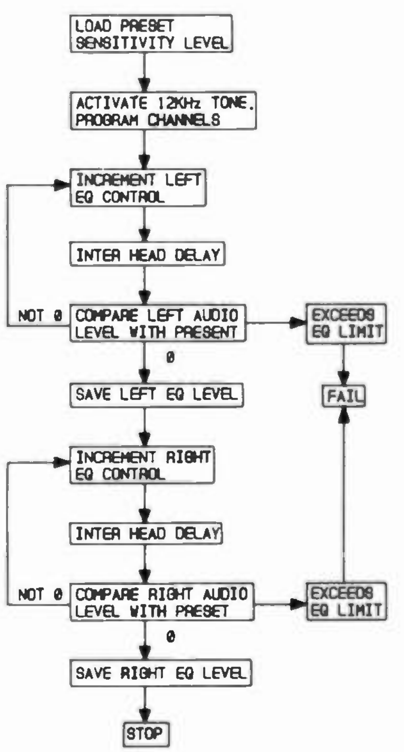

FIGURE 8.

FLOWCHART LEARN MODE EQUALIZATION ADJUSTMENT

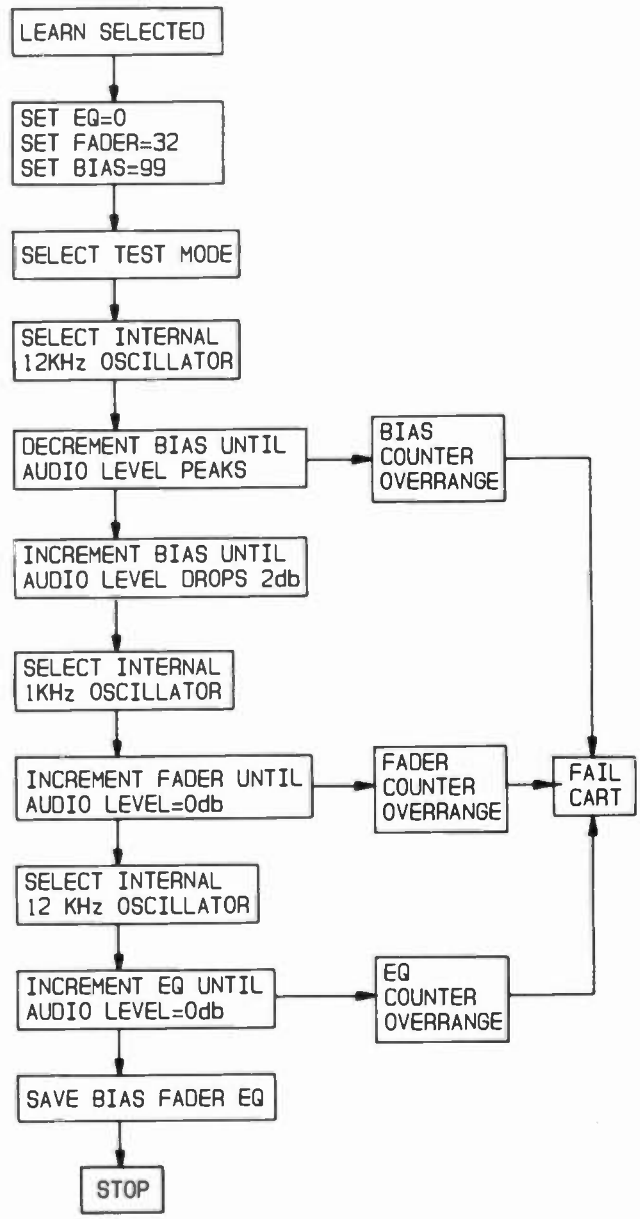

FIGURE 9.

SIMPLIFIED LEARN MODE FLOW CHART

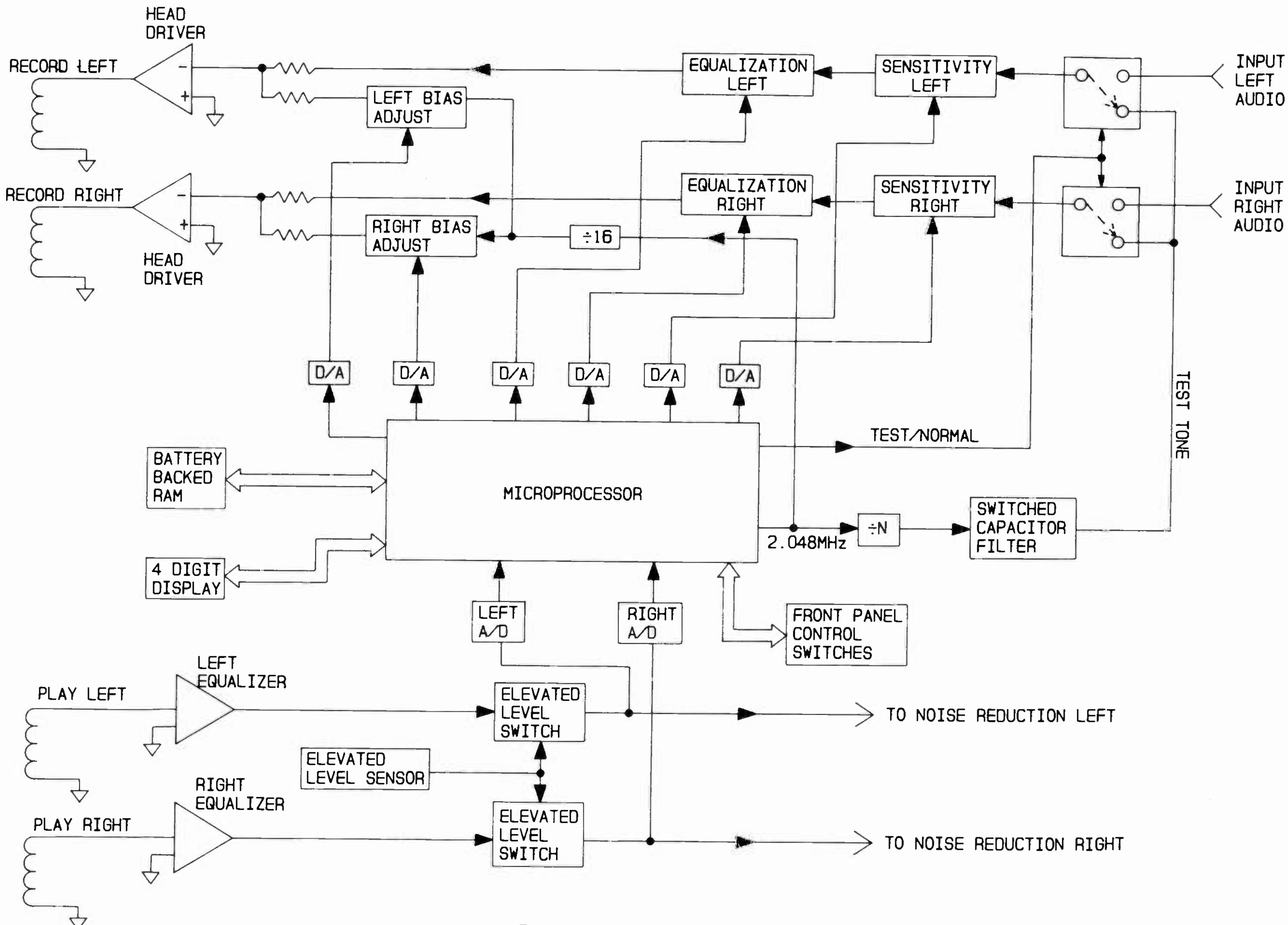

FIGURE 10.
SIMPLIFIED BLOCK DIAGRAM OF LEARN MODE

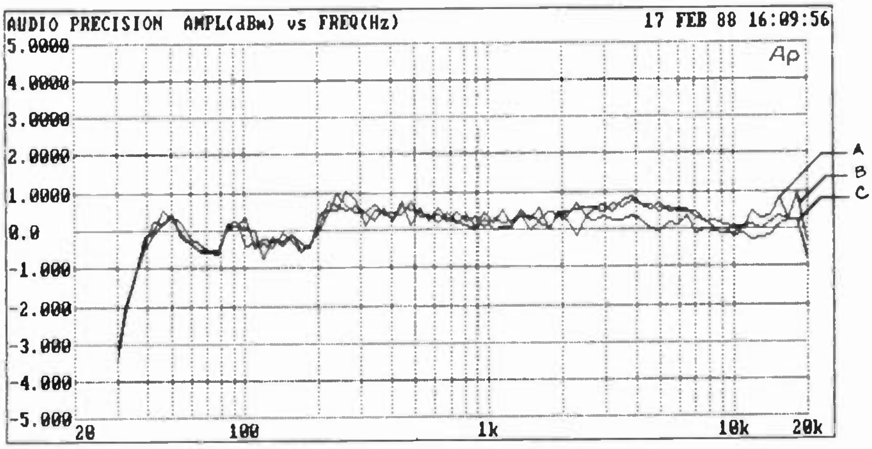

FIGURE 11.

POST LEARN, PERFORMANCE OF THREE DIFFERENT TAPE FORMULATIONS

MACHINE FEATURES

The inclusion of a microcomputer in the cartridge machine has been the means of cost effectively adding other useful features. The PT90RPS includes a prioritized three level access to features, a real time tape timer, a manual adjustment mode, a built in tone oscillator (with sweep mode) a four digit alphanumeric display, a digital FSK tone encoder and the already mentioned LEARN mode.

Three Level Function Access

To insure the data integrity of the 10 memory registers, access to some functions of the cartridge machine can be restricted. There are three levels of access which can be programmed on the CPU module:

1. Normal — Timer, recall memorized tape settings and produce recordings.
2. Learn — Timer, settings can be recalled from memory and the LEARN (LRN) mode can be initiated to learn and save new tape settings to memory.
3. Manual — All functions and settings can be accessed.

Table 2. gives a summary of the three levels of function access. The characters in parentheses are the defaults for the front panel alphanumeric display.

Timer

The four digit front panel timer is active in all modes when tape is running. It displays up to 59:59 minutes, but for the first 9:59 displays a "T" in the leftmost digit. The timer will freeze at the end of the EOM displaying the message time and will display the total length of the cart by depressing the START switch. The timer can be programmed to accumulate time for multiple cuts.

TABLE 2. LEARN MODE ACCESS LEVELS

NORMAL LEVEL	LEARN LEVEL	MANUAL LEVEL
Timer (T000)	Timer (T000)	Timer (T000)
Recall Memory (RM#n)	Recall Memory (RM#n)	Recall Memory (RM#n)
	Learn (LRNn)	Learn (LRNn)
		Fader (FLnn)
		(FRnn)
		Bias (BLnn)
		(BRnn)
		Equal (ELnn)
		(ERnn)
		Save Memory (SM#n)
		Oscillator (OFF)
		Alignment (nnnn)

Manual Adjustment

In order to permit individual tailoring of bias, equalization and level settings, the PT9ORPS permits access to these adjustments when the CPU module is in the "MANUAL" setting. Adjustment to Bias Left (BL), Bias Right (BR), Fader Left (FL), Fader Right (FR), Equalization Left (EL) and Equalization Right (EL) is accomplished by pushing the Function (Func) and Execute (Exec) switches on the front panel. A numeric reading from 0-99 is displayed on the two rightmost digits (For example FR50). As the UP or Down (DN) switches are pressed, the numeric reading increments or decrements. Once the adjustments are made they are then saved using the Save Memory (SM) and Execute switches. Once stored these settings can be used from any CPU level by using the Recall Memory (RM) function and the Execute switch.

Tone Oscillator

The PT90RPS divides the master microprocessor crystal to provide a very stable reference for cue tones and for a built-in tone oscillator. The oscillator gives front panel control of eight test tones (50 Hz, 125 Hz, 500 Hz, 1 kHz, 4 kHz, 8 kHz, 12 kHz and 16 kHz) with a frequency response of ±0.25 db and distortion of less than 1%. Level of the tones is controlled by the front panel fader level controls. Depressing the UP or DOWN front panel switch for more than 3 seconds in test oscillator mode will initiate a sweep through the remaining test frequencies in that respective direction.

CONCLUSION

A microprocessor controlled broadcast tape cartridge performance optimization system has been profiled in this paper. This cost effective system finds, stores and retrieves the optimum settings for a wide variety of tape formulations, permitting the engineer to optimize the stations' audio performance, without time consuming manual adjustment for each and every tape.

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Mr. Carpenter has authored numerous technical papers, including co-authorship of the NAB Handbook chapter on "Analog Magnetic Recording" and is a member of the AES.

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PROCESSING AUDIO FEEDS FROM REMOTE SOURCES

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The immediacy and flexibility of radio allows for the use of audio from great geographic distances to be called upon for airing at a moment's notice. Unfortunately, the aural fidelity of such feeds are not always pristine -- a problem of increasing magnitude as the listening audience becomes more enamored of higher quality audio systems. It therefore behooves the broadcaster to commit renewed vigor to the enhancement of such programming, thus increasing its listenability in the contemporary aural environment. This paper will outline the techniques developed by National Public Radio and others to do just that, in an attempt to prolong listeners' tolerance of the limited fidelity often required on remote audio feeds.

AUDIO PROCESSING OF DIAL-UP LINES

The biggest key to improved fidelity of phone-fed audio is audio processing on the receive end. This processing can be divided into the following categories:

1) Filtering
2) Equalization
3) Compression
4) "Aural Enhancement"

Filtering

Two types may be employed: static and dynamic. The "static" filtering refers to any standard filter set, such as the UREI 565T, which includes relatively steep (18 dB/octave) high-pass and low-pass filters, and two hi-Q, tunable notch filters. The high- and low-pass filters are used to filter out-of-band noise elements from the output of the telephone interface device. Typically, these are set at around 150 Hz and 4 KHz respectively. The notch filters are used to remove any audible oscillation tones ("sings") on the line.

Dynamic filtering refers to one of the so-called "single-ended noise reduction" systems. It is an optional element here, but may be useful on noisy long-distance calls. Criteria of choice for a unit in this genre include its audio performance on band-limited material (most are specifically designed for tape-hiss elimination on wideband recordings), its ease in set-up and operation, and how well it performs with your phone system. On some digital PBX's, use of a Dynamic Noise Filter (DNF) may cause quantization noise to be more audible than without its use, on relatively quiet local calls. In such cases, however, the DNF may still be of value on noisier long-distance calls.

Equalization

The single most audible improvement to a dial-up phone line may be made with proper use of an equalizer. Calls in which the caller's voice is transduced by the standard telephone mouthpiece microphone are especially helped in this fashion. These calls exhibit an excess of energy in the octaves centered at 500 and 1000 Hz, with falling response on either end of the passband. Typically, the telephone receiver earpiece corrects for this, and intelligibility is adequate. In the case of direct feeding through a telephone interface, however, such complementary interaction does not occur, and must be electronically simulated. No two phone calls will have exactly the same frequency response, but if a "typical" or "standard" curve were to be stated, it would have the following characteristics, listed in decreasing importance:

-6 dB at 350 Hz
-3 dB at 800 Hz
+3 dB at 2.5 KHz
+2 dB at 200 Hz

The general object is to use subtractive equalization where excessive line response renders audio "muddy", such as in the upper bass/lower midrange region from 300 to 900 Hz. A bit of boosting on either end of the phone-line's typical passband helps further. Note that simply shelving off the low end of the phone line below 800 Hz or so is not recommended, since it will render the audio "thin", and lacking in "fullness".

Any type of equalizer with boost and cut capability can be used, and although a fully parametric design would have the highest resolution, an octave-band graphic can perform adequately; in fact, graphic designs are usually preferred due to their quicker set-up, an especially important issue if EQ is custom-adjusted on each phone call (as is recommended for best results). Start with an approximation of the generic curve above, and tailor by ear for maximum intelligibility. A "paragraphic" design that incorporates the features of both parametric and graphic equalizers is perhaps best suited here.

Compression

Gain-reduction processing is a two-edged sword in this application. Compression is desirable to increase the loudness of a band-limited signal, and restore loudness lost in the subtractive EQ process mentioned above. This is especially important when the phone audio is heard right up against a wideband audio source, such as in a typical telephone interview, where the studio talent is heard via a high quality microphone, and the interview guest via a 3.5 KHz phone line. If both elements are set for equivalent maximum electrical levels on the mixing console's meter, the studio voice will often seem much louder to the listener. Either the studio voice must be lowered in level (thereby reducing the entire program's loudness), or the phone voice must be discretely compressed, without affecting the studio voice, until both element's loudnesses match, with meter readings remaining in the acceptable range.

Unfortunately, this same process can increase the noise floor of the telephone audio, unless a sophisticated, gated compressor is used. Moreover, in phone-interview applications, trans-hybrid loss is reduced by the amount of compression used, thus increasing the amount of studio voice that gets back into the mix through the telephone interface's output, coloring the studio voice.

Therefore, compression should be used with moderation and care, when necessary for increasing phone-fed audio's subjective loudness only.

Aural Enhancement

As a final touch, one of the devices generically referred to as "aural exciters" (although this term is a trademark of Aphex Systems, Ltd., one of the leading manufacturers of these units) may be used. Again, its settings should be adjusted to the particular audio in use, since it can help some phone lines and hinder others.

Stereo synthesis is generally not particularly helpful, since most of these units employ a band-splitting technique, assigning alternate bands to left or right channels, and the phone audio typically falls mostly into only one band (and therefore only one channel). Line noise, moreover, may be assigned to the other channel, and thus made even more noticeable.

Processing Chain Order

For best results, telephone audio should pass in the following order:

1) **Telephone Interface**
2) **"Static" HPF, LPF & Notch filters**
3) **Dynamic LPF** (optional)
4) **Equalizer**
5) **Compressor** (if necessary)
6) **Enhancer** (optional)

Other Dial-up Issues

A good telephone interface device, with high audio quality and adequate trans-hybrid loss is essential. Great advances have been made in this technology in recent years, and these should be taken advantage of. Trans-hybrid loss of 15 dB or more should become a standard criterion. Devices using digital techniques exist now which can provide up to **40 dB** of trans-hybrid loss.

A "mix-minus" backfeed must be provided to the telephone interface input when telephone interviews are conducted.[1]

AUDIO PROCESSING OF LEASED CIRCUITS

Leased circuits or "private lines" come in various bandwidths, typically with high-frequency cutoffs of 3.5 KHz, 5 KHz, 8 KHz, and 15 KHz. Audio processing needs are fewer on these, since their performance should be stable and guaranteed. Nevertheless, the following practices are often helpful, once the lines have been properly interfaced to and verified to be within spec.[2]

3.5 KHz Circuits ("D-lines")

These are often thought to behave much like dial-up circuits but they possess several important differences: they should be relatively flat within the passband, the passband goes down lower on the bottom end (generally to 100 or 150 Hz), and the S/N is lower. For these reasons, much of the above audio processing recommended for dial-up lines is rendered unnecesssary, and other possibilities are opened up, most notably the use of complementary noise-reduction (see below).

Static or Dynamic Filtering may still be of use on these lines when complementary noise reduction is not used.

Some mild EQ may be added to taste, but the radical compensatory techniques mentioned above should not be necessary here.

Compression may be helpful in increasing loudness, with less penalty in noise floor increase. Also, since these lines are not interfaced through a hybrid, but typically exist in a "four-wire" arrangement, trans-hybrid loss reduction is not a factor either.

Aural Enhancement may also be of significant value to these lines.

5 KHz lines

Some filtering or mild EQ may be useful here. Long-distance circuits may be especially improved by static or dynamic LPF's, or complementary noise-reduction (see below). Aural Enhancement may also be worthwhile.

8 & 15 KHz lines

These generally do not require significant audio processing, although for critical music applications or long-distance circuits, complementary noise-reduction may be helpful (see below). All pairs of lines for stereo use should be ordered as "stereo conditioned", i.e. phase-matched. Protection limiting at the **transmit** end may also be of value, since headroom is generally quoted as only 10 dB (+18 dBm). Additionally, a lower reference level than telco's +8 dBm may be used, thus trading S/N for additional headroom.

FREQUENCY EXTENSION

Dial-up lines, 3.5 and 5 KHz circuits can be improved dramatically by the process known as frequency extension. It is a complementary process, requiring hardware at both the transmit and receive locations.

Single-line systems

The simplest frequency extension scheme involves the addition of low frequencies to telephone-transmitted audio, at the expense of high frequencies, in a linear fashion. The advantage is that cycle per cycle, such a tradeoff presents a greater portion of an octave to be gained on the low end than is lost on the high end. For example, one typical system reduces high frequency performance and extends low frequency performance by 250 Hz. On a dial-up line, with 300 - 3 KHz response, the 250 Hz loss on the high end means a 1/6 octave loss, while the 250 Hz applied to the low end provides about **2 1/2** octaves of increased bandwidth.

The system works as follows. Using single-sideband heterodyning, audio is shifted up by 250 Hz just before being sent into the phone line. The phone line still only passes 300 to 3K, but what really gets through is 50 to 2750, masquerading as 300 to 3K due to the frequency shifting. A complementary downshifting is accomplished by a mirror-image heterodyning, and the audio that arrives at the output on the receive end has a bandwidth of 50 Hz to 2750 Hz. Similar extensions are achieved with other, wider bandwidth lines; devices in current use provide from 250 to 350 Hz on the low end. Note than any LF noise (typically, mains hum) added by the phone line will be downshifted as well, usually below the audible range.

Dual-line systems

Here, two lines (dial-up, 3.5 KHz or 5 KHz are appropriate) are used, to provide frequency extension on **both** ends of the passband. First, audio is bandsplit, typically around 2.5 KHz. Line 1, or the "low-band", operates on the audio below the crossover, functioning as in the single line system above. Line two, the "upper-band", operates on audio above the crossover, but in this case, a **down**shift of around 2 KHz is applied on the transmit end, thereby translating **2.3 to 5 KHz** into frequencies that will pass through a 300 to 3K line. On the receive end, complementary down- and up-shifting are performed to the lower and upper bands respectively, and the two bands recombined, providing a 50 to 5 KHz response on two dial-up lines. On wider bandwidth lines, the extension is 250 to 350 Hz below the passband on the low end, (depending on the particular brand of equipment used) and around 2 KHz above the passband on the high end.

Most of these two line systems also incorporate some sort of complementary noise-reduction system. As a result, their set-up is a bit more painstaking than the single-line system, and some noise-gating may be evident on extremely noisy lines, especially when half-speed transmission is used (see below). Moreover, access to two lines at the remote location is necessary, which may be a problem in typical dial-up use (**three** lines, if a backfeed for two-way communication or interview is required), and both lines must be time-aligned in the grossest sense, i.e. both satellite or both terrestrial, not one of each.[3]

Half-speed Transmission

When live or real-time feeding is not essential, field-recorded audio may be sent at half its original recording speed. It is a well-known fact that playing a recording back at half its original speed and re-recording it through a band-limited system will result in an altered frequency response when the receive-end recording is played back at the proper (i.e. double) speed.

What occurs is a one octave loss of response on the low end, and a one octave gain on the high end. As in the frequency extension above, the passband is effectively shifted, such that one end is traded for the other, but here there is no aural advantage, since the gain in octaves is exactly duplicated in the other end's loss, as opposed to the **linear** shifting's unequal octave redistribution. Since phone-lines' passbands are set not arbitrarily, but sensibly, for balanced frequency response on voice and music transmission (the so-called "Rule of 400,000, alternately quoted as the "Rule of 500,000", where multiplication of the two cutoff frequencies should equal 400,000 [or 500,000], as 20 x 20,000 does), such an octave shifting is of no value, in and of itself. Enacting such a technique on a dial-up line provides an unacceptably "thin" result. Combining this technique with frequency extension, however, can provide considerable improvement.

The process utilizes one of the frequency extension systems, which add approximately 2.5 octaves to the low end, as mentioned above. Since the low end is now so plentiful, one of these newly gained octaves can inturn be resacrificed for an octave of increase on the top end. (The full 2.5 octave LF increase clearly violates the Rules of 400,000/500,000, and often does sound quite bass heavy in practice. The loss of one of these octaves is generally beneficial, in fact, making half-speed with frequency extension doubly preferred over real-time frequency extension. Without the excess low end, however, half-speed transmission would not be possible.)

Step by step, the process works as follows. Assume we are using a system with 250 Hz of low end extension on a dial-up line with 300 to 3 KHz response. The response of this line is 50 to 2750 Hz after extension. If this was half-speed audio being transmitted, the 7.5 ips recording made at the receive end can be played back at 15 ips, and its response will be 100 to 5500 Hz, a fairly well balanced result, satisfying the "Rules" noted above. Note that this half-speed approach will yield about the same frequency response with one line that the two-line systems provide at real-time.

Of course, transmission time is also doubled, increasing (but usually not doubling) long-distance charges. Moreover, we have found that lowpass filtering the receive-end audio at 3 or 3.5 KHz **ahead** of the extension system's decoder is helpful in reducing unwanted noise for all extension system lines, but especially in half-speed applications.

15 ips recording and 7.5 ips playback is recommended for the remote end, with 7.5 ips recording and 15 ips playback for the receive end.

Half-speed may be used in two-line extension systems or with wider bandwidth circuits, shifting the passband up by one octave in all cases.[4]

COMPLEMENTARY NOISE REDUCTION

On leased circuits only, external noise reduction systems may be used to advantage. Among existing systems, those with the least sensitivity to varying levels (non-"threshold" systems), and those with the least amount of preemphasis, are the most appropriate. In addition to substantially lowering the noise floor, use of a system of this type more importantly lowers the audibility of unwanted coherent signals, such as crosstalk and impulse noise.

Two important caveats are critical. The first regards flatness of response. Most noise-reduction systems are quite sensitive to frequency response anomalies between encode and decode stages. If substantial variation occurs, the noise-reduction system can at the least exaggerate the anomaly, and at worst exhibit dynamic mistracking, causing gating, pumping, overshoot and noise-floor modulation.

Secondly, the noise-reduction system must be tailored for use with a band-limited system, such that audio is pre-filtered ahead of the encoder on the transmit end with a response that imitates the passband of the line being fed. Otherwise, the encoder would act on wideband material, and the decoder, acting on bandlimited material, could not act in a complementary fashion, regardless of the flatness of the line within the passband. Both of these concerns must be satisfied, and vigilantly maintained, for complementary noise-reduction systems to properly function on program circuits.

Leased lines are the only ones appropriate for this, since they are the only case in which the customer may demand flat response within the passband. Dial-up lines will vary widely in this respect, and the user is powerless to change this.

OTHER APPLICATIONS

Although the assumption has been made that the above aural improvements are to be made on terrestrial phone lines, the same techniques may be applied to satellite, microwave and STL or RPU applications. Refer to the LEASED CIRCUITS section with the corresponding bandwidth above for specific recommendations.

REFERENCES

1) **"Understanding Mix-Minus"** - Donna Carter & Skip Pizzi, Mix Magazine, September, 1987.

2) **"Dealing with Phone Lines, Pts. I & II"** - Skip Pizzi, Mix Magazine, April & May, 1985.

3) **"A Five-Band Companded Technique for Converting Telephone Quality to Broadcast Quality Using Two Voice-Grade Phone Lines"** - Daniel B. Talbot & John F. Cheney, Journal of the Audio Engineering Society, October, 1986.

4) **"Tips on Recording from the Telephone"** - Skip Pizzi, db Magazine, March, 1983.

AN ANALYSIS OF THE FCC'S FM STATION SEPARATION METHODS IN VIEW OF DOCKET 87-121

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Abstract

In May of 1987 the FCC initiated Docket No. 87-121, a proceeding to consider modification of the FM rules to permit reduced distance separation between existing commercial FM stations and allotments by utilizing directional antenna systems. While making site selection more flexible, the plan suggested by the Commission could also carry a future price of slightly greater interference between some stations. This paper reviews the basis of the present allocation system, discusses interference between stations providing minimal protection, and considers some of the tradeoffs involved if new rules were adopted.

Background of Docket 87-121

A Notice of Inquiry initiated by the Commission in Docket No. 87-121 suggests that an overhaul in the separation requirements for commercial FM transmitter sites is possible by using directional antennas and some form of signal contour protection. The Commission noted that directional antennas are currently used to avoid interference in special cases and are routinely permitted for the noncommercial FM service. If eventually adopted, the changes might eliminate the current protection requirements based on distance, in favor of signal contour protection equivalent to the minimum separation distances.

A number of issues must first be resolved before adoption of new separation rules for commercial FM stations. Among them are the specific method for determining contour protection, directional and nondirectional antenna performance requirements, determination of the protected contour for existing stations, and whether to alter the protection ratios for Class B and B1 stations (which differ from all other classes, including NCE-FM operations).

The Present Allocation System

For the past 25 years, the Commission has assigned commercial FM channels on the basis of distance separations rather than signal strength contour protection. This system fixes minimum geographical separations according to the class of each FM station, which ranks facilities within the six brackets of power and height shown in Table 1.

The six classes are listed in the table, for different levels of service. Each class has been assigned a particular range of operating characteristics, i.e., maximum to minimum allowed effective radiated power, and a maximum to minimum allowed antenna height. The distances to the F(50,50) 60 dBu contour are shown for maximum and minimum facilities within each class.

The present allotment system assumes each station operates with maximum facilities for its particular class (maximum effective radiated power and antenna height). The transmitting antenna is assumed to be omnidirectional, and its height above terrain is assumed uniform in all directions (based on the average height above eight standard radials).

The 60 dBu (1 mV/m) contour represents the protected contour for interference determination purposes for Class A and C stations. The 54 dBu (0.5 mV/m) and 57 dBu (0.7 mV/m) contours are the protected contours of Class B and B1 stations, respectively, although their usable service may not extend to these predicted contours. It is apparent that a wide difference in contour distances exists between maximum and minimum for facilities of the same class.

Interference Ratios Related to Existing Separations

The FCC's present distance separation rules were derived from a minimum cochannel signal-to-interference (S/I) ratio of 20 dB and minimum adjacent-channel S/I

Table 1 FM Station Classes and Contour Distances

Class	ERP (kW)	HAAT (meters)	F(50,50) 60 dBu km	F(50,50) 60 dBu (mi.)	(Class B & B1 only) F(50,50) 54 or 57* dBu km	(Class B & B1 only) F(50,50) 54 or 57* dBu (mi.)
A max.	3.0	100	24.3	(15.1)		
A min.	0.1	30	5.7	(3.5)		
B1 max.	25	100	39.1	(24.3)	44.7*	(27.8)*
B1 min.	3.1	30	13.5	(8.4)	16.1*	(10.0)*
B max.	50	100	52.3	(32.5)	65.0	(40.4)
B min.	25.5	30	23.0	(14.2)	31.2	(19.4)
C2 max.	50	150	52.3	(32.5)		
C2 min.	3.1	30	13.5	(8.4)		
C1 max.	100	299	72.2	(44.9)		
C1 min.	51	30	26.9	(16.7)		
C max.	100	600	91.7	(57.0)		
C min.	100	300	72.4	(45.0)		

ratio of 6 dB. Figure 1 illustrates how these ratios relate to the protected and interfering contours of each class of station. Table 2 list the F(50,50) field strength values of protected contours for each class and the associated F(50,10) field strenths for interfering contours.

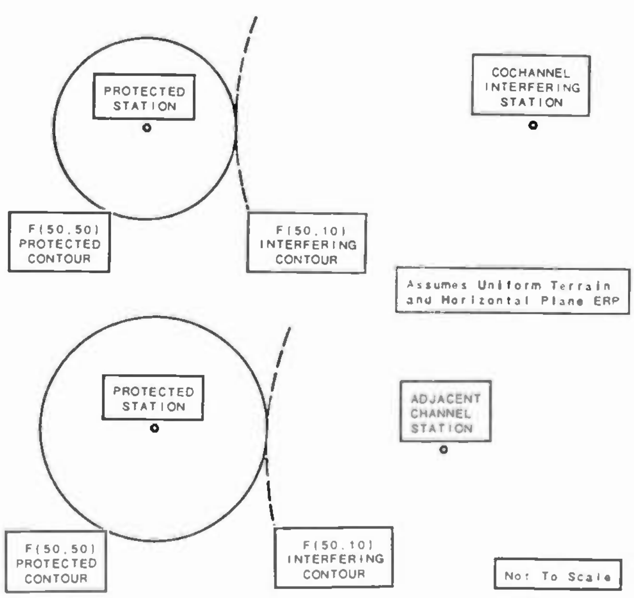


Fig. 1 - Pertinent FM Contours For Cochannel and Adjacent Channel

Table 2

Station Class	Protected Contour (dBu)	Interfering Contour Cochan. (dBu)	Interfering Contour Adj. Ch. (dBu)
C	60	40	54
C1	60	40	54
C2	60	40	54
B	54	34	48
B1	57	37	51
A	60	40	54

Table 3 is excerpted from Section 73.207(b)(1) of the FCC Rules to show the distance separations permitted between the various combinations of classes.

Table 4 shows the computed S/I field strength ratios between Class A, Class B and Class C stations, using the minimum permissible separation distances against various combinations of station size. Four sets of columns display the S/I ratios assuming the following combination of facilities for the existing station compared to the interfering station: maximum to maximum, maximum to minimum, minimum to maximum, and minimum to minimum. (Other combinations are possible, of course, but are omitted for reasons of space.)

A study of the first (max/max) column shows that the cochannel S/I ratios are close to, but slightly greater than, 20 dB. (The slightly higher values are probably due to metric conversion and rounding from the original English unit calculations, which were also rounded values.) Examination of the columns to the right show that the S/I ratios are substantially higher between stations which operate with minimum facilities. The higher ratios imply less interference between these services. Class B to Class B S/I ratios were also computed at the 60 dBu contour for the existing minimum separation distances, and show the expected increase in protection ratios.

Table 3 Required Separation

Relation	Cochannel (km)	Adjacent Channel (km)
A to A	105	64
A to B1	138	88
A to B	163	105
A to C2	163	105
A to C1	196	129
A to C	222	169
B1 to B1	175	114
B1 to B	211	145
B1 to C2	200	134
B1 to C1	233	161
B1 to C	259	193
B to B	241	169
B to C2	241	169
B to C1	270	195
B to C	274	217
C2 to C2	190	130
C2 to C1	224	158
C2 to C	249	188
C1 to C1	245	177
C1 to C	270	209
C to C	290	241

Examination of the adjacent channel S/I ratios reveals a similar situation: the ratios approach the nominal 6 dB value for minimum-spaced maximum facilities, and increase substantially between the services involving minimum facilities.

In its Notice, the FCC states that protected contour overlap criteria is complicated by the fact that Class B and B1 operations use different protect contour values. Accordingly, the S/I ratios for these services shown in the Table 3 are based on protection to the 54 dBu contour. Since these contours extend farther than the 60 dBu contour, protection is increased. It is evident that these operations enjoy significantly higher S/I ratios at the 60 dBu contour than Class A or Class C operations. It is also probable that Class B or B1 stations would be most affected if the Commission adopted a uniform protection contour of 60 dBu.

The FCC suggests in the Notice that any change in separation requirements would still include the requirement to contain the entire principal community of service within the 70 dBu (3.16 mV/m) field strength contour. Stations on adjacent channels have much shorter separation requirements than cochannel operations, therefore the same displacement away from the community reference coordinates (the FCC's nominal site for an FM station allotment) may result in significantly larger reductions in effective radiated power in some directions for adjacent channel operations than for cochannel operations. This matter should receive more careful analysis before rule changes affecting separation are proposed.

The Commission's Notice suggests that separations could be determined by either of two methods. In one case, the ERP/HAAT combination on the azimuth between the affected stations would be used to classify each operation. The classifications would then determine the minimum separations between these operations. As an alternative, the Commission suggests that protected and interfering contours be used to determine minimum separation, as is done presently with noncommercial FM stations.

The present approach to separating FM stations results in greater than minimum spacing for two reasons: First, some FM stations operate with less than maximum power or height for their class, which reduces the range of potential inter-

Table 4 Examples Of Signal To Interference Ratios Assuming Minimum Separation Distances

Facility Relation:	Cochannel Separation: Desired F(50,50) to Undesired F(50,10) Ratio (dB) max/max	max/min	min/max	min/min	Adjacent Channel Separation: Desired F(50,50) to Undesired F(50,10) Ratio (dB) max/max	max/min	min/max	min/min
A to A	21.3	40.0	26.3	43.8	7.4	31.3	14.5	35.8
B to B	20.8	28.3	26.8	34.1	7.3	14.7	14.0	21.2
C to C	20.1	24.8	23.5	28.1	9.3	16.0	13.2	19.4
Present Class B Protection Ratio At The 60 dBu Contour								
B to B	32.0	36.6	34.2	41.7	16.0	23.2	21.6	28.7

ference to adjacent and cochannel FM stations. Second, the FCC's allocation process has assigned channels to communities that are randomly spaced and therefore are likely to be at greater than the minimum geographic separation.

In a study of approximately 3,700 FM stations conducted in June, 1982, by the Technical Subgroup of the NAB's Radio Advisory Committee, the median mileage separations for Class A, B, and C channels were 105, 190, and 230 miles, respectively, for the nearest three cochannel stations. (See A Study of FM Station Mileage Separations Prepared For The Technical Subgroup Of The Radio Advisory Committee, dated November 17, 1982.) Then, as now, the FCC permited minimum separations of 65, 150, and 180 miles (105, 241 and 290 kilometers) for Class A, B, and C stations, respectively. On the basis of this difference, the Subgroup concluded that many FM stations were operating with less adjacent or cochannel interference than is permitted by the FCC Rules.

By employing a directional transmitting antenna, FM stations could reduce their radiation in the direction of a critical service to that equivalent to a minimally-spaced non-directional FM service. This approach would allow stations greater freedom to locate their transmitter sites, supposedly without causing more interference than would be technically possible between two stations at minimum separation with maximum facilities. The second article on MM Docket 87-141 will discuss the audio signal to noise ratio that results from S/I ratios of 20 dB (cochannel) and 6 dB (adjacent channel), the question of statistically how much interference is possible from minimally-spaced FM stations, directional antenna performance, and related matters.

A computer study of separation distances between 4052 FM stations in the continental United States was conducted for this paper. The results of the study are included as an appendix. These histograms show the number of licensed FM stations of a given class (named first) to licensed stations of a given class (named second). The distance brackets are five kilometer wide rings around the stations studied.

While data averages are not possible herein (since stations may be counted in more than one distance bracket) the graphs show the relative distributions of station separation, compared to the FCC minimum separation (indicated by a star on the distance scale). For example, Figure A-1 shows that a substantial number of Class A stations (235, to be exact) are located between 105 and 110 kilometers (the lowest permissible distance bracket) from other Class A cochannel stations. The number of stations separated by 100 to 105 kilometers is sharply lower (only 53), as are those stations separated by more than 110 kilometers.

Contrast the Class A to Class A cochannel spacing with Class B to Class B cochannel spacing depicted in Figure A-3. While dozens of Class B stations operate below the required 241 kilometer separation, the tendency of the distribution is toward far greater separation, peaking at approximately 325 kilometers.

It is the author's view of the data that the separation distance between stations tends to peak a much greater distances than the specified minimum distance for all classes of stations, with the exception of Class A to Class A cochannel operations. The conclusion of the Technical Subgroup in 1982 is generally true: stations operate at greater than minimum separation (and less than maximum mutual interference). The situation of Class A stations may have changed for the worse in that time, however, due in part to the large number of Class A allotments created by Docket 80-90. The extent of interference resulting from minimum cochannel and adjacent channel separations will be considered next.

Underlying Basis for Station Separations

As discussed earlier, the FCC's FM allocations technical standards are based, to a large degree, on a cochannel signal-to-interference (S/I) ratio of 20 dB. This ratio was established approximately 40 years ago when the only FM transmission mode was monophonic.

A 20 dB cochannel S/I ratio results in an audio signal-to-noise (S/N) ratio of approximately 50 dB for monophonic reception. For adjacent channel operation, the FCC expected that a 6 dB RF S/I ratio also results in a 50 dB monophonic audio S/N ratio (although this value is affected much more by receiver selectivity and capture effect than is cochannel performance).

For stereophonic transmission, the interference ratios are considerably different. Tests conducted for the National Radio Systems Committee concluded that a 50 dB stereo signal to noise ratio of 50 dB requires a 40 dB RF S/I ratio for cochannel stations, and a 25 dB RF S/I for adjacent channel stations. These values represented the mean for measurements of 17 receivers ranging from low cost to high cost models. (See Report to the National

Radio System Committee by the FM Subcommittee Task Force, July 24, 1981.)

Recent tests have characterized a 50 dB S/N ratio degraded by interfering signals as "perceptible but not annoying" for most radio programming. In the same tests, a 30 dB AF S/N ratio was generally characterized as "annoying". (See Subjective Evaluation Of Audio Degraded By Noise And Undesired FM Signals, A Report by the Technical Subgroup of the Advisory Committee on Radio Broadcasting, National Association of Broadcasters, November 17, 1982. See also FM Broadcasting Receiver Characteristics and Protection Criteria: A Report by the Technical Subgroup to the Advisory Committee on Radio Broadcasting, July 7, 1982.)

Stereophonic sound is the FM transmission standard today, and many receivers have neither manual nor automatic means to override the stereo mode when interference occurs.

The common international standard for FM stereo broadcasting supports this finding. The CCIR (the International Consultative Committee on Radio) believes a 50 dB stereophonic S/N ratio is necessary to provide a quality FM broadcasting service.

Little or no improvement has occurred in pertinent aspects of FM receiver performance since the adoption of the FCC's adjacent and cochannel allocations policy. One exception is capture ratio performance (a measure of an FM receiver's ability to suppress a weaker undesired signal), which has improved significantly in the past 40 years. The capture effect of an FM receiver is determined by the modulation index of the FM system being received, but this index is low for broadcast FM stereo.

The modulation index is the quotient of the peak deviation divided by the highest modulating frequency. For monophonic FM this is

$$75/15 = 5 \quad ;$$

for stereophonic FM the modulation index is

$$(75-7.5)/53 = 1.3$$

(where 7.5 kHz is deducted for stereo pilot injection).

Because stereophonic FM has relatively low modulation index, the broadcast receivers have a minimal ability to capture the stronger signal and suppress interfering signals. FM receiver adjacent channel performance (+/- 200 kilohertz) has generally not improved in the last few decades.

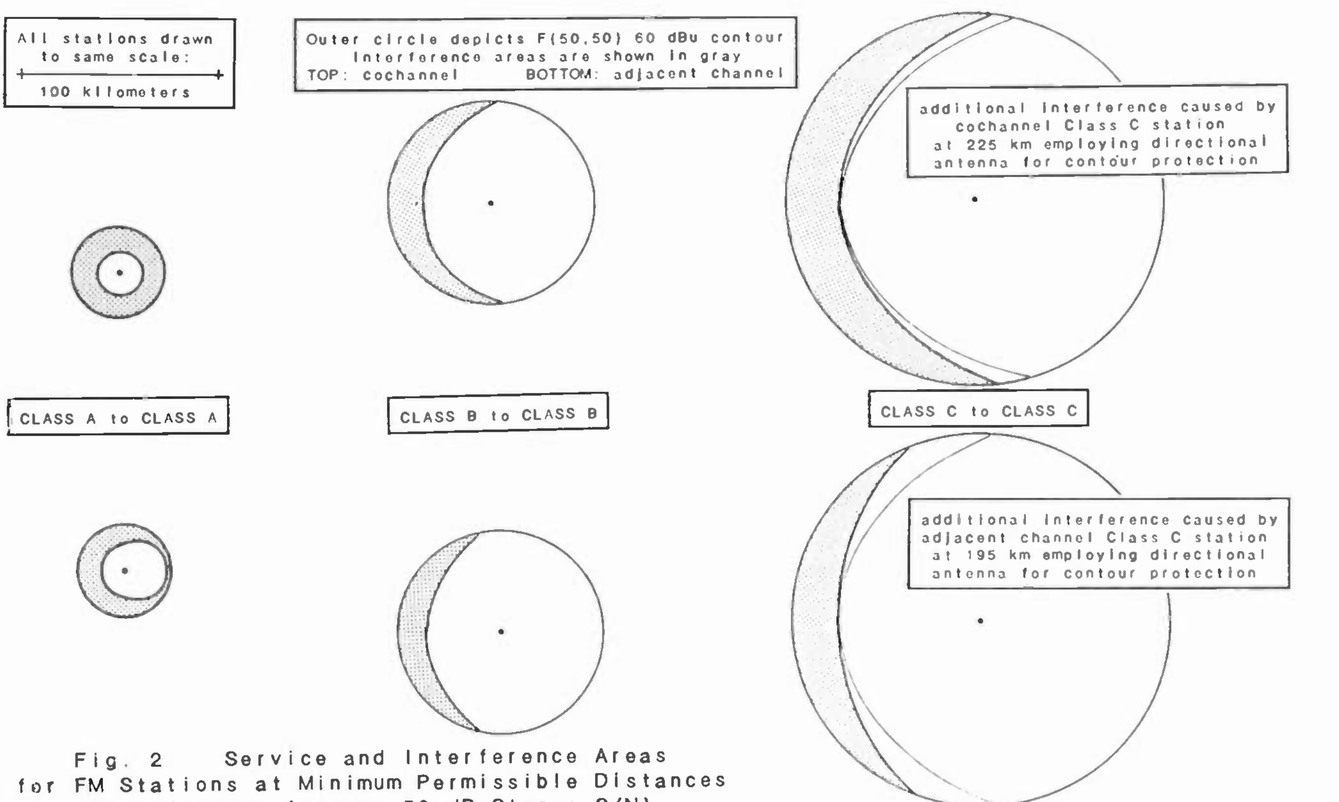


Fig. 2 Service and Interference Areas for FM Stations at Minimum Permissible Distances (Interference Assumes 50 dB Stereo S/N)

The demand for high fidelity equipment with low audio distortion and wide stereo separation has encouraged development of receivers with rather wide and flat filter characteristic. However, broadening a receiver's response across the desired channel reduces its ability to select against adjacent channel interference. (On the other hand, second and third adjacent channel selectivity of average receivers has improved, but this change is not exploited in the present proceeding.)

The diagrams of Figure 2 were prepared to illustrate the areas of interference that are predicted to result to three classes of adjacent and cochannel separations: A-to-A, B-to-B and C-to-C classes. (Of course, many other interclass combinations are possible). The separations are the minimum distances allowed under present rules, and are roughly equivalent to the contour protection methods suggested in the Commission's Notice.

The diagrams are based on the maximum facilities for each class and minimum separation distances described earlier. The outer circles depict the 60 dBu (1 mV/m) contour for each class of station. Omnidirectional transmitting antennas and uniform terrain are assumed for both the desired and interfering station. The distance scale is the same for each class so that service areas may be compared.

The shaded zone depicts the locations where the ratio of desired F(50,50) to interfering F(50,10) field strengths are 40 dB and 25 dB for cochannel and adjacent channel operation, respectively. These areas would result in a stereo signal to noise ratio of less than 50 dB for the desired station. (Note that in the case of Class A stations, interference sweeps fully around the site of the desired transmitter!)

If simple "contour protection" methodology were employed by a proposed station operating with less spacing than presently permitted, greater interference could occur to all classes of stations. As shown previously, the interfering contour of omnidirectional stations is theoretically an arc that is nearly tangent to the protected contour at only one point.

In the case of directional antennas, the effective radiated power of the new (interfering) station can be controlled in a manner that conforms its interfering contour to the protected station's contour across a broad arc. Although nowhere along the 60 dBu contour would the required S/I ratio be exceeded, a carefully designed directional antenna would extend the edges of the interference crescent further around the desired station's service area. This effect is illustrated in Figure 2 for a Class C cochannel interference case for the 50 dB noise-limited service area.

The service areas for maximum operations in each of the three sample classes is listed in Table 5. The interference areas are also listed, along with the percentage of service area receiving interference.

It should be noted that the interference areas depicted in Figure 2 and listed in Table 5 are the locations where the probability of interference exists to more than 50 percent of the locations for more than 10 percent of the time. There are more statistically valid ways of describing the interference to service, but these methods are beyond the scope of this paper.

It may come as a surprise to some that such large interference areas might occur between adjacent and cochannel operations. In reality, there are factors which limits the actual interference. As shown earlier, most stations operate at greater than minimum spacings due to the varying separation of communities chosen in the allotment proceedings. Also, some stations operate with less than maximum facilities, which reduces mutual inter-

Table 5 Examples of Interference Areas Assuming Minimum Separation Distances

Facility Relation:	Service Area	Interference Area			
		Cochannel		Adjacent Channel	
	(sq. km)	(sq. km)	(%)	(sq. km)	(%)
A to A	1855	1426	77	1037	56
B to B	8593	1733	20	1219	14
C to C	26417	5199	20	3210	12

ference. These factors could be easily dealt with in any type of contour protection method.

Pattern distortion of FM transmitting antennas is a common problem when mounted on a tower structure. Figure 3 is a measured pattern for a popular "omnidirectional" transmitting antenna leg-mounted on a tower having a face width of 36 inches. Stations which install such antennas may cause more interference than the FCC anticipates if the ERP exceeds the expected value in a particular lobe. Conversely, a station with this pattern may recieve interference if its ERP is below the expected value due to nulls. The FCC's present rules are not especially concerned with the performance of side-mounted FM antennas if they are described as "omnidirectional".

Figure 4 is a plot of Class B FM station's 70 dBu contour, determined in accordance with the FCC Rules. Overlaid with this contour is a plot produced by a terrain-sensitive coverage analysis computer program. This system evaluates terrain features which can affect signal propagation and adjusts field strengths accordingly.

The plot shows the radial segments of the same station for field strengths between 80 dBu and 70 dBu. It is readily apparent that some directions exhibit coverage far below the FCC-predicted 70 dBu contour, while other directions exhibit coverage beyond the predicted contour. Of course, this method could be applied to signal interference studies as well as service studies.

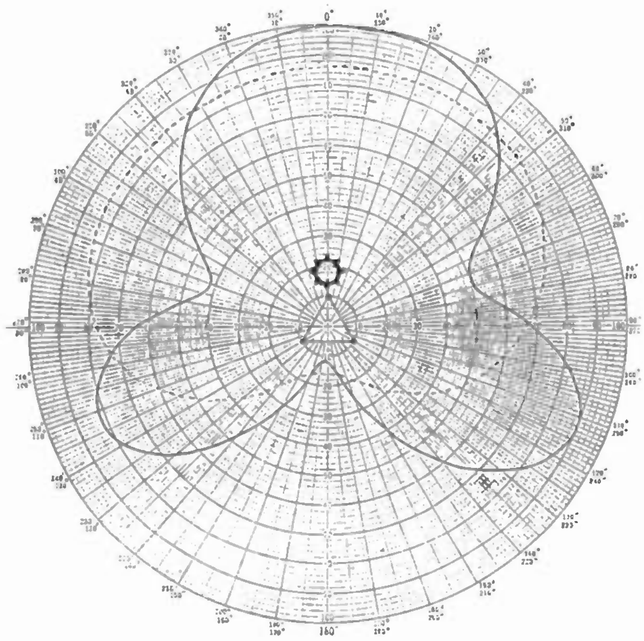


Fig. 3 Pattern Distortion Due to Side Mounting

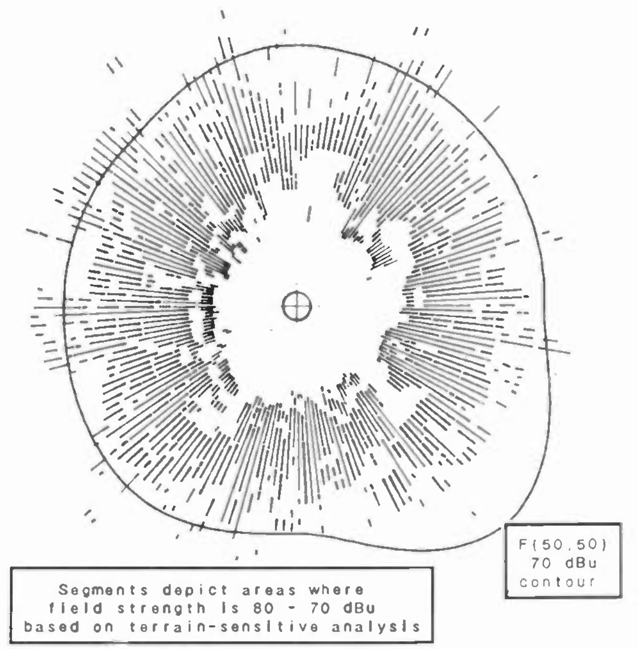


Fig. 4 Comparison of FCC Contour Prediction and Terrain-Sensitive Method

A more complex factor which reduces the interference in some cases is terrain shielding. Mountains, hills and ridges reduce the strength of FM signals, but these terrain features are not considered by the field strength curves. Therefore, desired signals may be weakened and interference increased over that assumed by the Commission's present coverage prediction method. Unusually flat terrain, such as lakes, deserts and plains is also ignored by the FCC's curves. In those cases, greater interference results between stations on adjacent and cochannels.

<u>Summary</u>

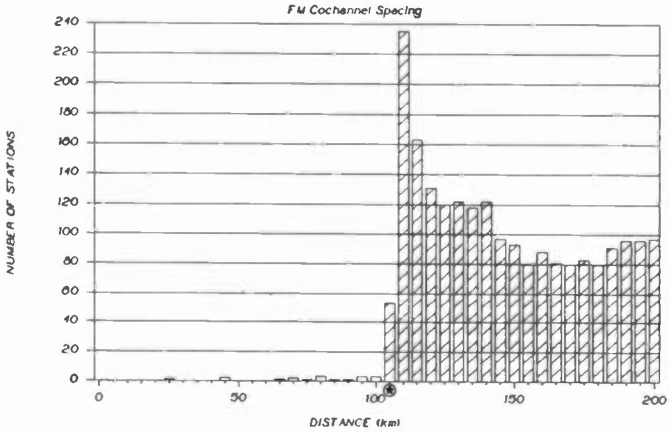
It is the writer's belief that consideration of terrain factors would be of significant benefit in protecting existing services from excessive interference, while permitting the flexibility of location possible with directional transmitting antennas. Equally important is the accurate determination of antenna pattern for omnidirectional transmitting antennas, as well as directional types. Field surveys of inteference levels around Class A stations should be conducted to ascertain whether significant loss of service is occuring, as predicted.

Acknowledgment

The author thanks Michael Degitz of Moffet, Larson & Johnson, Inc., for conducting the computer studies.

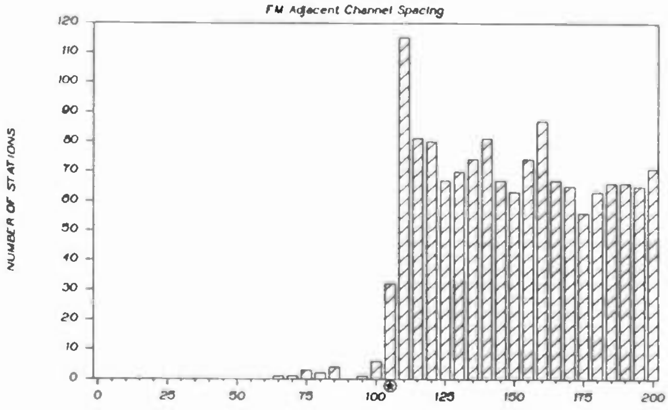
Class A to Class A

FM Cochannel Spacing



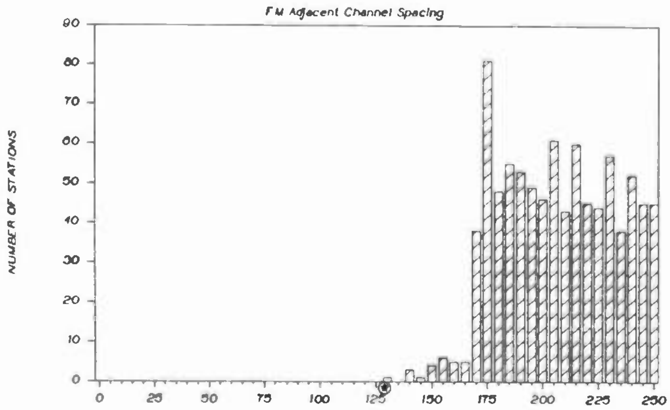
Class A to Class B

FM Adjacent Channel Spacing



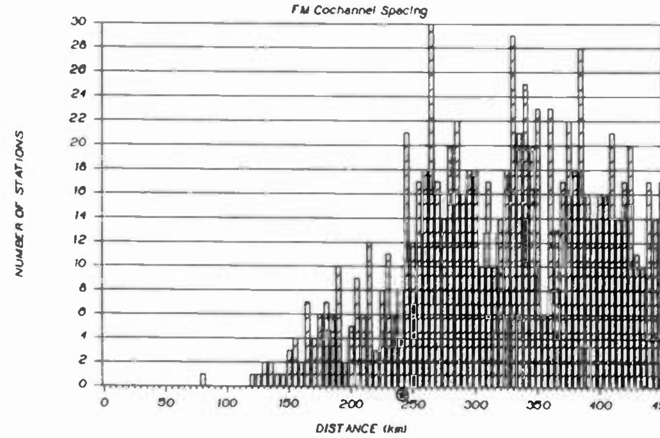
Class A to Class C1

FM Adjacent Channel Spacing



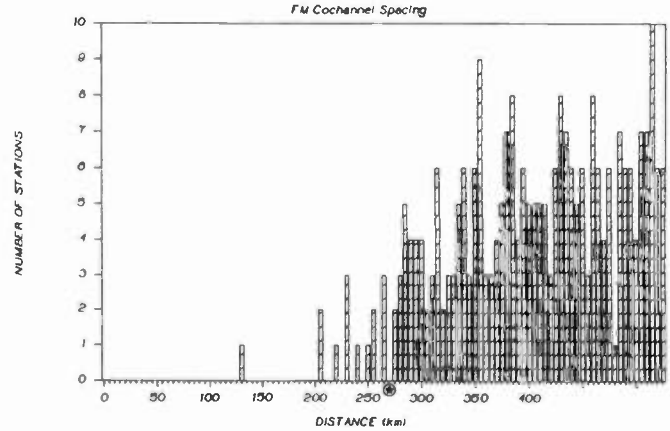
Class B to Class B

FM Cochannel Spacing



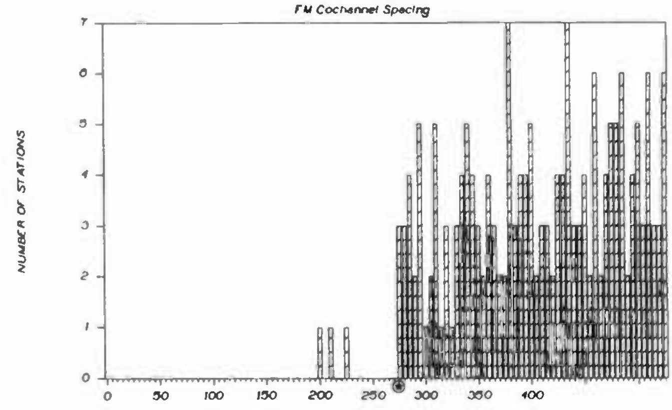
Class B to Class C1

FM Cochannel Spacing



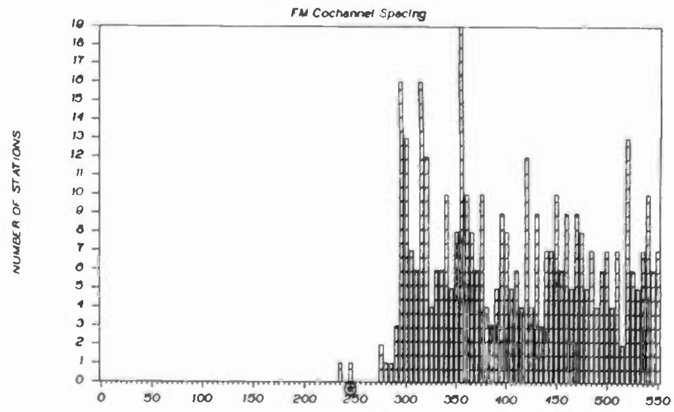
Class B to Class C

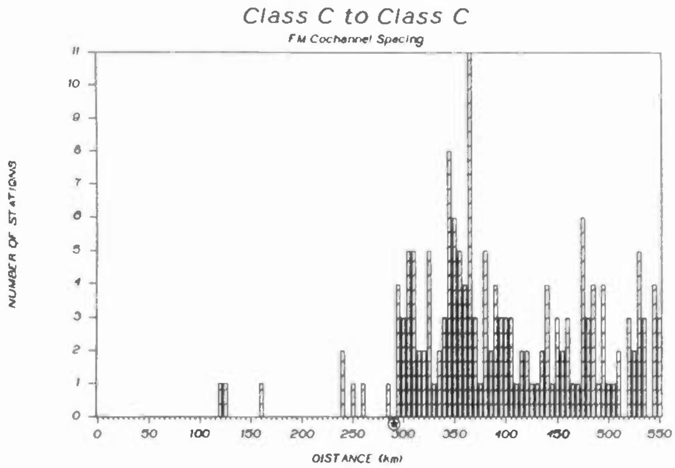
FM Cochannel Spacing

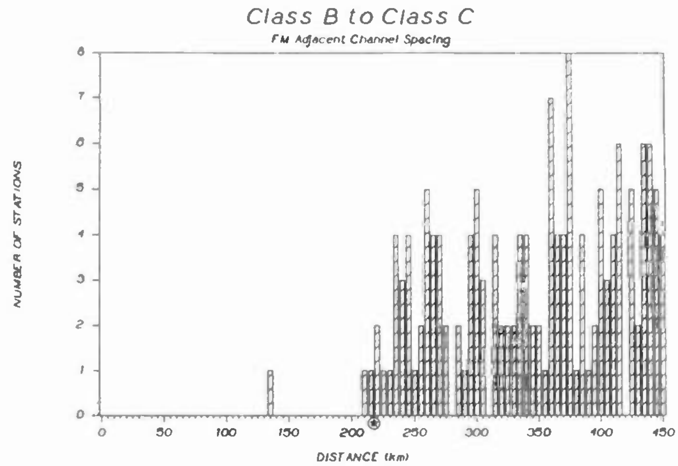


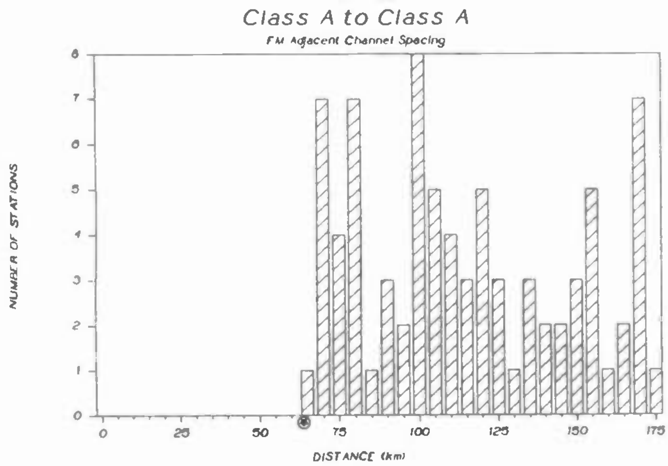
Class C1 to Class C1

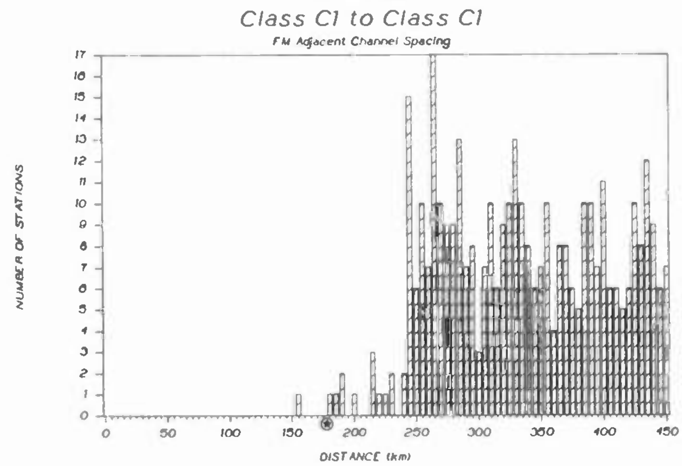
FM Cochannel Spacing

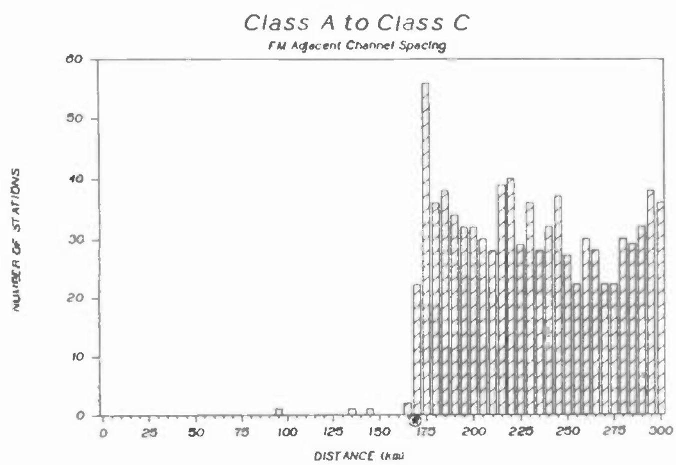


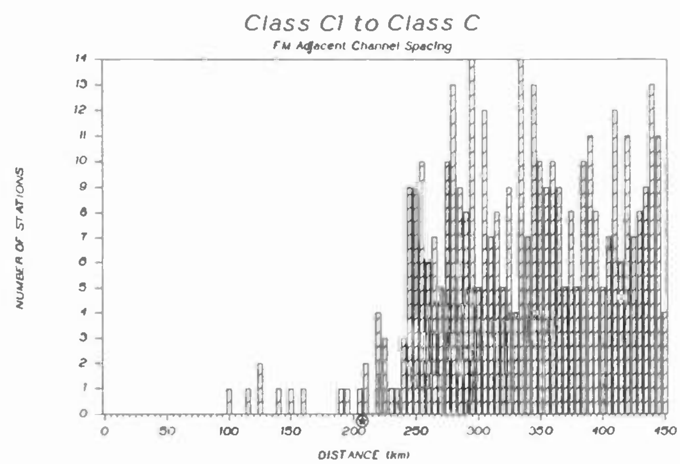

Class C to Class C
FM Cochannel Spacing
NUMBER OF STATIONS
DISTANCE (km)

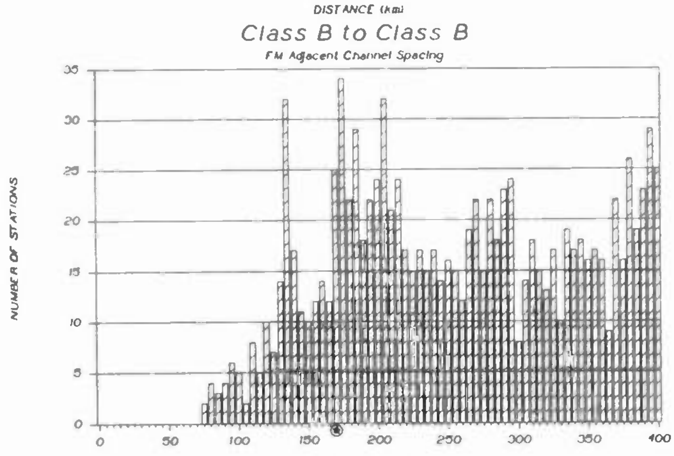

Class B to Class C
FM Adjacent Channel Spacing
NUMBER OF STATIONS
DISTANCE (km)

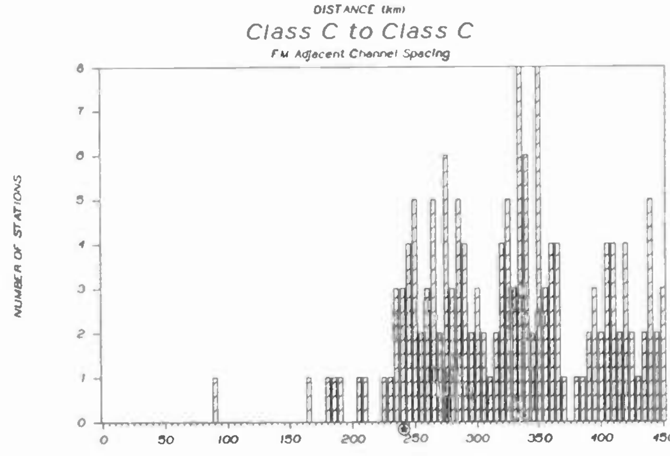

Class A to Class A
FM Adjacent Channel Spacing
NUMBER OF STATIONS
DISTANCE (km)


Class C1 to Class C1
FM Adjacent Channel Spacing
NUMBER OF STATIONS
DISTANCE (km)


Class A to Class C
FM Adjacent Channel Spacing
NUMBER OF STATIONS
DISTANCE (km)


Class C1 to Class C
FM Adjacent Channel Spacing
NUMBER OF STATIONS
DISTANCE (km)


Class B to Class B
FM Adjacent Channel Spacing
NUMBER OF STATIONS
DISTANCE (km)


Class C to Class C
FM Adjacent Channel Spacing
NUMBER OF STATIONS
DISTANCE (km)

MEDIUM FREQUENCY SKYWAVE PROPAGATION AT HIGH LATITUDES: RESULTS OF AN FCC SPONSORED STUDY

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Office of Science & Technology
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ABSTRACT

This paper describes the methodology and results of a Medium-Frequency (MF) Skywave radio/propagation investigation conducted at high latitudes. The field strength of transmissions from several standard AM broadcasting stations located in the northern tier of the U.S., in Canada and in Alaska were monitored for 5.5 years at Fairbanks, Alaska. The daily, seasonal, sunspot cycle and geomagnetic storm time variation of the field strengths are presented and discussed, along with various plots of median field strength of selected stations. This work was sponsored by the Federal Communications Commission and the Alaska Public Broadcasting Commission.

INTRODUCTION

If a radio communicator were to methodically search for a region on this planet where it would be the most difficult to communicate by radio (including interference effects) he would find Alaska high on that list! This would be true over *most* of the radio spectrum (ELF, VLF, LF, MF, HF, VHF and UHF). There are even *some* propagation problems in the *microwave* bands.

The salient reasons for these propagation problems are:

High Geographic latitude

The severe *climatic* conditions include:

- a large daily and seasonal *temperature variation* (expansion/contraction of antenna terminations, coax, connectors, etc.)
- *extremes* of temperature--down to -63°C)
- high *winds*
- *ice* accumulation
- *snow*
  - (blowing produces precipitation static)
  - accumulating--increases difficulty in access to installations for servicing and maintenance
- *surface and elevated temperature inversions* – these are some of the steepest inversions recorded in the world and can cause anomalous refractivity which sometimes affects propagation at UHF through microwaves.

Usually, high latitude locations for communication sites also have increased costs because of their remoteness from temperate latitude commercial supply centers. The auroral latitudes define that region of the ionosphere where the largest number of anomalies abound (large and small-scale irregularities in the E and F regions, enhanced absorption in the D-region, enhanced F1 layer effects, etc.). The polar ionosphere (~70°-90° Geomagnetic latitude) also displays considerably more anomalous behavior than the mid-latitude ionosphere. The effect of the polar and auroral latitudes on radio signals in the HF (3-30 MHz) portion of the spectrum are described in considerable detail by Landmark[1], Hunsucker[2], Lied[3], Folkestad[4], Hunsucker and Bates[5], Frihagen[6], Bates and Hunsucker[7] and Hunsucker[8].

Rugged Terrain

The variability of the terrain in Alaska is striking! One can go from sea water to freshwater glacier to precipitous mountain to alpine tundra in a horizontal traverse at $D \simeq 100$ km. The altitude range within Alaska is 6194 m from sea level to the top of Mt. McKinley compared to 4505 m from "Death Valley" to the highest point (Mt. Whitney) in the contiguous U.S. The conductivity and permittivity of these surfaces varies by several orders of magnitude and, of course, affects radio propagation in the VLF, LF, MF and HF bands. Quantitative values for the preceding parameters will be given later in this paper.

Logistics

Compared to a mid-latitude (temperate) communications site, high latitude sites have significantly higher installation and maintenance costs and decreased accessibility. Alaska has a population of ~400,000 dispersed nonuniformly over an area of

565,000 miles^2 (1,446,400 km^2). There are two cities of over 50,000 population and ~300 villages and settlements. No railroads exist from Alaska to the contiguous U.S., the one highway is at times marginal, and there are only a handful of ice-free ports for sea transportation. These reduced transportation routes and large distances from the communications equipment suppliers, of course, conspire to greatly increase the cost of installing and maintaining communication facilities. The hostile radio propagation conditions which are found in Alaska apply to many other countries and territories on the planet earth poleward of approximately the 60th parallel of geographic latitude.

The Federal Communications Commission (FCC) has sponsored a research program at the Geophysical Institute of the University of Alaska from 1981 - 1987 concerning medium-frequency (MF) skywave propagation at high latitudes. Assignment of broadcast band channels in the high latitude areas (e.g., Alaska) has, until now, been based on Part 73 of the Rules and Regulations of the Federal Communications Commission (FCC) which, among other things, contains MF skywave field strength curves. The data used to establish these curves were acquired essentially at mid-latitudes, hence these curves do not fully represent propagation conditions in the high latitude areas. The FCC and the operators of Alaskan broadcasting stations recognized this problem and the result was implementation of a research program at the Geophysical Institute of the University of Alaska Fairbanks to provide new MF skywave data over the appropriate paths. The study started at the Geophysical Institute May 15, 1981, and continued through 1987 in order to describe MF skywave behavior over one-half of a sunspot cycle. We have monitored the signal strength of selected standard broadcast stations in the contiguous US, Canada and Alaska during this project. This paper will discuss briefly some of the vagaries of high latitude ionospheric propagation, followed by a short technical description of our monitoring system and a more detailed description of the behavior of the MF skywave signal strength as a function of time of day, season, sunspot cycle and degree of ionospheric disturbance. Our "final product" to the FCC is median signal strength curves for MF skywave propagation at high latitude as a function of time of day, season, sunspot cycle and magnetic disturbance. These median curves should aid in the assignment of channels in the US standard broadcast Band for station in the northern part of the US and in Alaska--on a non-interference basis.

DESCRIPTION OF THE MONITORING PROGRAM

A. Location and Characteristics of the Field Site

Table 1 shows some characteristics of the "Ace Lake Field Site."

B. Description of Equipment

The equipment and antennas used in this experimental program are located at the Ace Lake Field Site approximately 4 km due west of the Geophysical Institute. The system is built around a commercial general purpose receiver modified for analog AGC output. The receiver frequency is automatically stepped through 16 channels every five minutes by the system programmer. Digital tape cassette recordings of signal amplitude are continuously made on ten or more standard broadcast stations. These data are then transferred on a regular basis to standard format computer tape for analysis on a VAX 11/780-785 computer. A noise source is also recorded continuously for regular system calibration. Aural identification of stations is made by an operator on a regular schedule. A top-loaded vertical antenna (TLVA) 32 m high with an extensive radial copper wire ground screen was utilized for most of the data acquisition on this project (see Figure 1). The TLVA was cross-calibrated with two other antennas used during the early years of the project to permit use of most of the data obtained over ~1/2 sunspot cycle.

Summary of the Primary Ground Characteristics Which Influence the Propagation of Radio Waves for Some of the Surface Types Found in Alaska

Type of Surface	Conductivity, σ in mhos (or Siemens) per meter	Permittivity (ϵ)
Coastal dry sand	0.002	10.0
Flat, wet coastal	0.01 to 0.02	4.0 to 30.0
Rocky land (steep hills)	0.002	10.0 to 15.0
Highly moist soil	0.005 to 0.02	30.0
Marshy	0.1	30.0
Hills (to ~ 1000 m)	0.001	5.0
Freshwater	0.001	80.0 to 81.0
Sea water	3.0 to 5.0	80.0 to 81.0
Sea ice	0.001	4.0
Polar ice (free)	0.000025	3.0
Polar ice (cap)	0.0001	1.0
Arctic Land	0.0005 to 0.001	23-34 for silts ~12 for dry sand
*Tundra underlain with permafrost		
(a) surface	~0.018 to 0.036	~25-42
(b) two feet below surface	~0.025 to 0.098	~33-788

RESULTS

Figures 2 and 3 are plots of receiver AGC voltage for various MF skywave transmissions monitored at Fairbanks. The location and frequency of each standard broadcast station are listed on the borders of the figures and universal time is plotted along the abscissa (Alaska standard, 150° west meridian time or UT - 10 hrs). The AGC voltage scale is included at the lower left corner and the calibration curve for converting receiver AGC to r.f. signal strength at the receiver input is given in Figure 4. The *diurnal* behavior of the MF skywave signals is shown for a magnetically quiet winter day for high sunspot number. For example, signals from KFAX and KGO (San Francisco) are received from ~06 – 16 UT or 2000 – 0600 AST in Figure 2.

FIGURE 1. THIRTY-TWO METER (106 FT) TOP-LOADED ANTENNA (TLVA) LOCATED AT ACE LAKE FIELD SITE NEAR FAIRBANKS, ALASKA. THE "TOP HAT" CONSISTS OF FOUR 22 METER WIRES AT AN ANGLE OF 45° WITH THE TOWER. THE TLVA IS FED AT THE BASE THROUGH A 16.5 μHY INDUCTOR, AND THE GROUND SCREEN CONSISTS OF THIRTY EQUALLY-SPACED 100 METER RADIAL COPPER WIRES.

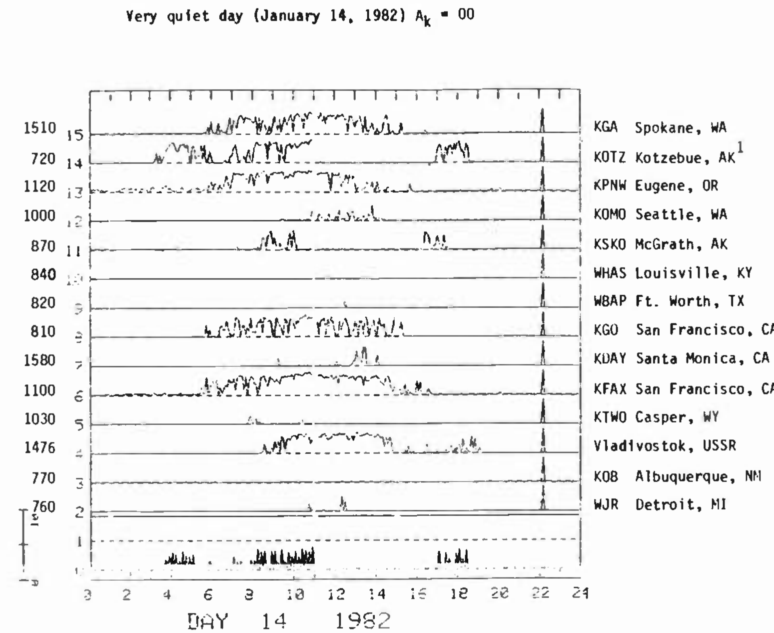

FIGURE 2. *DIURNAL* SIGNAL STRENGTH BEHAVIOR. PLOT OF RECEIVER AGC VOLTAGE FOR STANDARD BROADCAST STATIONS LISTED AT RIGHT RECEIVED AT FAIRBANKS DURING QUIET MIDWINTER PERIOD. SCALE AT LOWER LEFT IS RECEIVER AGC IN VOLTS. TIME ON ABSCISSA IS UNIVERSAL TIME (ALASKA STANDARD TIME = UT -10 HOURS). FREQUENCIES IN kHZ LISTED ON LEFT BORDER.

For a *disturbed* day the diurnal behavior changes considerably, as illustrated in Figure 3 when the only MF skywave signals received in Fairbanks were from Kotzebue and McGrath, Alaska. All signal propagation from the contiguous US disappeared.

The next four figures illustrate the *seasonal* behavior of the monitored MF skywave signals. Figure 5 is a fifty-one day median plot of all signals received for winter 1981 for high sunspot number and illustrates the consistent propagation during night of the signals from the western and northwestern regions of the contiguous US to Alaska. A rather dramatic contrast is shown in Figure 6 for summer 1982, when the only measurable signal strengths occurred for a few hours per night from Spokane, Washington; Anchorage, Alaska; and San Francisco, California.

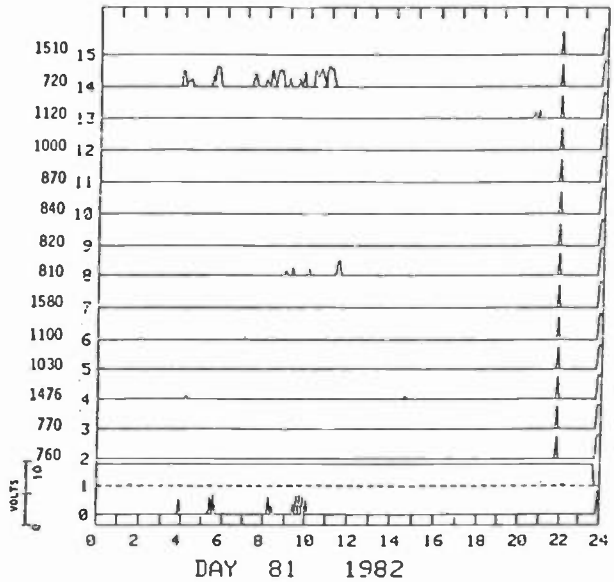

FIGURE 3. PLOT SIMILAR TO THAT IN FIGURE 2, EXCEPT FOR VERY MAGNETICALLY DISTURBED DAY DURING SPRING EQUINOX.

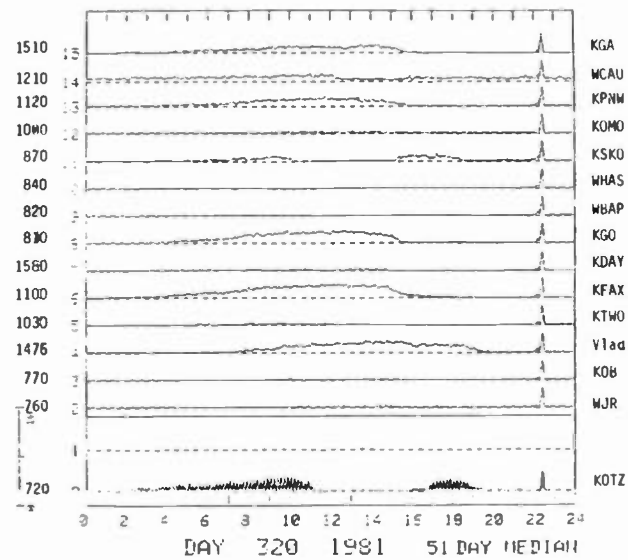

FIGURE 5. SEASONAL SIGNAL BEHAVIOR – MIDWINTER 1981-82.

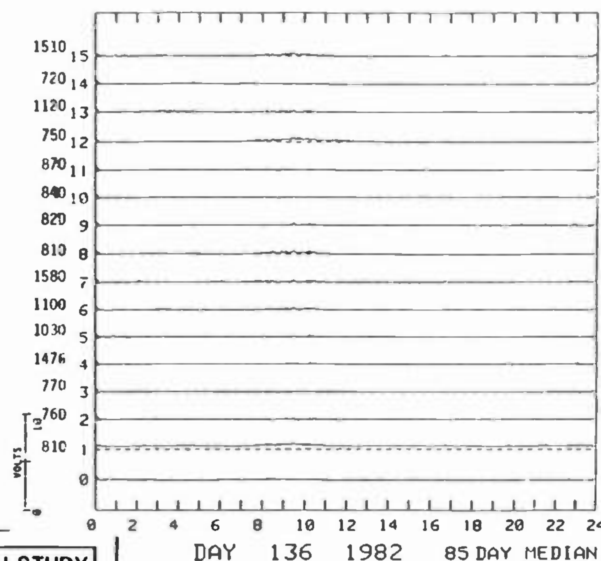

FIGURE 6.

SEASONAL SIGNAL BEHAVIOR SUMMER 1982.

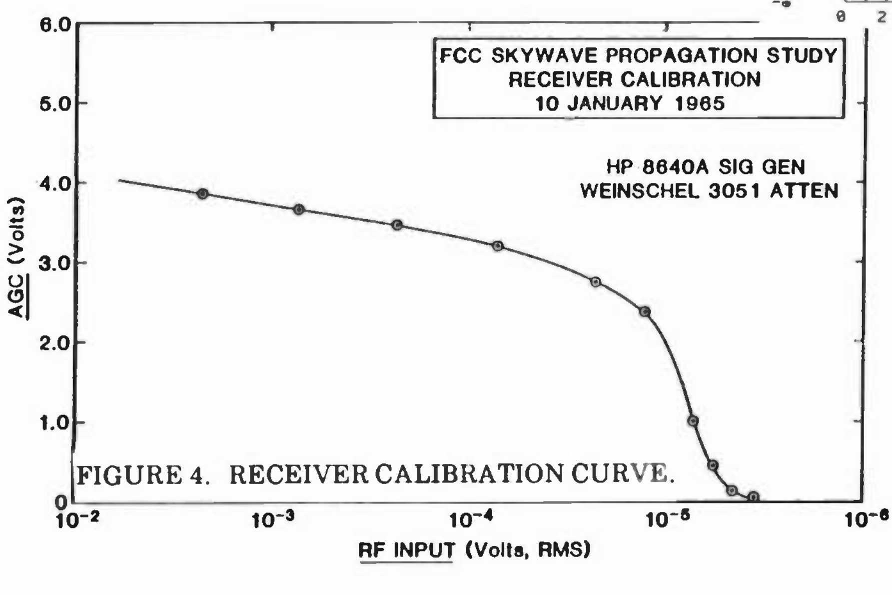

FIGURE 4. RECEIVER CALIBRATION CURVE.

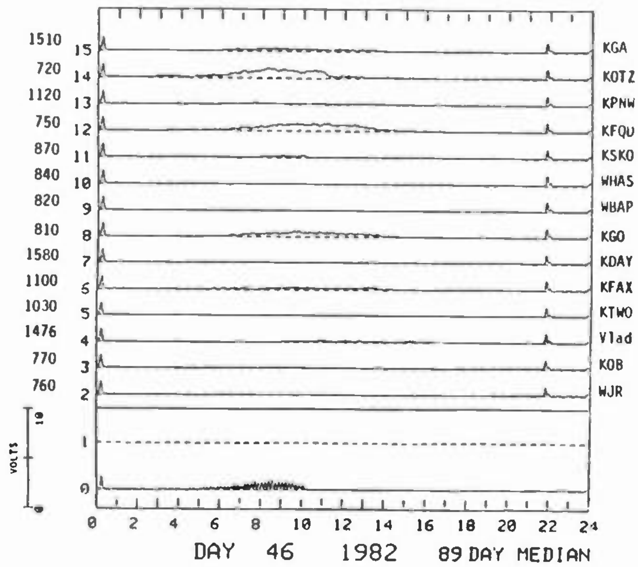

FIGURE 7. *SEASONAL* SIGNAL BEHAVIOR SPRING EQUINOX, 1982.

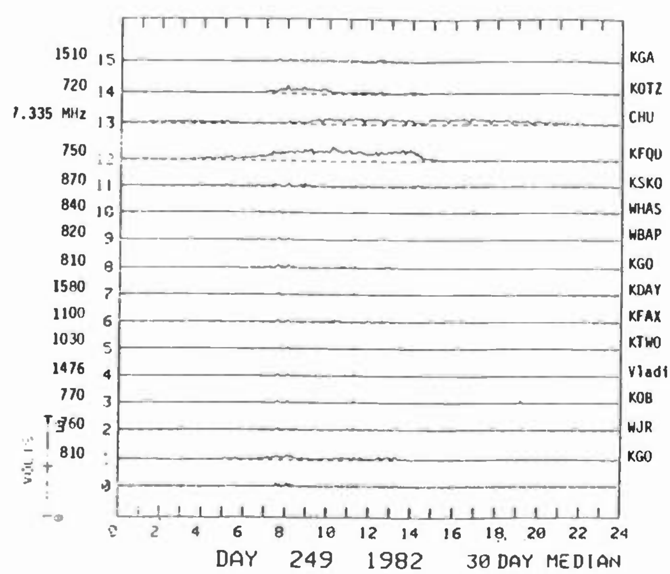

FIGURE 8. *SEASONAL* SIGNAL BEHAVIOR - FALL 1982.

The equinoctial period behavior is displayed in Figure 7 and 8 for spring 1982 and fall 1983 respectively, showing transitional signal strengths between summer and winter.

Sunspot Cycle Effects

Figure 9 shows a plot of solar cycle 21 with the interval of MF skywave monitoring indicated by a bar at the bottom. From midwinter 1981 to midwinter 1984 the relative sunspot number (RI) decreased from 147 to 18 and the 10.7 cm solar flux (SF) decreased from 208 to 76. This represents a decrease in RI by ~ a factor of 8 and in SF by ~ 3, which *should* have significant effects on MF skywave propagation.

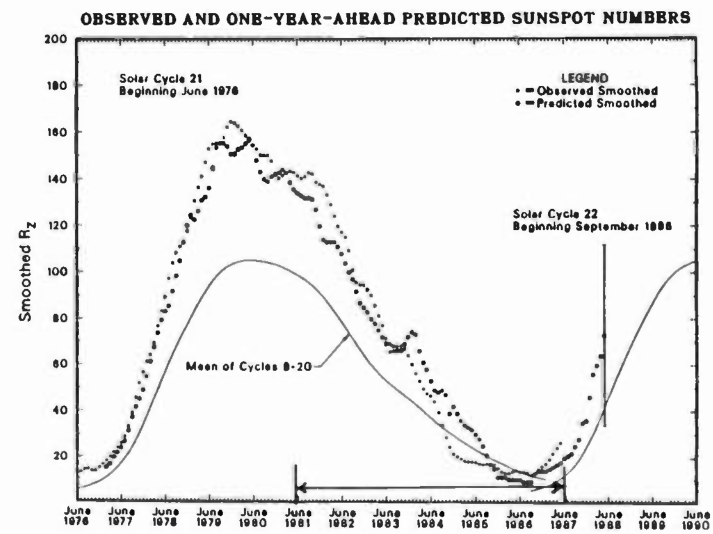

FIGURE 9. OBSERVED AND PREDICTED SMOOTHED SUNSPOT NUMBERS.

DISCUSSION

Figure 10 illustrates the uniqueness of the Fairbanks MF skywave monitoring station as a high latitude site, Hunsucker[9]. The figure depicts great-circle paths from most of the standard AM stations monitored at Fairbanks on an azimuthal equidistant map projection. The stipled sections depict the location of auroral ovals for midwinter near magnetic midnight (1200 UT or 0200 AST--Alaska Standard Time). The lightly stipled auroral oval section shows the location of the oval for disturbed magnetic conditions (planetary K-index, Kp=5.0), and the darker oval (further north) depicts the oval for quiet conditions (Kp=1.0). Only the portion of the oval affecting the paths is shown. The single letters denote standard broadcast band transmitters as:

N = Nome, Alaska	E = Edmonton, Alberta, Canada
M = McGrath, Alaska	C = Casper, Wyoming
A = Anchorage, Alaska	P = Philadelphia, Pennsylvania
S = Seattle, Washington	V = Vladivostok, USSR

As may be seen, during *very quiet* magnetic conditions (Kp=1.0) only the Philadelphia to Fairbanks path is directly affected--with the Chicago to Fairbanks path marginally affected. During disturbed conditions (Kp=5.0) all paths are aurorally disturbed (San Francisco the least). Most of the time 1.0 < Kp < 5.0, and most of the paths will be affected near 1200 UT.

EFFECTS OF A LARGE GEOMAGNETIC STORM

One of the largest geomagnetic storms in the last 40 years occurred on 8-9 February 1986 as a result of a protracted sequence of solar flares that occurred from 4-7 February 1986. Periods of Kp=9 were recorded from 1800 UT 8 February - 0300 UT 9 February, and the severity of this storm was equal to that of the storm of 13-14 July 1982. Effects of the February 1986 storm include propagation anomalies in signals from geosynchronous satellites, serious orbital shifts for certain navigation satellites, early reentries of debris from decaying orbits, HF radio communication link outages, impairment of telephone and microwave circuits and voltage surges on long power transmissions lines[10].

The purpose of this section is to document some effects of this storm on MF skywave signals propagated on selected high-latitude paths monitored at Fairbanks, Alaska and to relate signal variations to the auroral oval before, during and shortly after the storm. The maximum deviations recorded during this storm was H = -6110 nT, while the D-component change was 17.6°– which represents the larger values of these parameters measured at the College Observatory since operations started in 1949!

Figure 11 illustrate the storm-time behavior of a *long* "mid-latitude" path (San Francisco to Fairbanks). At the *top* of Figure 11 the MF skywave signal displays a diurnal variation characteristic for quiet time low sunspot number conditions, with ≈100% of the path lying outside the auroral oval. The *middle* plot in Figure 11 indicates a complete blackout of the signal near the storm maximum, with ≈40% path lying within the auroral oval. Since the region of auroral absorption extends several degrees of latitude equatorward of the oval[11], the blackout is most probably due to *auroral absorption*. The *bottom* plot shows that even on this "mid-latitude" path the signal has not recovered to its pre-storm level. Signal levels from Edmonton and San Francisco did not regain the pre-storm level until 14 February (5-6 days after the storm maximum).

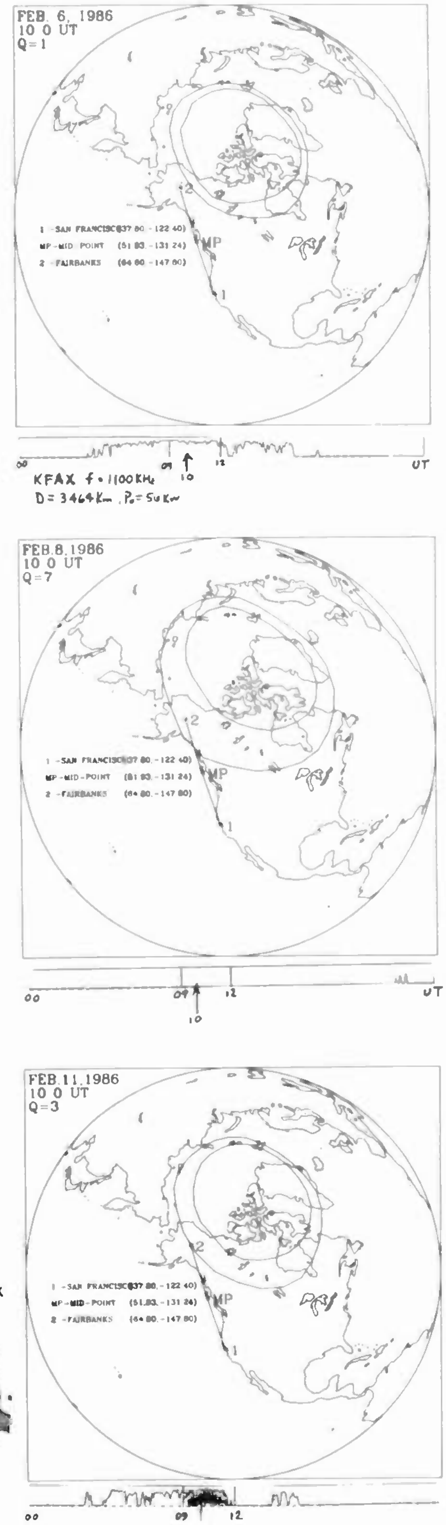

FIGURE 11. "STORM-TIME BEHAVIOR" OF SAN FRANCISCO - FAIRBANKS SIGNAL.

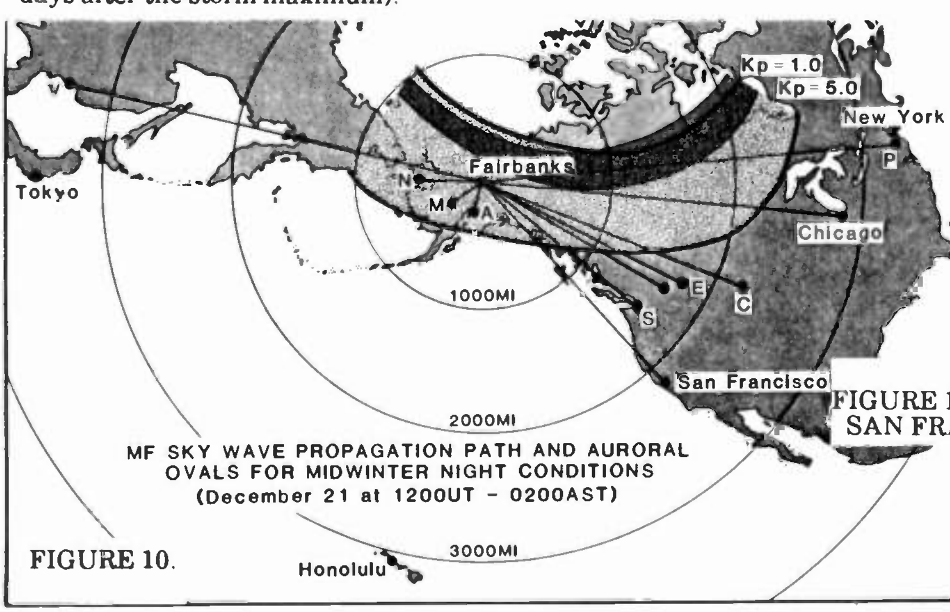

FIGURE 10.

Examples of Median MF Skywave Data

Figures 12 and 13 are typical examples of median MF Skywave data plots showing the *yearly* median values in μV/m and the ninth decile values (μV/m) for 1984, 1985, 1986 for the signal from KFAX, San Francisco received in Fairbanks, Alaska. These plots show the change in field strength as a function of the sunspot cycle. Figure 14 illustrates the *seasonal* behavior of the signal from KFAX for summer, fall and winter--especially showing the dramatic decrease in MF Skywave signals received in Fairbanks during the summer. Hundreds of station hours of field strength for MF signals from San Francisco, Anchorage and Edmonton received at Fairbanks are tabulated (along with selected plots) in our Final Report to the FCC.

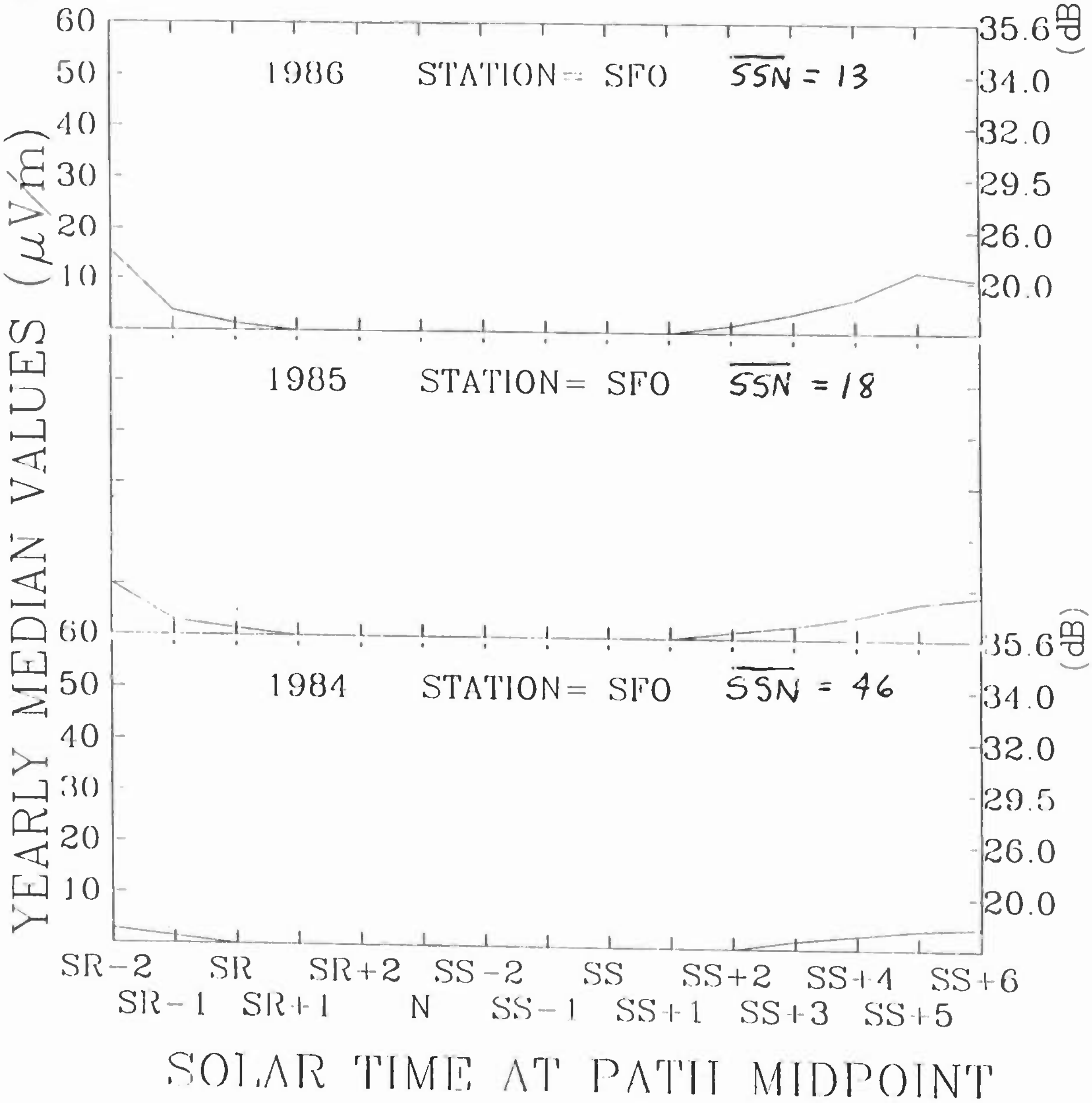

FIGURE 12. YEARLY MEDIAN FIELD-STRENGTHS (1984-86) OF SAN FRANCISCO SHOWING SUNSPOT CYCLE EFFECT.

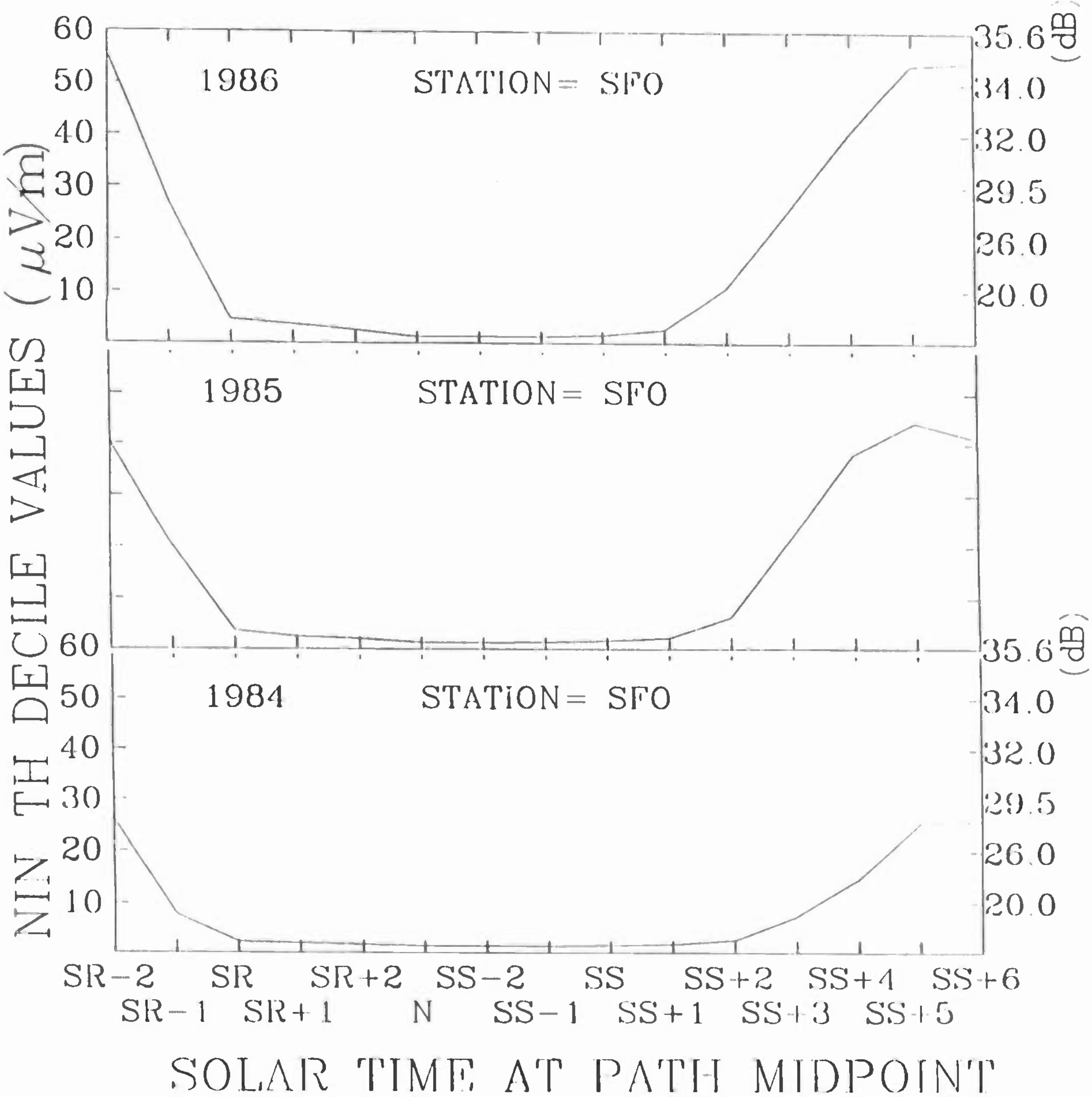

FIGURE 13. SAME AS FIGURE 12, EXCEPT 9TH DECILE VALUE ON LEFT.

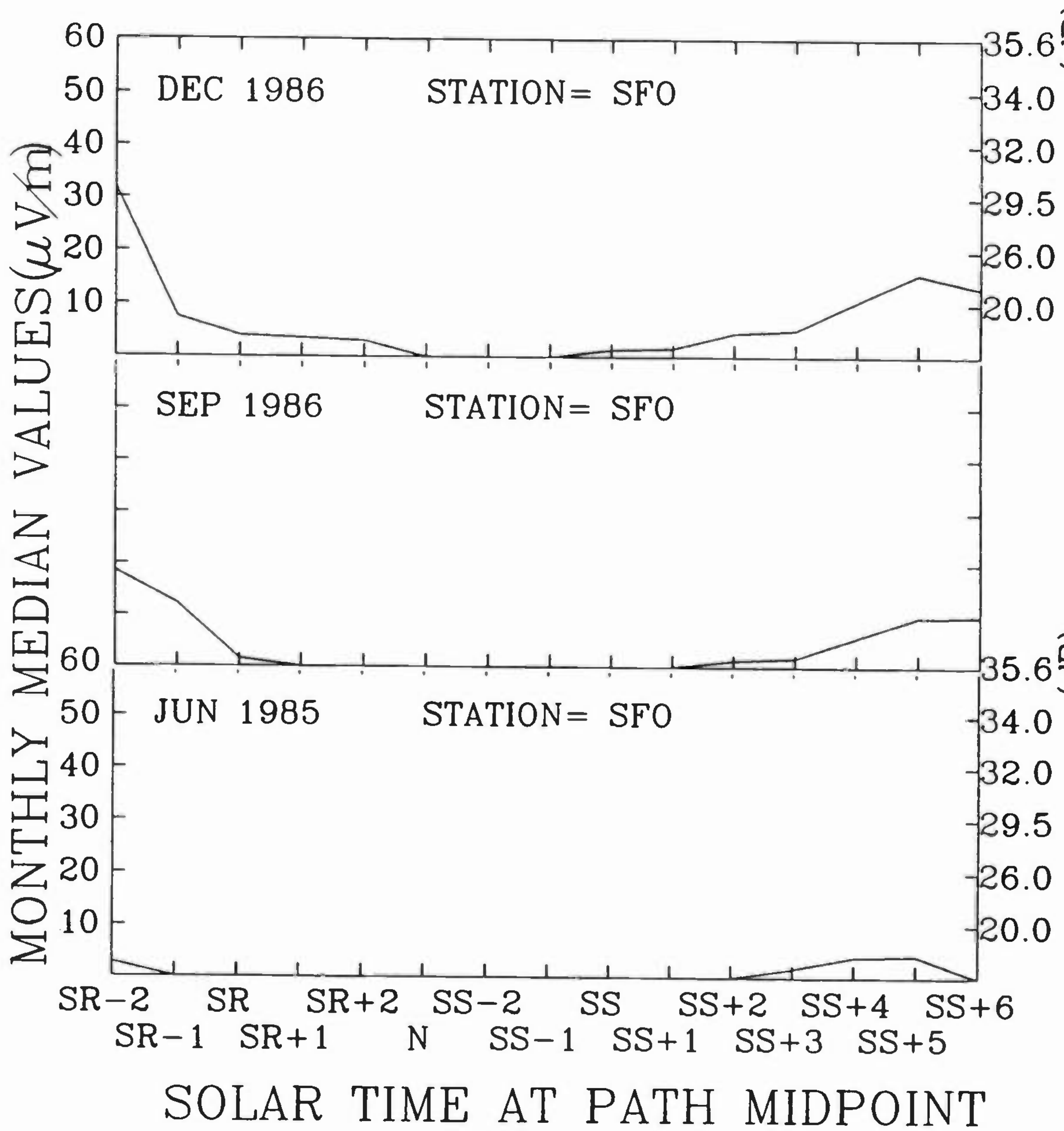

FIGURE 14. *SEASONAL* VARIATION OF SAN FRANCISCO MONTHLY MEDIAN.

CONCLUSIONS

Some tentative conclusions from this 5.5-year investigation of MF *Skywave* propagation at high latitudes are:

1. The electronically programmable receiver, noise calibrator, cassette tape data logger and top-loaded vertical antenna proved to be a very reliable system for continuous operation.

2. Regular aural identification of the standard AM broadcasting stations was essential.

3. At least 5.5 years of data acquisition is needed to characterize the *sunspot cycle* variation of field-strength.

4. The most consistent signal received at Fairbanks was from KFAX, San Francisco.

   Standard AM broadcasting signals (MF Skywave) propagating at high latitudes undergo some *profound variations*, such as:

   a. Very few hours per day of signal reception during summer months over most of the half sunspot cycle.

   b. Large change in field strength as a function of sunspot cycle--mainly attributed to D-region absorption.

   c. Sporadic-E ionization is important.

   d. Extreme variability of field-strength during geomagnetic disturbances.

   e. Very poor ground conductivity measured in the Fairbanks area (Hagn, 1983)[12].

Based on the results of this study (including the library of digitized data on tape, the data tabulations, and the median field-strength plots), we suggest that the FCC can now proceed to reexamine the standard broadcast band channel allocations in the northern tier of the contiguous US and in Alaska. More rational decisions on channel allocations on a non-skywave interference basis should be possible.

ACKNOWLEDGMENTS

We would like to express our thanks to Kay Lawson and Stan Schwafel for site maintenance and aid in data analysis and to Mr. A. G. Hiebert, President of Northern TV, who provided invaluable assistance. This work was sponsored by the Office of Science and Technology of the Federal Communications Commission on Contract FCC-0375 and by the Alaska Public Broadcasting Commission.

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AMERICA'S FIRST SOLAR POWERED FM RADIO STATION

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PREFACE

The saying goes, "Necessity is the mother of invention". In our case, it was more like desperation. KIHX-FM signed on September 1,1985 with it's primary signal emanating from a sixty foot telephone pole next to the studio. At a height above average terrain of minus 73 meters and in a mountainous region, the signal was heard in Prescott Valley, Arizona, population then about 6500. However, in nearby Prescott, population 25,000, no clear signal was receivable.

We began making plans before sign-on to elevate our primary transmitter site as soon as possible. The best available site was on nearby Glassford Hill, elevation 6160'.

Arizona Public Service, the local electric utility, told us it would cost over $100,000 to provide power to the proposed site on Glassford. Due to Glassford's location within the boundary of the State of Arizona Land Trust, state statute required the power lines to be brought underground or not at all, thus the large expense.

It was in October, 1985 that we contacted Gene Hitney of Photocomm, Inc. of Chino Valley, Arizona with the proposal to design and implement a solar power array sufficient to power a broadcast site. The state of Arizona gave the project its blessing and what resulted from months of work and research became known as America's first solar-powered commercial FM radio station. KIHX-FM is also the first known solar powered station built without government assistance in the world.(WBNO-AM in Bryan, Ohio is the first known solar-powered radio station. It was funded by a U. S. Department of Energy Grant and developed by the Massachusetts Institute of Technology in 1980). What follows is a story of innovation, determination and perseverance. It details the transformation of a money-losing proposition into a viable and thriving business.

UNDEVELOPED SITE FORCES ALTERNATIVE ENERGY OPTIONS

When KIHX-FM commenced operation, virtually no high elevation sites were available for development. We found that much of the higher terrain was controlled by the United States Forest Service, which had established an antenna farm on seven thousand foot Mingus Mountain. The problem for KIHX-FM was two-fold. First, the process of obtaining the necessary permits to broadcast from Mingus Mountain would take up to three years. Second, a Class A equivalent signal from Mingus would not encircle Prescott Valley within it's 3.16 mV/m contour. Clearly, an alternative site had to be sought.

We had toyed briefly with the idea of translating into Prescott, but that, too would have been a situation where the costs outstripped the benefits.

It was mid-September, 1985 when Terry Cohen, my wife and I heard that a local private radio company was seeking to develop Glassford Hill. We sought and were granted a lease for a one acre antenna site on Glassford in late October. This was certainly a record for having a state land lease processed. We walked the lease through every department and even presented it to the Commissioner for his signature. Thanks to Intermountain Communications, who had endured almost three years of proposals and counter-proposals to get the site approved, we were able to sail through with scarcely a hitch.

Now came the hard part. How were we going to bring power to this remote site? The nearest utility feed point was well over two miles away. The State required them to put the lines underground. Alternative energy was the only solution, but which kind?

Wind power was our first considered option. I had worked for the nation's only wind-powered station, KFMU-FM IN Steamboat Springs, Colorado, so my resistance to energy alternatives was perhaps far more reduced than your typical decision-maker. However, authorities at the University of Arizona told us that the wind was not sustainable for such a high-consumption project.

The next option we considered for Glassford Hill was a generator. We contacted an FM station in Barstow, California run totally on generators.

The owner told us that while the generator system was serving him well, the cost and frequency of of maintainence made that system cost prohibitive. Our site would be at the end of a narrow, winding, rocky path and frequent trips would bring even the staunchest of four wheel drive enthusiasts to their knees begging for mercy.

Something the professors at the University of Arizona said to us came to mind. Prescott, Arizona enjoys 310 partially to fully sunny days every year. We decided to investigate solar power.

SEARCHING OUT SOLAR POWER

Our next door neighbor on Glassford Hill, Intermountain Communications had established a solar powered site for it's two way and civil communications operations. Much of the equipment was of such low consumption that only twelve solar panels were needed, along with a battery bank to keep the site at full power. It was at the urging of Intermountain that we contacted Photocomm, Inc., a national disbributor of ARCO Solar Products, with the idea of designing and building a prototype broadcast transmitter site powered by the sun.

We convinced Photocomm that the broadcasting business was an untapped marketplace for their products and services. As a result of negotiations spanning several months, Photocomm agreed to build the solar power plant at no initial cost to us. Clearly, this fit into KIHX-FM's budget very well. The plan was to lease the power from Photocomm, similar to a utility, with the option of someday owning the power plant outright. Photocomm would establish an important outlet for research and development and the people of Prescott would soon receive a clear signal from KIHX-FM.

The agreement called for KIHX-FM to pay Photocomm $1000 per month initially, escalating to $1700 per month after the first year and increases tied to the Consumer Price Index every year up to ten years from commencement. Even though the cost was substantially greater than from a utility, the start-up cost (virtually nothing) was the only option KIHX-FM could afford.

MATCHING BROADCAST WITH SOLAR

KIHX-FM Chief Engineer Chuck Smith, along with Photocomm's design team set about putting the project together in February, 1986. Smith felt that the only way to make the idea work was to use a combination of low power, solid state transmitter, along with a multi-bay antenna for power gain. Despite the tendency of multi-bay antennas to add multipath distortion in mountainous situations, the only other option would have been to build a bigger solar plant and that would have been cost prohibitive.

Weighing the comparative advantages and disadvantages, Smith and I chose the CSI 500 watt solid state amplifier for it's compact design and low cost. Smith chose the Broadcast Electronics FX-30 exciter as superior in its class. Combined with the then newly released Gentner VRC-1000 remote control, KIHX-FM had a low consumption transmitter site package ready for implementation.

The last piece of the broadcast puzzle was antenna. Smith and I chose the Shively 5-bay circularly polarized FM antenna. With its gain factor of 2.55, all we had to do was drive the CSI amplifier to 411 watts to achieve our licensed power output of 1.05 kilowatts. With our antenna mounted at the top of a sixty-foot tower, KIHX-FM's signal now stood at 471 feet above average terrain. A clear signal into Prescott was certain.

CONSTRUCTION BEGINS

The construction crews at Photocomm began bringing supplies to the site in late February, 1986. All materials had to be transported to the site by four-wheel-drive vehicles. Concrete, rock, water, mixers, girders, panels, culverts, every inch of material by four wheel drive. The solar array was positioned at the southern end of the site with some 96 solar panels affixed to a movable set of light aluminum frames. This allowed the angle of the panels to be adjusted to correspond to the height of the sun as it varied through the seasons. The array measures some 60 feet in length and is adjustable between six and eight feet in height. The panels from ARCO are rated at 47 watts output each. The columns of panels are wired in parallel and then interconnected before terminating in the nearby building. Half of the building houses the storage batteries and back up generator system, the other half houses the broadcast equipment.

The broadcast equipment is cooled by a forty foot long culvert pipe buried in the ground at a downward angle from the building. Earth temperature air is fed through a filter into the broadcast side of the building and vented by a duct off the top of the transmitter. This ducted air is then diffused by an outdoor baffle to prevent standing warm air. In summer, the air fed into the building is an Earth-cooled 57 degrees. In winter, that same air is now a relatively mild 57 degrees. An electric fan at the entry point to the broadcast side of the building helps to draw the air inside.

The other side of the 10' by 20' modular steel building houses the so-called battery bank capable of some 3000 amphere hours of storage. The bank consists of a combination of two volt and six volt cells. The lead-acid, deep-cycle storage batteries are wired for 24 volts DC.

The broadcast equipment draws some 1300 watts or 50 amps through a DC to AC inverter system. Photocomm uses the Vanner 3000 watt inverter unit that converts the 24 volts DC to 120 volts

AC.

While the system did experience minor fluctuations in current, the recent addition of the "Boss" (achronym for 'Balance of Systems Specialists'), a 24 volts DC, 200 amphere voltage regulator has kept those fluctuations to a minimum. The "Boss" protects the batteries from overcharge at the hands of the solar array. If the batteries storage voltage drops below 23.5 volts, a Kohler 7.5kw industrial generator automatically switches on to recharge the battery bank up to 24 volts DC before shutting off. A small solar panel mounted on the roof of the building provides a maintainence charge on the generator's start battery.

The solar panels themselves are capable, under peak conditions, of producing some 4.5 kw of output. The panels are rated based on 1000 watts per square meter of solar insulation. That level is generally reached under clear conditions at or near mid-day.

"WHAT HAPPENS WHEN..."

The most commonly asked question of us is, "What happens to the radio station after the sun goes down?" With a propane storage tank on site and full, the system can operate for up to 21 days without direct sunlight. The longest period we have recorded without direct sunlight so far took place from October 21 through October 30, 1987. Prior to that period, we had not experienced more than four consecutive cloudy to partly cloudy days in a row since the project began. The system performed flawlessly and after the weather cleared up, we had discovered excess capacity in the propane storage tank.

It should be noted that in times of maintainence or parts replacement where the power plant had to be disabled for a short time, a Yamaha portable 5000 watt generator was placed on site with all equipment plugged directly into it. Both generator and equipment performed at par with normal solar operation.

As we enter 1988, the Gentner VRC-1000 is being equipped to monitor line voltage and signal alarm when and if the voltage drops below 23.5 volts. An RPU system has just been installed allowing total remote monitoring of the site by Chief Engineer Chuck Smith.

KIHX-FM stands prepared to see the project through to its contractural conclusion in 1996. Should the system continue to perform in the years ahead as it has in the past, the station can expect to remain solar-powered into the 21st century.

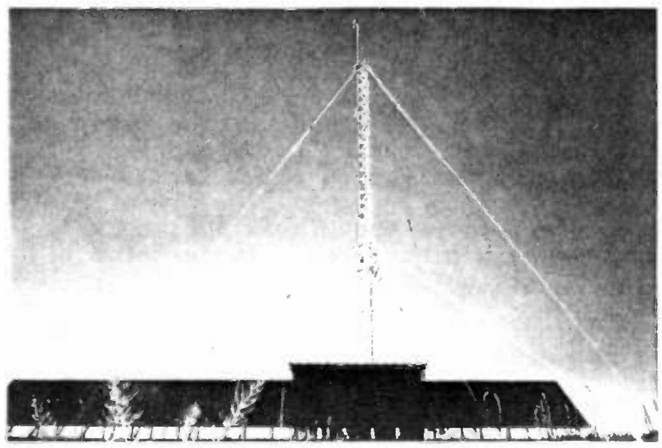

FIGURE 1

KIHX-FM solar array sits in the foreground with building just behind it and the broadcast tower towards the rear of the site

APPLICATION AND PERFORMANCE OF ROTARY PHASE CONVERTERS AS AN ALTERNATIVE TO UTILITY SUPPLIED THREE-PHASE POWER

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SUMMARY

A very common problem facing broadcast station owners is obtaining three-phase (3Ø) power service at prospective transmitter sites. Utility companies often charge exorbitant fees to extend service to remote areas. This often forces owners to select sites purely on the basis of affordable three-phase availability while compromising other desirable features of site selection.

The advent of the rotary phase converter in the early 1960's has contributed significantly to the broadcasting industry by providing the capability to produce three-phase power on site from any single-phase (1Ø) source. A rotary phase converter is an induction machine which operates on a single-phase supply and produces a true three-phase output. It is capable of supplying the full rated input requirement of any three-phase induction, resistance or rectifier load.

The cost savings to be realized by use of a phase converter can be breathtaking compared to the alternative utility costs. Documented savings of $100,000 or more are commonplace. However, the use of phase converters remains something of a mystery to many who could benefit from them most. Chief among the reasons for this are widespread anecdotal accounts of field problems and other misunderstandings of their application. Nonetheless, well over 300 radio and TV stations in the U. S. are successfully operating on phase converters with a service record exceeding 25 years.

The purpose of this paper is to explain the construction and performance of phase converters and characterize their application on broadcast transmitters. It is anticipated that a broader knowledge of converter capability and performance among owners and engineers would enable them to approach transmitter siting with a broader range of options.

TRANSMITTER SITE SELECTION FACTORS

A natural result of growth in the broadcasting industry is that the number of ideal transmitter sites is reduced while their costs continue to increase. Finding a good site where 3Ø power is available can pose a real problem in some areas.

Transmitter site selection often becomes an economic compromise of many factors, some of which are beyond the scope of this paper. However, judging from hundreds of interviews and conversations with station owners and engineers, a consensus view is that most site location issues fall into the following list:

* Availability of three-phase power
* Land lease or purchase cost
* Site Accessibility
* Potential Interference

Availability of Three-Phase

Virtually all broadcast transmitters rated 5 KW and larger require 3Ø power input. Equipment designers prefer 3Ø because its rectified output has much less ripple than 1Ø and requires less filtering to produce a clean DC output. However, in the real world, three-phase just does not exist everywhere and utility companies may not be willing to supply it within a reasonable cost or time frame.

Land Cost

Antenna farms or other developed sites where power is already available can be very expensive. By contrast there may be attractive undeveloped areas or inexpensive BLM leases which would be ideal sites if 3Ø were available.

Interference

This becomes a critical issue whenever locating near existing stations. The availability of 3Ø power must be weighed against the added cost of circulators, traps, expensive grounding systems, special antennas and other costs incurred to eliminate interference.

Site Accessibility

Even low cost sites which have 3Ø and are interference free may be unreliable locations if land owners are reluctant or unwilling to grant unlimited passage on private lands and roads. The availability or cost of access rights can drive up the price of an otherwise attractive site.

Final site selection may require a compromise on one or more of the above points. But it is unlikely that a site **without three-phase** could be seriously considered even if that choice were highly ranked in every other category. There are three alternatives which the owner can consider where three-phase power is not available. They are: 1) Request a utility line extension, 2) Install on-site power generation equipment, 3) Use a phase converter.

UTILITY POWER EXTENSION POLICIES

Every utility has its own policy on new service requests. However it is useful to understand the general issues which affect the utility's decision to extend power or the alternatives which they may offer.

The factor weighed most heavily is the proximity of the proposed site to the nearest sub-station or 3Ø line with adequate load capacity. A 3Ø line near the desired site is no guarantee the station can have a service drop if the line is loaded to its limit by other users.

In such cases, the utility company may choose to 1) increase capacity of the sub-station or distribution transformer, 2) install a new 3Ø line, 3) upgrade an existing 1Ø to 3Ø, 4) deny the request for 3Ø, or 5) offer a 1Ø service.

To convert an existing 1Ø line to 3Ø, the power company must string at least one additional cable and replace the 1Ø transformer with a 3Ø unit. In some cases, the utility may cut corners and simply add one more 1Ø transformer and thus supply open-delta three-phase. Open-delta is a very common practice in rural areas because it saves money on transformers. However it has very poor voltage stability and often undergoes wide voltage swings.

Three-phase line extensions are ideal but are very costly because they entail a complete new installation of poles, lines, insulators, supports and other hardware. The costs are further affected by terrain and accessibility to the new site.

As utilities come under increasing pressure from public regulatory agencies to justify their capital investments (which includes new distribution lines) when seeking rate increases, they have become very particular about where they extend new services. If the estimated investment of extending a line is not paid back fast enough through energy revenues, the utility will charge the customer for the new service. There are no absolute rules governing the calculation of these charges, however, it is widely accepted that line extension costs range from $30,000 to $90,000 per mile with $50,000 being a common average.

Engine Generators

The diesel or engine driven generator is a commonly considered alternative to a converter. However, generators are expensive and are usually not justifiable as the primary 3Ø source when compared to a phase converter (assuming an adequate single-phase service is already available). In addition to high initial cost, the logistics of fuel supply can pose a serious problem in some geographical areas. Further, the maintenance costs of engine generators tends to be quite high and downtime is longer than phase converters when outages occur.

PHASE CONVERTERS - THE FUNDAMENTALS

Types of Converters

A phase converter is simply a device which permits a 3Ø machine to be operated from a 1Ø source. It does so by producing a **manufactured phase** which becomes the third wire connection to the load. There are two types of converters, **rotary** and **static**. Static converters are less expensive than rotaries but are much more limited in capability. Static converters are useful on light duty motor loads but are not suitable for rectifier loads as found in transmitter applications. Therefore only rotary converters will be discussed here.

ROTARY PHASE CONVERTER SIMPLIFIED EQUIVALENT CIRCUIT

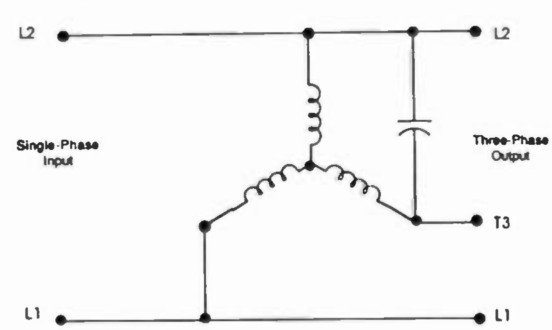

FIGURE 1

Converter Construction and Operation

A common misconception is that a rotary phase converter is similar to a mechanically coupled motor-generator set. In reality, the converter is a single armature device constructed much like a three-phase induction motor. It consists of a stator frame with a symmetrical three-phase winding and a specially modified squirrel cage rotor. A large capacitor bank is placed across a set of windings between one of the input lines and the manufactured phase. A simplified equivalent circuit is shown in figure 1.

When the converter is energized, single-phase power is applied to one of the winding groups. This produces an internal magnetic field proportional to the applied single-phase line. The capacitor bank provides a phase shifted voltage to another coil group which creates a starting torque on the rotor. As the rotor spins, it picks up a replica of the utility supply through induction. As the rotor passes each stator coil group (each separated by 120 mechanical degrees) the single phase field is replicated in the other two coil groups. The result is a three-phase sinusoidal output with each phase shifted by 120 degrees.

In this context, it becomes clear that a rotary phase converter is actually a rotating transformer where the rotor acts as a secondary winding on bearings.

Output Characteristics

Two of the three lines in a converter system come right from the utility (figure 2). Therefore, the important output characteristic is the behavior of the manufactured phase in relationship to the two utility lines.

The energy which flows into the manufactured phase passes across the internal air gap between rotor and stator. The greater the load, the more energy has to cross the air gap and the greater the effect on output voltage. The output diagram in figure 3 shows the behavior of the output voltages in relation to load. Note that at under no-load conditions, the manufactured phase voltage (L1-T3) is substantially higher than the incoming line voltage. As load increases, the voltage drops so that under full load conditions, the three voltages are quite closely balanced. The significance of this load dependent voltage will be discussed in further detail.

Ratings

Standard single unit converter output ratings are available up to 100 KW. But there is no theoretical limit to the total load size a converter can service. They may be paralleled indefinitely for any load. The only restriction is the maximum 1Ø load allowed by the utility. In practice the only time converters have to be paralleled is on large TV transmitters.

ROTARY PHASE CONVERTER TYPICAL CONNECTION DIAGRAM

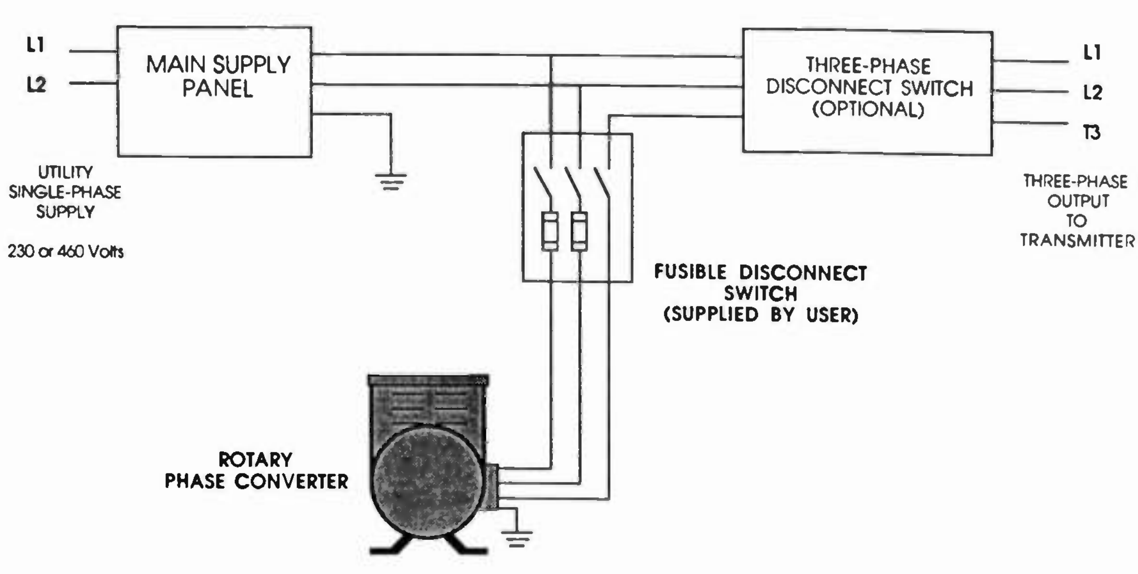

FIGURE 2

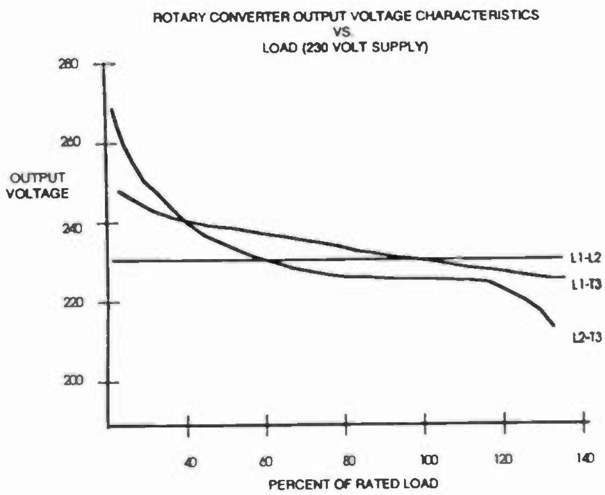

FIGURE 3

Except for the largest sizes, converters are usually dual voltage rated at 230/460 volts. Output is three wire closed delta.

Physical Size

Phase converters are dense and compact. The footprint of a converter for a typical 20 KW FM transmitter is about 24" X 24". It stands 30" high and weighs 800 lbs.

Design Operating Duty

Rotary converters designed for broadcast service are capable of continuous operation, 24 hours per day, every day. They need not be shutdown except for periodic scheduled inspection and maintenance. They can operate unloaded or at full load without effect. However losses are higher under no-load conditions.

Equipment Standards

Unfortunately because of the highly specialized nature of phase converter broadcast applications, there are no published industry standards governing their performance or ratings. The two best known approval agencies in North America, CSA and UL both have approval procedures for converters. However, these approvals are little more than equipment material and safety audits. They verify none of the manufacturers specifications or performance claims. In the absence of strict industry standards, the most reliable yardstick of relative performance is a comparison of the physical frame size and weight of the converter.

The severity of transmitter applications requires a physically larger frame and rotor than would be necessary to operate an induction motor load. If a motor is difficult to start, capacitors may be added to boost starting torque. Boosters are not effective on transmitters and are never a substitute for a larger converter frame size. It is recommended that users stick to converters which have been specifically designed for use on transmitters and have a demonstrable service record.

BROADCAST APPLICATIONS OF PHASE CONVERTERS

Operating Benefits

The most obvious benefit of phase conversion is the ability to operate 3Ø equipment while avoiding utility installation charges. But there are several other economic and technical advantages to converters which can be beneficial to owners and engineers.

1. Immediate Power Availability
A very important factor in the decision of whether to go with utility or a converter is the time required to extend the new lines. Depending on the distance or the utility work load, new services may take from weeks to months before the user can energize. The converter is a simple solution to this dilemma. It can be installed very quickly to allow start-up with minimal delay. Even if utility 3Ø is brought in later, the relatively small converter investment may save much greater revenues which would be lost in delays.

2. Elimination of Utility Demand Charges
The rate structure of 3Ø electric power often includes a component known as a demand charge. This charge seldom applies to 1Ø services. Over the years, many station owners have discovered that their power supplier serviced them with a 1Ø which did not include demand charges. The use of the converter was not only a satisfactory alternative to utility 3Ø but the rate structure was more favorable when purchasing as a 1Ø customer.

3. Reduction of Utility Line Transients
Almost anyone who has operated a transmitter is familiar with the phenomenon of utility line noise. Such transients result from system switching disturbances and load changes and can damage equipment or take a station off the air. Since a converter is capable of storing a large amount of energy in its rotating magnetic field, it can ride through momentary voltage sags by generating energy back into the system during the voltage drop. This type of event accounts for nearly 80% of all line disturbances and the effects are greatly reduced or eliminated by the converter. The converter also buffers voltage spikes.

4. Stabilization of Open-Delta Service
Open-delta service as previously discussed is a common form of three-phase which has very poor voltage regulation on one leg. By contrast transmitters require stable power supply voltage. Voltage swings are a very common nuisance to stations unlucky enough to have open-delta service. Phase converters are an effective method of closing the delta and stabilizing the wild leg and eliminating the unplanned downtime cause by open-delta service.

Converter Selection Criteria

When properly selected and installed, converters are capable of output and performance which is nearly indistinguishable from utility three-phase. The typical data required to insure a successful application includes the following:

* Type of transmitter, AM, FM, TV
* Manufacturer
* Power consumption in KW or Amps
* Day-Night output, if different

Once this data is known, selecting the proper converter is a straightforward procedure. The key to successful sizing is to match the converter output as closely as possible to the load. This requires knowing the true **input power consumption at the actual operating output of the transmitter**. On AM transmitters this is usually expressed at 100% modulation.

This sizing technique produces the best possible voltage balance. A properly sized rotary converter will provide operating load voltage balance ranging within 2-5% of the utility line.

Load matching is important to sizing because as discussed earlier, the output characteristics of converters vary with load. This phenomenon is only significant when attempting to operate a transmitter which draws considerably more or less than the rated output of the converter. The resulting voltage imbalance can cause an increase in AM noise level. This condition is easily avoided by proper selection. Field remedies are also available.

Load matching and voltage balance issues also affect AM stations which operate at reduced nighttime power. This situation is easily handled by splitting the load between two parallel converters (figure 4). One converter is sized for the night load only. During the day they run together. At night one converter is switched off and the other unit carries the entire reduced load.

ROTARY CONVERTER CONNECTION FOR PARALLEL OPERATION

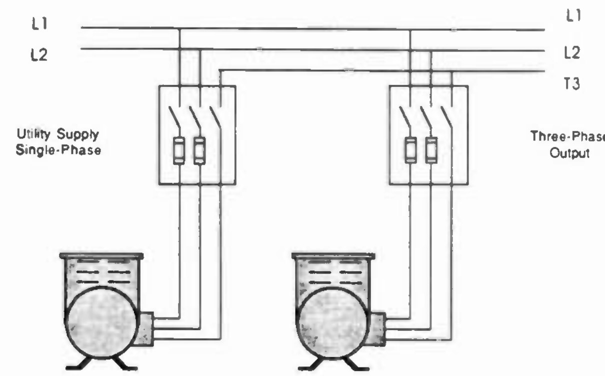

FIGURE 4

SITE ENGINEERING CONSIDERATIONS

Phase converters have clearly demonstrated the ability to handle high power installations on all types of transmitters. Nonetheless, the potential for misapplication is always present unless all system operating requirements are reviewed. When these issues are addressed, engineers and owners can approach the use of phase converters with total confidence.

Size of Utility Service

Undersized utility service transformers are a very common field problem. Engineers should work with the utility to insure the incoming service is adequate to supply the entire site load without excessive voltage drop. An FM transmitter with a rating of 20 KW will actually draw more like 35 KW at full operating load. If the site also has an air conditioning load, tower lights, etc. totaling an additional 10 KW, then 45KW becomes the total 1Ø demand which the utility must be able to supply. However the converter is sized only for the amount of connected 3Ø load.

Planned Operating Load Levels

A phase converter should always be sized for the maximum power which the transmitter consumes at normal operating levels. If a different output power is anticipated at a future date, consultation with the converter manufacturer is recommended to assure that converter design takes future needs into account.

Converter Control Systems

The most common and least expensive control scheme uses a fusible disconnect switch for isolation and short circuit protection. The converter is manually switched on and runs continuously until manually shutdown. However, there are operating conditions which call for a greater degree of control. For this reason, engineers may want to consider specifying automatic controls to start and stop the unit remotely.

Automatic controls use a magnetic contactor to start and stop the converter from a remote control station. A timing relay locks out the load until the converter reaches full speed and is producing three-phase (about three seconds). Automatic controls should be considered for sites which are unattended at any time or stations which sign-off at night. The convenience of this feature is widely appreciated by existing users.

Another key advantage of automatic controls is improved system reliability. Utility service in remote areas is notorious for frequent outages. Stations which use emergency back-up generators must have a way of isolating the converter when the transfer switch changes over to the emergency source. The automatic control provides this function as well as the ability to restart the converter in advance of switching back to the primary supply. It also prevents the converter from attempting to restart into a load when power returns after an outage.

ANALYSIS OF CONVERTER FEASIBILITY

The question of whether to consider a phase converter as an alternative to utility 3Ø reduces to an evaluation of economics and relative risk factors.

A complete economic analysis of the converter should include 1) Installed equipment cost, 2) Utility charges to bring in the three-phase line, 3) The difference in rate structure between single and three-phase service, 4) The operating cost of maintenance and losses, and 5) An analysis of the reliability record of converters.

Converter Costs

Approximate converter costs through 100 KW are shown on the graph in figure 5. Large rating units can be priced as parallel combinations Estimating prices of generators are shown for comparison. (Generator costs do not include fuel storage tanks).

A review of utility rate structures is strongly encouraged. The analysis of rates will reveal if they favor a customer metered at 1Ø or 3Ø.

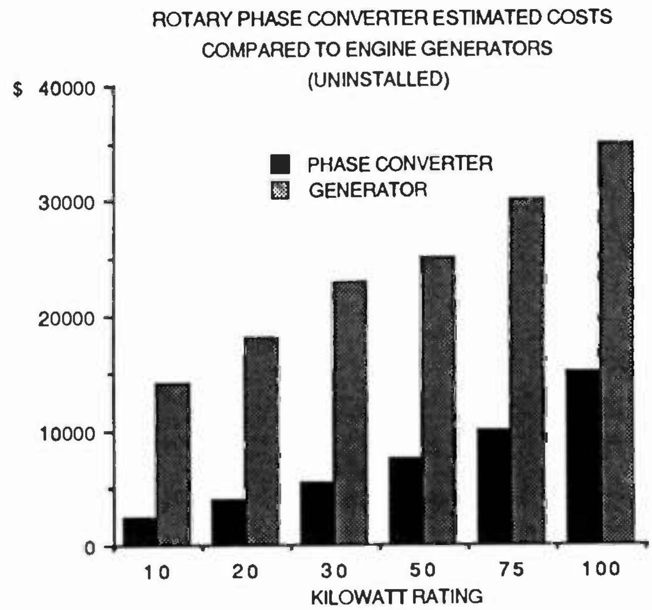

FIGURE 5

Efficiency

A common engineering concern is phase converter efficiency. The response to this question lies in understanding the difference between the converter efficiency taken alone and the overall **system efficiency**. The phase converter only sees one third of the system energy, so its efficiency taken out of context is not significant. Two of the three load connections come right from the utility supply **without passing through the converter.** Converters like most rotating machines are nominally 90% efficient at the upper end of their load range. Thus the full load **system** losses are approximately equal to 10% times 33% or 3.3% of the rated load.

Operating Costs

No load operation is the most inefficient running mode of a converter. As discussed above, the actual losses incurred by operating a converter system are quite small at full load. In fact, the large power factor correction capacitors in the converter will commonly improve the overall system efficiency enough to compensate for converter losses. Even without regard to these possible savings, the estimated cost of losses for a 25 KW converter running 24 hours per day would be less than $50 per month based on energy costs of eight cents per kilowatt-hour. Maintenance costs alone on a generator of this size would average 1.5 cents per kilowatt-hour or $270 per month.

INSTALLATION AND MAINTENANCE

Converter installation is neither costly nor complicated. Careful attention should be given to the manufacturer's instructions. Most start-up problems are traceable to installer deviations from recommended practices.

The converter is usually installed near the main power service entrance. An inexpensive fusible switch isolates the converter and provides its primary short circuit protection. A converter does have to be bolted down since it does not have to be braced against starting loads. A typical converter installation can be made in four hours or less.

Rotary converters require very little ongoing maintenance. The most effective maintenance program starts with strict adherence to recommended installation procedures. Once in service, the units should be inspected periodically. The ventilation slots must remain open and bearings lightly lubricated at intervals of 12-18 months. No other formal maintenance is required or recommended.

RELIABILITY RECORD

Many station engineers have observed the converter has far fewer outage occurrences than the utility supply. Of course this is not unexpected in many areas where weather conditions play havoc with rural distribution systems.

One make of converter has been used successfully in service on transmitters for more than 25 years. There are presently more that 300 stations known to be operating on these rotary converters. Many have been operating around the clock for five years or more without being shutdown.

The field failure rate averages well under one percent per year. This includes all types of failures regardless of cause or severity. Direct lightning strikes remain the most common cause of catastrophic converter failure. The addition of lightning arrestors as a standard phase converter accessory on transmitter applications has greatly reduced but not totally eliminated this failure mode.

By nature, phase converters are not service prone. They contain no contacts or switches. Apart from the rotor and bearings all other components are static devices. The most common natural failure modes of rotary converters are wiring and connector related. Far down the list in frequency of occurrence are shorted capacitors and bad bearings. The connector and wiring problems usually stem from abrasion of taped joints or connections which are not sufficiently tightened during installation.

Film type capacitors, if defective have a tendency to fail open. Capacitors rarely fail to a short circuit condition. Usually the only sign of defective capacitors is poor starting performance or sagging voltage on the manufactured phase. Both problems are extremely rare and easily corrected.

Bearing are the most commonly expressed concern of converter owners. In service, however, bearing failures are uncommon. The principle reason for this is that the bearings carry only the load of the spinning rotor. There is no external shaft extension on a phase converter and consequently no outside mechanical load on the bearings. Bearing life of 10 - 15 years is not uncommon with five year life a minimum expectation for installations operated continuously without shutdown.

A note of caution is in order here. Most rotary converters on the market have been designed for motor loads and are not well suited for transmitter service. Even subtle design differences in rotor construction and accessories can make a significant difference in transmitter performance. Owners and engineers will naturally want to examine and verify the service record of a manufacturer's experience on broadcast transmitter applications.

CONCLUSION

Specialized designs of rotary phase converters have been field tested for 25 years on virtually all makes of broadcast transmitters. The experience with these machines has clearly demonstrated them to be suitable replacements for utility supplied three-phase power without sacrifice of transmitter performance or reliability.

In view of their accumulated performance record and significant economic advantage, phase converters must be considered an important alternative power source which owners and engineers should not overlook when selecting transmitter sites.

SURGE PROTECTION AND GROUNDING METHODS FOR AM BROADCAST TRANSMITTER SITES

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INTRODUCTION:

AM broadcast transmitters are potentially more vulnerable to damage by surges than most other kinds of industrial electronics equipment. In addition to surges induced through AC mains connections, they are susceptible to damage from lightning strikes coupled from the antenna. With the recent proliferation of solid state transmitter designs, and the virtual elimination of ancillary tube equipment at a transmitter facility, it is becoming increasingly important for the broadcast engineer to understand methods of surge protection and how to integrate these with proper grounding techniques.

LIGHTNING SURGES:

A lightning strike will take the path of least resistance on its way to ground, and a broadcast tower becomes an excellent pathway, by nature of the fact that it is an excellent conductor and is typically the tallest object in the vicinity. Some stations have been successful in avoiding lightning strikes through the use of dissipation arrays on their towers. While an effective technique, this can be costly, and is out of the price range of many stations. Rather than trying to modify natural phenomena, a practical approach can often be to provide an attractive and direct path to ground for lightning currents, and take steps to make paths through sensitive equipment as unattractive as possible. To do this requires some understanding of how lightning-induced currents will behave in a conductor.

A typical strike can have a maximum current of 20,000 Amperes. This current will have a rise time of 5 u/secs., and will take another 40 u/secs. to decay to half amplitude. Because the earth has a measurable resistance, it is impossible to entice all of this current directly into the ground. Typical installations have an earth resistance of from 5 ohms to a few hundred ohms. Also, there are many paths to ground at the typical site, each having its own impedance. Thus, the path to ground will resemble a parallel resistor network, with different currents flowing in each leg.

Presume a ground resistance of 50 ohms. Ohms law tells us that with a current of 20,000 amps there would be one million volts peak voltage drop between the tower ground system and true earth. This voltage will be seen at the transmitter on the shield of the coax, and some of it will try to find a path to earth ground through the transmitter. Further, any current flowing in the coax shield will induce current into the center conductor, which will appear at the transmitter's output network. This is a frequent cause of lightning-induced arcing inside a transmitter.

DEVELOPING AN EARTH GROUNDING SYSTEM:

The most effective means of protection available to the broadcaster is through the use of a well planned grounding and bonding system. This requires treating the entire facility as a system, and thinking in terms of planned surge paths to a good earth ground. Proper grounding has a side benefit: reduced RFI and noise problems in equipment, which are often caused by differences in potential between two supposedly equal "grounds".

The primary objective of a grounding system should be to make a direct connection to earth ground, as low in impedance as possible, and then to make all other routes to ground through sensitive equipment as high as possible.

Be aware that there are two separate types of grounding at a transmitter plant: RF grounding and earth grounding. While some conductors may serve both functions, the facility should be analyzed separately with regard to its effectiveness for earth grounds. For instance, it should not be assumed that a tower is grounded adequately for lightning protection because it is connected to a radial ground system. Except in the wettest of sites, the RF ground system does not make contact with the water table. If possible, ground rods should be driven into the soil to at least this depth and bonded to the radial system.

It should also not be assumed that a single ground rod will be an adequate lightning ground -- in most cases it is not. Four or more ground rods spaced at least 10 ft. apart around the base of a tower and driven at least 10 ft. in the ground would be more appropriate in normal soil conditions.

SOME BONDING PRINCIPLES:

Conductors to earth ground should be short to be effective. The inductance of a conductor is the major factor its in impedance to a surge, and increases dramatically with length of the conductor. For instance, consider a conductor 10 meters in length of #6 AWG copper wire. It has a DC resistance of .013 ohms, and an inductance of 10 uH. For a 1000 Amp surge with a one u/sec. rise time, the resistive voltage drop is 13 volts, but the reactive voltage drop is 10,000 volts. Further, any bend in the conductor will greatly increase the inductance, and further decreases its effectiveness. A 90-degree bend is electrically equivalent to a quarter-turn coil, and the sharper the bend, the greater will be the inductance.

A large O.D. conductor should be used for all significant current-carrying connections. Four inch or larger copper strap, or #2 AWG solid wire is recommended. Because of the fast rise times, a lightning surge will behave more like AC than DC, and so the skin effect of the conductor becomes a factor. Use multiple conductors from the reference point to ground where practical. The use of multiple conductors lowers the impedance of the overall network, by decreasing the current flow and hence the voltage drop in each leg.

Use silver soldering or cadwelding for all connections exposed to the weather. Corrosion will dramatically increase the resistance of a connection.

If there are multiple earth ground connections, they should be bonded together to equalize any potential between them. These bonding routes should not flow through sensitive equipment. If local electrical code or other factors prevent direct bonding, then bond through an appropriate surge arrester to allow it to conduct only during a surge.

DEALING WITH PROBLEM SOIL CONDITIONS:

Sometimes the high earth resistance of certain sites requires special attention. Sandy or rocky soil has a high inherent resistance. Mountain top sites are a particular problem. Frozen soils or permafrost have poor conductivity as well.

One new method that can enhance the earth contact in poor soil conditions is a chemically charged ground rod. This is a copper pipe filled with earth salts or other chemicals. It has breather holes above the ground level to let air into the rod. Moisture condenses inside and travels down the length of the rod, picking up the salts along the way. It is finally released into the soil through another set of holes at the bottom of the rod. The release of these fluids at the base of the rod dramatically improves the rod's contact with the soil.

Another effective method makes use of a building's concrete foundation or a tower's concrete base pier to make the ground connection. Called a "Ufer" ground, this method was developed for the Army in World War II. The steel rebar of a foundation is securely welded together before the concrete is poured, and a conductor is brought outside for bonding to the equipment. The concrete acts as an excellent conductor between the rebar and the soil. Welding of the rebar is a must, to prevent the concrete from splitting or cracking during a lightning strike. Only cadwelding or tack welding techniques should be used, to prevent weakening of the steel and violating the U.B.C. building code.

AT THE TOWER:

The following are some of the important steps to be taken at the base of a tower to maximize protection against a lightning strike: (Fig. 1)

Weld each tower section together for at least one of the legs, to assure good lightning path to ground, as well as good RF continuity.

Adjust the ball gap at base of tower, located across base insulator. Make sure the contacts are close enough to be effective -- just beyond the point of flashover with modulation peaks. Typically, they should be set for 2/100 inch per 1000 volts peak. Orient the terminals horizontally to keep rain water out of the gap, and to assure the gap is self-quenching. Keep the terminals clean of corrosion, which would otherwise increase the resistance of the gap.

There should be an inductive loop in the connection from the A.T.U. to the tower. This creates an inductive reactance to the path, and encourage the surge to jump across the spark gap. This loop is usually made from copper tubing which has been wrapped around a cylindrical form of appropriate size for two or three turns. It helps to fill the tubing with sand first, to prevent it from collapsing as it bends. Run the tower lighting conductors, if any, through the tubing to give equal inductive reactance to both paths.

Install an air spark gap to ground at the output of the tuning unit. There should also be a static drain choke in parallel across this same path to drain off wind-generated static. This is an inductor consisting of many turns of #12 wire that is connected across the base of an insulated tower. It presents a high impedance to RF and yet a direct DC path to ground for accumulated static. A 100K ohm non-inductive resistor can also be used, about 200 watts. While effective for static electricity, these chokes cannot handle heavy currents, and typically will be destroyed by a direct lightning hit. They are readily available and inexpensive, and so it is a good idea to keep a spare on the shelf.

The ground radial system should be augmented by four or more ground rods driven ten feet apart. Also install a ground stake at each guy wire anchor, and bond it to a ground radial.

If there are any antennas mounted on the tower (FM, STL, etc.) their coaxial shield needs to be bonded to the tower at several locations, plus grounded to the tower base below isocoupler. If the antenna is for a receiver or low power transmitter, it is also appropriate to install any of several makes of coaxial surge arresters, which connect in series with the line and protect the center conductor.

All of these methods will provide a low impedance path to ground at the tower, and inhibit surges going to the transmitter. Despite these steps however, some currents will still flow to the transmitter through the outer shield of the coax, the tower lighting AC supply, and other paths. Further protection will be needed at the transmitter building.

SINGLE POINT GROUNDING:

In the transmitter building, it is important to establish one master ground point, and tie all inside ground connections to it with a branching system. Then, solidly ground this point to a ground rod network outside the building, using short, direct connections. It is most important that each piece of equipment offers no more than one path to ground. (Fig. 2)

An excellent location for this reference grounding point is at the AC service entrance to the building, where it can be bonded to the power company's ground (as allowed by local power codes). Bring all coaxial cables to this point directly from their entrance into the building, and securely ground the shields before routing them to the transmitter or other equipment. Surges will tend to enter the building via either the coaxial cable or power lines. The use of this method will provide a direct path to ground for those surges, one which will not flow through any of the equipment inside the building.

INSTALL SURGE PROTECTORS:

AC line surge arresters, if used, should be installed at this point between the AC service entrance and the reference ground. The arrester should be located as close as possible to the primary AC service entrance. This serves two functions:

1. It short-circuits any surge originating on power lines before it gets to the transmitter.

2. The AC neutral presents a low impedance path to ground for a lightning strike on the tower, due to its large conductor size and distribution to many ground connections upstream from the service entrance.

The surge arrester should be sized sufficiently to handle multiple 20 Amp surges without damage.

Most surge arresters install in shunt across the protected device to ground. In other words, they will share the surge with the protected device. In order to be effective, the path impedance of the arrester must be made lower than the path through the load. It is of primary importance to keep the lead impedance low between the arrester and ground, as the lead impedance and device impedance will add. The better surge arresters will also provide an inductance element in series with the power line itself, to increase the impedance of the path through the equipment. If this kind of arrester is not being used, an inductance can be added externally.

ADDING A SERIES IMPEDANCE:

Add a series impedance on the coaxial cable or power lines between the ground connection and the equipment. This is accomplished by passing the conductors through one or more ferrite cores or torroids. This will increase the impedance of this path to ground relative to the preferred path, and make the path through the equipment look unattractive to a surge. It can also slow the destructive rise time of the surge. If using this method, it is essential to pass all conductors through the same torroid. This will cause opposing AC currents to cancel each other and provide no impedance to the desired signal, and yet present a significant impedance to a surge. If all conductors do not pass through the core in the same direction, the ferrite will saturate and become hot. Besides dissipating power needlessly, its effectiveness against surges would then be lost. For coaxial cable, pass the entire cable through the center of a torroidal core and fasten it to the core with tie wraps or electrical tape. It is possible to find torroids of sufficient size to accommodate up to 1-5/8 inch line. For AC power lines, pass all conductors through the center of the core and, if possible, wrap one or more turns around the core, being careful to wrap all conductors in the same direction.

(The method of installing torroids on coaxial cable has not been found to be effective when the transmitter is situated right at the base of the tower. In this instance, the outer conductor is also carrying significant ground return currents to the base of the tower, and these tend to saturate the core.)

PROTECTING OTHER EQUIPMENT IN THE BUILDING:

In addition to power lines and coaxial cables, the same methods described can be applied to incoming telco audio or control lines, or any other connections to the outside world. On phone lines, connect an MOV of the appropriate voltage rating between each conductor and ground, as well as leg to leg for balanced lines. These are easily added to the punch block type of service panel. Connect the panel ground to the station reference ground. After they leave the service panel, wrap the conductors through a torroid as many times as possible before they are connected to the transmitter.

The single point grounding technique can also be effective to protect multiple pieces of equipment installed in a single equipment rack. Treat the rack the same as a building, and install a panel to serve as the entrance point for all conductors entering and leaving the rack. Install surge protectors at this point in shunt to ground, and install a series impedance between the panel and the equipment. Bond this panel to the building reference ground.

GROUNDING THE REFERENCE POINT:

Once all connections have been made to the master ground point inside the building, it must be bonded to an effective earth ground system outside the building. The same techniques described for towers can be used here. One additional method is to run a strap around the perimeter of the building, bonded to several ground rods spaced evenly around the building. In a large building with several independent systems inside, each system can have its own reference ground point established, connected at the closest convenient point to the perimeter ground. This is preferable because it avoids long runs of ground strap inside the building. It is important that ground connections between these systems be eliminated, and torroids or surge arresters be installed on the interconnections that cannot be avoided.

CONCLUSION:

These methods of protection are not difficult to implement at a new facility; however, adding them to an existing installation can be more difficult. Often, facilities have grown so complex and interconnected over time that reworking them becomes a monumental task. This brings rise to the question: How much protection is enough?

This question is answered by applying the reasonable risk concept: Given the statistical occurrence of lightning in the area, and the relative susceptibility of the equipment to damage from surges, one should expend enough time and money as is required to reduce the risk of damage to tolerable levels.

Some of these methods are inexpensive and easily implemented, and they are the most important ones to consider adding to your operation. Further, if you find yourself planning a new facility, you have a golden opportunity to "do it right" the first time.

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ACKNOWLEDGEMENT:

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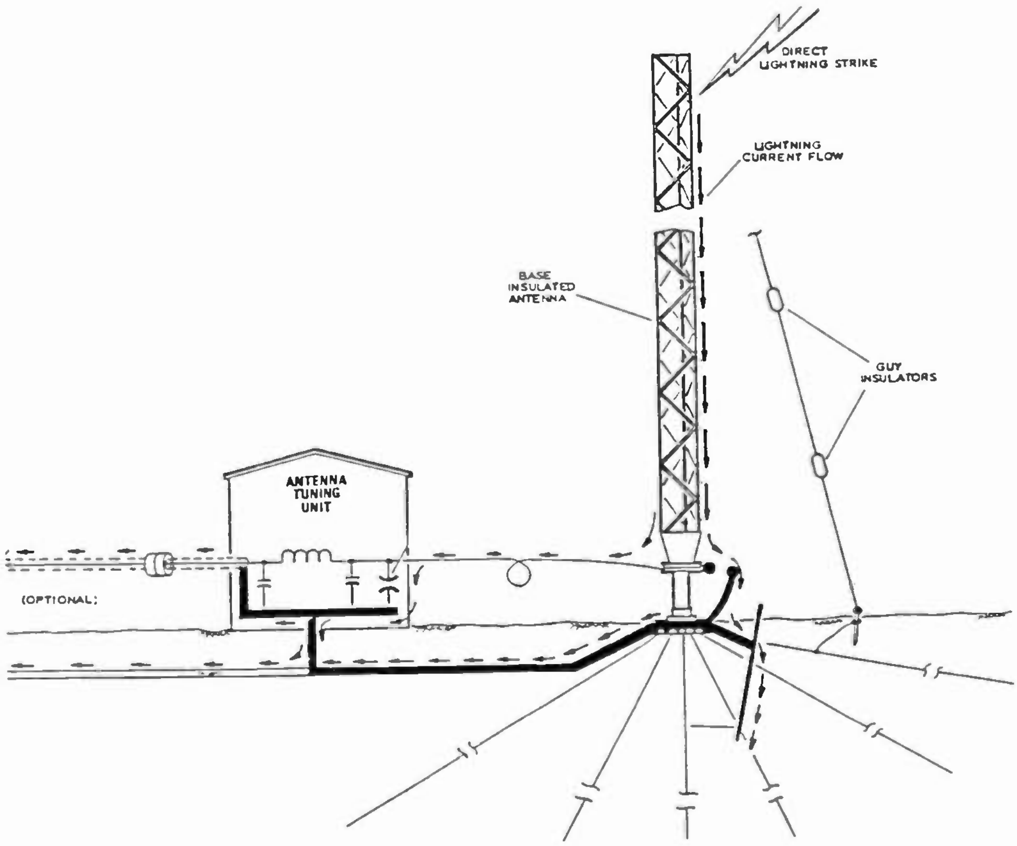

FIGURE 1.

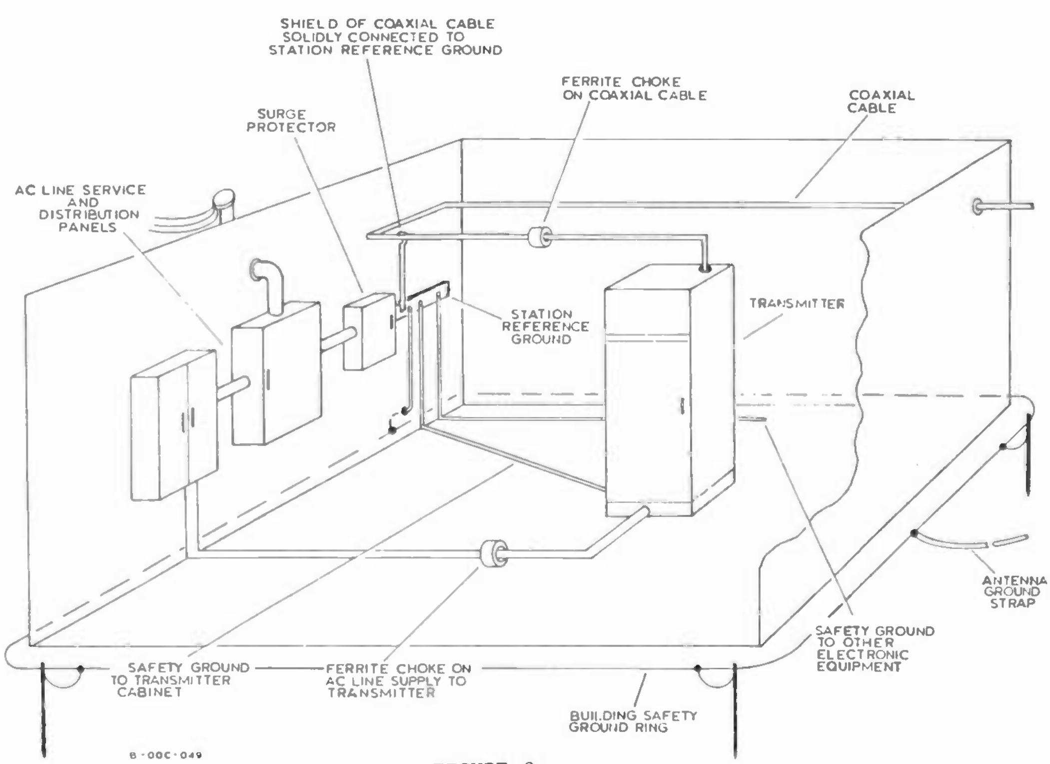

FIGURE 2.

UNINTERRUPTABLE POWER SUPPLIES FOR BROADCASTERS

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ABSTRACT

The technology of today's broadcast industry is one where the technical plant, the business, traffic, sales, newsroom, and other critical areas are controlled in great part by micro-processors, microcomputers, and minicomputers.

None of these computer devices tolerate power line problems such as transients/noise, impulse spikes, voltage dips, surges, under voltage/brownouts, over voltage/high line, split-second outages or, worst of all, total blackout.

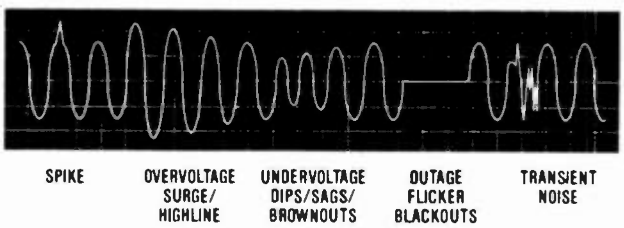

FIGURE 1

Many broadcast facilities have an emergency power generating system to cover total black-outs, but during the 10 to 15 seconds necessary for the emergency power to come on line, catastrophic things can happen to computer type devices.

This article will cover various types of "on line at all times UPS with battery back up and AC line conditioning."

INTRODUCTION

Bell Labs did a power line disturbance study collecting data from 24 sites. The samplings were obtained from a variety of locations monitored for this study, site monitoring time ranged from 2.1 to 22.4 months. The average monitoring time was 11.3 months.

RESULTS

The AC power line disturbances at the 24 Bell System sites is shown in Figure 2. Note that of all the disturbances recorded, sags occur most frequently, followed by impulses, power failures and, infrequently, surges. In fact, more sags than impulses were recorded during electrical storms. The reason is that the impulse-suppression equipment employed in the electric utility distribution system protects the system by momentarily shorting the power line. This produces sags that are seen throughout the system; whereas, impulses are quickly attenuated as they travel through the system's loads and other over-voltage protection devices.

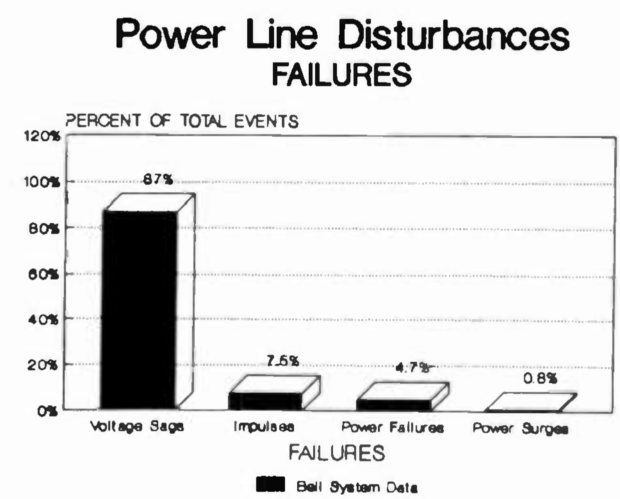

FIGURE 2

The duration time of sags and power failures are shown in Figure 3 and 4. Sags, with a voltage level of 96 volts or less and lasting at least 16 milliseconds, are used because at this level, computer equipment is prone to shut down. Note that the average sag duration was less than 0.53 seconds.

AC Line Disturbances
Power Line Sags To 96V Or Less

Percent Less Than Time	- Time -
10%	0.03 Seconds
25%	0.09 Seconds
50%	0.12 Seconds
75%	0.24 Seconds
90%	0.53 Seconds

FIGURE 3

AC Line Disturbances
Power Line Failures

Lasting Less Than Time	- Time -
10%	0.06 Seconds
25%	1.10 Seconds
50%	38.0 Seconds
75%	40.0 Minutes
90%	4.20 Hours

FIGURE 4

To illustrate the overall effectiveness of AC power-conditioning with "Uninterruptable Power System" (UPS) with 15 minutes of battery reserve, the percent of reduction in AC power line disturbances was 98%. The remaining 2% is assumed to be attributed to power failures lasting longer than the 15 minutes of battery reserve, with no standby generator back-up.

UPS TECHNOLOGY

WHAT DOES AN UNINTERRUPTABLE POWER SYSTEM (UPS) DO?

-Protects Against Electrical Power Disturbances and Power Fluctuations;
-Provides Clean/Regulated Electric Power to Load;
-Supplies Backup Power to Protect Against Black-outs.

BASIC TYPES OF UPS

UPS systems are confined primarily to three basic types. They are the Double Conversion (Static), Rotary, and the Ferroresonant type. The term static means the UPS is based on solid-state electronic technology. Rotary UPS is actually a combination of machine and solid-state technology, i.e., a motor generator set becomes the final power conversion device. The Ferroresonant UPS is a combination of a ferroresonant transformer and solid-state technology.

There is another type of system that is utilized for some small applications with loads up to 1000 watts. It is called a Standby Power System (SPS) that is not a true UPS system.

UPS comes in many sizes and configurations: from a single-phase 250 VA unit, to provide power for a single PC microcomputer to three phase systems, capable of several hundred KVA to power a large main-frame computer system or the complete technical operation of a television station. In most broadcast facilities, UPS would be used for smaller critical loads such as automation systems, character generators, electronic graphics, still store systems, computerized newsrooms, router/switching systems, traffic computer, telephone PBX, weather computers, PC microcomputers in our offices, etc.

The length of time UPS can provide on-line power during a blackout is determined by the battery capacity. The length of time on battery power available is usually 5 to 60 minutes. Where there is an emergency power generating system, a shorter length of time on battery power would be the norm.

Power Ratings of UPS

The power output of a UPS is usually rated in VA (Volt-amps) or KVA (Kilovolt-amps). Total watts of useful power from a UPS is defined as total volt/amps times the power factor.

Double Conversion Type of UPS

This is often referred to as the traditional approach to UPS. It supplies power 100% of the time from the utility AC line through its system to the critical load. See Figure 5. In this system, all of the AC input power passes through a static rectifier that converts the AC input voltage to a DC voltage. See Figure 6. A portion of the rectifier's DC output power recharges the bank of battery cells. The remainder of the DC power passes through the inverter that reconverts the DC voltage into an AC output voltage. The inverter provides continuous power to the critical load. During an AC input power blackout, the UPS's battery supplies the DC power to the inverter. See Figure 7.

Double Conversion UPS
Normal Mode

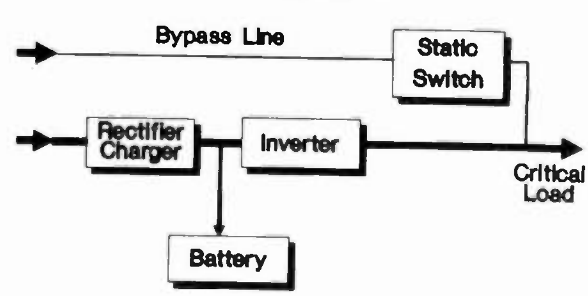

FIGURE 5

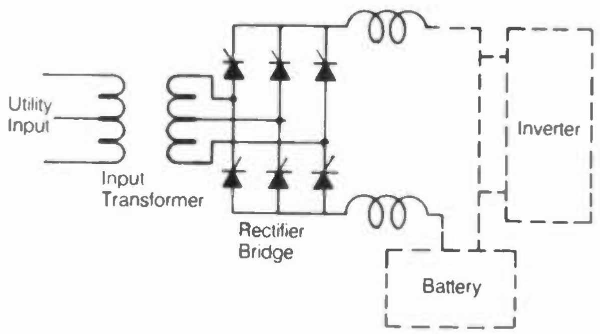

FIGURE 6

Double Conversion UPS
Emergency Mode

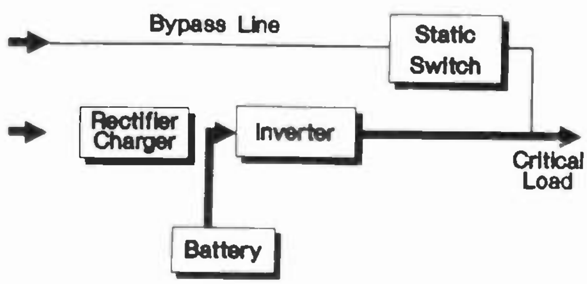

FIGURE 7

The most critical and complex part of a UPS system is the inverter. The inverter must reliably produce a clean, stable, regulated low-distortion sine-wave output. It must operate properly despite variation in load magnitude or power factor.

The latest inverter designs are based on pulse width modulation commonly known as PWM. The PWM technique synthesizes or creates a sine wave output from a series of pulses generated from the rectifier or battery DC sources. Rapid response to changing load conditions, coupled with size and weight reductions have made PWM the most common inverter design.

The output of the inverter is transformer coupled to provide the required output voltage to the critical load and electrical isolation between the utility power and the critical load. Also, on the transformer's output is an AC filter network to provide a computer grade sine-wave voltage waveform. For good isolation, the neutral on the output transformer should not be connected to the neutral of the utility power.

In the "Double Conversion UPS", a static bypass transfer switch is incorporated in the system to protect against rectifier or inverter failure. This will automatically bypass the UPS by switching the power directly to the incoming utility line. There is also a maintenance bypass switch to isolate the UPS system for maintenance. During the bypass mode, there is no protection from power line aberrations.

This traditional system for UPS has been used in critical applications for many years. Due to modern solid-state technology, its reliability and efficiency has greatly improved over the years. They are microprocessor controlled and use microprocessor-based diagnostic systems to assist in trouble-shooting and to minimize service time.

It is available in models for both small and large critical loads in a full range of single phase and three-phase ratings and voltages.

Typical Specifications for an 18.75 KVA/15 KW UPS System

(Specifications will vary for various manufacturers)

Input/Output Voltage: 208 volts, 3 phase

Three Cabinets: battery, rectifier, inverter

Dimensions Each: 72" high, 24" wide, 31" deep

Battery Type: sealed, maintenance free, lead acid or lead calcium

Battery Expected Life: 5 years

Battery Voltage: 150 volts

Protection Time/Battery Pack Weights in lbs:

5 minutes, 900 lbs.

17 minutes, 1360 lbs.

30 minutes, 1800 lbs.

System Efficiency:

85% at 100% load

83% at 75% load

78% at 50% load

Heat Rejection at Rated Load: 13,803 BTU/hr

Overload: 125% for 10 minutes

Harmonic Distortion: 5% total, 3% single harmonics

Audible Noise: 60 dBA

Rotary Type UPS

The Rotary type is a motor-generator system, See Figure 8. This system supplies power 100% of the time from the utility AC line through its system to the critical load. This type of system provides clean pure power to its critical load despite surges, sags, noise, and brief outages on the incoming line due to its rotating inertia.

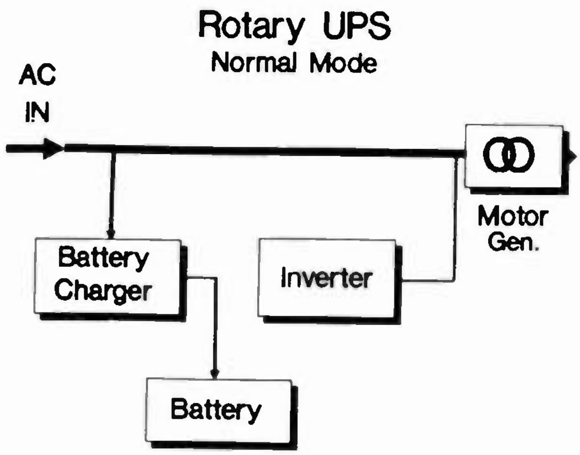

FIGURE 8

Upon loss of utility input power, the control logic senses the loss and immediately turns on the unit's static inverter system. DC power from the battery flows into the inverter and is converted to AC power and delivered into the motor-generator. See Figure 9. The rotating inertia (most systems have a large fly-wheel) of the motor-generator provides output power for the brief time required to switch the inverter system on line. Its efficiency is about 80%, and its cost is higher than other types of systems.

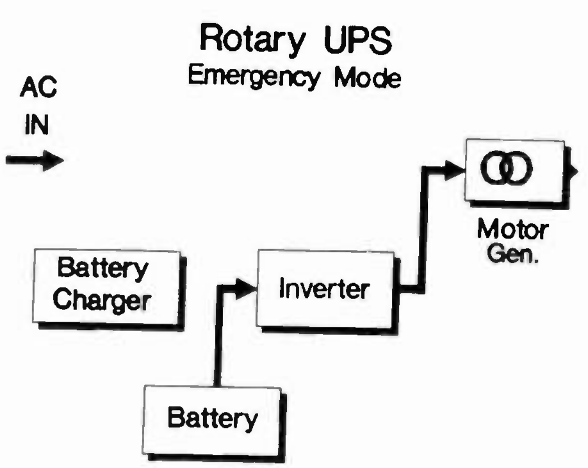

FIGURE 9

Ferroresonant Type UPS

The Ferroresonant approach to UPS (Figure 10) is a relatively new technology that has come into being during the past five years. It is becoming very popular for critical load applications in the range from 250 VA to 15 KVA. Presently, it is available only for single-phase input power. At least one manufacturer expects to have available in the near future, a system up to 20 KVA in a three phase configuration.

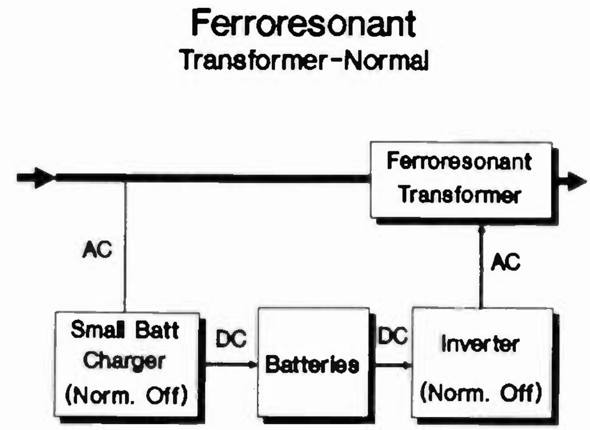

FIGURE 10

The heart of this system is the ferroresonant constant voltage transformer which conditions and stores energy. As the name implies, this device is a resonant LC network. Through the use of a combination of L and C in various resonant and/or buck-boost winding configurations, they seek to stabilize the output voltage at some predetermined voltage. It does provide a significant amount of single cycle fill-in. If the input power line fails, it will continue to supply power for 8 to 16 milliseconds before a significant voltage drop occurs. This is known as the "flywheel effect."

The ferroresonant transformer is widely used as an AC line conditioner. It can clean up raw AC line power, sags, surges, noise, provides fast dynamic response to line or load changes and high spike attenuation on the order of 250 to 1. It was invented about 50 years ago by Joseph Sola, a German engineer.

The modern computer-grade ferroresonant transformer did not arrive overnight. Originally, it had certain problems that limited its application. These problems included low efficiency, slow response time, inability to handle non-linear loads, and they generated a high audible noise. However, with new materials and advanced design techniques, most of these disadvantages have been eliminated.

When the Ferroresonant UPS is operating under normal utility line input conditions, only the ferroresonant transformer is in the active circuit. In its power class, it is more efficient when compared to other types of UPS that are on line at all times with active static systems.

This type of UPS is microprocessor controlled and detects power line problems. It brings up the static inverter in phase with no-break transfer line to inverter and return to utility power. See Figure 11.

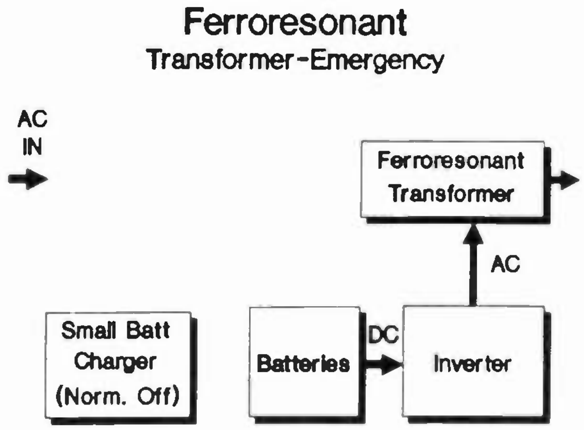

FIGURE 11

The efficiency of a typical system when operating under normal AC line conditions, is about 90 to 92% for the 3 KVA to 15 KVA units, in the inverter mode, the efficiency is 82 to 89%.

Harmonic Distortion: 5 to 6% THD, typically 3 to 4% THD
Acoustic Noise: 52 to 58 dBA

The batteries are sealed no-maintenance type with a life of three to five years. The battery voltage in a typical 3 and 5 KVA unit is 48 volts, in the 7.5 and 10 KVA units, 120 volts, in the 15 KVA unit, 168 volts.

The on-battery time with normal batteries for full load is 3 KVA, 20 minutes; 5 KVA, 15 minutes; 10 KVA, 10 minutes. At reduced load, the run time is increased, at half power the time is about doubled.

The heat (BTU/HR) for 3 KVA is 1400; for 5 KVA, 1900; for 10 KVA, 3000; for 15 KVA 4500 BTU/HR.

The inverter is a Pulse-Width Modulator (PWM) design, described under the Double Conversion type of UPS.

The dimensions of a typical 7.5 to 15 KVA is 42" high, 23" wide, 30" deep, with a weight of 600 to 750 lbs. The battery cabinet may be a separate unit and is somewhat larger with a weight (with batteries) up to 1000 lbs.

Parallel Processing

Another form of Ferroresonant UPS is called Parallel Processing. This type of UPS has the ability to process power from two sources of power simultaneously, commercial AC and batteries. It contains three modes of operation, Normal, Inverter, and Electronic Bypass. See Figure 12. Each mode can process power to the load exclusive of the other two modes. Normal Mode filters and regulates commercial AC, Inverter Mode provides battery supplied power, and Electronic Bypass connects commercial AC when the load starts up.

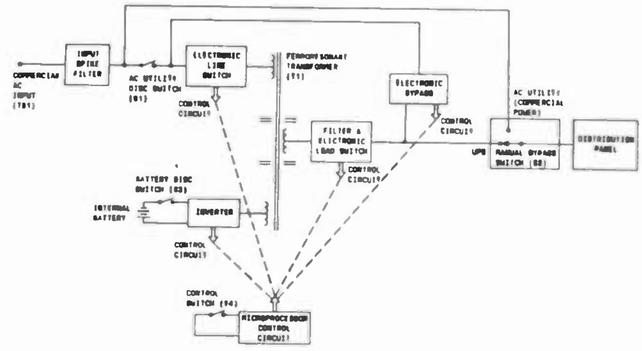

FIGURE 12

The inverter is controlled by a microprocessor. The inverter, ferroresonant transformer, and output filter combine to provide a comparable crest factor capability in the normal mode. During periods when commercial AC is acceptable, the microprocessor "idles" the inverter. Idling makes it possible to have the inverter in an on-line status. The inverter is inhibited from delivering continuous power while the UPS is in the normal or bypass mode. Only in normal mode is the inhibiting control removed during the few "hot" cycles when the start-up of a very fast load is at the threshold of requiring a transition to bypass mode. It is more effective to supplement commercial AC with inverter AC for a few half cycles.

In both normal and electronic bypass modes, the assumption is that there is acceptable commercial AC. The microprocessor has been programmed to prevent a transition from inverter to bypass mode. The only way you can get to bypass mode is to drop into normal mode and remain there for about 3 half cycles before making the transition to bypass mode. The electronic bypass in a microprocessor-controlled switch satisfies the load's temporary need for a large amount of inrush current during load start up.

This parallel processing is available in 3, 5, and 10 KVA ratings. This type of UPS's overall characteristics are very similar to a typical ferroresonant system. Its price is about 20% higher than a typical system.

The dimensions of a 10 KVA unit is 61" high, 20" wide, and has a depth of 30". This includes the batteries which are installed in the same cabinet. Total weight is 1000 lbs. The on battery time for a 10 KVA system under full load is about 10 minutes.

The ferroresonant type of UPS, as are other types of UPS systems, are made by a number of manufacturers and each has their own special circuits and technology.

Standby Power System (SPS)

In this type of system, the normal utility input power flows through a transfer switch directly to the load. The inverter is off, standing by, and is disconnected from the output. See Figure 13. A separate low-power charger rectifier floats the battery. If the AC input drops below about 85% of normal or fails, the transfer switch disconnects the AC input and connects the inverter to the SPS's output, usually in a break--before-make sequence. The inverter is then activated. See Figure 14.

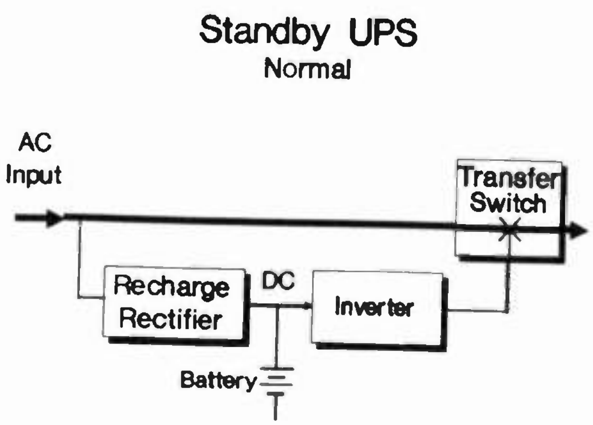

FIGURE 13

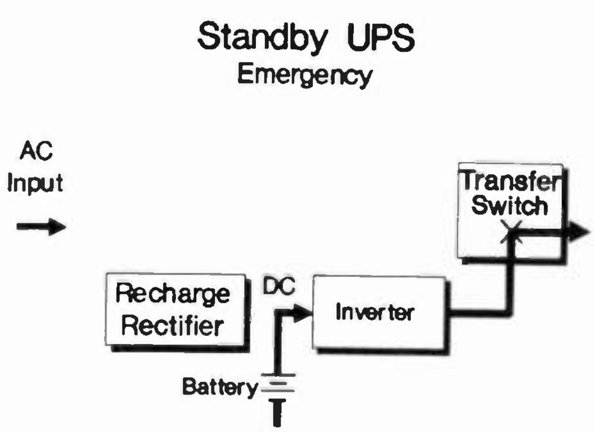

FIGURE 14

This type of standby system can be very small (for a 1000-watt unit about 15" high, 18" long, and 8" wide) and efficient because the inverter is normally off, the transfer switch has negligible conduction loss and the recharge rectifier's capacity is designed for only 10% of the system's rated power. Many low-power standby systems have recently appeared, most of which use either a squarewave or quasi-squarewave inverter. Their maximum total harmonic distortion is usually about 7%. They usually have no AC input voltage regulation and must synchronize their inverter to the AC input voltage for continuity of phase during transfer switch operation.

The majority of these types of systems have some form of transient suppressor protection. A standby system suffers from a long "transfer time" due to detection and switching action of the transfer switch's sequential operation. They are intended usually for a single type of equipment application. The upper power limits for this type of standby system is usually 1.3 KVA or 1000 watts.

The batteries in a 1000 watt SPS are typically 24 volts with a full back-up time of about 8 minutes or about 20 minutes under half load.

Their cost is about half the price of a true UPS system that filters and regulates the amplitude of the AC line input and shortens the break in power when transferring to battery backup.

CONCLUSION

The need for UPS has come to stay in the broadcast industry. With the greater use of computer type devices in almost all areas of today's broadcast operations, UPS will become a common necessity in a station's equipment needs.

The use of UPS systems will be determined by a number of factors: Any electrical machine, any electronic system, or computerized installation that could be operationally impaired, or the inconvenience caused because of unpredictable power line disturbances/blackouts may demand a need for a UPS system.

ACKNOWLEDGEMENTS

A special thanks to Jerry Plemmons, Vice President of Engineering and Leigh Moynihan, Outlet Communications, Inc. for their help in preparing this paper.

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ROBOTIC CAMERAS: THE NEWS OF THE FUTURE

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INTRODUCTION

Developments in the integration of robotic controlled broadcast cameras within the overall concept of full studio automation, have significantly advanced since the last engineering conference held in Dallas. At that time, developments in the computer control of camera systems were based on the requirement to exercise control of the pan/tilt position of the camera with zoom and focus of the lens, this method of control being more related to remote rather than robotic control.

Further developments considered by the paper given by Goldsmith and Wolfe(i) last year, emphasised the requirement to physically move, under robotic control, the camera across the studio floor, allowing use of multiple sets within a studio automation environment. This paper presents how such developments have been realised with the inclusion of video tracking techniques to enable complete hands-off control.

HISTORICAL

The remote control of broadcast cameras within a News or Presentation Studio is not new; such systems have been installed and operational for many years throughout Europe, the controlled parameters usually being restricted to height, pan, tilt, zoom and focus. Early systems were heavily based on analogue design techniques. However, the introduction of microprocessor technology enabled improved control capability. Typically, such control allowed :

1. Camera Positioning

   Remote control of the camera such that it could quietly and smoothly rotate 350° and tilt ±45° around the horizontal, whilst on-air. Additionally, zoom angle and focus under servo control could be modified, and the total system controlled by an operator physically remote from the studio. One operator was given the access necessary to be able to control camera configurations from one control position, the control room being either co-located, or many miles from the studio. A major operational benefit was the proximity of camera operator and programme director with the obvious advantage of simplified communications.

2. Preset Shots

   A main feature of remote control systems based on distributed use of microprocessors, was the ability given to the camera operator to pre-programme and store selected shots for a specific programme. These shots could then be actioned on demand, an essential feature being the repeatability of a shot to produce exactly the same result as that previously programmed. Typically, the level of positional accuracy sought would be within 1/20th of a degree.

3. On-Air Trimming

   The advantage of pre-programmed shots is that a camera can be instantly driven to a known position; unfortunately the object (normally human), is not pre-programmable, hence may have moved. This demands the system to have the capacity to trim the shot, often on air. Joystick control is utilised to provide smooth and accurate shot trimming. A feature of earlier Radamec Systems was that as the lens moved towards its narrow angle, the internal control logic became proportionately more sensitive, enabling small moves to be achieved with greater movement of the joystick.

4. Movement between Preset Shots

   It is important, to achieve the best visual effect, that all movement is synchronised to allow all axes to start and stop simultaneously. The movement which follows a preset timing pattern also incorporates start acceleration and stop deceleration profiles to enhance the smoothness of movement. Moves involving variation in all axes would be difficult or impossible for a cameraman to achieve; however, such moves can be easily incorporated in a remote system.

5. Continuous Movement

The more sophisticated control systems allow a number of preset shots to be linked together in a similar way to key frames. The broadcast camera can be programmed to undertake complex curved moves by interpolating between pre-programmed shots. This allows movement on fixed artwork, such as maps and photographs to be dynamically viewed.

NEW DEVELOPMENTS

The increasing acceptability of remotely-controlled cameras in the United States has resulted in the robotic approach to broadcasting. These developments can be determined by three distinct areas :

i) Robotic control of camera position around the studio floor - Tracking

ii) Centralised management

iii) Video Tracking

Tracking

The basic requirement is to physically move the camera around the studio floor quietly, smoothly and with a high degree of accuracy. This movement must be fully integrated with the control of camera height, pan and tilt positions. Such movement, when required, must be both fast for off-air shot acquisition and smooth to allow preselected timed on-air movement. Further, the cable management system must be unobtrusive but practical within a studio environment. The level of requirement can be achieved by either a fixed track-based system or a mobile robotic platform, both having advantages and disadvantages.

Track-Based

This method of robotic movement was designed and developed over a six-month period. It now forms an integral part of the most advanced robotic system currently installed for the use of News Broadcasting.

The Radamec EPO X-Y Tracking System utilises as the main camera, prompter and lens support, a Pan & Tilt Head mounted on a servo-controlled height mechanism as shown by Fig. A. The height mechanism is designed solely for robotic control. However, as the mechanical system is held in balance, only a small level of energy is required to change height. A small handle can be connected to allow the height to be manually altered if required.

The height mechanism is mounted on a support dolly which can be free-standing. In the robotic mode, motorised wheels are fitted. The dolly can be mounted and driven along a parallel track configuration, see Fig. B. In the Radamec EPO track-based system, the parallel tracks are themselves supported by a motorised wheel structure which allows both X and Y axis camera movement. This total configuration allows simultaneous movement of all previously stated remote control functions with a controlled X movement along the parallel track and a controlled Y movement 90° to the parallel track. Diagonal or curved moves can be achieved by this system. A single guide rail is positioned on the floor to prevent skewing in the Y direction and to enable accurate positional feedback to be received for positional purposes.

A single track system can be tailored to accommodate maximum X and Y movement within a prescribed area of the studio or newsroom. The structure of the system allows for multiple robotic cameras to be installed, sharing a single guide rail or using multiple guide rails. There is no interaction between camera systems.

A significant feature of the design is the cable management system which ensures that cables attached to the robotic dolly do not impede its movement nor do they have any adverse noise effect.

A major element considered at the initial design stage was the recognition that robotic control of any movement demands a reliable and effective safety system, designed to prevent injury to personnel or damage to equipment.

Two independent safety systems have been employed. The primary system uses infra-red (I.R.) beams projected parallel with the rail structure. This ensures that any obstruction encountered by the rail when in movement, breaks the beam prior to impact. The signal transmitted, when the beam is broken, is used to disable the drive mechanism in that direction. The use of out-riggers to support the I.R. sensors ensures that the movement ceases before impact. In the case of multiple camera systems, it may be necessary to move two cameras extremely close to each other. To achieve this, the operator can manually override the primary system, whereupon movement is allowed only at considerably reduced speeds. A secondary safety system using proximity sensors, shuts down motor power whenever obstructions are detected within two inches of the system.

An alternative approach to that of the tracking system described above, is the free roaming, mobile platform which uses the hovercraft principle. A particular case in point is the 'Robo-Glide' robotic platform by Elicon, La Habra, CA. Using air bearings to lift the platform, the system moves on a cushion of air, hence providing frictionless system operation. A proprietary drive system enables the platform to move forward, backward or sideways. Curved movement can be achieved within a minimum 4ft radius. Positional feedback for the air platform is based on a sensor system allowing it to follow magnetic tape laid on the studio floor. The tape is laid over the route which the platform is expected to follow. Tape intersections allow movement in two directions. Mounted on the platform could be the height mechanism used in the Tracking System, hence all facilities would be available. In the case of an air cushion system, care would be required when organising the cable management and safety systems to ensure freedom of movement is safely and noiselessly maintained.

Centralised Management

The requirement for News Studios which use 3 or more cameras, is that control of each camera must be operationally economic with regard to speed of control and flexibility. Whilst pre-programmed shots can be established and complete running orders scheduled, a News Studio operates within a dynamic and constantly changing environment. This demands that robotic controlled systems must be structured to allow for high pressure situations. Earlier systems allocated control panels for each remotely controlled camera. However, the demand for robotics has extended from simple control of the camera and lens to that encompassing co-ordinated control of vision, lighting and sound equipment. Simultaneous control of all areas allows composite cues to be established for each particular pre-programmed shot. In order that this form of co-ordination is properly managed, both in establishment and when changes are required, a centralised management Cue Computer System is essential. The primary function of a Cue Computer, illustrated by Fig. C, would be :

i) Simultaneous control of robotic camera systems

ii) Control of required external devices, eg., vision switchers, lighting control systems, character generators, etc.

iii) Cue selection and initiation from custom panels, data tablets, See Fig. D, or external stimulus such as Newsroom computers.

A cue would be pre-programmed and would relate to specific news items or sequence of items being covered or included in a tentative running order. Each cue would contain preset shot commands for each robotic camera, associated with the necessary instructions required to be transmitted to other external equipment; an example would be to call up lighting control system memory numbers. Access of that cue during the programme could then be initiated by an operator making the selection from the data tablet. Alternatively, it could be automatically called up by a vision switcher, newsroom system or station computer.

The display of cue information, method of access and selection has been designed to allow the operator flexibility of control, essential for the rapidly changing news room environment.

Video Tracking

It was earlier stated that whilst shots and cues can be established and repeated the subject, if human, will inevitably move from the desired position. This will require the camera to move with that person. In the past this has been achieved under joystick control whereby the camera position is trimmed to follow any movement. A major development within the introduction of robotics, is the inclusion of video tracking technology. A schematic illustrating a typical configuration is shown by Fig. E. Such a system enables, by closed loop control, the camera to 'lock-on' to a person or object. Signals from the tracker then take over control of the camera robotic system. Using sophisticated video processing and computing techniques, the system will use a high contrast edge to 'lock-on', such as the eye or face of the presenter. Once the camera is 'locked-on' to a pre-defined image, the tracker will output commands which control the Pan & Tilt Head supporting the camera, automatically directing it such that the desired picture is held, irrespective of any movement by the subject.

FUTURE DEVELOPMENTS

On-going developments in the area of robotics within News and Presentation Studios, are being targeted towards integration with main station computers. The benefits available to the broadcaster will be such that complete programmes may be structured from a terminal with interlinking between the editorial content of the show and the supporting associated broadcast studio equipment. While on a wider basis, further developments in video tracking are being targeted towards application in the broadcasting of sports.

Reference:

(i) B.J. Goldsmith & M.J. Wolfe

'New Developments in Computer Controlled Remote Control Camera Systems'

41st Annual Broadcast Proceedings

Page 284-288

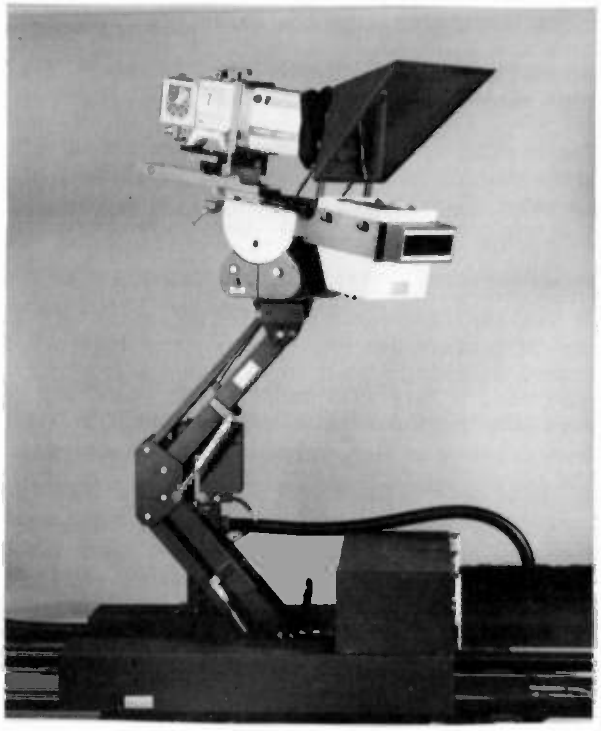

FIG. A:

RADAMEC EPD ROBOTIC HEIGHT MECHANISM

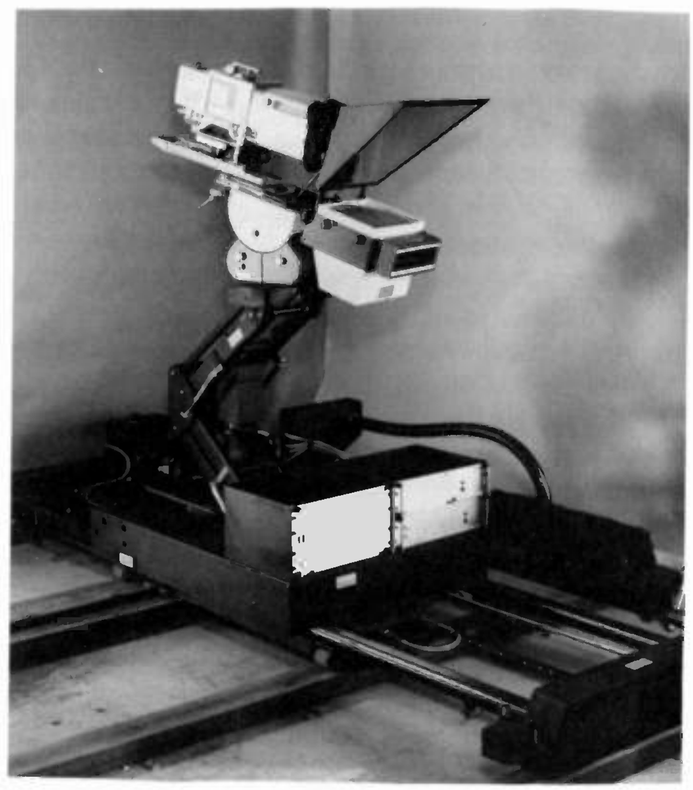

FIG. B:

X-Y TRACKING SYSTEM

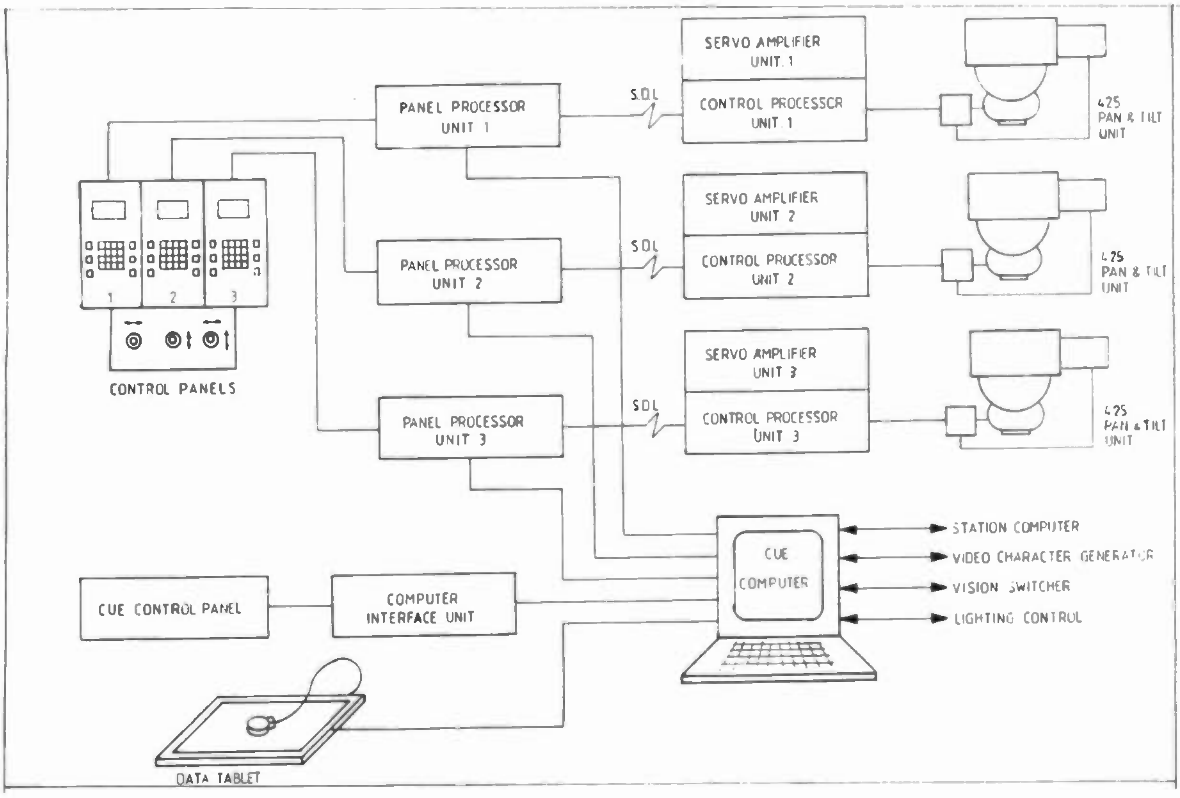

FIG. C:

CHANNEL ROBOTIC SYSTEM UNDER CUE COMPUTER CONTROL

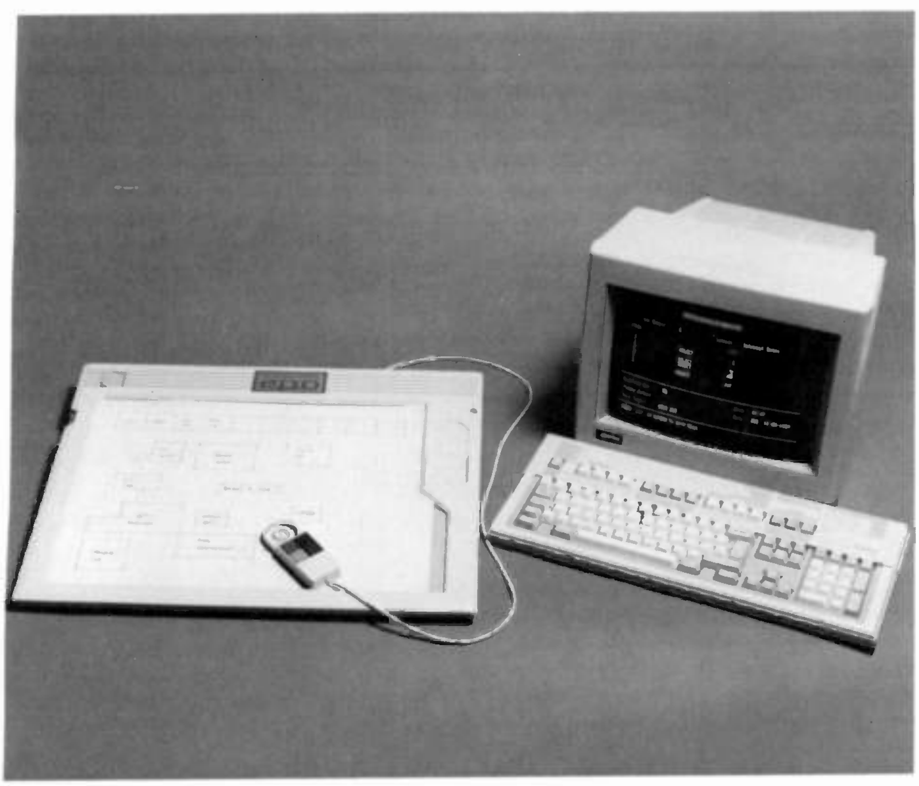

FIG. D:

RADAMEC EPO CUE COMPUTER & DATA TABLET

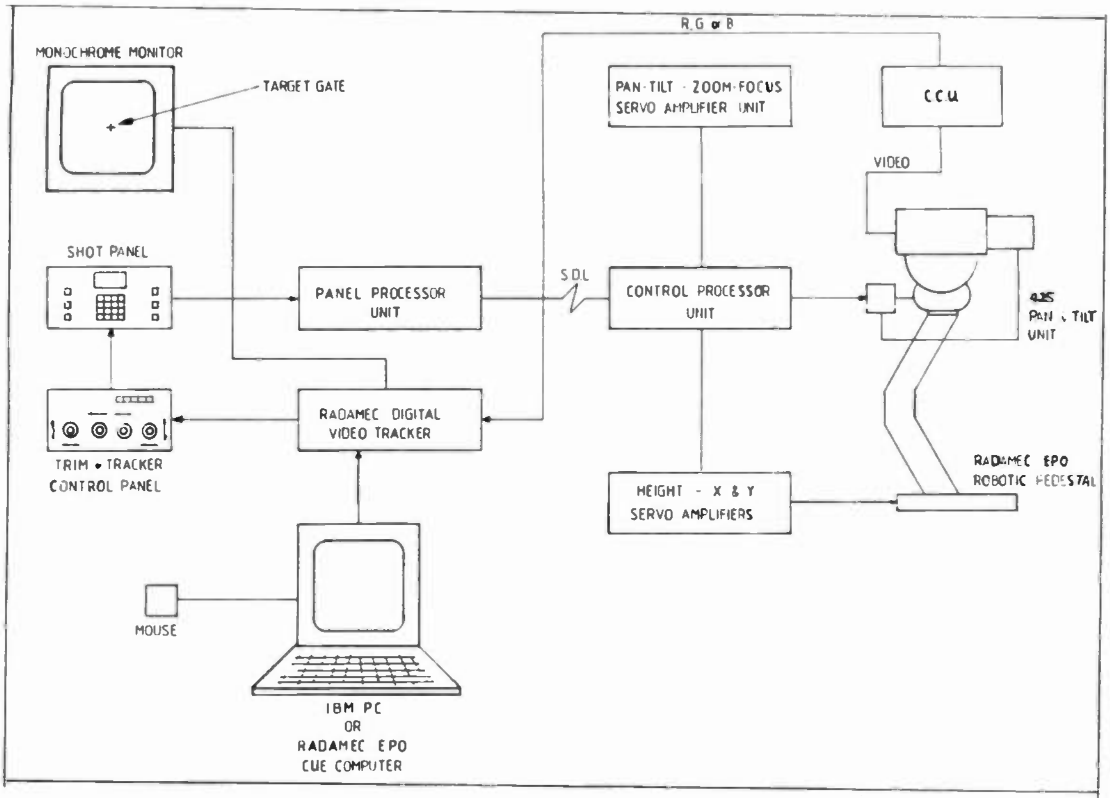

FIG. E:

ROBOTIC VIDEO TRACKING SYSTEM SCHEMATIC

NEWSROOM AUTOMATION OPPORTUNITIES

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ABSTRACT

First generation newsroom systems succeeded at providing automation of basic newsroom functions such as wire reading and filing, word processing and script production. But they failed to provide any benefits to newscast production. As a result, it was difficult to measure their return on investment, and sales grew only modestly.

Second generation "automation systems", now available, have been developed in response to broadcasters requests to "Improve the broadcast", and "Improve the bottom line". Such systems, by interfacing with other production equipment, are bringing automation to newscast production, and bringing a wide range of benefits to their users.

INTRODUCTION

What is Newsroom Automation?

The general term automation, when used in the broadcast environment, means different things to different people. This is not surprising considering that each vendor offering automation-related equipment views his or her product as the center of the automation universe.

If we assume that "broadcast automation" refers to automating all the steps and equipment needed to deliver a station's daily schedule, then "newsroom automation," as used here, shall refer to automation of the station's newscasts in isolation from the rest of the day's broadcast schedule.

During the eight year lifespan of newsroom systems, the concept of newsroom automation has changed radically. While first generation systems ended up covering the basics, today's second generation newsroom systems are bringing a broad range of benefits to broadcasters.

First Generation Newsroom Systems

First generation systems, first introduced around 1980, were primarily intended to bring "automation" benefits to the labor intensive and paper intensive tasks that occur before a news broadcast. These tasks included wire reading and filing, script writing, and script archiving. The automation of such functions, as proven by the commercial success of word processors in business offices, offered the promise of quantum jumps in productivity and organization within the newsroom.

In fact the newsroom looked like such a promising target for computers that many vendors chose to offer systems, each expecting to be swamped with orders. And the electronic teleprompter looked like the easiest part. The camera reading taped together scripts from a moving paper tray, and the operator of this device, would quickly be a thing of the past.

Vendors said they could do it all and more. Buyers said they'd all buy. But vendors didn't do it all, and so buyers didn't all buy. Today, as we enter the second generation of newsroom systems, more than four fifths of U.S. TV stations do not yet have systems. And today, as in the past, every news director still wants one.

The Interim Years

What the broadcast world got in first generation systems were the basics. These included wire reading, word processing, script writing, stacking of shows, some timing features and, perhaps, a one way download of scripts to the teleprompter.

The newsroom got quieter and more efficient. Scripts were better researched and better written. Newscasts could be edited right up to the last minute and then reliably printed on a high speed printer. Bulletins were more easily received and put on air.

As a productivity tool in the area of script production, such systems enjoyed strong acceptance. Immediate wire access, the ability to find wire copy fast, and the ability to edit scripts as they were written all proved to be popular capabilities that experienced users could never again live without.

But efforts to automate newscast production (ie. the teleprompter) failed miserably. The early systems actually prevented good newscasts instead of causing them. The computerized teleprompter was simply too inflexible. It couldn't easily handle all the changes that could occur. It got in the way. The computer was beaten by the camera and paper tray and razor and scotch tape. These basic tools, in the hands of a trained individual, could simply do it better.

Thus first generation newsroom system were relegated to the tasks of wire reader, word processor, and script writer. For all practical purposes, they could be turned off once the newscast started.

In spite of their productivity benefits and popularity among news directors, to the General Manager, such first generation newsroom systems were simply elaborate and expensive word processors with an unsatisfactory return on investment. Instead of spending $200,000 on a first generation newsroom system, broadcasters opted to invest in myriad other products that would provide a better return or improve the on air product. Hence today's low penetration levels.

THE PATH TO GENERATION TWO

Still, first generation systems were popular among news staffs and enjoyed consistent sales. Even the networks bought them. But sales were not high enough to support multiple suppliers, and marginal vendors fell by the wayside. Surviving vendors looked for ways to increase market acceptance of their systems.

As systems were installed in more newsrooms, users began asking for them to do more. Vendors were deluged with requests for enhancements. One such request would set a clear direction for future development.

Request 1: Improve the Newscast

The request heard early and most often was the one most difficult to accommodate and the one with the most wide reaching implications. It was "Help us improve our newscast. Help us reduce on-air mistakes." The answers vendors were looking for on how to spur additional sales quickly became clear. To enjoy greater acceptance and return greater value to their users, newsroom systems would have to venture into the never never land of on-air newscast production.

Request 2: Improve the Bottom Line

More recently, as TV stations' financial results have dropped with increasing competition, broadcasters have begun to ask another question. How can newsroom expenses and staffing be controlled without sacrificing the news product. Newsroom systems seemed to offer the most promise when discussions turn to productivity and efficiency improvements.

Two and a half years have passed since this company began its development efforts to respond to these market directives. What looked easy back then was not. The implications of fulfilling Request 1 were enormous.

Vendors learned what it meant to offer systems that prevent newscast mistakes. It meant that radical changes and enhancements needed to be made to the foundations of their systems. Newsroom systems couldn't "roll the right tape" until many other fundamental building blocks were in place.

SECOND GENERATION SYSTEM PREREQUISITES

Building a true second generation system with the ability to satisfy stated market objectives meant existing systems must improve in at least five areas: Functionality, Flexibility, Responsiveness, Reliability and Security.

Functionality

The newsroom system must control the show. It must provide all possible timing functions and be sure times are always correct no matter what changes are made to the rundown. It must feed the teleprompter the correct script for the anchors to read at the correct time.

It must allow for changes during the broadcast to accommodate late breaking stories, video not yet in, live shot not ready, broadcast too long, and broadcast to short. This means the system must be able to reorder stories, add stories, drop stories, and lengthen or shorten stories. Furthermore, each such change must be instantly reflected in the teleprompter and in the timing clock.

To be accepted by news professionals, the system must perform these functions more easily than the way they are done without a system. It must beat the scotch tape, razor, camera and paper tray approach to the teleprompter. It must beat the stopwatch approach to timing.

Flexibility

But every station produces its newscast a little differently. To be accepted in the marketplace, the system must adapt to each station's methods rather than vice versa. So, for example, the computer's rundown must be capable of mimicking a wide variety of formats.

Responsiveness

The system must be responsive. That is, when the producer gives it an on-air command that must be performed now, the system must perform it now. A delay of 3 or 4 seconds, caused by a busy central processor, is unacceptable. And in the hours just before the newscast, at the time when everyone is using the system, it must respond fast enough to aid, not hinder, productivity.

Reliability

If the system is going to have any involvement in the on-air production, then it must be very reliable and allow for emergency back-up procedures. Unlike other machines in a TV station, the newsroom system is used by potentially dozens of people at once, all sharing a common database. Hence it is not practical to have a second one on site as is the case with much other critical equipment. Instead it must have redundancy and reliability built into its architecture.

One solution is for the system to have discrete units that operate independently. The teleprompter could have enough intelligence, processing power and memory to contain all the scripts and to handle the rest of the broadcast if the mainframe went down during the newscast. The part of the system doing the timing could be a separate processor, able to continue timing functions if the mainframe went down.

Security

Because it is used by so many people, the newsroom system must provide a clear definition of who can do what on the system. Unlike the master control system that is operated by one or two people whose entire focus is on the job at hand, many people could be signed onto the newsroom system and be performing completely unrelated tasks to the newscast in a variety of locations. The potential exists for accidently destroying or even sabotaging a newscast if proper system security checks are not embedded into the system.

The logical conclusion is that the system must have an intricate security system that defines and limits each user's authority. Who is allowed to send scripts to the telepompter? Who can change the order of a line-up, once the newscast has begun? Who can drop a story from the line-up? Who can initialize the BetaCart?

Looking at all these requirements, is it any surprise that early first generation systems failed in the control room?

THE SECOND GENERATION SYSTEM TAKES SHAPE

Where True Newsroom Automation Comes In

Imagine now that we've built the newsroom system that performs all the tasks of first generation systems, but also allows the producer, through a workstation in the control room, to drop, add, or reorder stories at will, and have all the timings automatically changed, and all the changes automatically sent to the teleprompter. Now we have created shear chaos for the operators of the tape machines, character generators, still store devices, and cameras. Changes in the rundown can now be made so easily that the operators of the support equipment cannot keep up.

The answer lies in automating communications to those other machines -- linking them to the newsroom system so it can keep them advised of changes in the rundown order. Such a system would electronically:

Send the right script to the teleprompter;

Send the right "playlist" to the Cart machine;

Send the right Super list to the character generator;

Send the right shot list to the camera automation system;

Send the right scripts to the closed captioning device;

Send the right still store list to the still store device.

Having done all that, the system would then advise each device of changes to the rundown order during the newscast. And, it would have reported back to it error conditions from each such device. Communications must be two way.

If a system could perform all these steps successfully it would go a long way toward fulfilling Request 1: "Reduce On-Air Mistakes." Consider the environment now for the producer with such a tool.

How A Newsroom with A Second Generation System Might Operate

As the newscast start time is approached, writers are finalizing script wording for clarity and accuracy. Changes made to any part of the rundown (story additions, deletions, reordering, or arrival of video) by any user are automatically displayed on other terminals that are "tuned in" to the rundown.

The news director has a work station on his desk, tuned into the rundown. He sees it flash when story one is moved to segment two. He checks the cart machine status to see if video is in yet on the courthouse indictment announced an hour ago. He checks the clarity and accuracy of his new reporter's scripts. He looks at the overall show order to see if it flows logically.

The producer looks at the rundown on his workstation and reviews the status of all stories and all video in the show. Status indicators on his terminal confirm that the stories in segments one and two are all written, editorially approved and printed. (The stories are printed because no matter how good the electronic teleprompter is, anchors may always want hard copy!)

The cart machine has been initialized and the status column on the producer's workstation shows that the first and second video pieces are queued and ready for on air. Story slugs match bar code slugs on all video pieces in segments one, two and three. No video is in yet on a story scheduled for segment 4. No problem. It can be moved further down the rundown with a few keystrokes if the video does not show up.
The over/under clock shows that the newscast is over by 30 seconds. Something may have to be dropped or shortened.

Supers are all in and correctly spelled, except for the live shot planned for segment 4.

The anchors are ready, the automated cameras have moved to their open positions, and the teleprompter has been initialized with all the stories in the rundown.

The producer goes to the control room, about to start the newscast, confident that he can make any changes necessary to improve it during the broadcast. If a bulletin comes in, his terminal will be notified. He can read it instantly, file it in the rundown if he wishes, and immediately see the effect on show time. And, thanks to his second generation newsroom automation system, the cart machine, teleprompter, character generator, still store, captioning encoder and robotic cameras will all follow him like an orchestra follows its conductor.

The newscast starts and the producer puts his workstation in "On-Air" mode. System security puts him it complete charge of the rundown. The system clock starts, and the anchor reads the open.

The Second Generation System Defined

We now have a definition of what second generation newsroom system automation really means. It means a system which will:

(1) Control and automate all the equipment used in a newscast in such a manner that the staff can produce the best possible newscast;

(2) Give the producer the ability to change the rundown instantly as needed, during the broadcast, and have the related equipment also respond;

(3) Roll the right tape; display the right super; use the right camera shot; send the right story to the teleprompter; send the right story to the captioning encoder, no matter how the rundown is changed during the show.

SECOND GENERATION NEWSROOM SYSTEM AUTOMATION INTERFACES

The second generation newsroom system is an "automation system" with sophisticated interfaces to other newscast equipment (See Figure 1: Newsroom System Automation Interfaces). It not only automates the basic functions which occur before the newscast...the wire reading and filing, word processing, and script writing. With its machine interfaces, it also controls many of the important pieces of equipment used during the newscast.

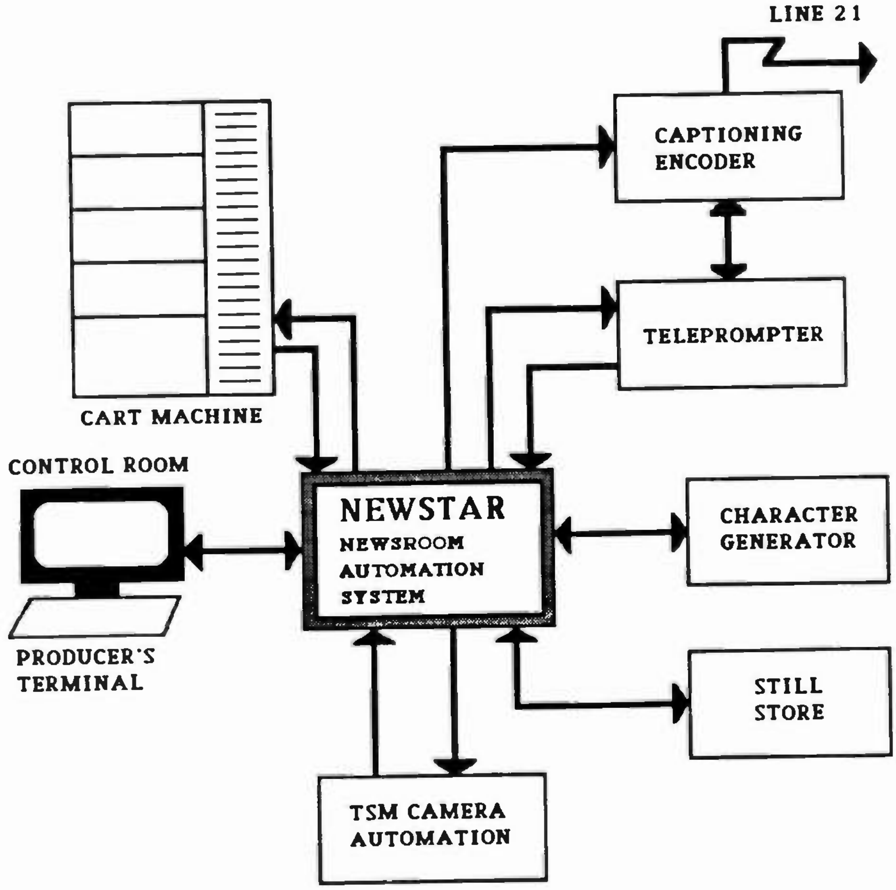

FIG. 1 NEWSROOM SYSTEM AUTOMATION INTERFACES

The Teleprompter Interface

The teleprompter interface enables the system to send the complete story rundown to it right before the show. The interface will synchronize with the producer's workstation, causing the teleprompter to access the current story to be smooth scrolled for the anchor to read. A separate speed control, operated by an individual in the control room, will insure that the scroll speed is always matched to the anchor's reading speed.

The teleprompter's internal computer will keep track of all stories and reorder them internally if the producer makes changes on his workstation. It will also delete stories if required and accept new ones. It will word wrap each line of script on the monitor to exactly match the word wrapping on the printed copy, so the anchor can easily find his place on the hard copy if he chooses to.

The teleprompter will indicate which anchor is to read each story, and will show how much time is left in each piece through a countdown clock.

Cart Machine Interface

The intelligent cart machine, with multiple VTR's and storage for dozens of mechanically accessible tapes, was originally intended to be used for playing commercials. But with the newsroom system's new ability to juggle story order on air, use of such cart machines for news pieces is logical and inevitable.

The cart machine interface is quite sophisticated. It must allow the following functions to be performed:

- Send the rundown play list to the cart machine;

- Attempt to match tapes to rundown stories using the information on the bar code labels;

- Provide a routine to allow the producer to match unmatched tapes;

- Reorder the play list as necessitated by changes to the rundown order;

- Feed back status conditions to the newsroom system including:

  a) Story ID doesn't match tape.
  b) Story ID matches tape.
  c) Tape matching story ID is cueing.
  d) Tape matching story ID is cued for output.
  e) Tape matching story ID is currently on-line.
  f) Tape matching story ID is currently on-air.
  g) Error detected on cart machine.
  h) No time code on tape.
  i) Tension released on tape in VTR.
  j) There are two tapes with identical titles.
  k) The tape is frozen on air.
  l) The tape may or may not be on air.
  m) The tape has played on air.

In addition, the system must be able to handle the cart machine's very high data transmission(38.4KB) rate without slowing down.

Character Generator Interface

One of the main functions of the character generator during the newscast is displaying supers. The supers are usually defined as the story is written, so almost all are available in the system long before air time.

The interface must allow the system to get the supers out of the story file and send them to the character generator. Ideally, the system should spell check them before the newscast. As the supers are written to the character generator's storage, the CG must return location numbers to the system.

As the producer adds, drops and changes story order, the system must report the correct new order to the CG. If an operator is used to press the play button on the CG, then the system must either display the next location number on a monitor near the operator, or cause the CG to reorder the supers so only the play next key is used.

The interface must also allow an operator to enter a new super for situations such as live interviews or remotes where the name is not known ahead.

Camera Automation Interface

The newscast represents a very logical and beneficial application of camera automation. Such automation becomes especially effective when a newsroom system is present to control the newscast. The camera automation system must, like the cart machine, have the intelligence to communicate in sufficient detail with the newsroom system.

The Camera Automation Interface must perform, or allow for, the following functions:

- Send the story event list to the Camera Control Unit (CCU).

- Send the story order to the CCU.

- Feed back status conditions to the newsroom system.

- Send story cue status to the CCU.

- Set a separate timing clock in the CCU.

- Reorder shots as the producer reorders stories. Re-position the cameras accordingly.

- Allow the CCU operator to take to air at the appropriate time for each story.

- Adjust the camera position if the anchor moves.

- Adjust camera position for over the shoulder video or still store.

Still Store Interface

The Still Store Interface operates much like the other interfaces discussed above, except that still store numbers are identified in advance of the newscast and fed into the rundown in the newsroom system. The system must send the correct still store number(s) to go with each story at the right time. The still store operator or the newsroom system then takes the image to air at the correct time.

Closed Captioning Interface/Benefits of Automatic Electronic Captioning

Captioning of local newscasts has always been of great interest to community service minded broadcasters. But in the past, the only way to encaption was by using a rather costly stenographic service offered by the National Captioning Institute (NCI). This method has not been widely accepted because of its high cost.

With the NCI service, a stenographer using a machine similar to those used in courtrooms watches the newscast and types what the anchor says. A computer system decodes the keystrokes and feeds the data into the encaptioning device. This unit puts the information in the line 21 interval for mixing with the broadcast signal.

When a second generation newsroom system with an electronic teleprompter is installed, captioning can be done as an easy and automatic byproduct of newscast automation. The information sent by the system to the the encoder consists of the scripts sent to the teleprompter, plus summaries or exact scripts from video pieces. The summaries are prepared by the writers earlier in the day. Live interviews in the newscast are not captioned.

The newsroom system interface to the EEG Encaptioning Encoder must perform or allow for the following functions:

- Transmit the scripts to the encoder with director's cues and other internal information deleted.

- Allow for information in the script to be coded to prevent transmission if desired.

- Start the encaptioned transmission of each new story when the anchor starts to read that story.

- Display the anchor's words at the exact time the anchor reads them.

- Spell check the scripts before the newscast to minimize misspellings.

- Include summaries or exact scripts of video pieces used in the newscast.

- In those cases were a piece has no captioning, transmit a message so indicating.

The benefits of automated captioning over stenographic captioning are significant:

(a) The incremental cost for electronic captioning is a modest one time additional charge to the newsroom system purchase price versus an annual cost of up to $50,000 per newscast for stenographic typing.

(b) The transmitted text represents the exact and complete words spoken by the anchors. The stenographic method represents a form of censorship since the typist must interpret what the anchor is saying and often paraphrases or omits information.

(c) The electronically captioned information is displayed on the TV screen almost exactly as the anchor says the words. With the stenographic method, transmission is usually one or two sentences behind.

The only disadvantages of electronic captioning are:

(a) The news staff must enter scripts or summaries for video pieces into the system.

(b) No captioning is available for live interviews.

THE STANDARDS ISSUE

Newsroom system automation and second generation newsroom systems will bring the most benefit to the most users at the lowest overall expense if vendors cooperate and standardize interfaces. If the newsroom system vendor has to choose which machine to interface to because standards do not exist, he will choose the one that is most popular in the marketplace.

It is probably too much to ask for the vendors of each product line to cooperate. The unfortunate conclusion, therefore, is that the buyer must pay very careful attention to which devices he expects to interface to his newsroom system.

THE BENEFITS OF NEWSROOM AUTOMATION

Second generation newsroom automation systems are providing significant benefits to broadcasters today. Such benefits include:

(a) The newsroom is much more efficient before the broadcast because of the automated wire reading and filing, word processing, and script writing.

(b) The staff can produce an up to the minute newscast, changing and adding stories right through the show.

(c) Breaking stories can be easily added to the newscast during the show.

(d) The newscast is produced with fewer mistakes. The right video, right super and right still store are displayed at the right time. The right camera shot is used.

(e) Better use is made of the staff. Machine operators are freed from repetitive tasks to work on more productive assignments.

(f) Staff morale is greatly enhanced.

(g) Overall newsroom efficiency results in better newscasts with the same or fewer people.

CONCLUSION

Although the surface of newsroom automation has barely been scratched, radical changes are taking place today in newsrooms everywhere. News Directors, faced with stiffer competition, are aggressively looking for ways to "Improve the Broadcast." Broadcast executives, faced with today's financial realities, are more and more inclined to replace labor intensive, inefficient methods with currently available improved techniques. Newsroom automation is becoming an increasingly accepted strategy for improving both areas.

IMPLEMENTATION OF SURROUND SOUND FOR TELEVISION

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ABSTRACT

Surround sound has been used for several years to enhance the realism of stereophonic motion picture sound by supplementing the left and right channels with a center monophonic signal, rear localization, "surround" ambience and a sub-woofer bass signal. It can also add these enhancements to television stereo sound. The icing on the cake for the broadcaster is that no additional audio signal paths are required.

The successful network transmission and local broadcast of surround sound requires care beyond that needed for stereo. However, many of the factors that contribute to high quality mono-compatible stereo also preserve surround sound.

What is Surround Sound?

Since the 1970's motion picture theaters have offered surround sound to enhance the realism of stereophonic film sound. In case this sounds suspiciously like quadraphonic sound, that notable flop of the early 1970's, the two are not the same. Quadraphonic sound was an attempt to spatially reproduce music in a full circle around a centered listener using four audio signals typically arranged as left front, right front, left rear, and right rear.

Surround sound also uses four audio signals, but arranged in such a way that the sound track of a video presentation is augmented by the presence not only of left and right stereo information, but also a "hard" center monophonic signal, the provision of both rear-located and general ambience information by use of a "surround" channel, and the reinforcement of bass information with a sub-woofer.

The addition of center and surround signals and enhanced bass contribute materially to the sonic realism of a movie sound track. Early stereophonic experiments often used more than the two audio signals that are conventional in stereo today. Indeed it may be argued that economy and convienence are the parents of the two-channel stereo convention, supported by the fact that music by itself does not require the accuracy in spatial location that can be afforded by more than two audio signals.

Stereophonic presentations in movie theaters, however, require a center speaker fed by mono or centered components of the sound track, rather than reliance on a "phantom" center image created by equal-level signals in left and right channel speakers. This center speaker is necessary to ensure accurate sonic localization for those not sitting near the center of the screen. Without the sonic image stabilization afforded by the center speaker, a person sitting off-center will incorrectly locate sound relative to the picture.

The large number of centered sounds in a movie sound track, including most dialog and many sound effects, plus the interrelationship of picture and sound make accurate location of front-stage sounds much more important in movies than in music listening.

Stereo television is another case of picture and related stereophonic sound, and the value of a center speaker applies here as well. This was borne out by some informal experiments conducted at NBC several years ago, which led to the conclusion that while two channels are good, three channels are better, particularly in the case of stereo music and sound effects accompanied by centered dialog.

The addition of rear location and general ambience provided by the surround speakers, plus the "larger than life" bass produced by a sub-woofer place the viewer in the middle of the sonic action, and significantly enhance the enjoyment of both movies and television. For the television viewer, it is a means of bringing the theater experience into the home.

Implementation

Seventy millimeter film records the four signals required for surround sound on four discrete magnetic tracks. Thirty-five millimeter film, however, is limited to two optical tracks, and a little ingenuity is required. The four signals are matrixed into two channels, which are subsequently decoded into the original four signals upon playback. This technique is called 4-2-4 matrixing. Because this process yields both stereo-compatible and mono-compatible results, it is well-suited to BTSC television transmission.

A description of the encode/decode process best begins with the decode matrix. The four audio signals consist of left, right, center, and surround. The decoded center channel is derived from the sum of left and right, while the decoded surround channel is derived from left-minus- right. Figure 1 is a diagrammatic representation of the four decoded channels. While the surround channel is represented as a single speaker, in practice this signal is diffused among a number of speakers arranged in a "U" shape around the rear of a theater. In home decoders, surround information is typically fed to two speakers after processing is performed to electronically "diffuse" the sound.

In the following description of the encode and decode process, simple letters refer to signals prior to encoding, e.g., "L" is the original left channel signal. A letter with a "prime" symbol designates a signal after decoding, thus "L'" is the decoded left channel. The symbols L_T and R_T designate the matrixed left and right channels as they appear on the film sound track or are transmitted. The decode matrix is expressed by the following set of equations:

$$L' = L_T$$
$$R' = R_T$$
$$C' = 0.707(L_T+R_T)$$
$$S' = 0.707(L_T-R_T)$$

In the 4-2-4 matrixing process, total isolation between signals is impossible and signals intended for a particular speaker are leaked into other channels. In the decode matrix, each signal has infinite isolation from only one other signal, and just 3dB isolation from the other two signals. Figure 1 shows the inherent leakage of the decode matrix, which provides infinite isolation between center and surround and between left and right, but only 3dB isolation between left or right and center and between left and right and surround.

The majority of the matrix's direction-encoding space is devoted to the front sound stage. The full separation of L' and R' enables the creation of a wide sound field, narrowed only about 25% from the full left and right speaker spacing by the presence of the center speaker, which is isolated by only 3dB from either L' or R'. Full isolation between C' and S' keeps centered dialog out of the surround speakers.

The decoded surround signal is delayed by 30 to 100 milliseconds, depending upon the size and geometry of the listening space, taking advantage of the precedence effect to solidly locate non-surround-directed sounds at the front. The surround signal is also low-pass filtered at 7kHz and subjected to level-dependent downward treble shelving to prevent dialog sibilance from bleeding into the surround channel.

The basic surround decode matrix provides complete center-to-surround isolation, symmetrical interior panning, a wide front stage, and appropriate ambience extraction, but by itself is not capable of stable image localizations over a wide listening area because of the lack of isolation between certain signals.

Surround decoders typically employ cross-coupling and leakage cancellation methods that, in effect, dynamically vary the decode matrix to cancel unwanted components of a directional signal. This is called directional enhancement. In this way the decoder overcomes the inherent lack of separation between certain signals.

Surround encoding equations are not quite complementary to decoding equations because of the need to include appropriate phase shifts to enable creation of "interior" sounds that appear equally in all four channels. The encoding equations are:

$$L_T = L+0.707(C-j_S)$$
$$R_T = R+0.707(C+j_S)$$

The -j coefficient causes S to be matrixed into L with a 90^0 phase lag relative to front signals and the +j causes S to be matrixed into R with a 90^0 phase lead relative to front signals. Thus C is mixed into L_T and R_T at a -3dB level, and S is mixed into L_T and R_T in opposite polarity (180^0 out-of-phase) at a -3dB level. This in-phase and out-of-phase encoding could be accomplished without the j coeffients, but the encoding of "interior" sounds would be prevented. For example, if identical signals without the j coefficient were fed into the C and S inputs of an encoder, they would subtract to 0 in L_T, effectively producing R-only encoding. With the j coefficents, the same encoding results in equal levels in L_T and R_T and a 90^0 relative phase shift between them, producing equal levels in L', R', C', and S'. Complete encode/decode equations are:

$$L' = L+0.707(C-j_S)$$
$$R' = R+0.707(C+j_S)$$
$$C' = C+0.707(L+R)$$
$$S' = -j_S+0.707(L-R)$$

Broadcasting Surround Sound

The 4-2-4 matrix surround technique may be readily applied to stereo television broadcasting. The L_T and R_T signals may be networked and broadcast in the same manner as conventional stereo. However, because the matrix encoding technique depends on manipulation of the amplitude and phase characteristics of the left and right channels, the encoded signals will suffer a certain amount of degradation in the transmission process, just as any stereo signals will. Because the left and right channels are matrixed into sum and difference signals, and the difference signal is further subjected to noise reduction companding, the transmission process is sending L+R and L-R over different paths creating the opportunity for their gain and phase relationships to be compromised. Impairments contributing to such compromise in the broadcast process range from the television transmitter, to multipath, to degradations introduced by the various components of cable television systems, to the receiver itself. For a comprehensive and informative treatment of such transmission impairments, I refer you to "Effects of Receiver Design and Transmission Impairments on Audio Signal Quality in the BTSC system for Multichannel Television Sound", by J.James Gibson, in the September 1986, issue of the Journal of the Audio Engineering Society.[1]

The networking process is carried out in the discrete left and right channel domain, and gain and phase relationship remain relatively intact. The broadcasting process, however, is carried out in the sum-and-difference domain, and here is where the potential trouble begins. A typical television stereo transmission system will permit the transmission of about 35dB of separation throught the mid band.

The typical stereo television receiver is capable of delivering about 25dB of mid-band separation. Transmission impairments can reduce separation, and it is not uncommon to see the ultimate separation delivered to the TV set's speakers reduced to around 20dB. This separation compromise is caused by the corruption of the gain and phase relationships between sum and difference signals. Figure 2 illustrates the effect of gain and phase errors between sum and difference channels on stereo separation. To provide the requisite 20dB of separation requires operation in the 1dB -10 degree region. These gain and phase errors between decoded stereo channels will of course have a negative effect on surround sound, because they will affect the decoded gain and phase relationships between L_T and R_T.

The result of these errors will be reduced separation between L', R', C', and S'; and errors in spatial location. The question to be answered is whether the degradation is sufficient to destroy the effect of surround encoding.

NBC has in the past networked and broadcast several theatrical films with surround sound encoding. In 1987, the Network aired a one-hour Amazing Stories program which was the first made-for-television program to be surround sound encoded. No formal testing was done in these instances, but those with surround sound decoders discovered that subjectively the effects survived the transmission process admirably.

NBC plans to make surround sound broadcasting a regular part of its network programming. We will also embark upon a plan to test the retreivability of surround sound under a variety of circumstances including the Skypath™ satellite distribution system, transmission on both VHF and UHF stations, and via cable systems. The purpose of these tests is to determine just how much degradation of surround sound occurs under the various conditions attendent to the television networking and broadcasting process.

We do know that since surround sound encoding is a process that involves altering the amplitude and phase parameters of the stereo signals, those factors that compromise the amplitude and phase relationships between left and right or sum and difference will also compromise surround sound. Thus all the precautions we take in the distribution and transmission process to protect the amplitude and phase relationships between left and right and sum and difference to insure mono compatibility are important to the preservation of surround sound. In short, if careful attention is paid to delivering good stereo and good mono, surround encoding will be transparently passed along to the viewer.

The use of stereo synthesizers must be re-examined in the face of surround sound, however. Synthesizers add a left-minus-right component to all parts of the audio program, including dialog. This means that all audio programming, including dialog, will appear in the surround channel as well as in the front sound stage. Dialog then comes from all over the room, and to compound the problem, it is delayed in the surround channel. This is a very disconcerting effect, and definitely one to be avoided. Even if an automatic switching stereo synthesizer is operating perfectly, it will still synthesize mono program elements such as commercials.

The message here is that if surround sound is being broadcast, any stereo synthesizer in the audio path should be bypassed. Stereo synthesis and surround sound decoders are enemies!

Surround sound is a material addition to the realism of motion picture sound, and to television sound as well. It will become even more important as we move into wide-screen and higher definition video transmission. High definition, wide screen television demands high definition, wide screen sound. Over half a million consumer surround sound decoders have been sold since 1985 and this number will surely multiply. There is a trend among high-end television recievers to build surround decoders in. Surround sound is definitely the next step in television sound's evolution. The broadcaster who is well prepared for stereo is ready for surround sound.

Reference:

1. Gibson, J.J. Effects of Receiver Design and Transmission Impairments on Audio Signal Quality in the BTSC System for Multichannel Television Sound, Journal of the Audio Engineering Society, vol. 34, no.9, September 1986, 647-660.

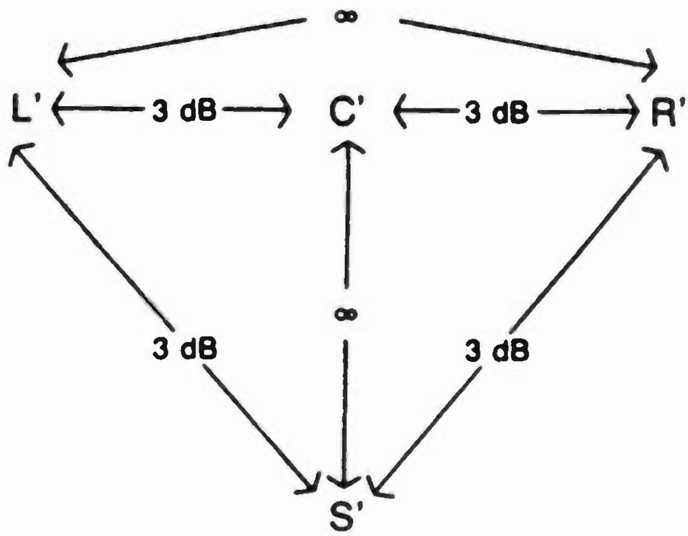

Figure 1.
SURROUND DECODE MATRIX

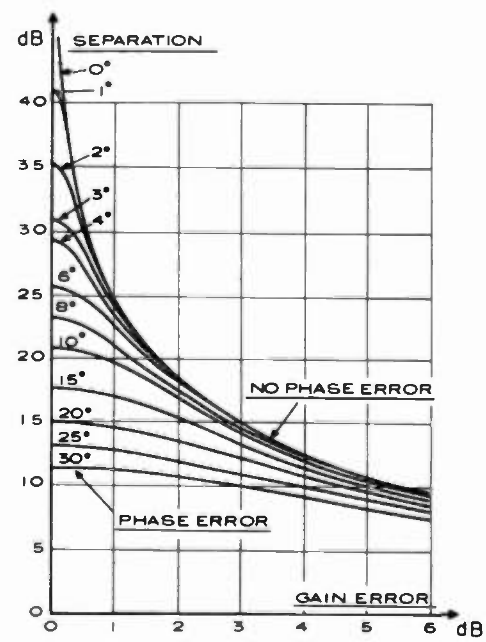

Figure 2.
STEREO SEPARATION AS A FUNCTION OF GAIN AND PHASE ERRORS IN L-R VERSUS L+ R PATHS.

EQUIPMENT SET-UP FOR STEREO BROADCAST ORIGINATIONS

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Audio for stereo video presentations is becoming more demanding as the quality of consumer MTS equipment increases. There is now the need for broadcast facilities to begin their own stereo production of local and regional events.

There are many types of programming that would benefit from a stereo mix, but many broadcasters would like some guidelines on how to prepare for the most popular types of stereo production.

Field Microphone Technique

Micing for stereo in the field obviously depends upon what the origination of the sound source is. Typically, news gathering usually involves the reporter on either a hand mic or a lavalier. Wild sounds are typically captured in mono through that mic along with the reporter's voice. At major events, it can make a big difference to have these important background sounds come home in stereo. They can add tremendous production value and added realism to the shoot.

Too complex of a set-up, however, would actually hinder the effectiveness and spontaneity of a live event.

This is why most stereo ENG shoots would benefit by adding a stereo mic mounted to the camera and used in conjunction with the normal reporter's mic. Since EFP shoots are more interested in creating a produced "look" and sound, mounting the mic on a fishpole gives some added flexibility, but requires an additional person.

In ENG, a news reporter's mic would be panned in the center, and the stereo background sounds would be picked-up by the camera mounted stereo mic. This mic would appear on its own two channels on the mixer and would be assigned full left and right.

The level of this mic, depending on the substance of the background sound effects, should normally be at least 10dB down from the reporter's voice. Remember that when this mix is heard in mono that the background sounds will sum together and appear louder than they do in stereo.

A small, battery operated portable stereo audio mixer is obviously necessary to do what has just been suggested. This writer would recommend one that is capable of 48 volt phantom powering for condenser mics and has a built-in tone oscillator. Provisions for a simple talk-back mic for slating tapes in the field and a built-in limiter/compressor is also a nice touch. Some portable mixers also have a low-cut filter on the microphone input. This is a very handy device and should be used to attenuate low frequencies from air conditioning rumble in buildings, hand held mics that are worked too closely—exhibiting too much proximity effect, and most outdoor situations where the slightest breeze can create enormous rumbles that cannot always be heard to their full intensity in the field through a pair of headphones.

It is absolutely necessary to use a light-weight, quality pair of headphones to monitor the source material. These phones should have ear muffs that help to seal out outside sounds and the console should be able to deliver a respectable output level to the phones. It's also a very good idea to use the same pair of headphones everytime to develop a reference standard you can become comfortable with. If your facility is large and there are many crews going out, standardize on one brand of headphones and equip each outfit with it.

Many of the best stereo mics are condensers of the MS type with an appropriate matrix box to allow adjustment between the elements. A stereo MS mic is one of the best ways to provide a compatible signal in mono. This is because there is always a mic facing foward in the design, giving a true mono signal. This mono signal can later be fully recovered if necessary in post production.

Altering the Image Size in Post Production

When field recording an MS signal onto the tape, it can be processed in post production through a line level matrix decoder, but only if the MS stereo mic is being used alone without other audio information mixed along with it.

The stereo image size can be adjusted if necessary, by rotating the control on the matrix box from MS to XY. This can help stereo audio width to better match the visuals. Adjusting the matrix changes the aspect of the listeners' position. More towards the MS side causes a "zoom" effect into center channel information, more towards the XY side creates additional stereo width. The added flexibility by being able to do this in post takes the pressure off the engineer to make decisions in the field that may not be the best once the tapes are viewed in the edit suite.

It is important for the MS signal to be recorded on a machine having perfect audio head azimuth alignment. Even a 2° misalignment can cause unusual matrix decoding errors, if an MS signal is recorded to be decoded in post. If this is a concern, it is advisable to simply dial-up the XY signal on the matrix box, record it that way in the field and avoid decoding in post.

Live Music Recording for Television

A goal to be set when attempting to record a good stereo signal is to create a wide image with good presence.

It is becoming more commonplace for local or regional broadcasters to become involved in coverage of special musical events. These events can range from the live taping of a symphonic production, to a well known pop or rock band. Importantly, a factor that will cause major differences in the way a performance will be miced is whether the performance will be amplified through a high powered PA system. In general, it is easier to handle music at the studio where all the available equipment can be at your fingertips.

Acoustical Performances

A desirable situation is when the performance is held within a hall having reasonable acoustics. In this case, a small number of quality microphones should do nicely—assuming there are no electronic instruments that are amplified into the building. This may cause too much leakage into your main microphones to achieve the sound you desire.

Let's assume it's a classical music session performing in a known hall with good acoustics. There are many ways to approach this task, but one way is to use a pair of mics in an XY configuration (the elements nearly touching and articulated near 90°), hung overhead near the front of the performers. This is sometimes referred to as the "conductor's ears" approach. Condenser mics are desirable in this situation, and additional mics may be necessary, depending on the size of the orchestra. Extra mics are sometimes used for augmenting softer instruments.

Setting-up the Recording Space

Many times the console will be located in a truck, or the facility may have a room where a temporary mix position could be set-up. It is necessary to record any complex musical performance in a separate room with monitor speakers in order to be able to provide a useable mix. This room should be equipped with an intercom feed, any support equipment for the console, and a high quality stereo program limiting system to feed the videotape recorder.

Video Monitoring

A video monitor connected to the program feed or individual monitors of each camera is also important. If several monitors are not available, a small vertical interval switcher or a simple A-B switch on the video monitor can be utilized so that the most important video feeds can be viewed in the audio room. In most cases, the cameras are the audio engineer's only eyes into the recording environment and are essential during the rehearsal and live performance.

The Console

The console should have adequate monitoring. By adequate, this means that all of the various mixes need to have switching available for soloing. It has always been important in broadcasting to have PFL (Pre Fade Listen) on all channels, but when creating a stereo mix, a post-fade, positional type of channel solo is extremely helpful. This will help to determine where a single microphone is panned within the stereo image. (Note that a pan control is not a stereo balance control, since it is only positioning a single source between the two speakers).

Additionally, the console should have comparison switching for listening to the signal over small speakers, and for listening in mono, which is important to check for unusual phasing problems. Many times a signal can sound alright when listening in stereo, but when summed to mono a problem can become apparent. A phase meter can be optionally purchased and installed within the meter bridge on many consoles. If the meter is a moving coil device, the needle will swing towards positive when the signals are mostly in phase, or towards the negative side if there is a possible cancellation problem. LED meters work very well too, but purchase one with a reasonable number of LED segments.

A phase meter can and should also be mounted in the engineering area to monitor the audio output at all times. Tektronics has an optional retrofit for a common vectorscope to allow it to display audio phase.

Noise Reduction

A further precaution if the performance is recorded direct to Type C one inch, is to use a noise reduction system for the videotape recorder, unless the program will be sent out of house to another facility without a compatible system. A companding type of noise reduction such as dbx type 1, can add extra headroom to the videotape that has a tendency to saturate quickly on transients.

Setting Up the Console

Our XY pair of mics should be assigned fully left and right on the console. Other mics can be panned to the right or left, depending on their positioning during the actual performance or if a wider stereo image creates a desirable effect.

Normally, if quality mics are employed, only minor EQ adjustments should be necessary. As mentioned above in the section on field audio, a high pass/low cut filter is a tremendous asset. The filter can be brought in without the need to switch an equalizer into the circuit. However, certain rooms or microphones may cause peakyness at some frequencies. This can be corrected through the use of a parametric equalizer. Many consoles now incorporate this type of equalization as a standard feature. It is normally better to attenuate the offending frequency than to boost others around it. Boost EQ also increases the gain of noise in the building residing at that frequency.

During rehearsal, it is wise to spend a little time in the building to listen with your ears and then return to the console and compare.

Natural reverberation and audience response can be picked up by placing a pair of widely spaced mics over the seating area of the auditorium and mixing these in very carefully during the performance. Too much ambience micing can ruin a good recording, but if a hall has legendary acoustics the results, if done with taste, can heighten the outcome of the production.

In any case, micing the audience is mandatory in most situations. The audience is one of your best sources of stereo information and can help the viewer to become more involved during the telecast.

Live Music with Sound Reinforcement

The above micing technique outlined can have a disastrous outcome if the same group is amplified through a sound reinforcement system. Microphones tend to pick up the direct sound from the instruments, the sound from the speakers, and any reflected sound, (whether initially produced by the actual sound of an instrument or that produced by a loudspeaker).

Because of this, the classical micing techniques we have outlined normally tend to sound hollow or tubby in this situation. The only recourse is to use close micing technique.

Close micing requires a console with a much larger mainframe size to accommodate the added microphone compliment. Some of the stereo separation may suffer if not carefully planned out when assigning channels on the console.

Direct Inputs

One cure for isolation problems is to use a direct feed from any electric instruments wherever possible. A good direct box is of the active type and provides unity gain while balancing the signal coming from the musical instrument. A ground lift switch is a common feature on a good direct box, and can provide a quick fix to a stubborn ground loop hum. Taking a direct feed from keyboards, guitars, electronic drums, etc., immediately give you an isolated signal from those instruments without leakage.

Acoustical Instruments

Most acoustic instruments fare pretty well when closed miced. But there will be some problem with instruments that play in a group and benefit from an "air" mix—like a string section for example. Without the close micing of violins in an amplified environment, there is little chance of hearing them at all. Even sectional mics will pick up the surrounding instruments as much as the strings. Several good lavalier mics carefully placed on the violins will work rather well, but good EQ'ing will be necessary.

Group Assignments

When assigning a console having eight sub-groups, they should be created in stereo pairs. Groups one and two—left, right; groups three and four—left, right, and so on.

This writer has become accustomed to setting up a live instrumental mix in the fashion shown in the example below:

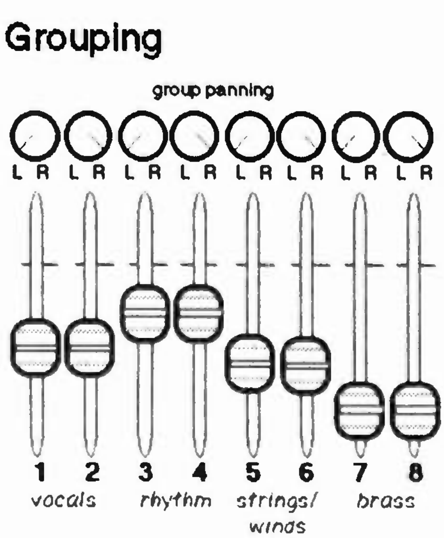

fig. 1

All Vocals = Groups one and two, L and R
This would consist of solos, choirs, back-up vocals, etc.

Rhythm = Groups three and four, L and R
A usual rhythm bed includes piano, keyboards, (synthesizers, samplers, etc.), drums, bass, guitar(s). tympani, glock, tubular chimes, etc.

Strings and Woods = Groups five and six, L and R
This grouping can include: violins, violas, cellos, basses, harp, oboe, bassoon, flutes

Brass = Groups seven and eight, L and R
Trumpets, trombones, French horns, tuba

Each of the above groupings utilizes instruments that play in a similar dynamic range. If the console has assignments for direct to stereo outputs, talk or announce mics can be directly assigned to the main outputs without disturbing the vocal sub groups.

Setting Up in the Studio

In most television studios, the cyclorama will wrap around at least three of the wall surfaces. Many times television studios will have a large panel with standard XLR mic connectors for the audio engineer to use. This requires a large bundle of wires to be pulled under the cyc curtain (unless a hard cyc is used), and can create a tangled mess of cables. Micing for stereo doesn't necessarily mean that twice the number of mics will be used, but it will no doubt mean that more lines will have to be run than before.

It is important to keep the sets clean of audio apparatus—except in those applications where the equipment itself adds a hi-tek look to the appearance of the set—like a rock band, for example. In most cases, though, it is wise to plan for as clean a look as possible, dressing all cables and mic stands with an occasional look through the viewfinder of a studio camera.

One way to accomplish a clean look is to either change the mic termination boxes in the studio to a multipin design, or add multipin connectors to the existing box, or boxes. It is desirable to have a box on each wall of the studio where a set could conceivably be placed. Each one of these boxes should be equipped with a multipin connector that mates with an appropriate mic snake. The snake can be pulled to the most appropriate place behind the set so that all of the mic cables can be plugged in without the need for extensions.

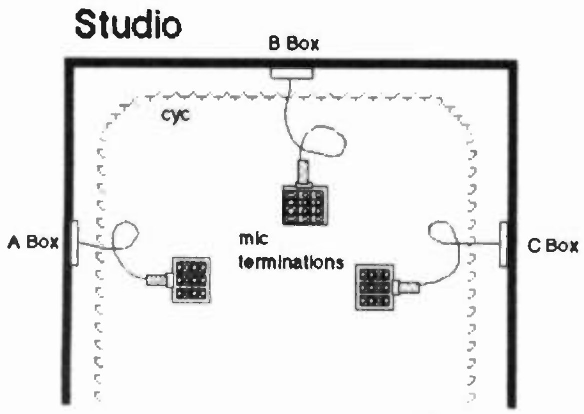

fig. 2

In the event of a hard cyc, a removable access door at the floor can be made just large enough to plug in the connector and pull the cable through. These covers could be snapped back into place after a production. Individual mic cables should be numbered at both ends in order to aid in quick identification if a change needs to be quickly made.

Each box can have the maximum number of inputs available on the console if patching or switching is provided. This prevents having to pull additional lines from another box on a different wall. If more than one set is used at a time, assignments can be made in the control room to determine which inputs should be used on each box. With this set-up, several sets can remain intact in the studio at the same time without time consuming audio re-cabling and re-patching.

Foldback in the Studio

Good foldback in the studio is important for the talent and for necessary communication from the audio engineer during set-ups and rehearsals as well as cues from the video director.

The system should have its own dedicated amplifiers and room equalizers, and be fed from an appropriate buss on the audio console. If several sets or areas of the studio are active during a live production, each area may require its own foldback mix, hence a completely independent amplifier and equalization network. Each system needs to be under control of the audio engineer to avoid deterioration of the TV mix due to overly loud monitors blaring somewhere in the studio.

Foldback Speaker Placement

Placement of the foldback speaker systems is important. It is important to always attempt to place a monitor speaker as close as possible to the optimum 180° off-axis of the primary mics on the set. Assuming that these mics are directional, (cardioid, hyper cardioid, or shotgun polar patterns), this accomplishes two objectives: 1) to keep leakage at a minimum, and 2) to avoid feedback at all costs.

Many times, a floor slope is used when a soloist is performing with a hand mic. Again, this is to observe the 180° technique. In comparison, when micing stereo audience response, it would be better to hang the monitor speakers since the response mics should be flown overhead and aimed downwards.

Split Mixes

If live music mixes become one of the mainstays of the studio, a separate monitor mix console and engineer may become mandatory. The use of two or more consoles will require a microphone splitter system, the most common ones being those that employ high quality transformers.

When using a mic splitter, the recording console in most cases should be assigned the direct feed, while the PA and/or monitor mix consoles should take one of the splits. The console that takes up the direct feed will also be responsible for supplying phantom power to the mics, since DC voltage will not pass through a transformer.

A good splitting system will utilize multipin connectors

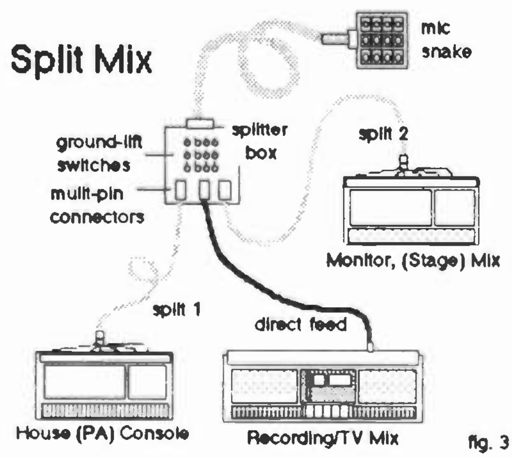

fig. 3

that feed multi-pair trunk lines to each of the consoles. At that point, if the console utilizes standard XLR mic connectors, a fan-out can be used to tie the trunk line into the board. In other cases, the console may have its own type of multipin connectors that the trunk can interface with. It is also advisable to have ground-lift switches fitted to each input of the splitter chassis.

Patching

The previously mentioned wiring scenario assumes that the console utilizes a built-in patch bay, which has become more prevalent in newer boards. Built-in patching is desirable because the shorter runs of mic level cables are also shielded by the console's mainframe and less likely to suffer from additional hum induced through electrical radiation. It is also better for channel insertion points, direct outputs, and so on to be on the shortest route as possible and be protected in the same way.

Audio Processing

In the case where multi-micing is necessary because of high PA levels in the building, digital reverberation and effects will be necessary. This is to enable the amount of reverberation to be controlled instead of depending on the room acoustics which will be unusable for the most part because of the PA system. The use of audience response mics will also have to be used with care and in most cases these mics will have to be brought in and out as necessary but probably never taken completely out of the mix.

When the master will end up on videotape, it is important to process the audio so that tape saturation is unlikely. It is also important that this limiting is done with some finesse so that all of the difficult work of doing a good mix won't be distroyed by an over active limiting system. It is also good to be aware of the fact that many types of limiting systems are employed at broadcast facilities to prevent peaks from slipping through the transmitter. Some of these systems are more sophisticated than others, so it is good to assume that the signal you have recorded will survive someone else's dubious limiting system without jeopardizing the quality of your product for syndication.

Fortunately, there are a few good systems available as a master processor. Even these can benefit from the use of other limiters during the production. Using a single limiting system for the entire signal to pass through can cause problems. A limiter will attenuate sound at the level its threshold is adjusted to. If this is the only limiter in circuit, it will bring down the entire mix to capture the one offending sound. If the instrument that is providing the transients happens to be a kick drum, then each time it is played the limiter will bring down the level on all the other instruments in the mix. The same applies to any other sound that effects the sidechain of the limiting system. This can be annoying when a vocalist is singing with live accompaniment and has a tendancy to work the mic too closely. The entire mix will tend to pump each time this happens.

To avoid this problem, an independent limiter should be used on each offending source. To avoid having to use a great number of devices, inserting a limiter through a sub group can many times handle several instruments at once without damaging the overall mix. In the group assignments above, groups one and two are assigned to all vocals. There may be several solo mics assigned to this group, but they may not all be used at the same time. Inserting a stereo limiter, (or two mono limiters operating with their sidechains strapped), into those groups will handle the problem without having to use individual limiters for each microphone. It is important when using two mono limiters for a stereo application to slave them together. Many mono limiters are equipped with this option.

Conclusion

This paper has outlined the basic set-up and preparation that takes place when recording stereo audio material for broadcast. There are many situations unable to be covered in a brief report, but hopefully this production oriented outline has helped to answer some basic questions on stereo technique.

Acknowledgements

The author would like to thank Jerry Graham from **Gotham Audio** for his advice on MS technique, and Jerry O. Horstmann from **Media One Productions** for the use of their facilities in assembling the videotape examples that accompany this paper. ■

TIME DELAY ERROR IN VIDEO AND FILM AUDIO

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Abstract

Inter-channel time delay errors (time skew) can occur whenever an analog stereo recording is created or played back. Such errors cause rotation of the stereo image and loss of mono compatibility due to the comb filter effect.

Phase encoded multi-channel audio, such as Surround Sound, is particularly sensitive to small time delay errors. The accuracy of the decoding process relies on correct phase information; any errors lead to leakage of center channel information to the surround (also known as "spitting").

Unfortunately, the broadcaster is not always in control of his source material, which itself may contain the time delay errors; it is not practical to attempt to adjust head azimuth to compensate for source material errors, and, in the case of digital recordings, (such as laserdiscs), there is no possibility of recalibration.

Over the years, there has been increasing awareness of such problems. Now, the Model 2300A Phase Chaser provides a solution to time delay and phase problems.

I. Introduction

The most familiar effects caused by inter-channel time delay errors are loss of mono compatibility and poor stereo imaging (1,2).

A pure time delay between the channels is equivalent to a frequency dependent phase shift for related signals. When summed into mono, this produces the familiar comb filter response, which is the loss of high frequencies due to the partial or complete cancellation of the correlated frequency components (i.e., the monophonic content) in the original stereo signal.

In a stereo environment, the perceptual effect of a time delay is an apparent rotation of the stereo image, without any frequency response degradation. Unless the time delay is quite large (greater than 100 microseconds) the imaging error is usually not apparent to the casual listener.

Recently, we have found that inter-channel phase shifts can also seriously degrade the performance of multi-channel sound systems which employ phase encoding (Dolby™ Surround Sound and the like). The frequency dependent nature of the spurious phase shift causes the decoder systems to route some parts of the audio spectrum to the wrong speakers.

II. The Origins of Time Delay Errors

Time delays are most often caused by mis-alignment of magnetic or optical transducers, such as record/playback heads in standard tape recorders or the solar cells in movie projectors. Regular calibration will minimize this kind of error.

Gap scatter in multiple pole head stacks is another source of time skew, but because the alignment error is built into the head assembly, it is generally not correctable using standard calibration procedures. (12)

If the same machine is used for both recording and playback, a fixed alignment error will tend to cancel out. However, this is usually not the case, so even a regimen of regular calibration will not be able to correct errors incorporated in the program material during recording. A playback machine could be re-adjusted to compensate for a particular tape (12), but this is not usually a workable method.

Other sources of time delay include the application of different audio processing to each of the stereo channels, and unequal paths encountered during transmission of discrete stereo signals. A recent example of an audio processing problem is the 11.34 microsecond inter-channel delay error occuring in CD players that have a single multiplexed D/A converter.

With the introduction of optical digital media, such as compact discs and laser video discs, there are no more heads to align. Any time skew which may have been present in the original master recordings are faithfully incorporated into the optical disc, and cannot be corrected simply by adjusting a screw.

Still another type of problem occurs in audio-for-video applications. It is not uncommon for a constant error to be present on the first reel of a film, another on the second reel, and so on; this problem has been observed in broadcast films, video cassette recordings, and consumer laserdiscs.

In similar fashion, it is possible for a single scene in a movie to contain multiple time delay errors, each due to an alignment error present in the particular source machine used in creating the multitrack master. For example, the dialog may be accurately recorded, while a featured piece of music may contain a large time delay error. In a situation like this, only "tweaking on the fly" would work.

III. Monaural Effects

The most well known problem associated with time delay error is the loss of mono compatibility (3). The amount of high frequency loss is directly proportional to the time delay, and has a characteristic comb filter contour. For a fixed delay, the loss will be constant, and the mono listener will usually attribute the poor sound to inferior program material, rather than to a technical problem in the equipment or the recording. However, with the proliferation of high quality audio on compact discs, audio consumers are developing more sophisticated taste, and are sure to be displeased when radio broadcasts and high quality TV music videos have comb-filtered audio.

In the case of fringe area radio reception, the problem is more obvious. A typical receiver will toggle between stereo and mono modes as a function of signal strength. A nominal time delay error may only shift the stereo image a little, but will certainly impair the sound quality when the receiver switches to mono. The sudden degradation of sound quality that occurs in mono will often cause the listener to change stations; if it happens on a regular basis, the offending station will no doubt lose a customer.

Poor audio quality due to time skew is by no means limited to broadcast audio. Whenver a recording originally produced in stereo is rendered in mono, loss of quality may result. In particular, this can affect the audio on video cassettes, which are often played on monaural machines. Similarly, problems occur when stereo movie sound tracks are re-recorded in mono for distribution to monaural movie theaters.

Regardless of the actual cause, however, inferior monaural sound quality in broadcast, cinema, or consumer recordings will eventually have a negative impact on revenue.

IV. Multi-Channel Sound

Time delay errors cause the rotation of the perceived stereo image. Because there is no actual mono sum, there is no attendant loss of high frequency response. For time delays less than 100 microseconds, the image shift is quite subtle and will often pass unnoticed by all but the most critical listener. Therefore, for strictly stereo listening environments, the problems caused by time delays are relatively minor (4).

By far the most interesting problems occur during the reproduction of any phase matrix encoded multi-channel sound, like Dolby™ Surround Sound (5).

Four audio inputs are matrix encoded into two channels, designated Left Total and Right Total. The left and right channels are recorded on LT and RT, respectively. Center channel information is recorded on both LT and RT. Material intended for the surround channel is split into two equal signals which are then differentially phase shifted by a cascaded series of allpass filters. All frequencies in the audio band are phase shifted by precisely 90 degrees: one channel is shifted $+90^{\circ}$, the other channel, -90°. These "out-of-phase" signals are then recorded separately on LT and RT (6). Therefore, when the final phase matrixed material is mixed into mono, there is complete mono compatibility for the Left, Right, and Center signals, although all pure surround information exactly cancels out.

During the decoding process, the LT and RT signals are subtracted, which leaves the surround signal neatly left over. With the addition of some clever processing for directional enhancement and noise reduction, the decoder box can produce four high quality outputs from the two input signals (6,7). Moreover, intermediate sound locations can be achieved by using proper mixing during the phase encoding process. It is the phase <u>and</u> amplitude relationships between signals on LT and RT which determine where the audio information is routed by the matrix decoder.

Because only two discrete channels are required for the phase matrix encoded material, full Surround Sound may be provided on all standard two channel media, including stereo radio and television transmission (9). As with all companding and matrix processes, the accuracy of the decoder is directly limited by the fidelity of its inputs. In the case of phase matrix encoded material, phase fidelity is the critical parameter: any phase errors will lead to sound localization errors (8).

If a spurious time delay is introduced between the LT and RT channels, a frequency dependent phase shift will exist between correlated signal components. In addition to the expected degradation of mono compatibility, signals experiencing a sufficient phase shift will be routed to the wrong output channel (typically a front to back reversal), with the problems being manifested first in the higher frequencies (see Figure 1).

Typically, these artifacts are most obvious in sections containing center channel dialog. A fixed time delay produces increasing phase error with frequency, so it follows that the higher frequency components of the dialog (sibilence and consonants) come out of the surround speakers, while the rest of the voices stay up front. For larger time delays, this effect, known as "spitting," becomes quite pronounced. And, with an extreme amount of delay, the spectrum at the surround speaker will have a comb filter shape! (See figure 1)

Another problem area concerns sound effects containing significant high frequency components. It can be disconcerting to see a jet plane land in front of you, and yet have the whine of the turbine behind you.

The necessity to eliminate front-to-back crosstalk, particularly with respect to dialog, has been generally recognized (6,7), and now there is a cure.

V. The Solution

The ubiquitous nature of time skew requires a solution that works anywhere in the production-playback-broadcast chain. Because the precise nature of the time delay error depends both on the program material and the hardware, only an automatic correction system can adequately address all the possible error conditions.

An elegant solution to the dilemma is provided by the Howe Model 2300A Phase Chaser; it can detect and correct time skew in real time without any prior preparation or encoding of the source material. The operation of the Phase Chaser relies on two facts: first, that stereo audio contains a significant monaural component, and second, that the apparent stereo image is centered between the speakers (9, 10).

For a centered stereo image, there will be a net zero time delay between the correlated (i.e., monophonic) information on the channels. Any displacement in time between the left and right mono components can therefore be attributed to an unintentional time delay. Conversely, the width of the perceived stereo image is related to the amount of dissimilar material between the channels. As the degree of dissimilarity increases, a point will be reached where the image will no longer fuse, and will instead appear as a pair of mono sound sources.

In the Phase Chaser, a cross-correlation technique is used to measure the time skew between the monophonic components of the two channels; this information drives a negative feedback servo loop operating in the time domain (Figure 2). By applying differential time delay correction to the audio channels, the servo loop operates to hold the average phase error at the outputs at zero (2).

If such a servo loop is allowed to operate unchecked, it will cheerfully correct the very phase differences which characterize a stereo signal. In general, any system which just corrects short term phase errors will tend to compromise the stereo image. Two special circuits (patent pending) have been included to prevent such errors of commission.

The Window of Zero Correction allows the Phase Chaser to discriminate between incidental stereo phase fluctuations and systematic phase errors caused by fixed time delays. The unit will rapidly react to changes in time delay error while ignoring short term stereo phase variations. Therefore, the device will respond appropriately to changing error conditions which typically occur at splices and scene changes, yet will not collapse the stereo image into mono.

In Surround Sound encoded audio, intentional phase differences are built in so the decoding matrix can assign the signals to the proper channels. During passages which contain primarily in-phase information (left, right, and center), the Phase Chaser behaves normally, applying the appropriate amount of time delay correction to keep the average time delay error zero.

However, when out-of-phase information predominates, another proprietary circuit switches the unit into HOLD mode: the most recent time delay correction remains in effect until the level of the surround information subsides, at which point the unit reverts to the normal ERROR TRACKING mode. If it were not for this circuit, surround information would be routinely shifted to the front. As an added benefit, the correction being applied at the time the unit shifts into HOLD also serves to keep the surround information from leaking into the center channel.

VI. Conclusions

Inter-channel time skew is responsible for a number of troublesome problems throughout the recording and broadcast industries. With the introduction of the Model 2300A Phase Chaser, real time correction capability is available in a convenient package.

The device has application both as an error monitoring tool (11), as well as an on-air insurance policy. Because the device relies on information contained in the program material itself, it can be used reliably at any stage in the production and broadcast process. It is not intended to be used as a substitute for good engineering practice, but in situations where time constraints or technical limitations prevent correcting time delay errors at the source, the Phase Chaser can be an invaluable tool.

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Figure 1. Surround Sound Leakage vs. Time Delay Error

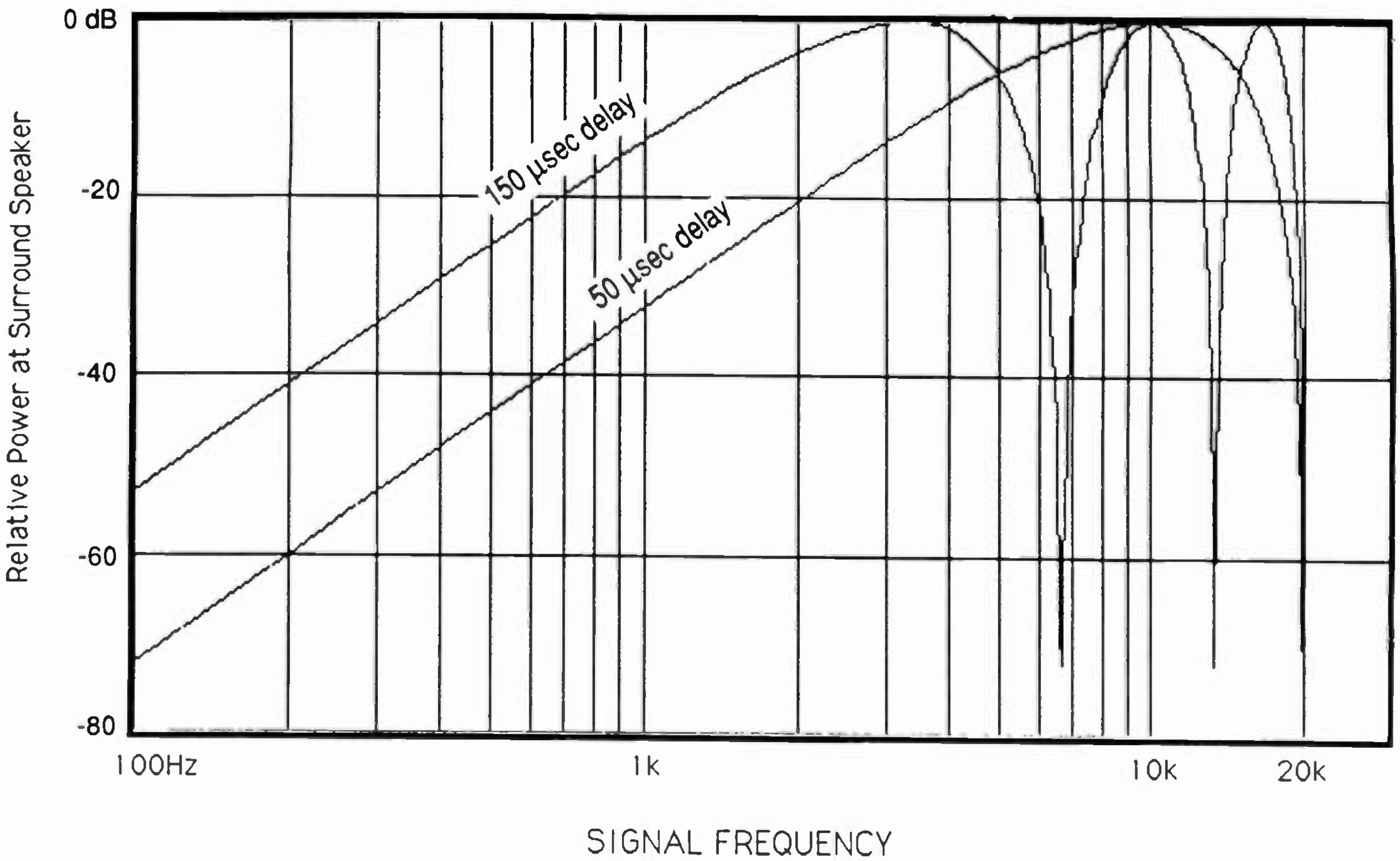
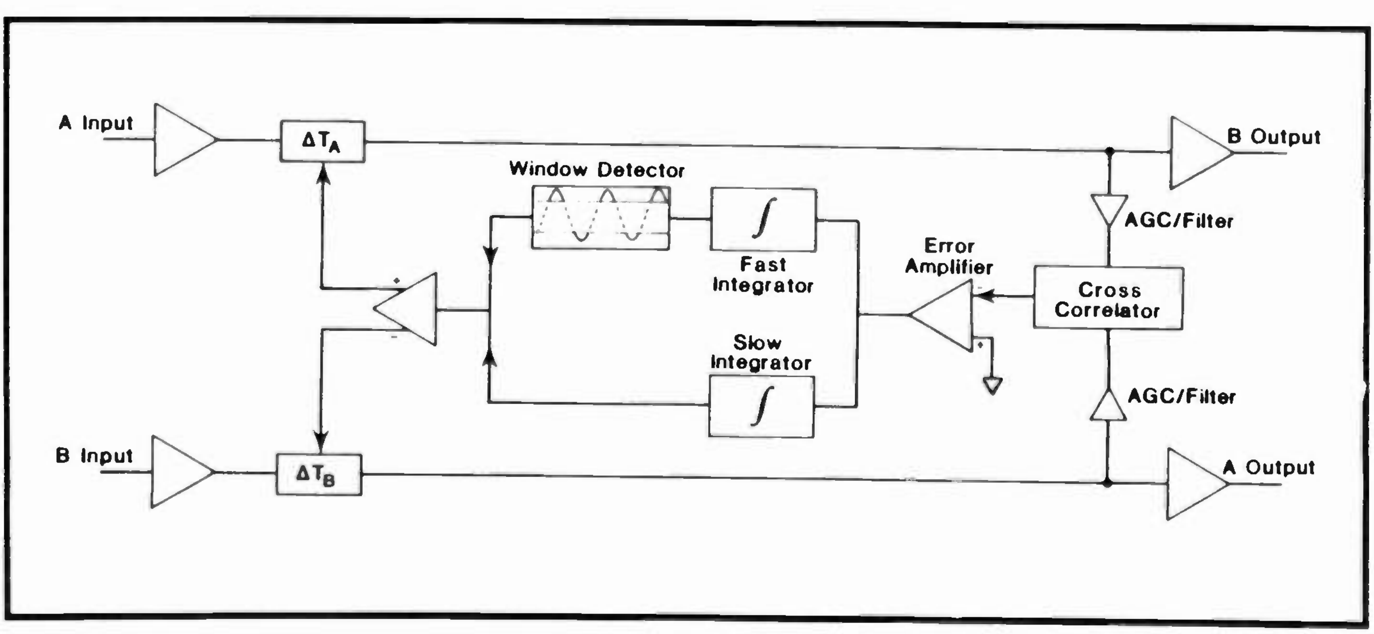


Figure 2. Phase Chaser Model 2300A Block Diagram



RECORDING FIELD AUDIO ON PCM PORTABLE VHS RECORDER PROVIDES AUDIO AND TIME CODE

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ABSTRACT

This paper will discuss the development, technical specifications, and operating features, of a battery portable, digital audio recorder, used for field acquisition of high quality, double-system stereo sound with SMPTE time-code, for either film or tape. Digital sound production and post-production techniques used by Airfax Productions in Chicago will also be discussed.

HISTORY

In 1984, Airfax Productions, a producer of both films and videotapes, installed a time-code based, 3/4" off-line, A-B roll edit system. The goal was to save post-production dollars by doing the creative work inexpensively on our off-line "flatbed", and utilize an on-line suite as the finishing "laboratory". It was our intention to create an off-line edit decision list (EDL) which, when massaged through various sorting routines written for an IBM PC, could be used to rapidly auto-assemble a television program in a 1" on-line suite.

The idea was to adapt the techniques of film editing (such as building separate dialogue, music, and effects tracks) as much as possible to editing on tape- even if the program material was shot on film.

At the time, editing sound on videotape was limited to just two audio channels; the EDL format was limited to Video, Audio 1, and Audio 2. But in practice, eight or more lists may be needed; A and B rolls of picture, music, dialogue, and effects.

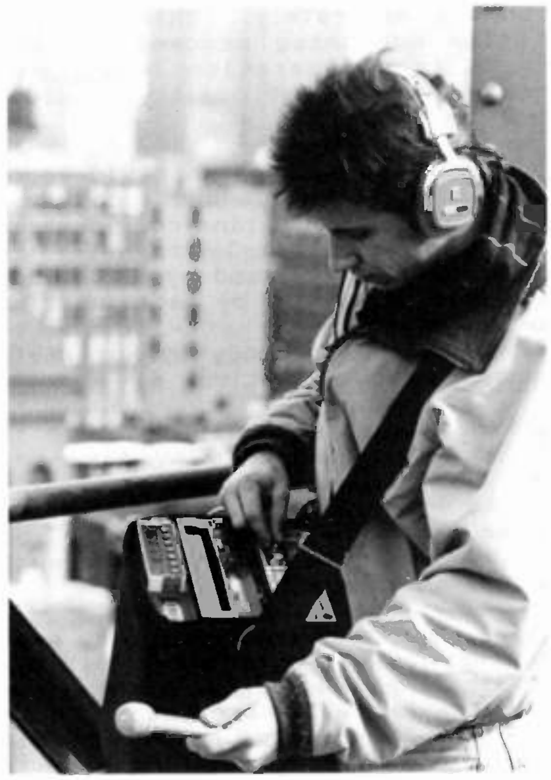

Figure 1: On Location with the Airfax Digital Audio Recorder

As development proceeded, it became apparent that auto-assembling pictures was simple. Using techniques pioneered by Los Angeles editor Rich Uber, and software that we wrote, we were able to assemble the picture portion of a half hour program containing over 500 cuts and dissolves, and from twelve separate source reels of 1" tape, in under three hours.

The stumbling block, though, was in the sound assembly. Recording studios generally have no capacity for a CMX style edit system and its EDLs. And for other than the simplest of sound tracks, it was difficult and expensive to mix a high quality sound track in the video edit suite.

There was a further complication in adapting the film double-system production process: because the sound and picture are recorded separately and must be synced up later, it is usually necessary to make a workprint of the negative, transfer the original sound from 1/4" audio tape to magnetic film, and sync each take separately. This expensive and time consuming process not only results in a generation loss, but a transfer to a lower quality recording format as well.

We wished that somehow we could encode the original negative and sound track with identical SMPTE time-code numbers. Then, we could simply transfer the negative to 1" tape, and sync the audio track directly to it by matching time-code numbers in our off-line suite. And we would save money by not having to workprint the negative.

After editing, we could separate the various audio segments by time-code numbers and lay up the first generation sound to separate tracks of a time-coded 24-track audio tape. And finally, mix in a top audio house and preserve the original sound quality.

In 1985, we discovered that Sony had a consumer audio product with technical specifications exceeding anything else on the market. We thought it ironic that this inexpensive consumer device should be capable of audio quality far superior to anything offered to the professional production community.

Sony's PCM-F1 was capable of encoding 16 bit digital audio, with 90 db dynamic range, .005% distortion, 80 db of channel separation, and unmeasureable wow and flutter. It recorded stereo sound by digitizing it into a video picture and recording it on a VHS cassette. (Figure 2.) In fact, PCM was intended for use with low-end consumer VCRs and, as we subsequently discovered, tapes made on professional 3/4" VTRs (such as the BVU-800) were actually more difficult to deal with.

Undaunted, we decided that a system using PCM recordings, in a video format with SMPTE time-code, could solve all of our audio problems.

Because DAT (then RDAT) had not even been announced, and Nagra's time-code audio recorder and playback machines were unavailable to us in Chicago, we decided to build our own portable digital audio recorder. And in 1986 we first used it to produce a half hour, filmed, television program.

The audio in this program was mastered digitally, mixed on 24-track with Dolby-A noise reduction, and then transferred to the 1" edit master as a Dolby-A track, thus maintaining the original high quality sound throughout the post-production process.

TECHNICAL SPECIFICATIONS AND OPERATING FEATURES

The Airfax digital audio recorder basically consists of a compact packaging of the Sony PCM-F1, a Skotel portable time-code reader-generator, a Hitachi portable consumer VHS VCR, and rechargeable batteries. It is packaged in a case with a shoulder strap and weighs about twenty-five pounds.

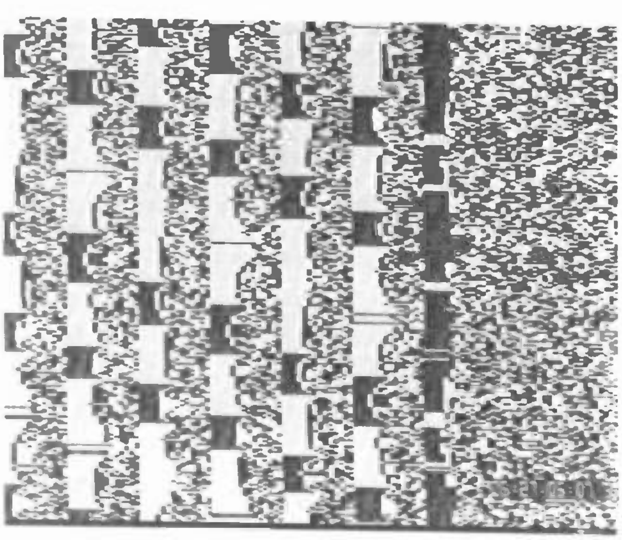

Figure 2: One frame of PCM encoded video with time-code (in lower right corner.)

An additional circuit board was designed and built into the PCM battery compartment to interface the assembled equipment; two panels for connectors and switches were fabricated. The Sony and the Skotel units required modification, but the Hitachi VCR was unmodified.

The following is a list of the features we designed into the unit:

1. The Airfax digital recorder records two channels of PCM digital audio on a standard VHS cassette, providing two hours of continuous record time on a T-120 cassette at the SP speed.

2. It also records two additional VHS hi-fi tracks which may either be recorded independently of the digital tracks, or may be switched in parallel with the digital tracks.

   These could provide a backup analogue recording, or a scratch track for transcription, for example. Thus the recorder is really a four track recorder, plus

3. It also records SMPTE time-code (and user bits) on a fifth, longitudinal audio track.

4. The PCM-F1 encodes 16 bit digital audio, with 90 db dynamic range, .005% distortion, 80 db of channel separation and unmeasureable wow and flutter. (Sony specs.)

5. The Airfax Digital Recorder operates on a switchable 60 hz. internal crystal for double-system film recording, or may be synced externally for video (59.94 hz.) use with loop-thru BNC connectors.

6. The PCM is vertically locked and time-code is always phased locked to sync. The phase may be shifted so that dubs may be edited on BVU-800 recorders with no loss of the PCM run-up code and resultant audio dropouts. This allows PCM dubs to be edited from machine to machine.

7. LED indicators confirm that external reference is present and PCM vertical lock integrity.

8. It provides in-the-field playback so that "keeper" takes can be decoded and checked.

9. Time-code (and user bits) may be preset and loaded.

10. A switch selected option allows the time-code generator to run continuously, as in a time of day operation, or to run only when the recorder is in record.

11. Playback time-code is automatically jamsynced and displayed, and the time-code reader can also be jamsynced to external time-code.

12. An XLR connector is provided for a time-code output to drive a time-code slate or a computer note logging system.

13. Both Line and Microphone level inputs are provided, with low noise preamps and 15 db input pads.

14. There is a built-in monitor amplifier, headphone jack, volume control, and slate microphone.

15. There is an internal tone generator to set record reference levels. The tone generator is locked to the internal 60 hz. crystal so that by simply looking at the PCM encoded playback video on a monitor it is possible to verify whether the recording was made at 60 hz or locked to external 59.94 reference. (The digital picture will appear to "drift" if recorded with the latter reference.)

16. It functions as its own "re-solver" when used in the dub mode for transfers to 3/4" videotape.

17. The system uses 12v. D.C. supplied either by an internal rechargeable battery, (which provides approximately two hours of continuous operation,) or by an external battery belt or A.C. power supply.

18. The whole unit draws approximately 30 watts in record but only 3 watts in standby, or "autopower" mode.

19. A switchable battery-saver feature automatically powers up the PCM when record is initiated on the VCR, yet still allows E-E monitoring of the input audio for set-up when not in record.

20. Lock up time is fast: two seconds in full power, (only one second if the VCR is in "standby" mode,) and less than ten seconds in "autopower" mode. The cassette can be changed and recording again in under ten seconds. Of course, there is no need to wind the tape off first.

21. An LED indicator warns of low battery voltage. The PCM-F1 turns itself off well before the voltage drops to a point that would result in an unplayable recording or a loss of the counter time on the VCR.

22. When external power is removed, the unit switches to the internal gel cell batteries so that the time-code generator will continue to count, and the VCR will "remember" the counter time.

23. Error checking is robust enough that recordings are impervious to all but the most severe tape dropouts. In fact, dubs with a visible time-code "window" in the PCM video signal itself have been decoded with no audible artifacts, at sizes up to 20 scan lines high.

24. The unit weighs twenty-five pounds and can be carried over the shoulder; it fits under an airplane seat.

25. The Hitachi VCR has an independent tape remaining counter, to warn of tape runout.

Figure 3: The AIRFAX Digital ATR

FIELD PRODUCTION

As a portable recorder, the Airfax Digital recorder is functionally equivalent to a Nagra reel-to-reel machine with a two hour load of tape. One operator can carry it over his shoulder and view record level indicators simply by looking down. Playback of any take is simple; time-code is an instant reference.

In the fall of 1986, we had our first opportunity to field test our new recorder. We were shooting a half hour documentary narrated by Louise Mandrell, on farms in several states.

The transition from Nagra to Digital recorder was easy for soundman Tim Turner, and cameraman Ric Lee. After a short orientation, field operation proved quite simple. As in any film shoot, the soundman would roll first, wait for the lock light, and say "speed", at which time the cameraman would roll film, and say "speed".

Now, one more important element was introduced. We built a slate with a time-code reader and display that was wired to the Airfax recorder's time-code output. (We'd have bought one of several coming out on the market at that time, but they were back-ordered for three to six months.) The cameraman would shoot a few frames of the slate for the sync reference, and we clapped the slate as a backup - just in case the time-code display on our new slate was inaccurate for any reason.

Since playback of the VHS tape would be in a video edit suite, though, it was necessary to wait five to ten seconds after the camera rolled so that the playback machine would have enough heads for pre-roll.

There was a big advantage in shooting with a two hour tape load. In our traditional film experiences, the camera (with its eleven minute film magazines) or the Nagra (with its fifteen minute tape reels,) would run out stock at different times. It always seemed like we were waiting for someone to reload. But with two hours of record time on a VHS cassette, (and four hours if we used the "LP" mode!) it never happened. And if the cassette ran out, replacing it took less than five seconds.

We shot in dusty fields, rain, and on gravel roads in a truck without shocks. We expected video dropouts to result in audio dropouts. They didn't. We waited for the humidity sensor in the VCR to shut us down. It didn't. We expected vibration to unlock the VCR. No way. In fact, we shot over 13,000 feet of 16mm film (about six hours) and five hours of sync sound over several days, without a single technical problem. Except one.

The Hitachi VCR was a consumer deck, and, as we discovered later, had an automatic rewind when the end of the tape was reached. Once it did, but the soundman didn't know it - and we overcut some scenes. Fortunately, we didn't need them, so what could have been a "gotcha", turned out to be of no consequence.

A related problem arose during our first field trip. We were shooting with two different film stocks, and thus reloading some short ends later. The post-production process required ascending time-code on all of the 1" film transfer reels, so that auto-assemble searches could work without interruption. But if the film roll was changed while the time-code generator continued to run for the next roll, when we later re-used the prior stock again, the time-code would be out of sequence.

The simple solution? Tape is inexpensive, so change both tape and film at the same time. Increment the hour digit in the time-code to the next hour, label and store them together, and reload them together, later. The jam-sync feature of the time-code generator would regenerate the correct sequential time-code.

MOS shooting presents another logistical problem. Because the PCM audio tape carries the time-code reference for the transferred picture, no time-code space would be allotted for picture-only shooting. There are three options:

1. Advance the time-code generator by at least the amount of time the film rolled MOS, (warning-math required here) or

2 Switch the time-code generator to "free-run" mode while the camera is rolling, or

3. Roll sound anyway - you might just capture something for the Christmas party reel.

Conversely, shooting voice over, presence, and sound effect tracks, (no picture,) was simply a matter of assigning one VHS cassette solely for this purpose.

One of the interesting sidelights of shooting audio with such an unprecedented dynamic range, is that "riding gain" almost becomes a technique of the past. Of course, you still need a skilled soundman to place the microphone and listen for clothing rustle, but once tape rolls, by setting the level control mid-range, he can almost go to sleep. There's plenty of dynamic range left to "fix it in post".

The quality was so good that when we shot an interview in front of a crackling fireplace with a condenser shotgun microphone, the pops sounded like gunshots - the audio headroom was so great.

One other production technique is worth mentioning. We also fed time-code to a Radio Shack Model 100 laptop computer with Comprehensive's "Log-It" software installed. When we clapped the slate, we entered the time-code number, scene and take information, and production notes into the computer with the word processor. At the end of the shoot, we simply printed a hardcopy of the log with time-code numbers. And those field generated numbers were valid right through all phases of post-production, both sound and picture. No math required here, either; no troublesome offsets to compute.

And during slow periods in the shoot, the soundman could entertain us with VHS movies rented from the local 7-11.

POST PRODUCTION

When we were finished shooting, we transferred all of the negative to 1", without sound or time-code. The transfer house was surprised at the idea of a tape without time-code, but its presence would only complicate things later.

We made 3/4" work dubs of the VHS/PCM cassettes on BVU-800's (no loss in quality - it's digital) and used them to sync up the sound to the picture. This step required editor-controlled 3/4" and 1" VTRs. We would dial up to a slate on the 1" transfer roll, read the time-code number displayed on it, enter the number into the edit controller and cue the 3/4" PCM audio roll to that number.

The audio and time-code were then edited directly onto to the 1" film transfer roll. We could even transfer just the analog scratch track (and not the PCM digital track) since we would only be using this audio for off-line, and not for the finished mix.

A minor glitch arose during the syncing process. One of the nice things about shooting film, unlike videotape, is that the usual five to ten second delay to provide enough time-code for the playback VTR to preroll for the edit doesn't exist. In a documentary a ten second delay may result in losing the shot. And once in a while, we didn't get the full ten seconds.

But by letting the camera roll on for a couple of seconds after each take ended, we could just add five seconds of heads to the number displayed on the slate (and to the same number on the PCM 3/4" transfer roll) and back-edit into the tails of the previous scene during the syncing process.

Another small problem would crop up if we couldn't read the time-code number on the slate for some reason. In that case we just used the audio cue of the sticks clapping as our sync reference, (and picture of the sticks closing) just like in film syncing.

The synced-up film transfer master was dubbed to 3/4" A and B roll cassettes for the off-line edit, with jam-synced time-code and "window burns" of the time-code. For some weeks, we massaged the cut inexpensively on our 3/4" off-line system. When it was finished and approved, we began the list management process.

Basically, we separated the EDL into separate video and audio lists. The picture portion, (and scratch track audio,) all 500 or so transitions, took less than three hours to auto-assemble, with on-line editor Tom Evans. We spent another two or three hours adding titles, slates, and various video optical effects to complete the picture portion of the program, and made a safety-master dub.

The audio list was further divided into music, dialogue and effects lists and each separate audio event in the program was transferred from the 3/4" PCM rolls to a designated track on a 24-track ATR with Dolby-A. (Of course, today this transfer could be to a studio digital multitrack recorder.) A specially coded piece of 2" tape had time-code identical to the off-line master, and each sound byte was placed in its appropriate time sequence using the slaved 3/4" off-line master as a visual reference. The layup process was unsupervised and took about ten hours; it could have gone still faster if the 24-track had been under CMX control-we could have auto-assembled from our audio EDLs.

Steve Wilke equalized and mixed the half hour show in another five hours, adding additional sound effects from his Compact Disc library.

Lastly, we transferred the mixed track to both the 1" master videotape and to the safety, replacing the scratch track audio. Now both masters had the same generation (first) of 1" tape sound. End of project. Make the dubs and ship.

Though the program specifications called for a monaural mix and release, the 24-track tape could as easily have been mixed in stereo with just a little more time spent in the mix.

CONCLUSION

Since its first use, we have been able to use our digital audio recorder instead of a Nagra on a wide variety of film and tape projects, from television commercials to videodiscs, whether shot in studio or on-location. In addition, it has greatly expanded the audio capabilities of our edit suite.

But there are still a few things that we never quite had time to add to our box.

One would be a wireless link for time-code data to the slate. It is always a nuisance to be dragging the cord around; just one more cable to trip over. Two, actually - there is another feed to the computer. Each setup requires moving these cables.

A small internal loudspeaker like that in a Nagra would have been useful to check content and performances, but we simply ran out of room in the box.

And perhaps phantom powering for the microphones.

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Looking to the future, as random access off-line editing equipment capable of much more sophisticated list management becomes available, the process we developed could easily be adapted to a film release as well, simply by converting time-code numbers to feet and frames It will become practical to shoot film for its look, edit on disk (or whatever) for speed and convenience, and release on film for the big screen.

With the coming DAT audio revolution it will be possible to produce vastly superior audio, on location, and to preserve that quality right through release.

We're still waiting for DAT - but we'll know how to use it when it arrives.

ABOUT THE AUTHORS

Neal Kesler is President of Airfax Productions, a thirteen year old company that produces television commercials and corporate programs, with 1"/inter-format post-production facilities. For several years prior to that, he was a cameraman and videotape editor at WTTW in Chicago. A graduate of the University of Illinois with a B.A. in Mass Communications, Neal also directs for Airfax.

Jim Swick, Chief Engineer at Airfax, was formerly Assistant Chief Engineer at WTTW, and has spent the last twenty years in broadcasting, television production, and equipment design. Jim played a key role in the development of Broadcast Stereo, and was awarded an EMMY for his early TV stereo efforts. Jim attended the Illinois Institute of Technology, and last presented a paper to the NAB in 1985.

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A LONG-TERM MTS SYSTEM DESIGN FOR THE BROADCAST STUDIO FACILITY

Rick Craig
WGN Television
Chicago, Illinois

ABSTRACT

In order to continue to be competitive, most television broadcasters will sooner or later be faced with the task of converting their facility to handle Multichannel Television Sound. Although conversion of the transmitter can sometimes present a challenge, it is the studio facility where the greatest investment of time and money will usually be required. Most television facilities have been constructed with limited attention to audio, and almost none are prepared for stereo distribution. The audio portion of most plants will have to be redone from scratch, and many broadcasters will want to take advantage of the Secondary Audio Programming (SAP) channel.

INTRODUCTION

In designing the studio facility, many questions must be addressed in order to provide a flexible and user friendly system. How will out-of-polarity audio or channels that are left-to-right reversed be corrected? What if there is time code or some other unwanted audio on one of the designated stereo channels? What if the audio on the designated stereo channels is not really stereo at all, but instead needs to be summed together to produce monaural audio? If a discrete system is chosen, is there the possibility that in the future a matrixed program source might need to be decoded, or vice-versa? If a stereo synthesizer will be used on nonstereo material, how will it be controlled? How will the source of audio to the SAP channel be switched? Will there be silence on the SAP channel during commercials and other times when secondary audio is not available?

Many broadcasters currently transmitting MTS stereo have devised primitive schemes to simply "pass" network audio and switch a limited number of stereo sources to air. At the WGN Television Broadcast Center in Chicago, a decision was made to develop a stereo and SAP distribution system to fulfill the long-term needs of the facility. The design is innovative and makes use of unique distribution hardware. This is the story of that conversion.

HISTORY

WGN-TV began stereo programming in April of 1986 with the broadcast of Chicago Cubs Baseball from Wrigley Field. Since then, more than 120 home games have been produced and aired along with many other sports and entertainment specials produced by WGN-TV. Among these were The Bozo 25th Anniversary Special, Notre Dame football and basketball games, De Paul basketball games, and the Chicago Emmy Awards.

On December 20, 1987 WGN-TV aired Chicago's first bilingual stereo program using the Secondary Audio Program (SAP) channel. That program, the Hollywood Christmas Parade, was produced by our Los Angeles affiliate, KTLA. It had a Spanish translation, plus effects, mixed down onto the 3rd audio channel of a one-inch VTR.

GOALS AND OBJECTIVES

Long before that first stereo broadcast came the engineering design phase of our facility. The first stage of that design was to define just what we wanted in a stereo environment. The completed system should allow us:

- To air any incoming satellite, microwave, or other remote program source in stereo and/or SAP.
- To air any stereo program media from any playback machine in the house capable of stereo reproduction.
- To handle any in house routing of stereo for the purpose of recording and dubbing.
- To route any live source to air.
- To generate and/or route the appropriate program feed to the SAP channel.
- To make detection and correction of common routing, level, and polarity errors so fast and simple that it is routinely done.
- To easily maintain the system.

CONVENTIONS

Certain in-house conventions needed to be established. Some of the standards that we adopted are:

- A discrete Left and Right audio system. A matrix system was discussed and rejected for two primary reasons. First, we felt that maintaining the required precise control of relative sum and difference channel gains in order to preserve a reasonable degree of channel separation was an unnecessary burden and could, at times, be difficult to achieve. Second, the need for a matrix decoder at every monitoring location was deemed an unacceptable complication and expense.
- Left audio would normally be recorded on Channel 1 of all VTR recordings and Right on Channel 2.
- We need to be able to reverse these channels (i.e. Right on Channel 1, Left on Channel 2), without patching, in the event of a channel reversal in the program source. This is particularly useful with incoming remotes where you have no control of this situation.
- Secondary audio would be recorded on the 3rd channel of one-inch VTR's.
- To have the ability to invert the polarity of one channel of any program where Left and Right are out of polarity with each other.
- To have the ability to sum the Left and Right channels. This is particularly useful with news tapes that have the sound tracks mixed separately on both channels, but not in stereo.
- To have the ability to synthesize any and all programs and commercials that have not been produced in stereo and to externally control the synthesizer's bypass/synthesize operation with a minimum of operator intervention.

  Some research led us to the conclusion that a synthesizer's automatic sensing circuits leave something to be desired. It is difficult for any circuit to detect the difference between a monaural program with audio on both channels in the center, and a stereo program with dialog in the center.
- To transmit a sum of Left + Right on the SAP channel at all times when there is no secondary language to be broadcast.
- To provide good quality metering facilities in all critical technical areas.
- To utilize constant voltage (60 ohm sources and non-terminating loads) distribution throughout the facility.[1, 2, 3]

  The 60 ohm source impedance provides the isolation required to prevent capacitive loading of the output stage from causing destructive overshoots, while providing good impedance matching to the actual 56 ohm

1 "Voltage Transmission for Audio Systems", Richard Hess (ABC-TV), Paper Presented at the 67th AES Convention, October 1980, Preprint # 1708

2 "The Audio Side of Videocassette Duplication - A Tutorial", James W. Brown, SMPTE Journal, March, 1987, pp 235-236

3 "Interconnecting Audio Equipment", Allen Burdick, Broadcast Engineering, March, 1986

impedance of real-life audio cable.[4, 5] When an individual cable run is long enough to dictate the application of transmission line techniques to it, the receiving end is terminated in 56 ohms. No change is made at the sending end, which is already optimally matched to the line. If the potential exists for the long line to be noisy, the receiving location uses a matching transformer; otherwise a fixed 56 ohm termination resistance is used. In either case, the gain of the receiving amplifier will need to be increased to compensate for line loss and the power lost in the impedance match.

- To operate with a fixed unity gain structure throughout the plant, so that the only level control in the distribution system occurs at the source.

  The constant voltage distribution system is ideal for a fixed gain structure, since there are no variable gain controls and termination losses (particularly from multiple terminations) to complicate matters. Maintenance is much simpler since DA's may interchanged with no effect on system setup.

SELECTION OF EQUIPMENT

After laying out exactly what it was that we wanted to see in a stereo and SAP distribution system, the process of selecting the equipment to meet these requirements began. The search for a stereo distribution amplifier and a system to provide the switching capabilities that we wanted led us to a unique system manufactured by Benchmark Media Systems of North Syracuse, New York.

The distribution amplifier is designed around a voltage distribution standard, based on a 60 ohm output impedance and a properly designed bridging (i. e. high impedance) differential input. Each channel's output stage is a balanced 40 watt 8 ohm power amplifier, isolated from its loads by the 60 ohm buildout resistance. The system is capable of +27 dBu[6] with an unweighted noise floor of -103 dBu, providing a real dynamic range of 130 dB. Bandwidth is 150 KHz; distortion, crosstalk, and common mode performance are equally impressive.

Each channel of the stereo card provides five 60 ohm balanced outputs and one special output that may be ordered configured as 1) a direct, unisolated output of that channel, 2) a sixth 60 ohm output of that channel, or 3) a 1,000 ohm output which is a sum of the two channels. We chose to utilize the latter configuration, the 1,000 ohm sum. In order to provide good isolation between outputs, the output driver provides a very low balanced output source impedance (<< 1 ohm) -- it can even be used to drive a speaker. Isolation between outputs is the ratio between that output impedance and the 60 ohm isolation resistance. The cards may be strapped for fixed unity gain.

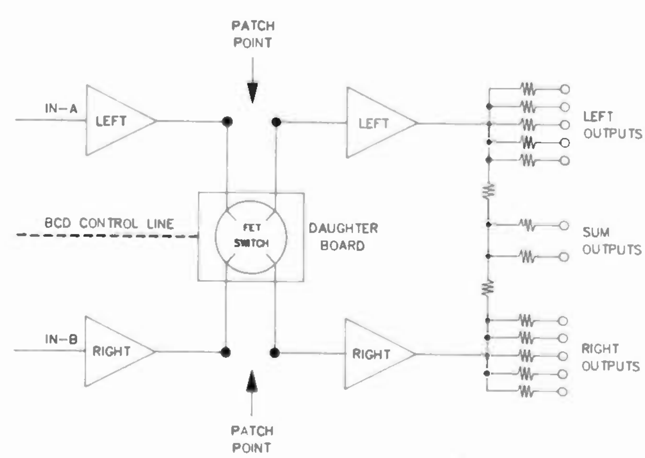

Figure 1a

The Stereo Distribution Amplifier card. Each channel's signal path is interrupted after the input buffer but before the output line driver and routed through an optional daughterboard. When no daughterboard is in place, DIP switches can be set to allow each channel's audio to normal through, be inverted, or be routed as a mono sum, matrix, etc. The patch point may also be accessed at the card edge connector for insertion of external equipment.

4 "Wired For Stereo", David Bytheway, Robert Bosch Corporation, Broadcast Engineering, September, 1986, pp 22-32

5 "Long Line Application", Dean Jensen, Jensen Transformer Application Note, Jensen Transformers, North Hollywood, CA., 1987

6 0 dBu = 0.775 volt into an unspecified load impedance

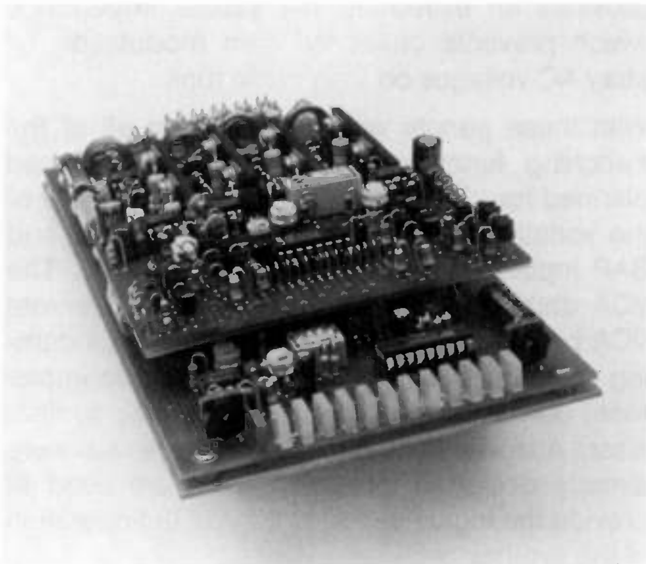

Figure 1b Stereo DA Card with Daughterboard

What was even more interesting, however, was the unique daughterboard system that Benchmark had conceived. The system is based on the belief that broadcasters need to be able, on an instantaneous basis, to do all of the switching and routing of two-channel sources we had outlined and perform a few other functions as well. One interchangeable daughterboard provides on board switching, matrixing and de-matrixing, and can be switched to all of the desired modes by a binary coded decimal (BCD) logic signal. Another daughterboard provides VCA capability and the ability to remotely configure the output to be a sum of the two inputs channels. A precision oscillator and parametric equalization are implemented by utilizing other optional daughterboards, allowing access to the system for testing and limited equalization of program circuits.

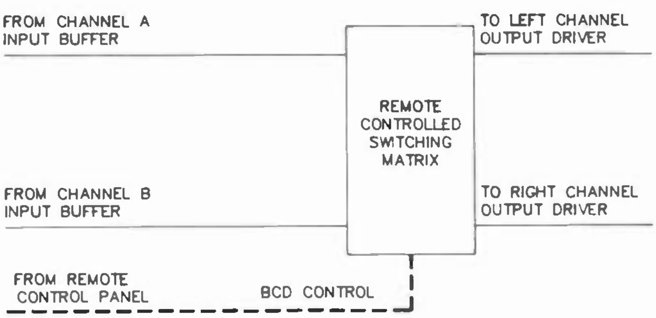

Figure 2 The routing matrix daughterboard.

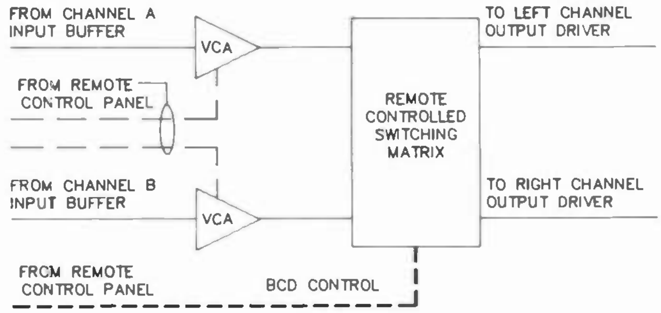

Figure 3 The VCA and matrix daughterboard.

The next question was one of density. With our desire to handle 70 stereo sources in the initial construction phase and be able to add 40 more with expansion, the space requirements of the system were a major consideration. The design of this system allows for 12 stereo cards to be inserted into a single frame. We use redundant power supplies in conjunction with a diode combining unit to supply power for the entire DA system. A single bipolar 15 volt, 15 amp power supply is conservatively rated for up to 144 DA cards.

Each DA card has two complete input and output stages, as shown in Figure 1a. Using dip switches, either input may be routed to either or both outputs or summed, in or out of polarity, with the other input to either or both outputs. The switching/matrixing daughterboards allow the switching to be done remotely by a BCD control signal. We chose to generate the control signal using a control panel for each program source, physically locating the panel adjacent to the source.

Figure 4a The Type A Control Panel

Figure 4b The Type B Control Panel

Figure 4c The Type C Control Panel

Three variations of control panels are used, depending on the functions required. Figures 4a, 4b, and 4c show the three types of control panels.

- The Type-C panel controls the matrix output switching for the Left and Right outputs of a stereo DA assigned to sources which have only two audio channels. For these sources, a sum of Left + Right is hard-wired to the SAP channel, so no switching is provided for it.
- The Type-B panel provides the same switching functions as Type-C but with one exception. There are two additional buttons labeled L+R and CH-3 on the panel that control a second amplifier which feeds the SAP channel. With these selections, the operator can choose to feed either the 3rd audio channel of a program source out on the SAP channel, or feed a sum of Left + Right if secondary audio is not available. This way, there is never silence on the SAP channel. Type B panels are used for all in-house sources with three audio channels.
- Type-A panels provide all of the above features with the addition of VCA control of the amplifiers. These panels are used with all incoming remote sources, such as networks, satellite and microwave receivers, etc. Unfortunately, the VCA daughterboard does not retain the capability for polarity inversion and channel reversal provided by the matrix daughterboard. Because these functions must be provided on remote lines, we built our own electronics to provide these functions for inputs which have VCA control, using the contact closures from the switches on the panel.

The switches on the control panels are dry contact closures that provide a switched voltage for the amplifier's BCD input. The VCA controls on the panels drive an OP amp which provides a variable DC control voltage for the voltage controlled amplifier on the DA card. The OP amp provides an extremely low source impedance which prevents unwanted gain modulation by stray AC voltages on long cable runs.

With these panels we could perform all of the switching functions, and more, that we had planned for. The matrix is able to perform all of the variations in routing of the Left, Right, and SAP inputs and outputs outlined above. The VCA daughterboards controlled by the remote VCA panels provides control of gains of incoming remotes. Custom logic was used to implement our scheme to control the stereo synthesizer. Auxiliary contacts on switches which were already designed into the panels are used to provide the logic needed to control the operation of the synthesizer.

Precision metering and monitoring is provided in all primary control and operating locations. The operator is provided with precision peak/average level monitoring and an X-Y stereo scope. Compact powered monitor loudspeakers located at the control location allow the operator to hear the results of his switching decisions and the content of all channels for confidence, cueing, and the correctness of the monaural sum.

THE SYSTEM DESIGN

A simplified signal flow diagram illustrating the Channel 1 and Channel 2 outputs of a typical two channel source (in this example, a VTR) is shown in Figure 5. The Channel 1 output is fed to the Left input of a stereo DA, and the Channel 2 output to the Right input of the same DA. In fact, this is the case for all audio sources in the facility, including satellite and microwave receivers, telco lines, audio consoles, etc. In the case of monaural sources, the single channel audio is strapped to both inputs of the DA. This system, therefore, provides audio on both of the channels in the facility at all times, even if it is monaural.

Figure 6 shows how the system accommodates the SAP channel (the Left and Right channel DA outputs to the master control and routing switchers still exist, but are omitted from the drawing for clarity). The same type of stereo DA card is used for both stereo channel and SAP channel functions. The summed output of the

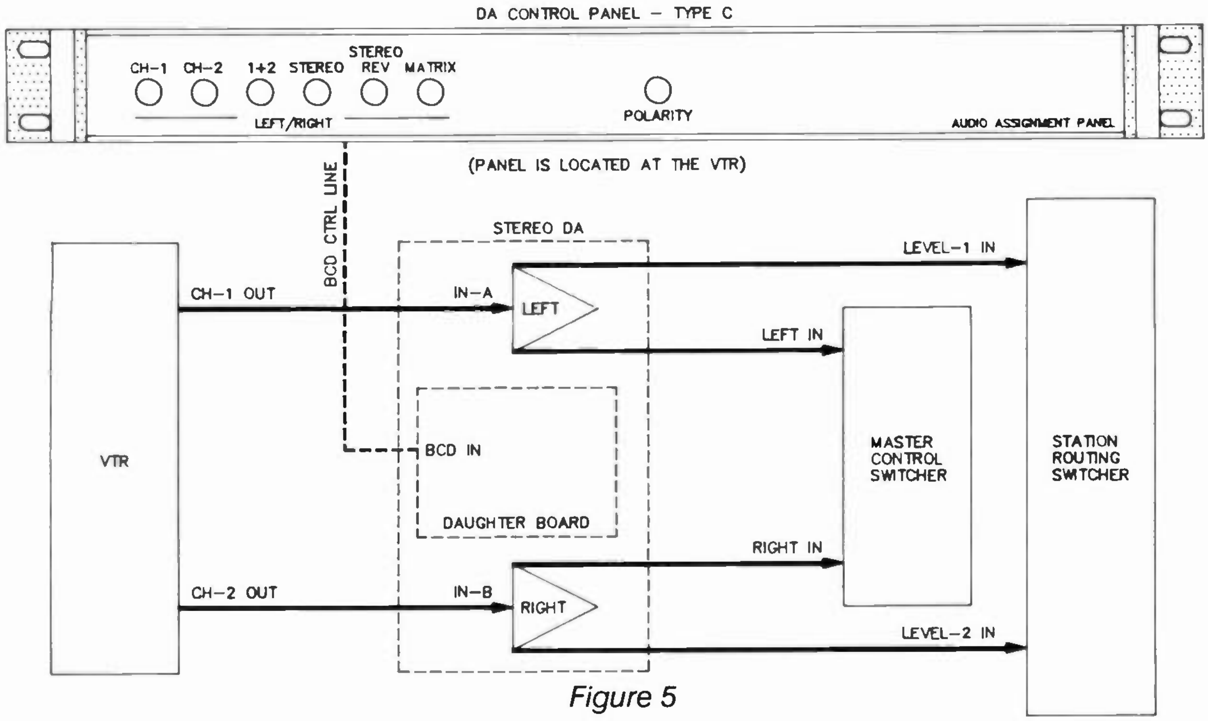

Figure 5

Signal and control flow for the stereo channels of a typical two channel program source. The daughterboard is controlled by the BCD logic generated by a Type C control panel remotely located with the source. Two outputs of each channel feed the station's Master Control and Routing Switcher's Left and Right inputs. The remainder are available for other user applications.

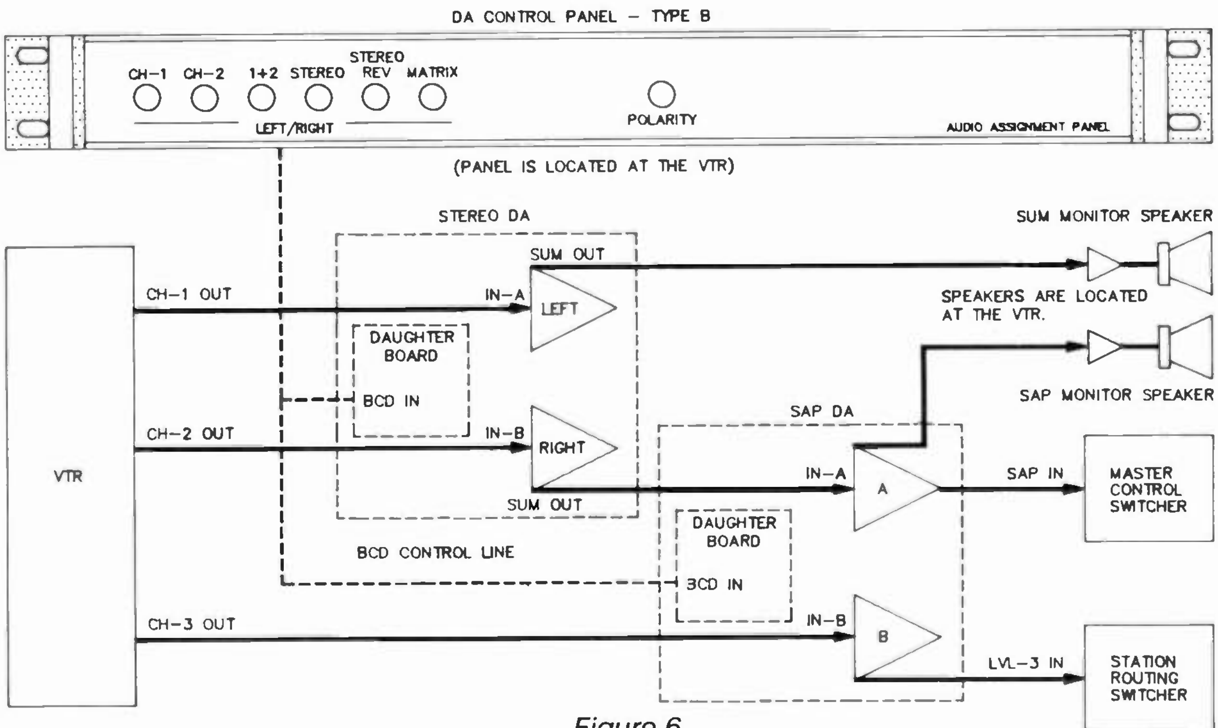

Figure 6

Signal and control flow for the SAP channel of a typical three channel program source, with details of the stereo channels omitted for clarity. The daughterboards are controlled by the BCD logic generated by a Type B control panel remotely located with the source. Two outputs of each channel feed the station's Master Control and Routing Switcher's third level input. The remainder are available for other user applications.

stereo channels (reflecting the switching matrix decisions made by the operator) feeds the A input of the SAP DA. The B input of the SAP DA is fed by Channel 3 of the source (VTR or remote). Either input may be selected to all of the outputs of both A and B SAP DA's.

Remote program sources are also configured as in Figure 6, but with the addition of VCA control of DA gains. The Type A panel, which provides the variable DC voltage to control the VCA gain in addition to the BCD control for the switching matrix, is located at the local control point for the remote source (the satellite or microwave receiver).

With program sources whose audio does not need to be switched, such as a monaural source or an audio console, the optional daughter board and control panel are simply not installed. In this case the DA performs like two discrete amplifiers. Where switching is necessary, a daughter board is installed on the DA, and a control panel is located adjacent to the source of audio (i.e., VTR, satellite control point, etc.).

A DA slot was wired for each of the 64 inputs of our house routing switcher. If an input wasn't used, a DA was simply not installed, but the frame was wired for future use. This provides for post-switched audio to any of the recording or monitoring devices located on the outputs of the switcher anywhere in the facility. One output of the DA (there are 5 outputs per channel) feeds the switcher. Another output is wired to the master control switcher's Left and Right inputs.

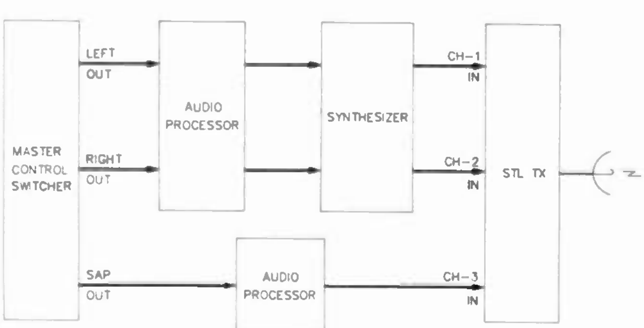

Figure 7

The Master Control Switcher output system feeding the STL Microwave Transmitter.

Figure 7 shows a simplified audio flow diagram of the master control switcher output system. The Left, Right, and SAP channel outputs of the switcher feed a compressor/AGC amplifier. The Left and Right outputs of the amplifier feed the inputs of the synthesizer. The synthesizer then feeds 2 audio subcarrier channels of the STL microwave transmitter. The SAP channel is identical, but without the synthesizer. Since all channel switching has already been performed prior to the switcher's inputs, these signal paths contain the desired audio for transmission.

STEREO SYNTHESIZER CONTROL SYSTEM

Most synthesizers can be set up or modified to synthesize from the Left Channel only, and we have chosen to follow this convention in our plant. The synthesizers have a built in automatic sensing circuit designed to detect incoming stereo or monaural audio and synthesize only when the source is monaural. Unfortunately, these circuits do not perform well in the synthesizers we have evaluated, and we have chosen to utilize the external control input which some manufacturers have provided for user control of this function.

This external input could be controlled with the station's automation system, assuming that it has one. Our automation system does not provide a field for this purpose, so early on we devised a scheme to use an auxiliary relay from the automation system. Although our design worked, it was labor intensive because our business service could not address the field. The Master Control engineer had to enter, by hand, a stereo mark in the auxiliary field for every single stereo element that aired. This is where the DA control panels came to the rescue.

Since the operations engineer has to select an appropriate button on the control panel each time a program source is set up, the decision has been made by the operator as to whether or not the program source is in stereo or monaural. Translating this decision into a control signal for the synthesizer is then a simple matter.

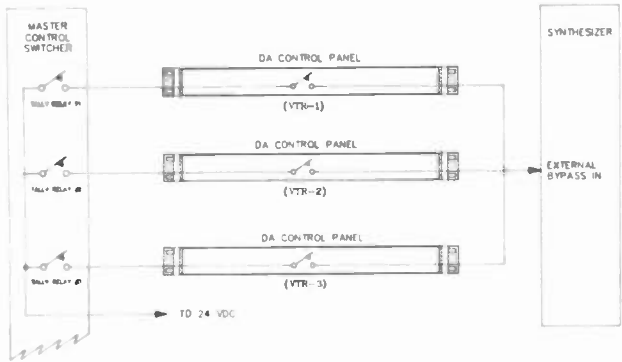

Figure 8
The Synthesizer Control System.

Figure 8 illustrates the technique we used to control the synthesizer. All master control switchers have tally relays. This may be in the form of a dry contact closure, or a switched DC voltage. This tally signal from the switcher is routed to its respective control panel. Each control panel then provides a dry contact closure which corresponds to the selection that has been made on the panel by the operator. If a stereo selection has been made, the contact is in the closed position. If it is monaural, it is open.

When a crosspoint has been made on the switcher, the tally output of the switcher will be routed thru it's respective control panel. All of the outputs of the switches in the control panels are bussed together on one common control line. There can only be one output at a time on the buss since there can only be one tally voltage at a time out of the switcher. The buss is then fed to the synthesizer's external bypass control input. When the operator makes an audio selection on the control panel, he or she is simultaneously controlling the operation of the synthesizer.

CONCLUSION

This unique system of switching has provided us with not only the flexibility of complete audio control, but with an audio distribution system of superior quality. In the two years it has been in operation no failures have occurred, nor has there been a situation where it did not meet all of our needs. The concept of unity gain voltage distribution has proven itself to be troublefree and much simpler to maintain. Since the system uses a binary coded decimal logic signal for its switching control, it stands ready for a future when every station has an automation system able to directly address it, eliminating the need for the control panels and operator intervention.

ACKNOWLEDGEMENTS

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WEATHER & NEWS GRAPHICS SURVEY RESULTS & INTERPRETATIONS

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The results of a survey of weather and news graphics is presented. Included is a review of available graphics and their sources together with a breakdown of availability according to image resolution, (i.e., low, medium, high and ultra high). Examples of these different kinds of images are presented in video tape. Emphasis is placed on the technical aspects of weather and news graphic production and on the distribution of these graphics, i.e., whether by 1200, 2400 or 9600 baud telephone connection, through satellite delivery or through some other media. furthermore, there is discussion on the handling of these images once they are received at the station, i.e., Stillstore, ChromaKey, etc. Statistics are presented showing the breakdown of these various factors according to market size. There is also a presentation of trends and expected new developments as predicted by the many dozens of engineers participating in this survey.

Over the past few years, on-air graphics have become a bigger and bigger part of our professional lives. With greater use in each passing month and easier production capabilities, all of us have seen more and better graphics over the past few years at both our own stations, and at other stations all across the country.

In the next few minutes, I will provide my perspective on the trends around the nation in the use of news and weather graphics and also what the outlook is for change over the next year or two. This information is based on the experience we at Accu-Weather have had in the marketplace, and a random sample telephone survey that we conducted of more than 50 television stations around the country. These stations are both Accu-Weather clients and non-clients representing small, medium and large markets.

We will take a look at the decisions that news directors and engineers have made to improve their look; their selection of in-house or out-of-house graphics creation and what is available if you go outside for your graphics. We will look at the different means of distribution of these graphics and will take a look at market differences. Lastly, we will look at the future trends in the use of graphics.

Within the area of improving your look, there are two choices: one is better hardware and the other is better graphic design. Better hardware results in higher resolution and more colors and will allow you to render objects in a more realistic manner and provide the means for better design capability. In most markets, high resolution is an absolute requirement for news graphics. On the other hand, medium resolution graphics continue to be the norm for weather graphics with the exception of the largest market stations.

One decision having to do with hardware is whether to get a single graphics computer that can be used for both news and weather or to have two or three graphic computer systems devoted to different tasks. In most small and medium market stations, news and weather share a single graphic computer. This means an outside source is a must for all weather graphics, so the in-house graphics computer can be used for producing news graphics. As you know, a variety of good weather graphics will take several hours of top talent time to produce before each and every show.

The other major choice for improving a station's look is in better graphic design. In large markets, where money is less of a problem, stations typically have larger graphics departments and excellent artists at work. In small and medium markets, sometimes producers/directors make the graphics after doing reports out in the field. The extent and the price range of graphics capabilities is far-reaching, and continues to expand. The four bit, 16 color medium resolution machines such as the LiveLine III and newer PC and Amiga units, are able to receive weather graphics from an outside source or produce medium resolution graphics in-house. The higher grade units such as LiveLine IV's, Artstars, Dubners, Vidifonts, Cubicomps, Chyrons and Quantels produce high resolution graphics from 256 colors on up, but need more artistic talent to get the most out of them. Not only does the cost of the equipment have a big range, but the cost of operating the equipment also increases with more colors, more resolution and more capability.

At many stations, the graphics department has become more important than ever to news and production. In the case of one station with whom we talked, everything now goes through one graphics department. Many stations, at one time, had two separate graphics departments: one for news and one for production. Now it is usually done in one to better utilize resources. When you are striving for better graphic design, you have some important choices to make. Are you going to do the graphics in-house? Are you going to get them done outside? If you remain in-house, you can control your own look directly but there are the disadvantages of increasing equipment and personnel costs, and the difficulty of finding good design people at reasonable wages. If a medium or small market station finds a good design person, all of their eggs may be in one basket. If that person leaves, or even is ill for a period of time then the station's look may wind up changing.

If you go with an outside source, there are specific advantages. In most cases a greater variety of graphics are available to you, because an outside service can produce more graphics than your staff can, so you can choose from a larger inventory rather than the more limited production that your people can turn out. In the case of weather graphics, there are a number of additional advantages in using an outside source. First, you are drawing on a body of expertise in a meteorological firm that is looking at the weather on a minute-by-minute and hour-by-hour basis. Furthermore, we have found the graphics we produce are more accurate than the graphics created by many top 10 stations that do them in-house, even though the meteorologists at those stations are top notch. Weather graphics at outside services created in real-time and there is much more variety of ideas than could be generated by one meteorologist working alone at a station. Also, when a major weather event is occurring, the availability of many special graphics from a private service can be a major asset. With a major weather graphics house, like ours, you can even have specialized local graphics and control your look through specifications given to the outside staff.

Let's discuss now the differences between news and sports graphics on one hand and weather graphics on the other. These are two very distinct areas of graphics. How they are produced, how they are accessed and how they are used is quite different.

News graphics, both the "over-the-shoulder" or full screen ones are created in different ways depending upon hardware. If you have a 24 or 32 bit machine everything can be done in the computer, from rendering backgrounds to doing basic grabs from video tape and a character generator and then sent to the still store. If the graphics computer allows you to use 256 colors or less, then backgrounds may be done on the computer, but layering words and people over each other will be done in the still store. As your resources lessen, only freeze frames may be used with words over them. Outside news graphics can be received from the networks or private suppliers in slide form or as a daily fideo satellite feed.

The two areas, news and sports on one hand and weather on the other, seem to have gone their separate ways mostly because news graphics originated in-house, and are designed to compliment an anchor or live on-the-scene video, while weather information and weather data have always come in from outside and the graphics are the full weather story representation. There is not usually live weather footage and weather graphics usually take up more of the screen than news graphics.

Another interesting point is that there is often a different look between the news and weather graphics. Stations themselves tend to develop their graphic "look" in the news and sports area while in many cases, it is the top market stations or the weather services that have primarily created the "look" that is used in weather. Even here, however, the look of weather maps and graphics is a constant evolutionary process, which is the result of feedback from stations over time with regard to a variety of weather situations.

Suppose you decide to see what is available from an outside vendor to augment what your graphics department is doing in-house. In addition to the daily feed from the three networks, there are private vendors such as G and G Design and Communications which has a slide service and recently started a daily satellite service. Also, there is Conas and Newsfeed. These, of course, help with your late cast and national stores if you don't have the ways or means to make them, but at 6 o'clock when most of the news is local, you are still on your own.

In the area of outside weather graphics, there is a greater variety of material available from private vendors, and you have more of a choice. From the networks, basically all that will be received are national or large regional satellite pictures, and in some cases national maps. Furthermore, the national feeds from the networks are outdated. They are several hours old, and in some cases they are prepared as many as eight hours before showtime to meet network feed deadlines, so they are often inaccurate at showtime because they are history by the time they go on the air. Fewer and fewer stations are going on-air with network weather graphics because of these problems. The weather graphics from the networks are in video form, but most outside weather graphics are sent as digital computer images. The private vendors include Weather Central, WSI, ESD, Kavouras and Accu-

Weather, all provide continuous 24 hour access. But some services update their graphics more frequently than others.

There are significant aesthetic and content differences between weather graphics and news graphics, differences which also play a major role in the selection process. A news graphic is used with a given story, primarily to create an impression and go with live footage. Weather graphics, on the other hand, are the "live footage" and try to communicate several facts and figures all at the same time. The patterns are often complex and the person watching needs to be oriented to geography and science. There obviously is a different goal for news versus weather graphics. Also, the audience is much more familiar with what they are seeing in news graphics.

News graphics can run into bias questions, making sure that they don't prejudice the viewer to the story in the graphics. For example in a story on abortion, should you show a fetus, a pregnant woman, a doctor or some other image? Each can create a different response to the story.

In weather graphics, there are different kinds of problems. The audience is less familiar with what they are seeing. Many weather graphics try to cram in too much information and so impart none. This is one area where Accu-Weather has pioneered in the look and concept of weather graphics. Better weather graphics are accomplished in a number of ways. One, by what is shown on the map, by the color combinations used, by simplification of the weather patterns and through selection of font types and weather symbols.

1. What we choose to show on the map. On a weather map with highs, lows, fronts and precipitation, why show clouds? Clouds are shown by the satellite picture and if clouds are added to this type of map we have found, through surveys and focus studies, that it makes the map too complex for the average viewer to understand, especially in the few seconds that it is on the screen.

2. Color combinations. We have found that certain colors mean certain things to viewers and certain color combinations are more confusing than others. As a result, we have selected the colors that are most easily convey a given weather pattern.

3. Simplification of the weather pattern. Low pressure is normally associated with stormy weather; however, in the Desert Southwest in summertime, the pressure is low simply because of the intense heat. Most official weather maps show a low pressure symbol in that area all summer long, yet skies there are clear and the weather is good. The low center never moves and it is not responsible for any stormy weather. It simply confuses viewers who are conditioned to expect bad weather with a low pressure symbol and expect it to move, bringing bad weather with it. As a result, we take scientific license and remove that misleading low. Maybe there is a cold front in the midwest that technically has seven low pressure areas along it because the pressure doesn't vary that from one spot to another. However, in two hours the main low will develop in Missouri and from there you will be able to track it eastward to Virginia tomorrow and explain the weather pattern easily. In such a situation we wouldn't put seven low pressure areas on our map; there will be less confusion by putting one low in Missouri.

In summary, the weather patterns are the most complex thing being delivered in the entire newscast. As a result, the graphics need to be crisp, clear and obvious. Furthermore, they need to be accurate and up-to-date. One of our stations, WBRE in Wilkes-Barre, uses our forecasting service. For a while, they used network weather maps but found that they were so far out of date and out of agreement with current forecasts that they switched to our graphics service.

Let's now talk about the distribution of graphics. How do the various services get the information to the stations that subscribe to them? There are again, two major divisions; video distribution and digital distribution. On the video side you have satellie feeds from the networks that are sent along with national story packages several times a day. Generally, stations videotape them and then transfer the frames they wish to use to still stores for on-air presentation. Another approach are slide services. The large market stations have their own look and will use these sources as concept ideas or to "cut and paste" in their own style. The smaller stations will tend to use them directly on-air.

This is also true of weather graphics. For example, some of our major market stations take our graphics and change them to their style by altering colors, fonts and backgrounds for a distinctive look. The advantage here is that they have the latest information that has been double checked for content; plus, the flexibility available from a digital feed.

In digital image distribution, there is dial-up delivery for medium and high resolution graphics at 1200, 2400 and 9600 baud, also satellite delivery, which is a continuous data stream into a Front Door unit or graphics computer. The graphics are transmitted digitally. Either way, dial-in or satellite delivery, the stations receive graphics that can be used in the original form or customized for their own needs.

Now let's put this all in perspective by market size. In the largest markets, money is not a major concern. The key here is a distinctive look and design, one that if possible will not be found elsewhere. To do this they need the best equipment and good graphic designers that are on the cutting edge. In the major markets these special looks can be seen in both news and weather graphics. Through the top stations have the resources to render whatever they want for their weather segments, this accuracy base is usually obtained from an outside source.

In the medium markets, there seems to be the greatest range of differences. Money is a more important factor here so there tends to be less expensive equipment. But ease of operation is not as inherent in the less expensive equipment which requires station personnel to have to devote more time to obtain a desired result. Medium market stations tend to have less of a distinctive look. Personnel costs here tend to be an important factor. Some medium markets have graphics departments, in others the reporters/producers/directors of the stations actually do the graphics. Medium markets also tend to use outside graphics sources; such as the networks, for national stories; and, various weather services for their weather graphics, allowing them to use staff and equipment for local news graphics. Most of these stations tend to have one graphics unit that doubles for both news and weather, and in some cases it may also be used as the character generator in the newscast. This is another reason why these stations find it helpful to have an outside source of weather graphics, so that they are not tying up the graphics unit trying to create weather graphics at a time when they need to be creating news graphics.

At the small market level, the tendency is even less on design and more a question of equipment and priorities. In other words, there is less design in-house and more dependence on out-of-house graphics such as network feeds and video tape freeze frames with character generator over them. Fewer of these stations have ADO's, still stores and graphics computers. Often they are working with monitors on the set, using one inch video tape as freeze frames. These smaller market stations take all the material they can get from the networks and reshape it for local use.

One of the most interesting parts of the survey dealt with what the future may hold for graphics. The various market size concerns still seem to hold true. The larger markets are more interested in design elements; that is, what is the next thing coming over the horizon. While the medium market stations are trying to work up to what the larger market stations are doing as that becomes easier and equipment costs come down. They are also more concerned with other aspects of the newscasts such as content, in stories and graphics. The same thing holds true for the smaller market stations. Bit by bit small market stations are getting new computers and in the process they have changed the look of their newscasts and are moving up to the status of where the medium markets are now.

One of the trends that has been showing up on the larger market stations is the design of a particular motif for the newscast and the weather. Yet, at the present time, many stations have a slightly different look early morning, noon, six and 11. This can be seen when viewing the different newscasts that run around-the-clock on CNN. They present the same stories, but graphics that are used at different times have a slightly different look.

In conclusion, several points can be seen. Because of the difference between news and weather graphics, out-of-house weather graphics will be needed in most markets for some time to come. Knowing the advantages of each weather graphics service will be a big plus in selecting what your station needs.

The evolution in equipment will make it easier to receive out-of-house graphics at all levels of market size, but getting the right equipment for the right task will require careful consideration.

Also, smaller and medium markets will be able to narrow the gaps that have separated their presentation, from top markets, especially in the area of news grapics.

ELECTION COMPUTER SYSTEMS FOR LOCAL BROADCASTERS

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ABSTRACT

The use of computer systems to provide data-driven displays of vote totals and projections has become commonplace among local broadcasters. In this paper, the various types of election systems will be enumerated. Emphasis is given to PC-Based systems, which are supplanting other types of systems for this application. Also explored will be some of the factors that go into choosing, configuring, and using a computerized election system.

INTRODUCTION

For the local broadcaster, Election Night is a competitive and challenging environment, where often all the resources of the station are mobilized to cover the event. Increasingly, computer systems have been used to drive electronic graphics equipment to display election returns and projections.

Data-driven graphic displays of election returns have been in use for more than a decade at the major networks. Initially, only the networks had the resources necessary to mobilize the mainframe computers, custom software, and exotic electronic graphics equipment needed to do the job on Election Night.

However, recent advances in computer hardware, software, and electronic graphics systems have put data-driven election displays within practical reach of local broadcasters. Specifically, PC-based computer systems have supplanted the use of mainframe and minicomputers, providing more computing power at a fraction of the cost. At the same time, electronic graphic equipment manufacturers have responded by making systems more easily interfaced to host computers.

At first, the software that ran on the PCs was highly customized to each end-user. Requirements differed for each station. A variety of data sources needed to be coordinated, including poll-watchers, and wire services. The services of a programmer might be required to make software changes to handle unanticipated situations on election night. There were also speed limitations imposed by demands on the PC operating systems and Local Area Network (LAN) operating environments; systems could slow down under actual operating loads. Graphic design limitations were also imposed because the host computer-character generator interface was rudimentary. With further advances in PC technology, software, and graphics equipment, these problems are evaporating.

The result is that some of the once-custom software packages have evolved into commercially-available products which are configurable and not in need of customization.

In this paper, we will explore the technical and operational considerations faced by local broadcasters in selecting an election computer system.

THE COMPONENTS

Regardless of the particular hardware used to do the job, all election computer systems share the same basic components.

Host Computer

The host computer is the "brain" of the system. Its basic function is to run software which compiles, stores, and formats election data for on-line retrieval. The host computer accepts data from entry terminals, or via modem, and interfaces to graphic display devices for use on the air.

Terminals/Workstations

In most election systems, multiple terminals connect to the host computer.

The election system software operates terminals which serve the following users of the system:

1. Data Entry personnel are responsible for manually inputting election returns and exit poll data which are not entered directly via modem. The number of data entry terminals can range from one to twenty or more depending on the particular circumstances, and the volume of data which needs to be entered manually.

2. Political Analysts scrutinize the incoming data for trends, and project winners.

3.Producers use status displays on the terminals which often show multiple races at a glance. The producer must be informed immediately when a new winner is projected by the analysts. (Being first on the air with winner predictions is of the essence in local election coverage)

4. On-Air Display is usually controlled from a terminal. The on-air display program may incorporate features to allow preset sequences of races to be displayed without having to call them up individually.

Some systems bypass the need for an on-air display terminal by allowing recall to be initiated from the character generator keyboard. In this case, either the screens have been previously loaded into the character generator as a background process, or the CG keyboard data triggers a screen recall in real-time.

Graphic Display Devices

1. Character Generators are the mainstay of the graphics display devices used in election systems. Most current character generators incorporate a serial data interface (RS-232 or RS-422), which can be used to connect an election computer in one of two ways:

The first way to interface a character generator is by keyboard emulation. In this case, the host computer, which may actually be connected in parallel with the CG keyboard, "types" in the keystrokes necessary to put the names and numbers on the screen. This method has the advantage of getting the latest numbers on the air as quickly as possible.

In the past, keyboard emulation interface was a problem because the CG keyboard ports were not designed for rapid data transfers from a host computer. Many current CGs provide buffered serial inputs which are designed for computer interface.

The second method of interface is off-line entry. The host computer enters the data necessary to build updated screens into a separate data port, as a background process. The completed screens are recorded on the CG disk and recalled from the CG keyboard by the operator.

2. Electronic Still Stores are the second most common graphic display device used in with election computer systems. Still stores are generally used to access predefined background images over which CG text and graphics are added. As with CGs, a serial data interface is used by the host computer to control the still store.

3. Other devices, such as paint systems, video disk recorders, and production switchers, can be interfaced to an election computer for special applications, but within the scope of the local broadcaster, it is the character generator and the still store which predominate.

TYPES OF SYSTEMS

Various types of hardware have been used to implement election computer systems at the local level.

Mainframe

The use of a outside mainframe time-sharing service to implement the election computer system dates back to a time when this was the only choice available. While some of these systems may still be in use because of an existing software base, mainframes are on their way out, given current technology.

Minicomputer

Minicomputers are still seeing much use as the basis for election computer systems. The software has been around for a while, and minis are good at supporting the multiple terminals necessary to do the job.

The disadvantage of minicomputers is their high cost relative to PC-based systems whose computing power rivals that of a minicomputer.

Interestingly, some of the established base of minicomputer election software has been ported to a PC-based system.

Supermicro

This category of computers has been used as the platform for many data-driven real-time display applications. Supermicros handle multiple terminals and tasks, at a price which is less than a minicomputer.

The supermicros also have the disadvantage of being relatively nonstandard hardware for the local broadcaster. Unlike the PC-based systems, the cost of a supermicro-based election system cannot be amortized by using the hardware for other purposes throughout the year.

Internal to CG

In this category, the election computer system is built into the character generator itself. The advantage of this concept is a high degree of integration of with the graphic display aspects of the system.

The disadvantage of this concept is that the election computer system is tied to one particular brand of character generator, and it may not happen to be the brand you own. There is also more room for flexibility with the election software itself if the election computer is a stand-alone device.

PC-Based Systems

PC-based election systems are emerging as the replacement for all the previously mentioned hardware types.

The advantages of a PC-based system include low cost, commonly available hardware which is usable throughout the year, outside of elections.

However, it is only within the last few years that PC-based systems have become powerful enough to use as election computers.

One major problem is that special measures must be taken to allow PC-based systems to support multiple simultaneous users to access the election database. Several different approaches have been taken to solve the multitasking problem on PC-based systems.

1. Local Area Networks, or LANs allow multiple PCs to access the election database on a central file server.

Advantages of the LAN approach include distributed processing, minimal degradation, built-in provisions for file sharing, and a modular approach to addition of workstations. PCs which already exist at a station can be used.

The disadvantage of a LAN is a high cost per workstation. A separate PC is required for each additional user, and each of these PCs must be equipped with a network adaptor card and software. LANs can also be rather complicated for the end-user to configure.

2. Multiprocessor Systems use additional CPU cards that plug into the PC bus to allow for more users. While this method would in theory allow for faster data interchange than a LAN, the cost per workstation is higher than a LAN, and election computer systems are not that throughput intensive.

3. Multiuser Systems timeshare the CPU on the PC to service concurrent tasks or multiple users. Each user on the system uses a terminal, connected to the PC via a serial port.

The terminals are significantly less expensive per workstation than networked PCs. It is also relatively easy to connect with terminals at a remote data entry point by means of a modem.

Unfortunately, MS-DOS, the major operating system used by PCs is not capable of handling multiple tasks and users, and Election system developers have taken various approaches to get around this problem.

One way to get a PC to operate as a multiuser system is to use a multitasking operating system, such as Xenix. While Xenix is an excellent and very flexible operating system offering good performance, the difficulty here is that not many end-users are familiar with Unix-based operating environment.

Another approach is to use multitasking software, such as Multilink, which coexists with MS-DOS, and allows multiple concurrent users on terminals. The performance is not as good as with a multitasking OS, but compatibility with MS-DOS is maintained, simplifying development and implementation.

Regardless of the method used, any of the multiuser systems will suffer from degradation as the number of terminals on the system increases. Without adequate testing, one is at risk of having the system slow down with degradation on election night.

Some recent advances in PC technology may make the multiuser PC systems even more practical as election computers. For the new 80386-based PCs, multiuser operating systems have become available which are compatible with MS-DOS based software.

IMPLEMENTING AN ELECTION SYSTEM

What follows are some logistic and technical considerations involved in choosing and implementing an election computer system

Data Sources

The source of the election returns and polling data varies greatly in each case, depending on locale. The election system should handle the types of input you will actually be using, which could include:

1. Election Wire Services - Will the system interface to the service you need to use?

2. Keyboard entry of reports from poll watchers or other source - Will the system provide a sufficient number of entry terminals to handle the peak activity? Does the software provide data entry checks to prevent entering incorrect data or decreasing vote totals?

3. Dialup data from the Network news computer - Is this supported by the system?

4. Remote terminal input via modem - If remote terminals are used, is there security built into the system to prevent unauthorized data access?

5. Cumulative vs. Totalized Data - Will the system handle the mix of data particular to your location?

Data Analysis

The system should provide the data analysts with tabulated information from exit polls, to allow for winner projections - Does the software allow you to tailor the breakdown of data to your needs? Does the system allow the analysts to enter the winner projection into the database for on-air display?.

Producer Status Display

A particular consideration here is a status display which alerts the producer to new winner projections, or noteworthy new totals - Is there such a facility on the Producer Terminal?

On-Air Display

1. Real-Time vs. Off-line CG update - Be sure to consider the tradeoff between the two methods of CG interface in your particular application. Building screens in real time may is a good option, assuming you can dedicate a character generator specifically to election results. However, if you only have one character generator which must be used for election returns and for normal production supers, off-line entry is probably better from the logistic standpoint.

2. Still Store - If the election system is interfaced to the still store, contention between election graphics and production stills is also a logistical issue.

3. Sequencing - Does the software provide a means of sequencing groups of screens for on-air display?

Training

Does the supplier of the system provide adequate on-site training for the "administrators" and "users" of the election system? What support is provided when setting-up and testing the system? What support is available on election night should something unforseen happen requiring the supplier's attention?

Backup

Configuring an election system represents a sizable investment in time. Does the supplier provide a convenient way of backing up the data? Do you have a backup plan for displaying election returns in case a critical component in the system should fail?

Testing the System

The importance of testing the election system thoroughly before use cannot be stressed enough. Election systems have been known to "test out fine", only to exhibit problems under actual operating conditions. It is unfortunate that many bugs in election systems have been discovered in the heat of live election coverage, when it is often impossible to recover.

There are some specific recurring sorts of problems which can be brought out by testing the system in advance:

1. **Degradation** - The most common problem is that the system "slows down" when multiple users are accessing the system. Some degradation is inevitable, but the only way to evaluate the impact is by testing the system under a load.

2. Character generator screens that look fine in testing can suddenly go awry when actual numbers are inserted in the CG formats. Be sure the CG interface is tested with all possible combinations of numbers.(A particular problem are large numbers which exceed the tab field allotted by the CG and cause subsequent numbers to be displaced).

Rehearsals

One way to test the overall system is to stage rehearsals, and rehearsals may be the only way to test the system under load. However, rehearsals never have quite the impact of actual "battle conditions". It is often very useful to use the computer system in a less critical situation, such as a local primary leading into a major election. This can help to bring out any glitches in the system so that the major election coverage will proceed smoothly.

CONCLUSION

Improvements in PC technology, low cost, and the availability of off-the-shelf software packages have made PC-based election systems a viable option for local broadcasters. The computing power available to the local broadcasters allows the production of data-driven displays which would have been the exclusive domain of the major networks only a few short years ago. PC-based election systems will no doubt dominate the field in the future.

GRAPHIC PREPARATIONS FOR THE OLYMPICS

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The broadcast of the Olympic Games occurs every four years and is one of the major programming efforts by ABC or any television network. The Graphics portion of the Olympics is a large element of this effort and is treated as the most important project. Careful planning and preparation try to make this element as informative and artistic as possible.

Electronic Graphics are used in the modern major television program in several ways:

1. The graphic opening is the headline, the scene setter which has to gain the interest of the audience and introduce the show. For this reason, it is usually the most carefully conceived 3D animation using the highest and latest technology, effect and style. It is elaborately produced on the most sophisticated equipment, carefully crafted and usually rendered at the field rate to make the motion as smooth as possible. The ABC Olympic Openings, conceived by Roger Goodman, have traditionally been the most ambitious and artistic that technology permitted, and draw on major producers in the graphics industry.

Openings of individual Olympic events and introductions to locations are pre-recorded elements of a similar nature. They serve as chapter introductions and are only a step below the Olympic Opening.

2. The second function of graphics in the Olympics is to provide names and nationalities of the athletes, scores, performance times, distances, speeds, individual and team standings and statistics. Flags are used extensively to identify the nationality of the athlete throughout the event, even though they may originate from different locations and venues.

Many venues use rental trucks from all over Canada and the United States and have a variety of equipment aboard. Graphic fonts, logos and flags had to be prepared at ABC in New York to work with various manufacturers and models of equipment and to appear the same on final airing. These will be used for international distribution as well as shown in the United States.

3. The third major category of graphics are so called "Technical Pieces." They consist of stills and many animations in 2-D and 3-D which explain technical details to the audience, and thus make the viewing of Olympic events more enjoyable. They cover such items as: how do downhill skis, jumping skis and cross country skis differ; how is a bobsled constructed; what do judges look for in ice skating school figures; what is meant by a 70 meter and 90 meter ski jump; or how does a biathlon rifle look; what is the best path for a bobsled to enter a hairpin curve, etc. In many instances these questions can be explained much better in an animation or artistic still rendering than in a live shot of the event. We always started with the real thing for the artist to copy, and for a while the graphics room looked like a sporting goods store. Accuracy with simplicity is a must in the "Technical Pieces."

Of course, other graphics such as "Promos", "Stay Tuned", etc. are also used throughout the broadcast.

All these graphic needs required a year of planning and pre-production.

In today's tight economic environment, they had to be fulfilled for about one-half of the budget for previous Olympics with a substantial increase in quality and quantity. When ABC-TV in New York built its new Graphics floor in 1986, the prospective need for Olympic and Special Events programming was an important criterion of the design.

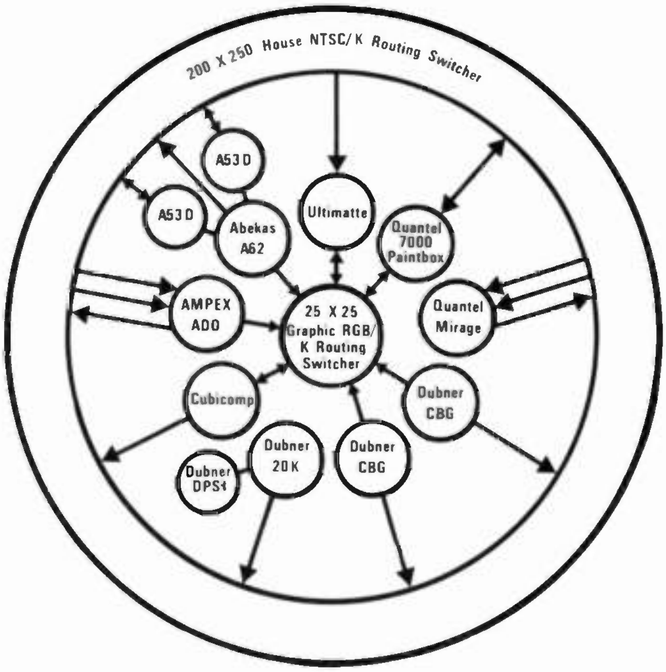
This type of programming requires a flexible combination of graphic facilities which are reconfigured for each event.

It was therefore decided to build a modular system which kept the rack equipment in a central environmentally conditioned equipment room, with all of the operating rooms located nearby for convenience of control and artistic and operational interaction. Connection between the two was created by designing a standard interconnect cable bundle. This cabling can be crosspatched in a custom master interconnect panel.

This system allows any graphic generator to be operated from any of thirty-eight different locations. Some of these locations are individual rooms of 9' by 9' size interconnected by sliding glass doors, some are located in control rooms when it is desirable to have the graphic operator in the production control room. The arrangement is completely flexible and responds to show needs. All video interconnection is via two routing switchers, one with two video levels: NTSC and key, and the second with four video levels: RGB and key. All are fully timed. Twelve of the operating locations are in 3 Multiple Equipment Smart System (MESS) rooms. These MESS rooms (affectionately referred to as Mess) were first conceived by Creative Director Goodman during the 1984 Olympics and were then expanded and fully developed for the 1988 games.

The Multiple Equipment Smart System (MESS) for the 1988 Olympic pre-production consisted of an Abekas A-62 Digital Disc Recorder with two A-53 Effects units, a Quantel Mirage with Floating Viewpoint and Starlight, a Quantel Paintbox, two Dubner CBGs with anti-aliasing, one Dubner DPS1 paint system, one Dubner 20K character generator, one Chyron 4100 character generator, one Ampex ADO, one Cubicomp with "race" accelarator and one Ultimatte.

The synergistic effect of this combination was truly astounding. It allowed the artists and operators to work in parallel on separate projects as well as together for complicated animations in 3-D, 2-D, rotoscoping, terrain modeling, stills, etc. The team, under the leadership of Peter Diaferia, functioned efficiently. Special software programs written by Katherine Dillon, allowed unique moves and effects on Mirage.



OLYMPIC MESS

The impact on the artists, engineers and programmers was an eye-opening, memorable lesson in how talented people can inspire each other and thus develop a higher level of performance. Even so, we believe we have much to do to carry the effort further, and in the words of Mr. Diaferia, we are "Ever so slightly past the Wright Brothers' Age of Computer Development in Graphics." Perhaps we are in the "Barnstorming Stage."

Each Olympic producer was assigned the MESS and crew to create the "Technical Pieces" for his or her event, as well as work for the overall Olympic program.

Some of the equipment used were "old friends" that we had worked with at ABC for many years, others like the Abekas A-62, Mirage with Starlight, Dubner 20K, the new Cubicomp, were new items, purchased for the Olympics. These required the usual "learning curve." Both formal training and practice time were assigned, allowing for operators to acquire familiarity and high skill levels. After the peak effort of the Olympics, we will train a wider group of other employees in the company. Thus the Olympics will act as a spearhead for improving not only technical facilities, but also personnel skills throughout the staff.

Some of the examples of work done in the Olympic MESS were: a three dimensional computer model of Mount Allan and Mount Canmore, the main downhill event areas.

The model was created with Mirage. Surface features were obtained from aerial photographs and adapted on Paintbox. This presented the base on which the various ski paths were then animated by the CBG-2, giving the viewer a concept of the locale and terrain where the events would take place. The effect was introduced as a folded map, which then unfolded and metamorphized into a 3-dimensional view of the mountain with a fly-over effect. All moves were combined and executed on the Abekas A-62 and then off-loaded to a 1-inch videotape. All software for the Mirage moves was written in house by Katherine Dillon.

A slightly different approach was used in the creation of a 3-dimensional animation for the bobsled/luge run. The presentation of this run was a very tricky project which was first tried by various means, none of which quite gave the effect the art director desired. We finally tried it on the Cubicomp 3-D animation generator which had just been upgraded to a Picturemaker 60R using a Compact 386 Computer with 387 math coprocessor and rendering accelerator (RACE) board. It had also been modified to have a simultaneous Alpha Channel output. This was important in speeding up the integration with Paintbox/ Mirage background in the Abekas A-62 Digital Disc Recorder.

A technique to achieve cleaner keying was developed. When creating the "foreground" frame over black, it was difficult to avoid a thin black outline around the key in the final signal. By painting in a base of an average value around the foreground, we could eliminate this black edge which cries out "key", and achieve a much more natural blend in the final picture.

For all the pieces created, the producer's need was first discussed with the art director and a preliminary storyboard developed. The storyboard was then refined with input from the artists, programmer and electronic graphic operators for the best way to achieve the desired results. The result, again approved by the producer or his or her representative, was then put into production. Further adjustments often had to be made when better opportunites presented themselves or difficulties forced modifications. Thus the final piece is usually a composite of ideas by all members of the team without losing sight of the original intent.

As in most ongoing productions, experience makes modifications desireable. In order to have a unified look for Olympic presentation some rework was required in the final product to take advantage of later developments.

A particularly effective method was found in making technical explanations in the style of a blueprint. This lent itself to showing the equipment as well as the athletic technique. The latter was created by rotoscoping athletes' movement from videotape clips by using the Quantel Paintbox and assembling the frames on Abekas A-62. It is a technique we had previously used on other shows where accuracy of movement was required.

An ABC development for the Olympics is the "Stamp Wipe." It is a video/wipe in any direction in the shape of a logo or pattern. The shape and video are generated in the Dubner CBG and played back by the 20K Animation Playback Generator, and the wipe is built into the Grass Valley 300 Switcher. Any shape which can be built as an animation can create the effect.

All of the equipment used in preparation for the Winter Olympics was sent to the ABC Calgary TV Center where it was further supplemented by additional similar equipment which had been shipped earlier in the installation period and was used to check out interconnection and distribution on site. Thus when the final shipment arrived it could readily slip into preconstructed slots and could be brought on line quickly. This resulted in a minimum interruption of the artistic process. The production team was able to continue in its endeavor to illustrate the 1988 Winter Olympics without losing its momentum.

At the Olympic venues, where the events take place, the remote vans are a mixture of ABC and rental units. They have Dubner CBG-2 or Chyron 4100 Character Generators aboard which do lower third supers, stats, performance times, judges' scores, time remaining clocks, etc. In order for these graphics to be uniform, all discs were prepared by the New York Font Create group. This is particularly important for the flags which are used with the athletes' names to identify the country they represent. At a previous Olympic game, one country's flag was depicted slightly wrong, almost causing an international incident. Also required are the sport disciplines, those little stick figure icons which identify each sport. Most important, of course, is the absolute accuracy of performance times and judges' scores. The official timing is done by Swiss Timing for the Olympic Committee. All scores are maintained by the official Olympic computer. A special interface was developed by Walter Bohlin of ABC to take these values and translate them directly to the graphics generator for television display without the intervention of an operator. Thus no errors in copying numbers can creep in and the latest official score is instantly seen by the audience.

The purpose of Graphics in an Olympic event is to enhance and explain the events presented as quickly and as clearly as possible. The assembly of a talented team of graphic artists, operators and programmers as well as the construction of an efficient graphic facility for their use is a challenge that comes to us only once in four years. I hope that we were able to meet this challenge and that the graphics presented during the 1988 Winter Olympics at Calgary were worthy of the spirit of these great games.

COMPUTER ANIMATION IN BROADCASTING
NEW DIRECTIONS FOR THE BROADCAST INDUSTRY

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Computer generated animation has found useful application in the broadcast industry where the pressures of deadlines, the demand for quality images and a desire for creative freedom are uniquely blended. Continuing development of computer software dedicated to graphics production is opening new possibilities for the animator, offering entirely new concepts such as behavioral animation along with faster, more easily accessed systems. Some of these developments are already commercially available, others have been demonstrated in research programs and will soon be in the hands of users. This paper discusses how these new developments are impacting the industry and how broadcasters can most effectively utilize the capabilities of computer graphic animation.

TECHNOLOGY ADVANCES

Briefly, the current state of the art in computer animation systems, can accommodate a wide variety of animation projects. Whether it's a short term, quick turnaround graphic for the news, or a very highly refined artistic network promo piece, the chances are that the same graphic workstation can be economically used to produce it. The difference in output relates more to the amount of time you have, the number of people available to work on it, how well the job is organized, and the difficulty of design content. The modern animation workstation is not the limiting factor in terms of quality. In fact the spectrum of products that can be generated on a single system is amazing.

Modern systems incorporate paint functions, 3D animation functions and supporting capabilities in one package. Earlier approaches often involved assembly of a number of components and software modules by the user. Current systems offer ease of use, flexibility of output, and are capable of high quality productions which enable broadcasters to approach a variety of projects with confidence.

However, experience tells us that every product or technology has some limitations and animation workstations are no exception. In the case of current graphics production equipment, there are two primary issues that constrict what can be done within desired budgetary and scheduling limits. The first of these, and the most often cited, is rendering speed.

By its nature, the rendering of graphic images is a voracious consumer of computer time. So much power and memory is required to render quality 3D animation on a computer that in the past only the larger processors could even tackle the assignment. Now that less expensive but more powerful computers and specialized rendering processors that increase their capabilities by orders of magnitude are becoming available, rendering is becoming more affordable and less of a scheduling bottleneck. And for those with really large rendering requirements, the availability of "render facilities" that are commercially operated, offer the power of multiple parallel processors on demand without the expense of ownership. The joint result of these two developments is that rendering is rapidly becoming a non-issue.

The second issue is a much less recognized problem: the design and specification of animation. The time and effort required to do this can be very large. The operator has to design all the components, every detail, of an object and in modern computer generated commercials there are many, many objects. The operator then must design the animation element by element, move by move. For sophisticated motion such as a scene with multiple human figures, this task can be very painstaking. It can involve literally hundreds of man-hours to produce a few minutes of finished animation. The technique of providing more workstations so that more operators can work simultaneously has its limitations as a solution. It is in this area of animation design and specification that the greatest potential for system improvement lies. What is needed is another means of telling the computer how objects are built, how objects are described and how animation is described. Without such assistance, the next order of magnitude of scene and animation complexity will not be achieved. Rule-based modeling and behavioral animation are recent developments that are addressing this problem.

Rule-based modeling enables designers to specify rules for the construction of an object, say a building or a tree, that permit the computer to generate any number of buildings or trees that have similar characteristics but are not identical. The rules specify how this is controlled at different sizes, different scales and different functions. The computer does the variations. Examples of this technique have been demonstrated in Computer Aided Design (CAD) situations showing that with rule-based modeling, variations in designed objects can be defined much more readily. Those techniques will migrate into animation modeling systems. The next step is to get objects to interact in that environment. A similar technique that specifies rules for how animation can occur offers one solution.

As a new technological development, rule-based animation, of which behavioral animation is one example, promises significant future benefits to animators. This concept uses intelligence within the animation system to allow users to specify animation at a higher level. Work already done at Symbolics demonstrates that instead of specifying paths of action for every object in a scene, the animator can define rules for the behavior of classes of objects. The animator thus takes a giant step away from the minutiae of designing animation, and instead defines the aggregate motion of a whole class of objects. This offers a remarkable payoff in human productivity. When the artist can richly express himself more quickly, the level of product can be produced at a given price within a given schedule is greatly improved.

Symbolics demonstrates how this object-orientated programming technique can be used in the animated film "Stanley and Stella in Breaking the Ice." In this film up to 80 fish and 80 birds appear in the more complex scenes. Explicitly animating the paths of up to 160 characters would be tedious at best, would be prone to errors such as collisions with other characters and with the surrounding scenery, and would be unlikely to result in a naturalistic representation of the motions of a school or flock. By encoding the behavioral characteristics of birds or fish in each of the "computer-generated extras," the individual character acts like a fish or a bird when told to take some generic action.

This technique should find application not only in animal character animation, but in human characters and other multiple-element images across a wide gamut of applications from robots to weather patterns. The potential is limited only by the imagination of the users. In this connection it should be noted that the real problems in computer-assisted animation continue to be the gaps between user and machine, between vision and image, between concept and product. The tools offered by the Symbolics Paint and Animation system, relying on a tagged virtual memory architecture, symbolic processing, and object-oriented programming, are providing one means of rapidly narrowing those gaps. Behavioral modeling can help to make animation more realistic, more detailed, and visually richer. But most important, the computer will free the artist to spend more time and creativity on the real problem of animation: creating quality dynamic art that communicates to other human beings.

With regard to quality, it is important to recognize that in an animation system, quality derives from more than just the technical specifications such as signal-to-noise ratio in the digital to analog conversion, or the resolution of the frame buffer being used to produce the image. Usually it has more meaning in terms of the design content of the piece and how well the production is organized to make maximum use of the facility and the people. Getting these things in order as a first priority means that technological advances such as we have just discussed can more readily be exploited for artistic and production gains.

ORGANIZATION AND CONTROL

Time constraints and airdate constrained deadlines are an everyday problem in graphics production for broadcast use. The solution is to avoid being caught by these immovable deadlines by carefully organizing the job and efficiently managing it. Moving the project from storyboard to on-screen implementation requires efficient controls that enable the producer to track changes in the concept until the job is completed.

None of the technical advancements in computers will get the job done on time if the job is not planned to begin with. When a tight deadline is imposed, operators are wise to make sure that the scope of the job isn't beyond what can be done in the allotted time. Having a talented designer as well as an operator in the process from the start will assure production design inputs that operators using advanced equipment can implement.

If the equipment and controls are in place and the designers and operators know how to use them, emphasis can then be directed to content. The potential is there to produce computer animated graphics that have warmth and human feeling. Remember that the sophisticated tools discussed above are still only tools. Human direction is needed using those tools to input emotion, character and personality into the production and fulfill the original definition of the word animate - to bring to life.

OPTICS PLUS COMPUTERS: THE GIANT LEAP FOR ZOOM LENSES

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In addition to the usual combination of optics and mechanics, Angenieux has developed the use of the computer inside the zoom lens. Certainly one of the most spectacular advantages is the possibility to sharply maintain focus at all distances, from infinity to the front lens element. This could not be done with any zoom lens before. It will considerably expand the director's operational flexibility. It is in fact the first of a new generation of optics whose astonishing possibilities we have begun to discover which will allow us to enlarge the field of Broadcast zoom lenses applications. The first lens is a versatile 40x, now in production since the summer of 1987.

This paper describes the latest state of the art in development for zoom lenses for Broadcast Television cameras.

SYSTEM EXPLANATION

1.- Conventional zoom lenses and minimum focusing distances.

What we require in the first instance of a long focal zoom lens geared to outside broadcast is to make images at extreme long distances. For that a high level of luminosity is required as control of the light level at these distances is not often possible. These requirements dictate large lens elements, typically up to 200 mm. in diameter.

In addition, because these long focal lenght distances must be highly chromatism corrected to obtain an image well defined, and that this aberration augments as the focal length increases, these front elements must be made out of low dispertion glass of low refractive index. As a result the diopter's curves are very important and therefore would have a high spherical aberration if we did not divide their power between several lens elements. Doing so, weight, volume and cost would increase.

Zoom lenses, so far, are usually based on the so called mechanical compensation system, as shown in figure 1.

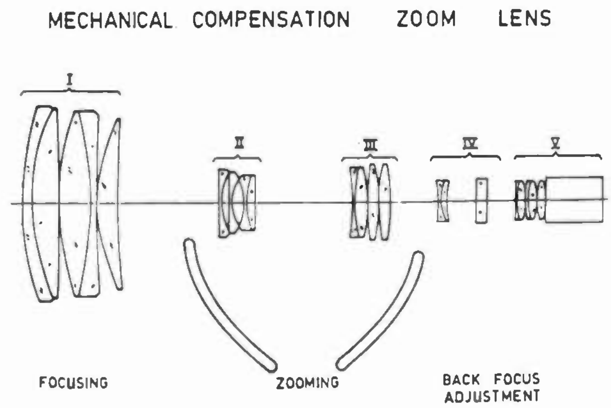
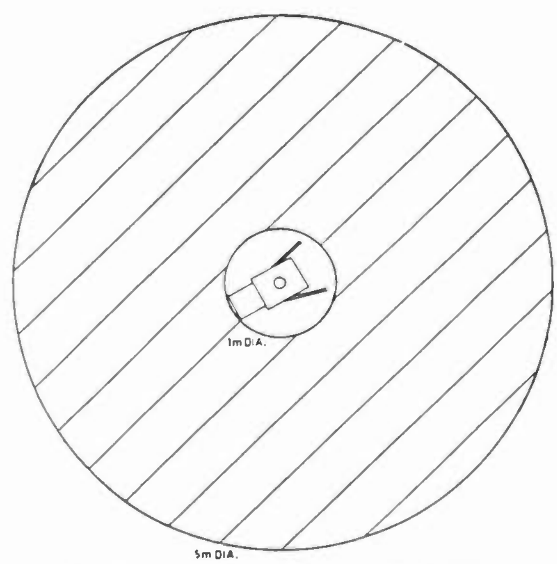


FIGURE 1

As a result minimum focusing distance for OB lenses is over 8 feet (2.5 meters). This added to the lenght of the lens and the distance of camera plate to axis of pedestal rotation creates a blind sphere of 20 feet diameter (6 meters).



TRADITIONALLY FORBIDDEN AND NOW PERMITTED ZONE AROUND THE CAMERA

FIGURE 2

2.- Overcoming the minimum focus limitation.

To reduce this restricted area another solution had to be found. The solution consists in moving certain lens elements, notably those in back of the iris whose position normally remains fixed.

GENERAL LAYOUT OF THE NEW ANGENIEUX ZOOM LENS

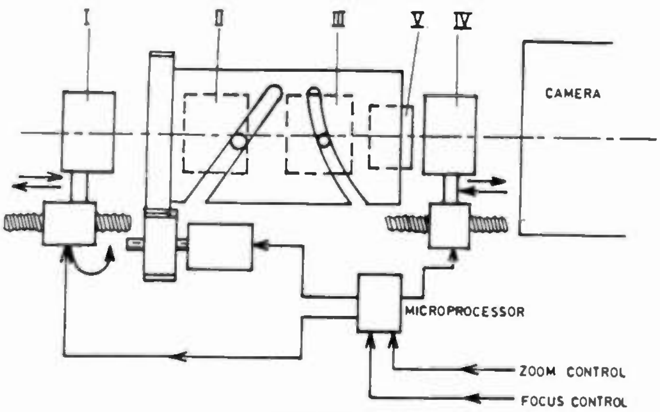


FIGURE 3

Under these conditions the lens does not operate with the mechanical compensation system and the position of elements is a function of both the focal length and object distance.

The position of each mobile element is measured by means of high resolution optical encoders. The computer then calculates and controls the necessary movements as a function of the desired focal lenght and focusing distance.

An important benefit of this concept is that it allows further correction of optical aberrations such as distortion. It also helps stabilizing the incident angles of light rays on the dichroics mirrors in the beam splitter.

Along with the task of managing the movement of the elements, the microprocessor allows other improvements such as displacement of the zoom converter, rotation of the turret's range extender and opening of the iris. It allows us to reduce to 0 (zero) the residual focusing variations because of mechanical tolerances when zooming occurs.

From now on, the microprocessor will allow the use of accessories such as wide angle attachments, shot-boxes with preselected and preprogrammed sequences... and also opens the possibility of communication between lens and camera.

3.- Operation principle:

The system is described in Figure 4.

GENERAL LAYOUT

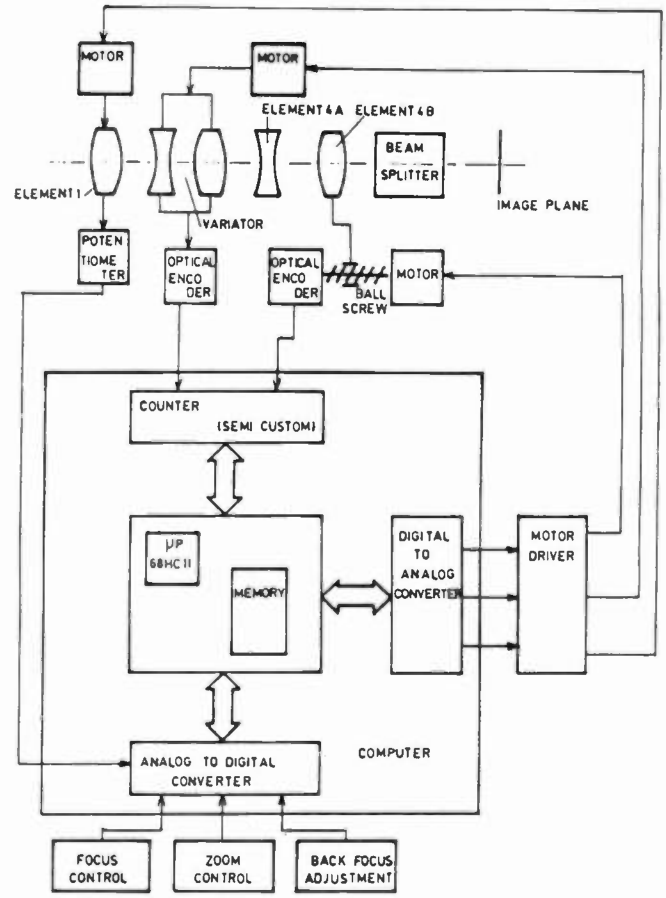


FIGURE 4

The position of lens groups is measured by optical encoders and fed to the central processing unit. It is compared to the desired values. An analog signal is then sent to the actuators.

Fail safe:

A mechanical cam has been left. In the event of an electrical failure flexible cables can always be used. The lens operates then in a conventional way.

4.- Lens characteristics and performances.

The 40x is an OB lens with full studio capacity as it is wide angle and can focus from front glass to infinity.

The specifications are as follows: (for 2/3" format).

- Focal lenght: 9.5-380 mm
  20-800 mm with standard built in 2.1x extender.

- Range: 40 x

- Horizontal angles: 1 1/3° to 49 3/4°

- Aperture: F/1.3 for F 9.5-195 mm
  F/2.1 for F 380 mm
  F/4.4 for F 800 mm

- Weight: 20 kgs

- Acoustical noise: less than 38 dBA (for full zoom range and full focus in one second).

- Field of view:

Distance	Smallest field	Largest field
0.1 m	75 x 100 mm	105 x 140 mm
1. m	75 x 100 mm	690 x 920 mm
2.5 m	19.5 x 26 mm	1.5 x 2 m
10. m	82 x 110 mm	6.9 x 9.3 m
1000. m	8.2 x 11 m	690 x 930 m

- MTF

F	9.5	50	380
center	90	95	70
corner	85	80	70

(maximum aperture, green channel, 5MHZ)

5.- Shutter for slow motion:

Among the many new possibilities offered by microprocessor in the lens is the control of a rotating filter in the iris plane, which acts as a shutter and is synchronized to camera scanning.

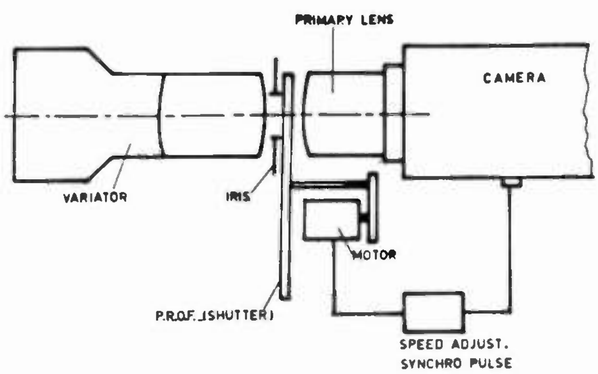


FIGURE 5

An optical opening in the rotating filter provides for the equivalent of a fast shutter speed of a photographic camera. The image is received by the camera sensor during 1/250th. of a second.

As the device is in the iris plane, all the points of the image are affected together, to the contrary of other systems placed near the image plane which can produce an undesired "curtain effect" for images moving at the same speed as the shutter.

6.- Achievement and perspectives.

The first lenses as briefly described here have been operating in the field since last summer when they were used in Rome for the World Athletic championship. They are now in regular production both for larger cameras and for EFP.

Based on the same principles we are now introducing a new lens at this NAB: a 20 x 8.5 for studio, OB and EFP.

We do believe this new approach development will lead to many further advances, some of them not having yet been thought of. We are looking forward to addressing this exciting new challenge.

angēnieux zoom 40 x 9.5 · f/1.3 for 2/3" cameras

THE ADVANTAGES OF USING VERTICAL INTERVAL TIME CODE OVER LONGITUDINAL TIME CODE

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INTRODUCTION

Video tape editing is the process of re-arranging the sequence of events on a video tape by re-record those events in the desired order. Thus you must define what portion of the source material you wish to re-record, and where that event is to occur on the final version of the tape. Next you must have a way to communicate the beginning and end of the source material to the computer that controls the video tape recorders in the edit system. Time code is one way to solve this problem. Time code gives each frame of video on the tape a unique identification or address. The counting sequence used to identify each frame of video is very similar to the sequence used to keep track of real time, but time code is only equal to real time when non-drop frame time code is being used with black and white video. This is because only black and white video runs at 30 frames per second. Drop frame time code attempts to rectify the difference between color video, which runs at 29.97 frames per second and real time. While drop frame time code is accurate enough to time the length of a segment of video on a tape to within approximately 60 milli-seconds; a time code generator running in the drop frame mode with color video as its reference will not keep acceptable real time over a period of more than a few days since the error grows at a rate of +75.442 milli-seconds per day.

The General Form of Video Editing Systems

All of the preceding is true for both Vertical Interval Time Code (VITC) and Longitudinal Time Code (LTC), however there are some very significant differences between LTC and VITC. The numbering sequence for editing systems that use LTC as their edit control code is placed on an independent audio track, while VITC is a part of the video. Figure 1 illustrates the general form of an editing system that uses LTC as the edit control code. Based on that diagram you would expect an editing system that uses VITC as the edit control code to look like figure 2. The LTC edit system reads time code from an audio track, while the VITC edit system extracts its time code from the video. However the user of the VITC system will be very unhappy with its performance because this system will encounter the "VITC Erasure Problem."

The VITC Erasure Problem

During a video insert the VTR will erase the previously recorded video, then record the new video in its place. Since the VITC is part of the video whenever the video is erased the time code is erased; rendering the time code discontinuous. On the record tape we now have the record time code before the edit, the source time code during the edit, followed by the record time code after the edit. For simplicity, instead of using the time code numbering sequence let us number the frames of video from the source 1 through 14, and the frames on the record tape 101 through 114 as shown in figure 3A. Then define the edit event with a source-in of 5, a source-out of 11, and a record-in of 105. After the edit is performed, the record tape time code numbering sequence will read "...103,104,5,6,7,8,9,10,11,112,113, ..." as shown in figure 3B. If on some future edit we try to use the same record-in point of 105, the edit controller will never find that address, on the other hand, if the tape is parked at 104 and we use 5 as the record-in point, once again the edit controller will be lost. Whenever this resulting discontinuous time code occurs it is termed "the VITC erasure problem."

The Original VITC Editing System

The first VITC editing system was patented in 1979 by Katsuchi Tachi, and assigned to the Sony Corporation. This system avoided the VITC erasure problem by requiring each frame to be identified twice; first with VITC and then with LTC having the same address. During the performance of a video insert the output of the LTC track is fed to a VITC inserter so that it can be added to the source video just before the insert is recorded, forming a type of feedback loop from the LTC out to the video in. The functional diagram from this patent is reproduced here as figure 4. The time code insertion is accomplished by compressing the normal period of the LTC waveform from one frame to one line and inserting the result into the vertical interval. Since the passage of the original VITC patent, the SMPTE has updated the LTC standard to include a different waveform specification for VITC. The specifications for the VITC waveform are shown in figure 5; while the corresponding specifications for LTC are shown in figure 6. The only effect this will have on the original patented process is to require a LTC-to-VITC converter to be installed between the record machine's LTC reader output (replacing the VITC reader or the SMPTE generator in figure 4), and the jam sync input of the VITC generator(which is labeled VITC inserting circuit in figure 4). In addition this same process must be duplicated when recording the original source material, otherwise the time code will be discontinuous between takes, or a generation of video must be lost adding VITC to the tape[1].

Since LTC cannot be read at very slow speeds or in the still mode; VITC was invented to allow an operator to identify the time code address of a frame located during slow motion or the still mode by simply pushing the mark-in or the mark-out button for the record or play VTR. The problem of reading time code during slow motion did not become a problem until C-Format VTRs brought these features to broadcasting[2]. There are two alternate methods of determining the address of a still frame, they are:

1) Burned in time code.

2) The control track mode.

Burned in time code is the process of superimposing the LTC on audio 3, into the active picture area while making a work print of the original. Thus when the tape is moving too slow for the LTC reader to resolve the time code directly, the operator merely reads the time code directly off the video monitor, and enters the address via the numeric key pad. The final cut is auto assembled from the edit decision list using the original tapes.

An edit system is said to be editing in the control track mode whenever it is identifying frames on a tape without directly reading time code. Most of the time code edit system currently in use, read LTC at or near play speed, and revert to the control track mode at or near the still mode. The use of burned-in time code requires too much time to make work prints, the old VITC editing system is too complicated, on the other hand the modern control track editing systems are very accurate they just don't offer the benefits of an edit decision list, or any of the many other advantages of using time code.

JUSTIFICATION FOR THE CHANGE FROM LTC TO VITC

There are many advantages to using VITC over LTC. Match frame calculations occur when an operator wishes to dissolve or wipe between a scene that is already on the record machine to a scene on the source machine. If the frame to be matched was recorded from a source that was playing at standard speed, then the address of that frame on the source machine can be found by using interpolation on the appropriate time code addresses. That is:

$$TC_{play\text{-}in} = (TC_{match\text{-}frame} - TC_{record\text{-}in\text{-}old}) + TC_{play\text{-}in\text{-}old}$$

Where $TC_{match\text{-}frame}$, is the address of the frame on the record machine, $TC_{record\text{-}in\text{-}old}$ and $TC_{play\text{-}in\text{-}old}$ are the respective in-points of the event that originally recorded $TC_{match\text{-}frame}$ and $TC_{play\text{-}in}$ is the in-point for the new event that will dissolve (or wipe) to another scene. When the original scene was recorded at non play speed then $TC_{play\text{-}in}$ will be a non integer at certain speeds which can cause a glitch at the edit point. In this case the above formula becomes:

$$TC_{play\text{-}in} = (TC_{match\text{-}frame} - TC_{record\text{-}in\text{-}old})\ PS + TC_{play\text{-}in\text{-}old}$$

Where PS is a fraction which represents the play speed. PS is greater than one for fast motion and less than one for slow motion.

On the other hand, if the source machine's time code were assigned to a different pair of lines in the vertical interval than the record machine; then there would be no need to delete the source machine's time code. During a normal edit, the VITC reader for the record machine would read the lines in the vertical interval that contain the record time code address, and during a dissolve (or wipe) the VITC reader would read the lines that contain the play time code address. The two time codes could be distinguished by counting lines in the vertical interval or by reading reel numbers in the user bits.

Another advantage of using VITC over LTC is that VITC has color field identification, which means you are no longer limited to frame edits as with standard LTC. However if you want to perform a real frame accurate edit (normal edit systems are only accurate to within +/- a frame) or even a field edit; then a video processor must be added to the video output circuit of the play VTR. This video processor must have the ability to determine the original color of the video, then strip off the chroma and add new chroma with the same color referenced to the new field.

The most profitable use of VITC over LTC, is the use audio 3 for something other LTC. For example if it were possible to use VITC as the sole edit control code, then there would be no need for the LTC on audio 3, and audio 3 could be used for the SAP or any other purpose. This would mean that a station in New York could broadcast a Spanish translation on the second audio program channel of the English on the stereo channels, that could increase its potential viewers by two-million people. By now I am sure you are all very familiar with the aural baseband specification for multi-channel television sound, reproduced here as figure 7. Of the stations that do broadcast stereo, few use the second audio program channel on a full time basis. Since most programs originate on video tape, and there are only two audio program channels available in the C-Format, there is normally no place to put this additional channel.

SOLVING THE VITC ERASURE PROBLEM

With the proper modifications to your video editing system, VITC can be used as the sole edit control code. These modifications can be divided into two basic categories:

1) The feedback solutions which are applicable to all helical scan video tape recorder's (VTR) but designed with E-Format broadcast video cassette recorders (VCR) in mind.

   a) The VCR solution.
   b) The time code generator solution.
   c) The time code reader solution.

2) The non-feedback solution which is applicable only to C-Format broadcast VTRs.

In this paper edit systems which conform to the specifications set forth in the Sony patent are referred to as the first generation of VITC editing systems and are distinguished from other systems by the presence of LTC in the feedback path. Edit systems from this authors feedback solution are referred to as the second generation and are distinguished from other system by the presence of VITC in the feedback path, while the non-feedback solution is referred to as the third generation of VITC editing systems.

VITC Editing System With VITC Feedback Paths

The feedback solution can be further divided into three categories as indicated above. However only the VCR solution will be discussed here. For a detailed discussion of the other two, see the paper I presented to the 129th technical conference of the Society of Motion Picture and Television Engineers, preprint No. 129-114 "A Method For Recording And Reproducing The Second Audio Program On A Standard C-Format VTR In A Stereo Environment." It should be noted that all of the VITC feedback solutions have a circuit topology similar to figure 8. From this illustration we can see that if we have a way of determining the address of a frame on the record tape, that address can be inserted into the source video so that the same time code is re-recorded. Even the sony patent can be approximated by this circuit topology if we replace VITC reader#2 with a LTC reader, who's output would be the "source of VITC pre-set" after conversion to VITC as described above. Figure 1 illustrates the corresponding LTC circuit topology. It should be noted that the most significant difference between LTC systems and, the first and second generation of VITC editing systems; is the presence of a time code generator. While the non-feedback VITC editing systems that make up the third generation have a one to one correspondence with LTC editing systems as displayed by figure 9.

The VCR Solution

In the VCR solution, modifications are made to the scanner of the record machine which permit it to provide the time code address of the next frame to be erased by the flying erase head. This information is called the pre-erase out. The VITC is extracted from the pre-erase video output and the feedback loop is completed when this VITC is inserted into the record machine's video input during the recording of a video insert. This system is displayed in figure 10, the pre-erase video output can be achieved in one of two ways.

Adding a Pre-Erase Head

By adding a video playback head to the scanner upstream of the flying erase head, then routing the output of this head to the playback circuit, so that the video on the tape can be converted back to base band video before it is provided as an output of the tape machine. In this way the time code on the tape is read just before it is erased. This pre-erase output is feed to a VITC reader where the time code is extracted as displayed by figure 10. The extracted time code is feed to the jam sync input of a VITC generator. The video input of this generator is the source VTR's video output. The video output of the VITC generator is the record VTR's video input. Thus we have created a feedback path which inserts a time code address into a frame of video from the source VTR which is the address that frame will have on the record VTR. The position of this pre-erase head is the same as the sensing head specified by Mr. Harry Kaemmerer in U.S. Patent #4,163,262 figure 3 heads 360 and 340 where 305 is the direction of rotation of the scanner. Figure 11A is a copy of that illustration, while figure 11B demonstrates the effect of this pre-erase head on the video tracks. However since Mr. Kaemmerer's invention is essentially a line accurate servo system it does not provide an output from his sensing head to the exterior of the VTR. The form of feedback loop is similar to the upper loop shown in the original VITC patent by Mr. Tachi, however no mention is made of a pre-erase output, so one is left to assume that a standard VTR is being used. With a standard VTR in this circuit the VITC erasure problem mention earlier will result.

Modifying the AST Head

Alternatively the pre-erase video output can be achieved by modifying the AST head so that this head is capable of scanning the track adjacent to the track that is being scanned by the flying erase head. On an E-Format VCR this would require the piezoelectric deformation to be towards the top of the scanner, while a C-Format VTR would require deformation towards the bottom of the scanner. These differences are required because tape motion in one format is in the opposite direction of the other format. The location of the adjacent track can determined by first initializing the AST head to the same track as the record/reproduce head. Initialization can be verified by comparing the output of these two heads for congruency. Next the output of the AST head is checked as the base of the head deforms towards the desired track. This head should first reach a minimum at the guard band, then reach a second maximum on the desired track. It should be noted that the confidence playback capabilities of the VTR are sacrificed with this approach.

THE NON-FEEDBACK SOLUTION--PROCESS OVERVIEW

The non-feedback solution has only one implementation. However it is the author's opinion that this solution is far superior to the feedback solution. The non-feedback solution utilizes a feature which is unique to the C-Format. The specifications for the location of recorded information on C-Format video tape, is defined ANSI (American National Standards Institute) C98.19M-1979. Figures 12A and 12B are reproductions of illustrations from that standard. The unique video information is encoded on the long diagonal tracks located at the center of figure 12A, and is labeled video. A sixteen line segment of the vertical interval is recorded on the short diagonal track located on the lower portion of the tape and is labeled sync.

IF the VITC is constrained to the ten line segment of the vertical interval between lines 5 and 14, it will only appear on the sync track, see Figure 12B. By doing this we have created a dedicated time code track for VITC in the same way that audio 3 is a dedicated time code track for LTC. However, we must further constrain the VITC to lines 10 through 14 so that it does not interfere with vertical retrace, otherwise it will be seen on the monitor as a white or gray flash in the active picture area, and some monitors may even lose vertical sync.

As illustrated in figure 13, this is because a logical one in VITC is well into the white or gray area of the luminance portion of the video waveform. Line 10 through 20 are located at the top of the screen out of view. Line 10 must also be avoided in systems that use Grass Valley Group video production switchers, since these switcher switch on line 10 and clamp the signal at line 10 in the process. When performing a video insert we merely turn off the sync head in the same way that audio 3 is off during an edit in systems that use LTC as the edit control code.

The Non-Feedback Solution--Hardware Overview

The apparatus required to make the concepts of section above materialize are as follows:

A. Video and sync head input and output switching.

B. Independent video and sync head insert control circuitry.

C. Circuitry which will allow the VTR to record video and read VITC from the tape simultaneously.

Item A above deals with the normal operation of the video and sync heads. As illustrated in figure 12A, the video head begins each field at the lower left end of a particular track and ends at the upper right portion of that track. Then that head loses contact with the tape, circles around to the lower half of the tape to begin the next track. The information missed while the video head is not in contact with the tape is recorded by the sync head. Therefore at least one head is in contact with the tape at all times. Thus a means by which the incoming video can be routed to either the sync or video heads at the proper point in the recording cycle, is required.

The information contained in the sync track is so repetitious that the heads used to erase, record, and reproduce these tracks are never provided as part of a standard C-Format VTR. Since this information is never recorded; there is a gap in the video playback waveform called "format drop-out." A special type of sync generator replaces the video output at the format drop-out during playback. In fact I am told the only reason a sync track exist is that certain members of the original committee which wrote the specifications for One Inch C-Format Helical Scan VTRs thought it would be wise to provide a space for the information missing from the format drop-out in case some future situation requires the entire waveform to be recorded and reproduced.

Item B from the list above deals with the ability of a C-Format VTR to control the signal going to the sync head and the video head independently of one another. This is crucial if the VITC erasure problem is to be avoided. The Ampex VPR-2 and the VPR-3 both have switches labeled sync; however, the manuals for these machines indicates that the sync switch is not applicable to the NTSC television system. The European standard for the C-Format allows a fourth audio track to be added to the tape in the area normally allocated for the sync head[3].

Item C from the list above relates to the problem of reading the VITC from the sync track while recording new information on the video track. In machines equipped with confidence playback the confidence playback output can be used to monitor the VITC.

References

1 It would be possible to avoid losing a generation of video if the VITC could be added as a line insert edit, however at time of this writing none of the VTRs available are capable of performing such an edit. It is this author's opinion that a standard VTR must be modified to include a line accurate servo like the one invented by Mr. Harry Kaemmerer (U.S. patent #4,163,262 assigned to AT&T) instead of the field accurate servos currently being used, if a VTR is to be capable of line insert edits. In addition, high speed switching must be added to enable the VTR to erase and record single lines instead of frame(s); the details of these modifications are beyond the scope of this writing.

2 A New Edit Room Using One-Inch Continuous-Field Helical VTRs, by William C. Nicholls, The Journal of the SMPTE, volume 87 number 11 November 1978.

3 Videotape Recording, page 73; by Joseph F. Robinson, Focal Press, London and Boston.

AURAL BASEBAND SPECIFICATIONS FOR MULTI-CHANNEL TELEVISION SOUND

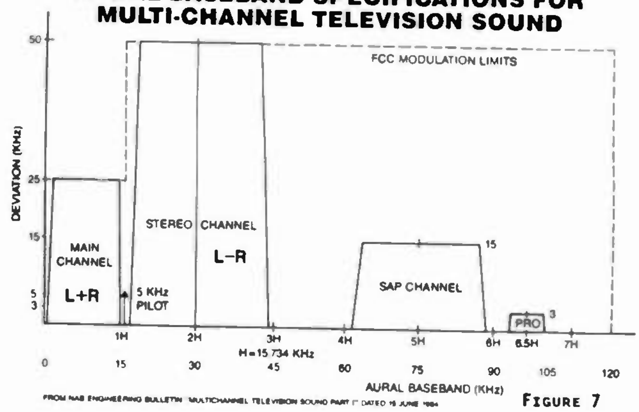
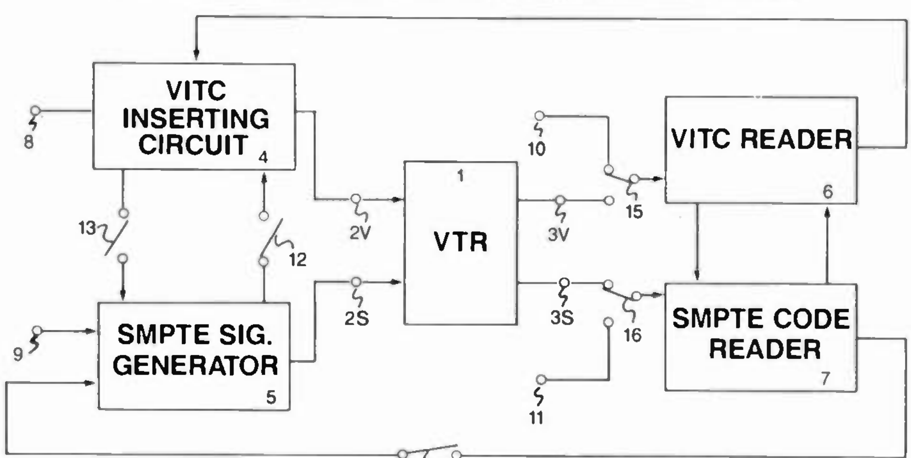


FIGURE 7

FUNCTIONAL DIAGRAM OF THE ORIGINAL VITC EDITING SYSTEM
(Invented by Katsuchi Tachi, assigned to Sony Corp.)



FROM SHEET 5 OF 5 FIG. 7 OF U.S. PATENT #4, 167, 759. SEPT. 11, 1979.

FIGURE 4

GENERAL FORM OF A LTC EDITING SYSTEM

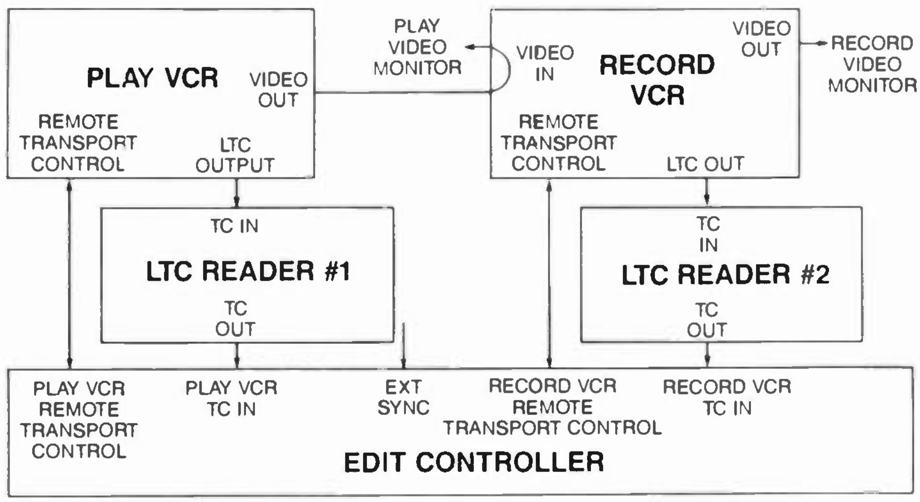


FIGURE 1

GENERAL FORM OF A VITC EDITING SYSTEM

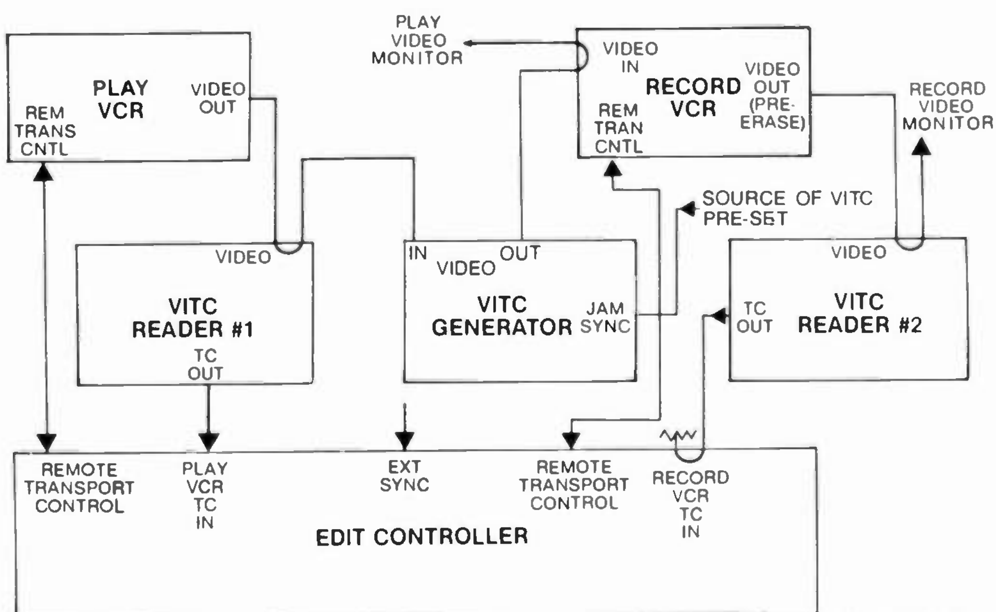


FIGURE 2

Discontinuety of the VITC Numbering Sequence Caused by a Video Insert

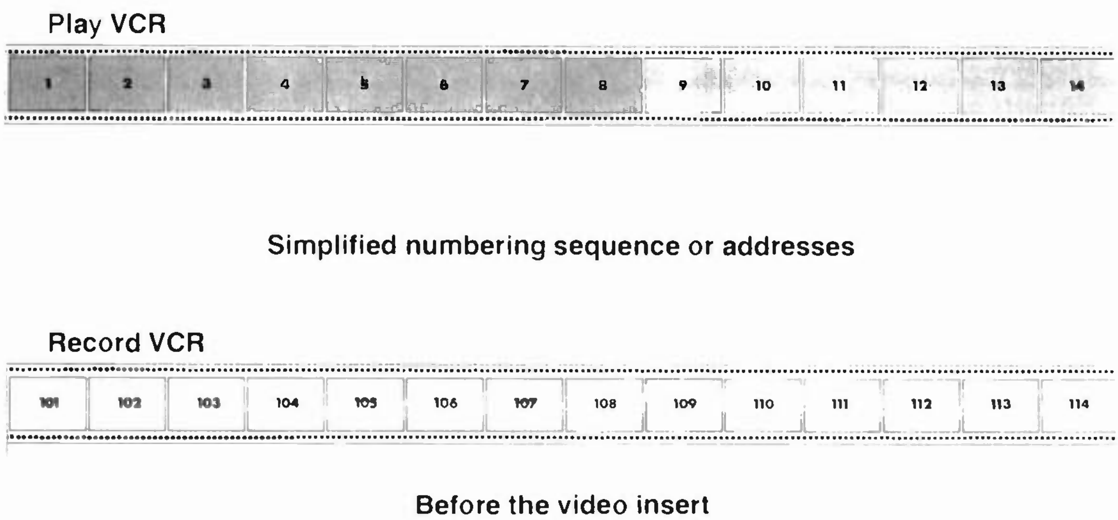


FIGURE 3A

Discontinuety of the VITC Numbering Sequence Caused by a Video Insert

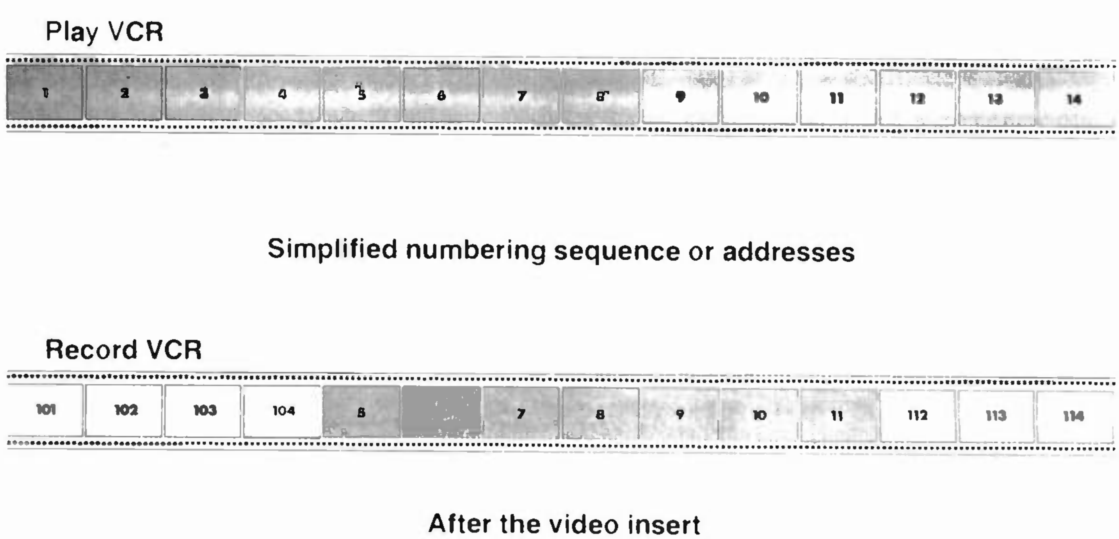


FIGURE 3B

VERTICAL INTERVAL TIME CODE WAVEFORM

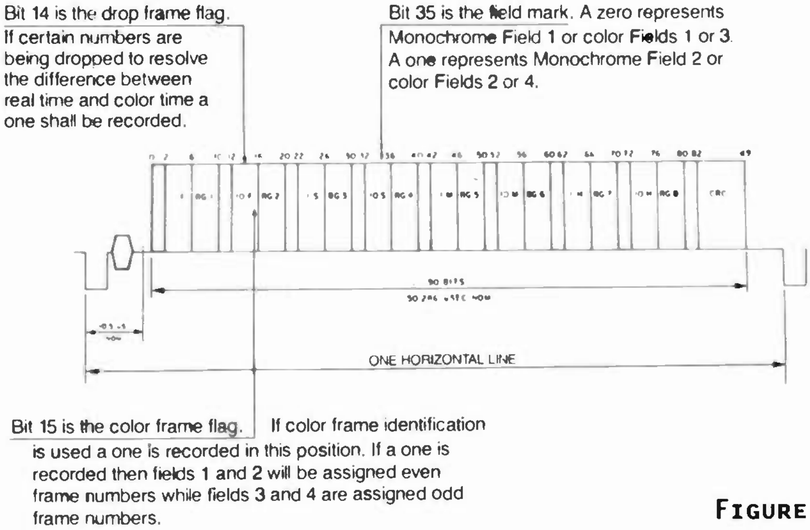


FIGURE 5

LONGITUDENAL TIME CODE WAVEFORM

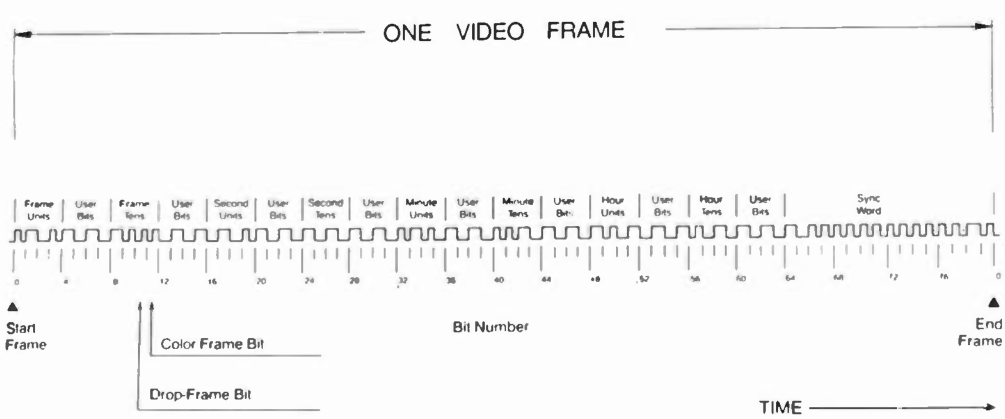


FIGURE 6

VITC EDITING SYSTEM FOR 1 INCH C-FORMAT VTR'S

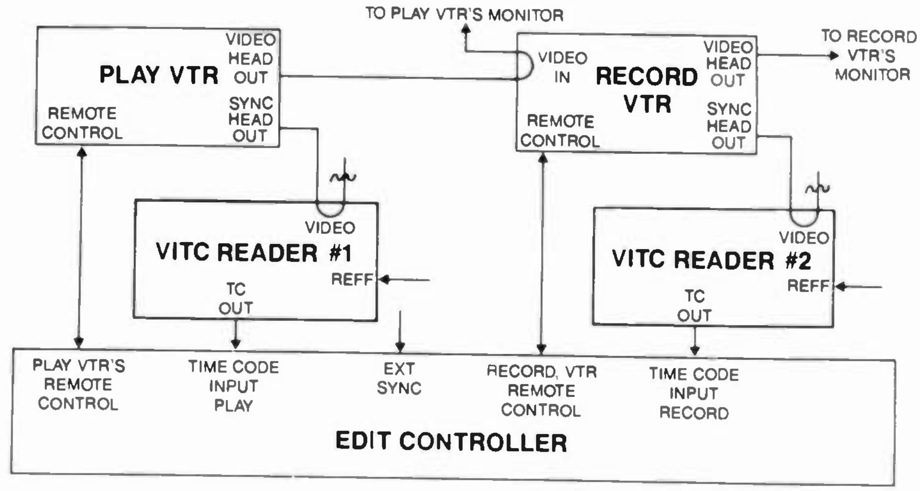


FIGURE 8

GENERAL FORM OF A VITC EDITING SYSTEM

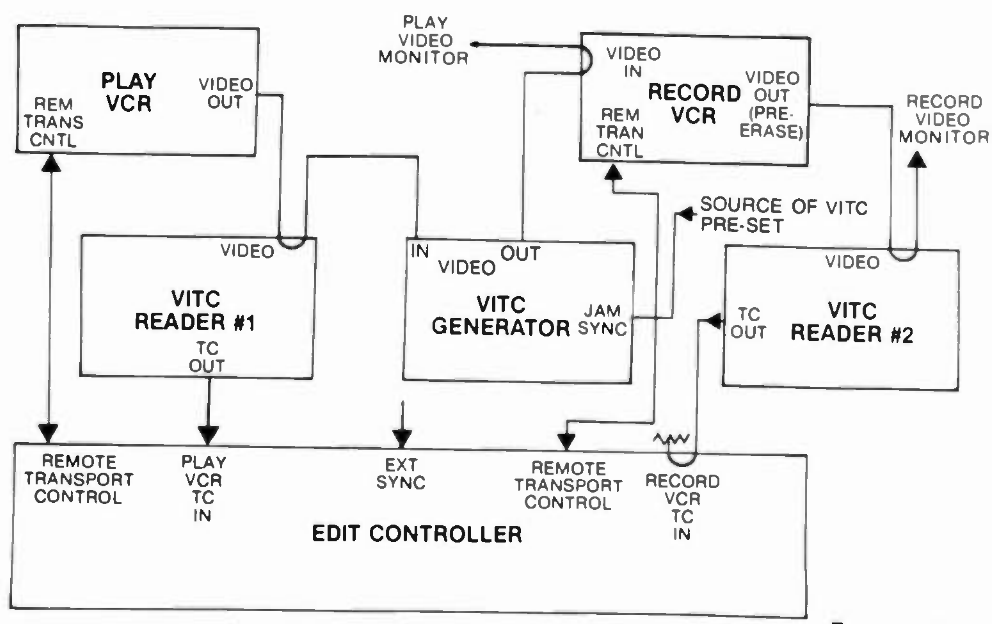


FIGURE 9

HEAD POSITION ON SCANNER

340, 360 POSITION OF PRE-ERASE HEAD OR SENSOR HEAD

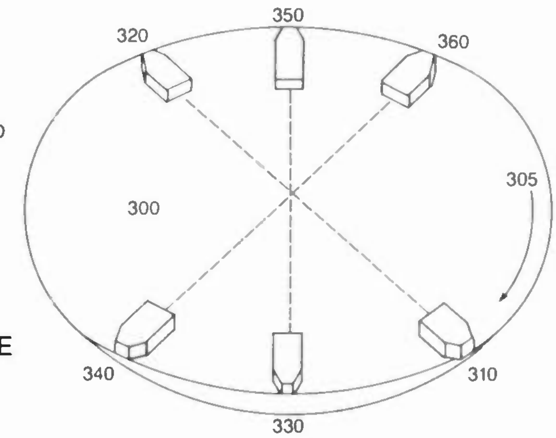
330, 350 POSITION OF ERASE HEAD

310, 320 POSITION OF RECORD/ REPRODUCE HEAD

300 SCANNER

305 DIRECTION OF ROTATION OF SCANNER

FIGURE
A



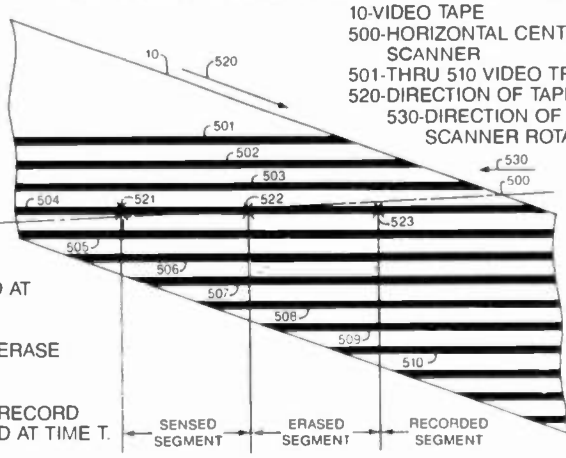
U.S. PATENT 4,163,262 JULY 31, 1979; FIG. 3 & 4 SHEET 2 OF 3

FIGURE 11A

POSITION OF HEADS ON TAPE WITH RESPECT TO PRE-RECORD VIDEO TRACKS

FIGURE
B

10-VIDEO TAPE
500-HORIZONTAL CENTER LINE OF SCANNER
501-THRU 510 VIDEO TRACKS
520-DIRECTION OF TAPE MOTION
530-DIRECTION OF SCANNER ROTATION



521 POSITION OF PRE-ERASE HEAD OR SENSOR HEAD AT TIME T.

522 POSITION OF ERASE HEAD AT TIME T.

523 POSITION OF RECORD REPRODUCE HEAD AT TIME T.

U.S. PATENT 4,163,262 JULY 31, 1979; FIG. 3 & 4 SHEET 2 OF 3

FIGURE 11B

VIDEO AND SYNC RECORD LOCATIONS

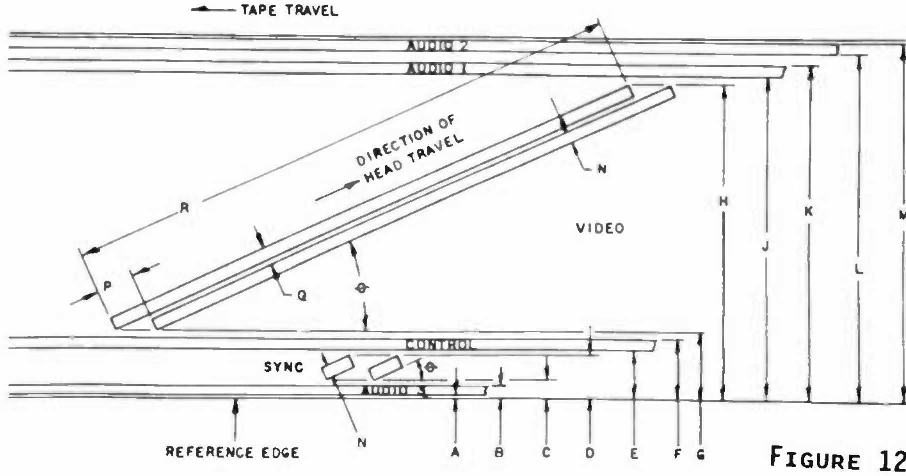


FIGURE 12A
C98.19M-1979

RECORD LOCATIONS AND DIMENSIONS

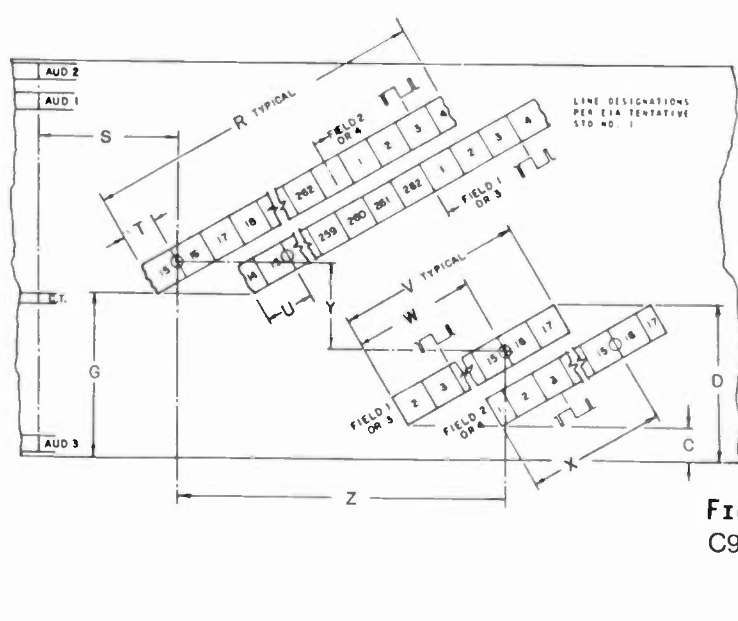


FIGURE 12B
C98.19M-1979

VITC EDITING SYSTEM UTILIZING MODIFIED VCR

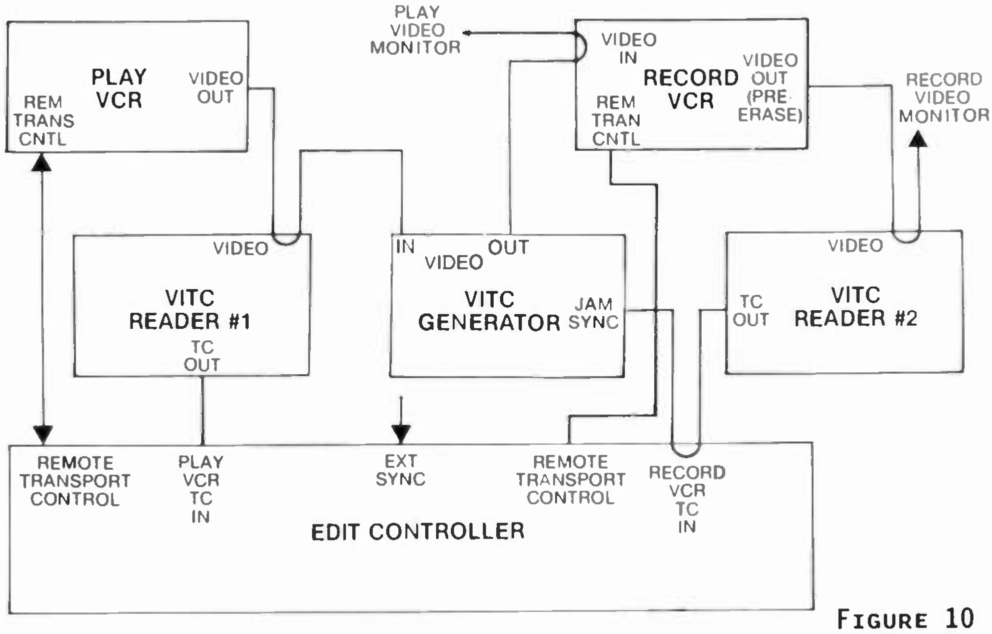


FIGURE 10

WAVEFORM SPECIFICATIONS FOR VITC

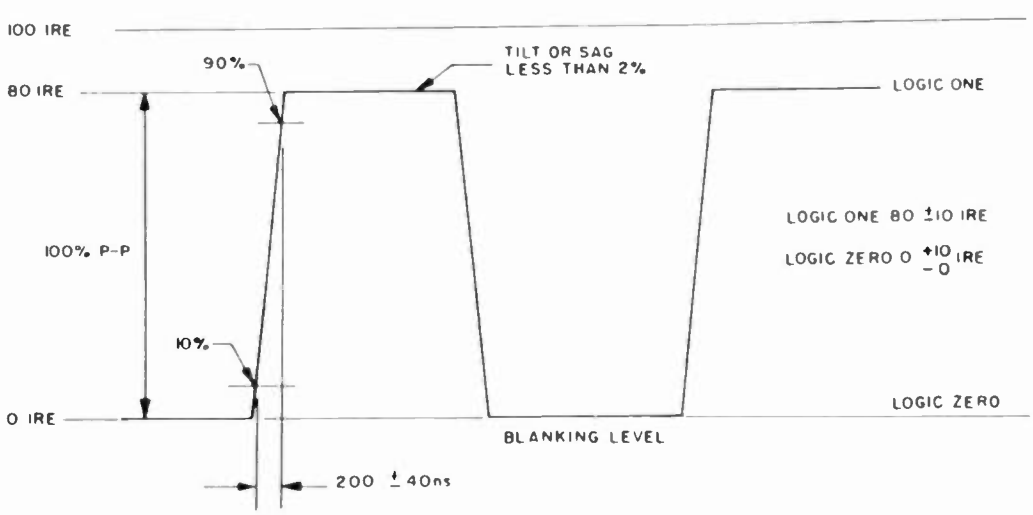


FIGURE 13

A NOISE REDUCTION SYSTEM FOR NTSC COLOR TELEVISION LINKS

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A new technique that provides significant noise reduction for color NTSC signals transmitted via FM links is described. This technique dynamically and separately modifies the video level of both luminance and/or chrominance so as to always take maximum advantage of the channel bandwidth and the transmitter FM deviation.

The color NTSC source signal is analyzed, picture element by picture element, by means of a complex signal transformation and arithmetic calculations, then instantaneous but independent video level adjustments are applied to both luminance and chrominance. The modified signal still has the NTSC protocol and still remains within the maximum luminance and chrominance levels specified for NTSC, yet it contains much higher video RMS energy.

At the receiver, the opposite process is performed, the video energy is reduced back to the level of the original source signal, but in so doing, any link-introduced noise is also reduced. The system is capable of providing better than 6dBs of noise reduction with most pictures. This opens up the possibility of carrying two video signals though a single standard satellite transponder.

This technique is also suitable for FM recording systems, and it can also be used with AM links, albeit with less startling results.

INTRODUCTION

Satellite links and other FM links for video contributions and distribution are often operating in adverse environmental conditions and/or power limited conditions resulting in marginal or relatively poor performance in terms of the received picture signal-to-noise ratio. It is not unusual to find that a given FM link will often deliver pictures with unweighted S/N's of only 40 to 45 dB's or even worse. This is particularly true of ENG links such as RADET.

S/N's of 40 to 45 dB's result in pictures having marginal quality. Pictures with S/N's in the 30 dB range exhibit very poor quality.

Unfortunately, conditions which result in poor link performance cannot always be avoided. They may be due to weather conditions, antenna size and power available, receiver sensitivity, and, in the case of ENG links, on distance and terrain between transmitter and receiver.

To remedy sub-par performance under the above conditions,one could double the transmitter power, or double the receiver sensitivity for each 3 dB of desired S/N improvement. With a satellite link one could also consider doubling the power of the transponders. All of these approaches are expensive and, at present, quite impractical. They would require very large investments to upgrade terrestrial equipment and/or millions of dollars for more powerful satellite transponders.

At the other extreme, one may find that some links are extremely good, delivering excellent quality video signals with plenty of power margin. They are so good in fact that one may feel they represent overkill, and thus an inefficient utilization of the channel.

Under those conditions, one may wish to consider transmitting two video signals through a single satellite channel,(i.e. one could cut FM deviation in half, back off power by 3 to 5 dB's, and squeeze two signals through one channel). The penalty of course would be a 9 to 11 dB S/N reduction for each signal. That could be enough to reduce the quality of the received pictures to a marginal or unacceptable level.

Both scenarios can greatly benefit by the use of a system that provides S/N improvements of 6 dB's or better and thus permit one to either improve the quality of marginal links to acceptable levels or transmit two video signals through a single channel with minimal quality losses.

SOME TECHNIQUES TO REDUCE NOISE IN FM LINKS

A number of methods have been used or proposed to improve the performance of FM links. Virtually all of them pre-emphasize the high frequencies before modulation and apply a complementary de-emphasis after demodulation. Either linear or nonlinear methods have been used with some degree of success. It is well known that the luminance signal of TV pictures has most of its energy concentrated at low frequencies. On the other hand, most noise introduced by an FM TV link is at the higher video frequencies. The received FM signal noise spectrum, even after the standard FM receiver de-emphasis filter, exhibits an essentially triangular noise spectrum for frequencies above 1 MHz. Thus it appears possible to apply additional pre-emphasis to high frequency, low level luminance, and complementary de-emphasis at the receiver to obtain improved S/N's. This type of pre-emphasis is normally non-linear which means it generates harmonics which are often beyond the original signal bandwidth and thus do not go through the link. It results in video distortions. They are normally not visible or not objectionable for a luminance signal, however, when one deals with color NTSC signals, they would be completely unacceptable in the chrominance signal. Any distortion of the chrominance signal translates into hue or saturation errors which are very visible.

Some present solutions employ separate component transmission, e.g. the different MAC systems. However, when one deals with color NTSC video links, it is not desirable to change the video protocol. Since in a color NTSC signal the chrominance is located at the upper video frequency band, just where the noise is highest in FM links, even being able to provide a significant improvement in the luminance S/N by means of any technique, the improvement is masked to a large extent by the chrominance noise.

To be effective then, a noise reduction system for FM links must provide significant noise improvement for both luminance and chrominance.

Another well known technique to achieve noise reduction, which is effective for both luminance and chrominance, is based on the use of recursive filters utilizing a frame store. They are known as digital noise reducers. Such filters however do introduce some artifacts, particularly motion artifacts and the "dirty window" frozen noise effect. They are not therefore the panacea they were first believed to be. Another technique employes multiline comb filters for noise reduction. It provides a significant noise reduction when measured on a noise meter, but it also changes the character of the noise to low frequency vertical noise, resulting in a very marginal subjective improvement.

We believe we have found an effective and useful solution for improving FM links S/N performance with NTSC color television signals.

A SOLUTION FOR NTSC COLOR SIGNALS

A basic block diagram of the system is shown in figure 1. A pre-processor is used ahead of the transmitter to maximize the luminance and chrominance levels, consistent with the NTSC signal protocol.

Optimized filters are used to reversibly separate the high frequencies from the low frequencies as well as the luminance from the chrominance components of the composite NTSC color signal. The signal is analyzed, in the digital domain, picture element by picture element. By means of a signal transformation and complex arithmetic calculations, the instantaneous actual RMS value of the signal is determined, (see figure 2). This information is then used to apply an appropriate signal gain, on a near instantaneous basis, to the video. This maximizes video deviation at the transmitter, without ever exceeding the maximum permitted level.

Dynamic level optimization is applied to both luminance and chrominance separately and independently. A code identifying the pre-processor operating mode is inserted in vertical blanking to enable the receiver post- processor. A most unique feature of this system, absolutely essential for chrominance pre-emphasis, is that the dynamic level optimization, although a nonlinear function, is linearly applied to the chromance signal. One could think of the process as being carried out by a superhuman, very fast operator, looking at the chominance signal on a color subcarrier cycle by cycle basis, and riding a chrominance gain pot up and down as an inverse function of the incoming chrominance level. This prevents most non-linear distortions of the color subcarrier,which could result in phase errors.

The output signal is a composite NTSC color signal which, from the viewpoint of the transmitter and receiver, is a standard NTSC color signal. This signal however will exhibit, if viewed on a TV monitor, a pronounced enhancement of low level high frequency details and increased saturation of low level colors.

At the receiver, a similar but complementary process is applied to the NTSC color signal. That restores the picture to the original levels. However, as luminance and chrominance levels are dynamically reduced to their original values, any noise introduced by the link is also dynamically reduced. In general, greater than 6 dB signal to noise improvements are possible with this technique, without introducing objectionable picture distortions.

A 6 dB improvement in S/N means that a channel that is presently capable of delivering excellent quality pictures could carry two pictures instead of one with acceptable quality.

A 6 dB improvement in S/N means that links that presently deliver marginal picture quality will be capable of delivering good picture quality.

A 6 dB improvement is equivalent to a four fold power increase in a satellite transponder!

All of this can be obtained by adding an inexpensive pre-processor between the NTSC signal source and present FM transmitters, and a complementary post-processor at the output of the FM receivers.

CONCLUSION

We have developed a technique that dynamically modifies the video levels of both the luminance component and/or the chrominance component of an NTSC color signal so as to always take better advantage of channel bandwidth and deviation in FM video links.

A unique feature of this system, particularly important for chrominance, is that dynamic video level changes are determined by the actual input signal level within a given picture area, and thus it avoids chroma phase and/or saturation errors.

The application of this technique to video FM links, as well as to FM recording systems, can provide better than 6 dB of signal to noise improvement in both luminance and chrominance.

The video processing system that utilizes these principles is now being manufactured by Intelvideo.

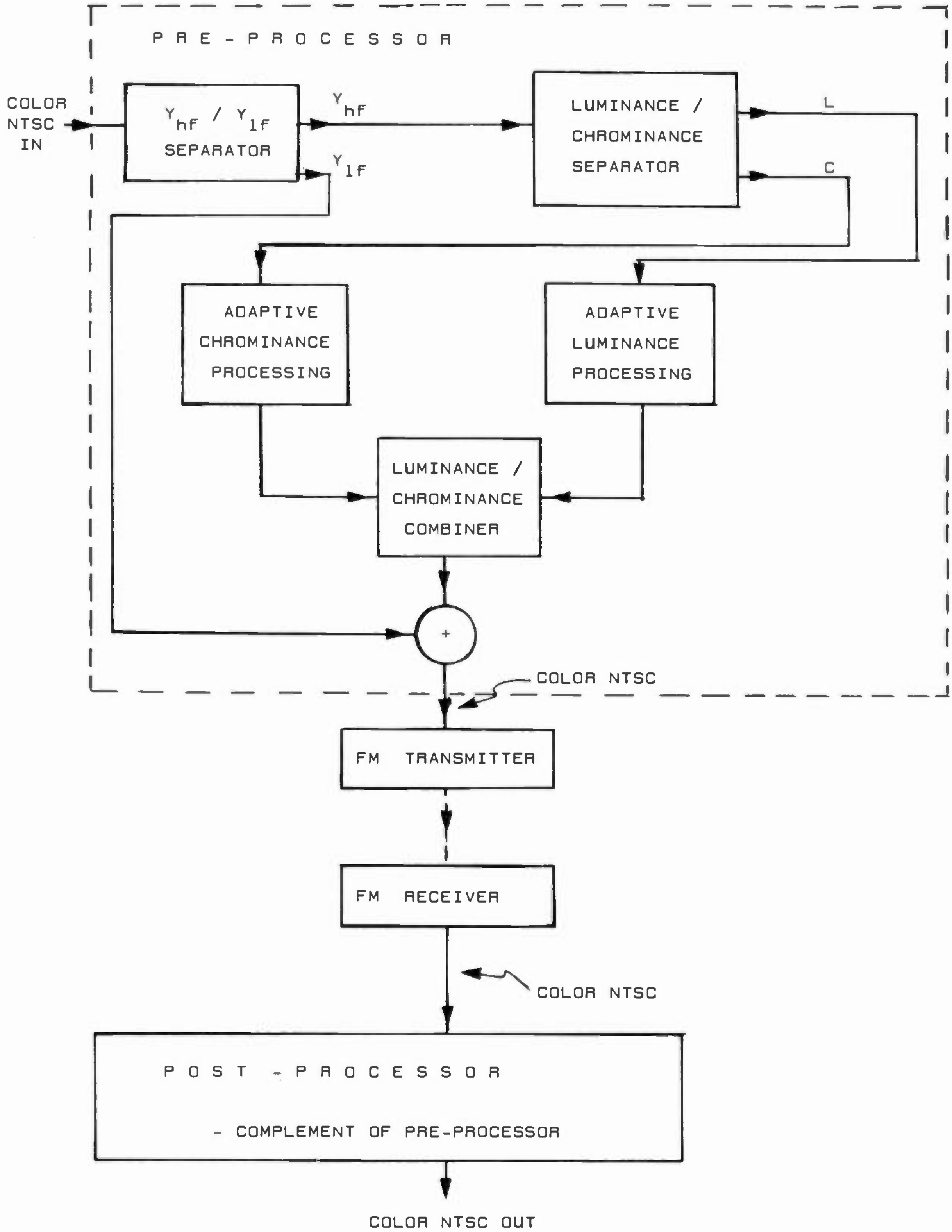

Figure 1. Block diagram of Adaptive FM Noise Reducer for NTSC Color Signals.

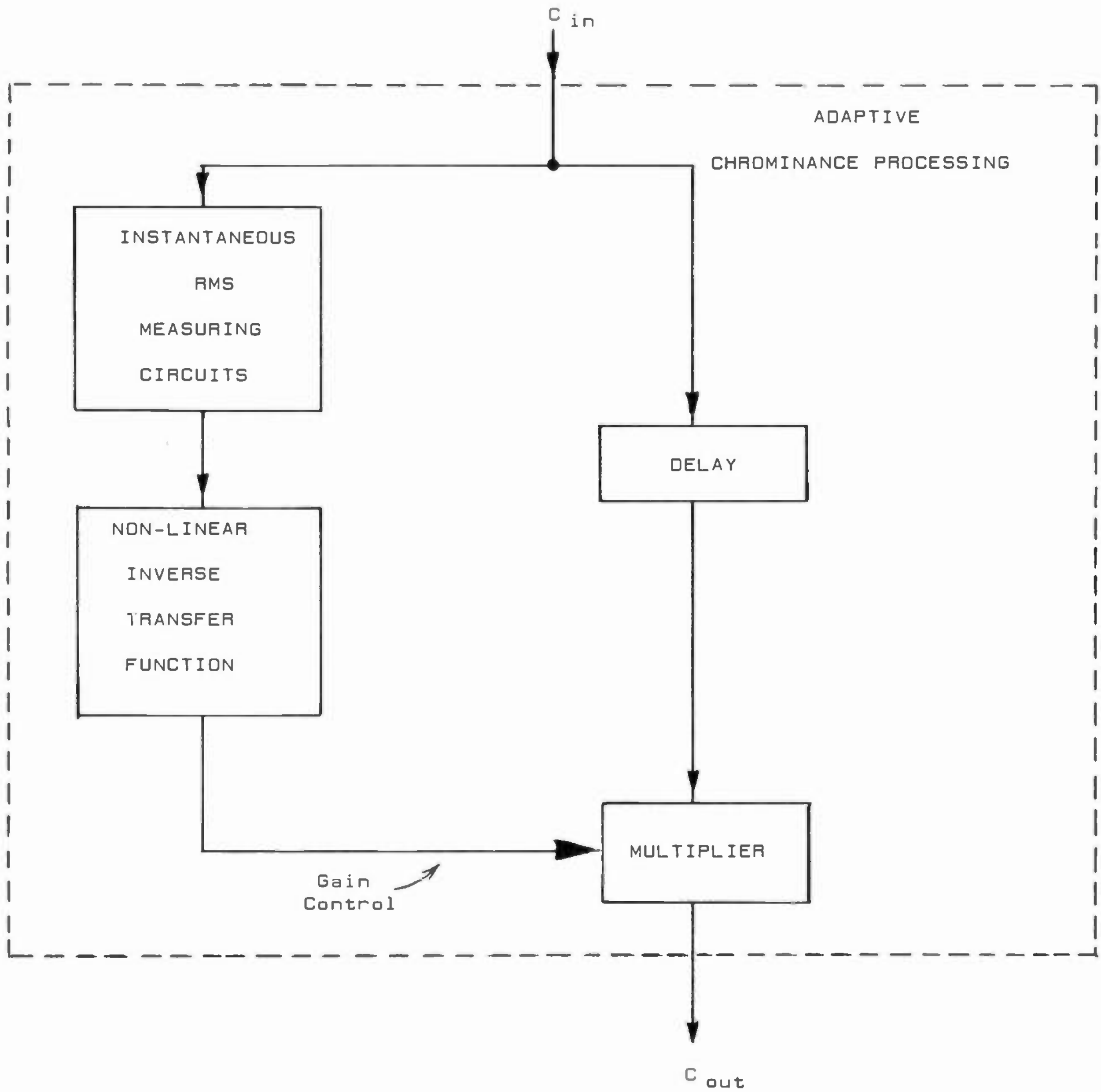

Figure 2. Block Diagram of Dynamic Chrominance Level Optimizer

A DIGITAL AMPLITUDE MODULATOR-TRANSMITTER

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Amplitude modulation is the oldest form of impressing information onto an "electromagnetic carrier" wave of energy with its beginnings taking root in the early days of spark. With spark came 100% amplitude modulation with the adoption of the Morse Code and, later the International Morse Code. And still later, voice amplitude modulation which has come to be known simply as AM. AM usually refers to full carrier amplitude modulation with a single set of in-phase sidebands containing the information to be transmitted and received, but it should be understood that any modulation system which causes the instantaneous composite amplitude of the waveform to be varied in accordance with the information transmitted is and should be termed AM. This includes suppressed carrier single sideband and vestigial sideband broadcasted visual television as well.

AM can be generated in may ways, but always may be expressed by the well known trigonometric identity

$$(1+\cos\omega_m t)\cos\omega_c t = \cos\omega_c t + \tfrac{1}{2}\cos(\omega_c-\omega_m)t + \tfrac{1}{2}\cos(\omega_c+\omega_m)t \quad (1)$$

where ω_m = modulation freq.
ω_c = carrier freq.
t = time

when the modulation form is a simple sine-wave. More complex modulation waveforms may be expressed as a Fourier series of sine or cosine terms, but the carrier and sideband terms would retain the same form and the modulation coefficient, m, would modify the amplitude of the $\cos\omega_m t$ term. Any method used to generate the carrier and associated sidebands of (1) is fair game as a method to generate AM. The purpose of this paper is to present what is believed to be a new method of generating pseudo-continuous amplitude modulation at any modulation index between zero and one at any carrier frequency using any class of amplifier, (Class A,B,AB,C,D,H,S, tc) as an RF source. First, however, it is necessary to review the operation of the basic quadrature hybrid power combiner/splitter.

The Quadrature Hybrid Power Combiner/Splitter

A device not familiar to all electrical engineers is the guadrature hybrid power combiner/splitter. This is a device that appears to be well known to certain circles and completely unknown to others. It finds favor in the microwave community because of its small size at these frequencies and because it is a simple practical way of adding the RF power output of many signal sources together to effect a much larger signal than that available from a single source. It tends to be unknown to the RF designer below about 50MHz because the combiner gets to be prohibitively large and expensive, although its theory applies to all RF frequencies. Throughout this paper it is referred to as a "hybrid", a "combiner", a splitter, or a combination of these terms.

A quadrature hybrid combiner is a four port device which consists of two (or more) parallel conductors placed inside a common outer conductor such that the two lines share the same E and H fields. For this basic definition; no restriction is placed on such things as characteristic impedance of this coaxial arrangement or where the terminations are located, but simply stated, two conductors sharing a common field will mutually induce current in each other according to some physical law. A vector analysis of the device may be found in most NAB handbooks in the television section. For the purpose of this review it is sufficient to discuss its significant properties.

In Fig. 1, coupled lines are shown placed at the center of a common outer conductor (ground). The four ports are identified where appropriately sized connectors may be attached. Schematically, the hybrid in shown in Fig. 2. If the combiner/splitter (shown as a splitter in Fig. 2) is of the correct dimensions for a given frequency, input power at port 1 will be equally split to ports 2

*Application has been made for a patent for the digital modulation method described here.

and 3. Port 2 will have the same phase as port 1 (except for very small propagation delays) but port 3 is at -90° with respect to ports 1 and 2. Whether or not power is split equally depends on the electrical length of the lines and the degree of coupling governed mostly by the shape of the lines and their proximity to each other. Characteristic impedance depends on the cross sectional geometry of the whole structure. If the two lines and outer conductor are circular in cross section, such as with copper pipe or tubing, the ratio of the inside diameter of the outer conductor to the outside diameter of one of the lines should be 4.0 for a characteristic impedance of 50 ohms at each of the four ports. If the two lines are circular in cross section and the outer conductor square in cross section, the ratio of the inside length of one side of the outer conductor to the outside diameter of one of the lines should be 3.5 for a characteristic impedance of 50 ohms. The degree of coupling is determined by the spacing between the lines while the length of enclosed line determines the frequency range over which the degree of coupling is maintained to be reasonably constant. It should be pointed out that a combiner of this type will maintain the same degree of coupling over an octave bandwidth within a few tenths of a dB while maintaining a nearly constant 90° phase shift to the quadrature port. Outside the octave bandwidth, coupling decreases in both directions and the phase angle departs greatly from 90°.

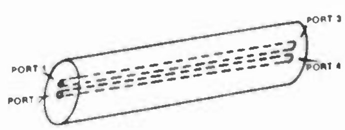


Fig. 1. Two coupled lines are shown inside a common outer conductor which is normally grounded. The quadrature hybrid is a four port device with very interesting coupling characteristics.

If ports 2 and 3 are properly terminated in the characteristic impedance of the hybrid, no power is coupled to port 4. If the same magnitude and phase mismatch is provided at both output ports 2 and 3, they also become input ports to the reflected waves produced. Fig. 3 shows that all reflected power ends up in port 4. Port 1 doesn't see the mismatch so that a perfect termination is maintained at the expense of lost power to port 4.

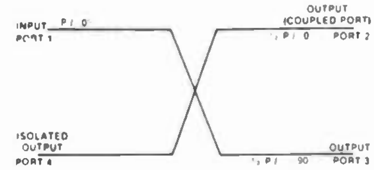


Fig. 2. Schematic representation of the quadrature hybrid power splitter. If RF power is fed into Port 1, it is split equally between ports 2 and 3 with the noted phase relationship. Ideally no power leaves port 4, the isolation port.

For mismatches at ports 2 and 3 that are not alike, some power will be reflected back to port 1 because amplitude and phase cancellation cannot occur at port 1. This can be a problem when used as a power splitter should the load connected to port 2 or 3 change in some way. This usually turns out to be less serious than as it first appears because when used as a splitter, power levels are much lower than when used as a combiner and any reflected power is usually easier to contend with. Port 4 is normally called the isolated port where a dummy load is connected to absorb reflected power. It must be sized according to the worst case reflected power expected. If the loads connected to ports 2 and 3 are always expected to be perfect, a dummy load at port 4 is not needed meaning that this port may be left unterminated.

A very important property of the hybrid device is that the two output ports so labeled in Fig. 2 ideally do not see each other. A mismatch may occur in one output port (port 2 or 3) and the other sees no power as reflected power. This is illustrated in Fig. 4 where in (4a) a shorted port occurs and in (4b) an open circuited port appears. This works because for a signal going back into ports 2 or 3, the opposite output port, port 3 or 2, becomes the new isolated port for the reflected wave. Since all other ports are properly terminated, no power makes it to the new isolated port. The conclusion is that the output ports are isolated from each other. In real practice the degree of isolation is within the range of 20 to 30dB.

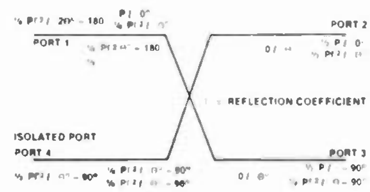


Fig. 3. For the same type of mismatch at ports 2 and 3, all reflected power is transfered to port 4. Reflected powers back to Port 1 cancel each other due to the phase relationship shown.

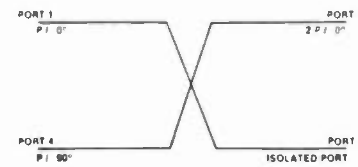


Fig. 5. The quadrature hybrid used as a power combiner. The device is the same as the splitter, but connected as shown.

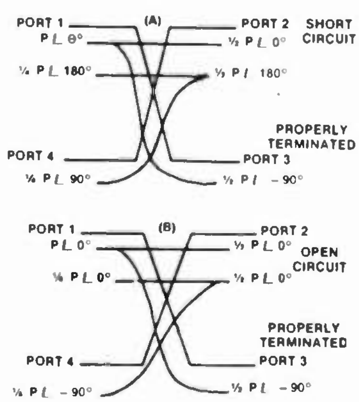


Fig. 4. Each of the output ports, ports 2 and 3 are isolated from each other's mismatch. (a) shows the signal paths when port 2 is terminated in a short circuit. (b) shows the signal paths when port 2 is terminated in an open circuit.

Fig. 5 shows the hybrid used as a power combiner. It corresponds to the configuration shown in Fig. 2 where is is presented as a splitter. Because it is a reciprocal device, the analysis is the reverse of the splitter. Two equal amplitude signals 90° split in phase are fed into ports 1 and 4. They are combined in port 2 while port 3 becomes the isolated port. It retains all of the properties of the splitter and likewise affords input ports isolated with respect to each other.

If a quadrature hybrid is constructed to be a 3 dB splitter or combiner, power will be split equally in amplitude to two ports or will combine completely from two equal amplitude ports. Should the power levels be unequal in the two input ports of a combiner, some power will be lost in the isolation port dummy load. (This load is often referred to as a reject load.) Since vector voltages are combining to produce power, the power output, P_o, is related to the input power levels P_{in1} and P_{in2} according to:

$$P_o = \left(\sqrt{\frac{P_{in1}}{2}} + \sqrt{\frac{P_{in2}}{2}}\right)^2 \qquad (2)$$

The power sent to the isolation port becomes

$$P_{iso} = \left(\sqrt{\frac{P_{in1}}{2}} - \sqrt{\frac{P_{in2}}{2}}\right)^2 \qquad (3)$$

It is easily seen that if $P_{in1} = P_{in2}$, (2) reduces to become the sum of the input powers while (3) goes to zero. If either P_{in1} or P_{in1} is zero, half of the power of the remaining active input goes to the output port while the other half is sent to the isolation port and dummy load. Again the two input ports remain isolated from each other.

More than one quadrature hybrid may be connected in various ways to achieve the purpose desired. For example, Fig. 6 shows how they may be connected to split drive power into four output ports to drive four amplifiers. The outputs of the amplifiers may be combined with a like combiner configuration to sum the powers to a single port.

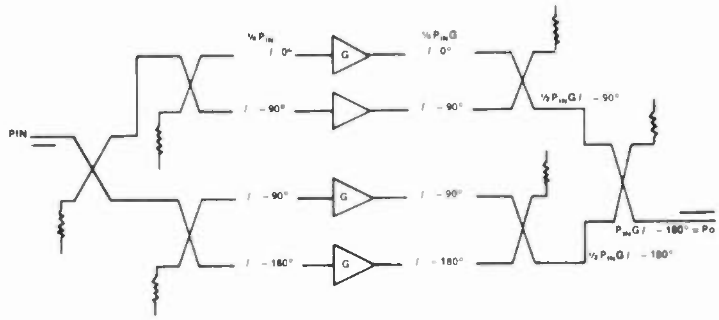


Fig. 6. The use of three splitters and three combiners properly connected to preserve phase are shown to drive and combine four smaller identical amplifiers to look as one.

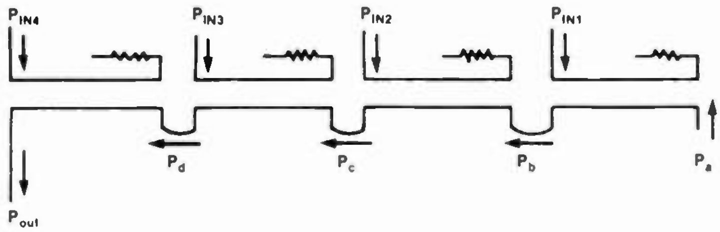
The Digital Amplitude Modulator, a Numerical Example.

Making use of the properties of the basic quadrature hybrid splitter/combiner and the relationships of (2) and (3) allows for the configuration shown in Fig. 7. Instead of the traditional symbol used for the hybrid thus far, a departure is made in Fig. 7 to reflect more the actual construction of the device. This is helpful in preventing crossed lines interconnecting the cascaded combiner. In all hybrids showns, the reject or dummy load is connected to the isolated port with respect to the two input ports. Power input doubles moving from right to left so that $P_{in2} = 2P_{in1}$, $P_{in3} = 2P_{in2}$, etc. Power levels are chosen so that they sum to one, but the important relationship is that all input power levels are equivalent to the weight of a binary number made up of ones and zeros. Moving from right to left in a binary word, each succeeding charactor has twice the numerical weight of the number to the immediate right. In Fig. 7 all input ports are on. That is to say, if all input ports were controlled by a one or a zero in a binary word, they would all be the same, one or zero. For the sake of easiest association, the convention is made that a logic one means ON and a logic zero means that corresponding input power is OFF. P_a is always on and is not considered to be controlled by a binary digit since it merely represents the total power coming from all previous stages, should they exist.

By numerical example an amplitude modulator is derived that could be controlled by the binary representation of the instantaneous value of the amplitude of an arbitrary waveform. Obviously a mixture of ones and zeros will result for any sample taken. For a modulator of this type, it is necessary for the power output of the combiner to produce the proper RF power level representing the modulation level sampled. Unless this is true, modulation will not be linear and distortion will result. Fig. 8 through 12 show a numerical example of the resultant summed power with one or more inputs turned off by a logic zero assuming the same configuration of Fig. 7. For the sake of simplicity of analysis, the summed power will be taken to be one unit of power or one watt. There are four input ports for a four bit word of sixteen possible states. The lowest power input becomes one divided by sixteen or 0.0625 watt. Using (2) and (3), each intermediate power level may be found. Fig. 8 for a logic word of 1101 has an output level of 0.765 power units. The sum of P_{in} is 0.875. For Fig. 9 the logic word is 1011 for a power output of 0.5625 and a power input total of 0.75. Fig. 10 shows an output power 0.25 while the input sum is 0.5. It is interesting to note and essential to the linearity of the modulator that the output power is numerically equal to the square of the input power, ie $(0.875)^2=0.765, (0.75)^2=0.5625, (0.5)^2=0.25$ and if carried through for all sixteen cases of the four bit word, the relationship holds. At first it appears that the square law relationship renders the combiner useless as a linear amplitude modulator, but the truth is that digital voltage representations of an abitrary waveform are just that -- voltages, that are controlling powers which automatically square the voltages into powers so that the squaring is cancelled. In effect the digital voltage word is squared into a power word which is perfectly linearly proportional to the power at the summed port, P_{out}. The linearity of the modulator does not depend on the type of power sources as long as available power to each port remains precisely double that of the next lower power input port. The proper amount of waste power automatically finds its way to a reject load and not P_{out} to maintain this linearity.

It may also appear, at first glance, that the modulator is terribly inefficient because power is dumped into reject loads to make it work. The combiner/modulator is of course theoretically 100% efficient with all inputs on (see Fig. 7). Efficiency decreases as some inputs are off, but so does total consumption.

Showing that an example set of combiners works as a perfectly linear amplitude modulator is by no means proof that the modulator retains all of its properties in the most general case. A rigorous mathematical development of all important parameters is the effort of the remainder of this presentation.



$P_b = P_a + P_{IN1} = P_{IN2}$

$P_c = P_b + P_{IN2} = P_{IN3}$

$P_{out} = P_d + P_{IN4} = P_a + P_{IN1} + P_{IN2} + P_{IN3} + P_{IN4}$ $\quad$ $P_d = P_c + P_{IN3} = P_{IN4}$

Fig. 7. Four power combining hybrids are connected as shown with the output of each feeding one input port of the one to its immediate left. Input power at each input port doubles from the previous one moving right to left so that they are all summed at P_{out}.

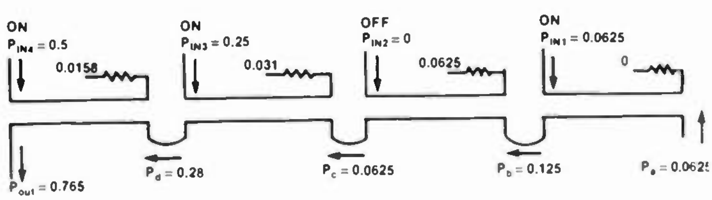


Fig. 8. Fig. 7 reproduced with a numerical example with P_{in2} turned off. Power levels shown represent a relationship to a total power of one unit if all inputs were on.

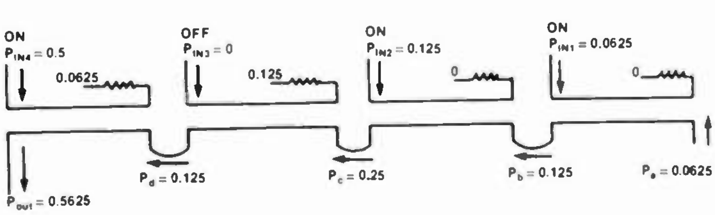


Fig. 9. Fig. 7 reproduced with a numerical example with P_{in3} turned off. Power levels shown represent a relationship to a total power of one unit if all inputs were on.

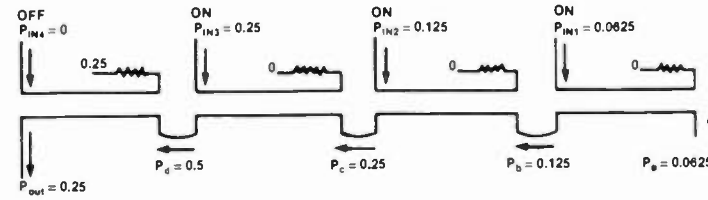


Fig. 10. Fig. 7 reproduced with a numerical example with P_{in4} turned off. Power levels shown represent a relationship to a total power of one unit if all inputs were on.

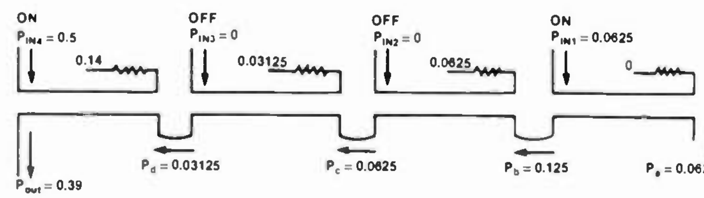


Fig. 11. Figure 7 reproduced with a numerical example with $P_{in2} = P_{in3} = 0$ or turned off. Power levels shown represent a relationship to a total power of one unit if all inputs were on.

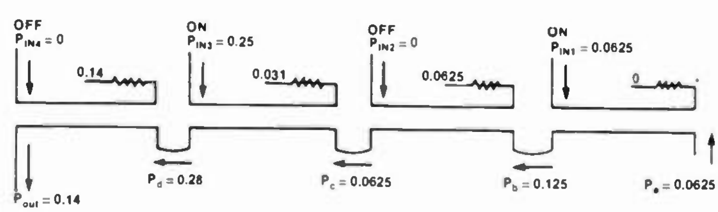


Fig. 12. Figure 7 reproduced with a numerical example with $P_{in4} = P_{in2} = 0$ or turned off. Power levels shown represent a relationship to a total power of one unit if all inputs were on.

A General Derivation of the Digital Modulator.

For the general case, let n be the number of bits in the binary word such that n also equals the number of power input ports to an n port combiner. Since the first port will hence forth be referred to as the zeroth port, the ports are numbered from zero to n-1. Fig. 13 represents the n port combiner. p is the least significant power level to the "carry" input from imaginary lower level combiners. Let P_{ci} be the total power into the combiner from those amplifiers gated on.

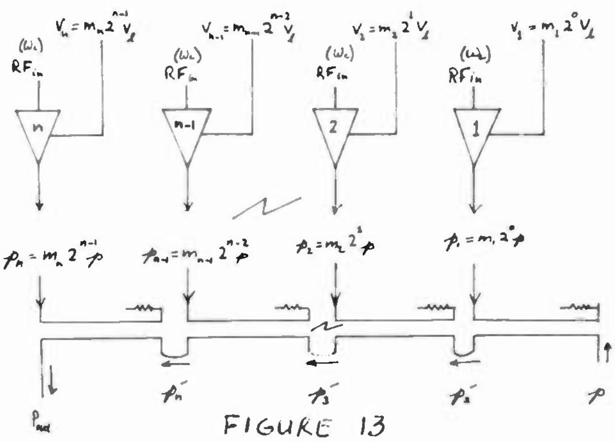


FIGURE 13

Fig. 13. The general case of hybrid combiner/ modulator is shown driven by n power sources represented by p_1 through p_n and gated on or off by m_1 through m_n. V_ℓ represents the logic voltage level output of an analog to digital converter. p_2' through p_n' are intermediate power levels working towards P_{out}.

It is apparent that

$$P_{ci} = p + p_1 + p_2 + \cdots p_K + \cdots p_{n-1} + p_n \qquad (4)$$

or that

$$P_{ci} = p + m_1 2^0 p + m_2 2^1 p + \cdots m_K 2^{K-1} p + \cdots m_{n-1} 2^{n-2} p + m_n 2^{n-1} p \qquad (5)$$

where $m_K = 1$ for the Kth amplifier on
$m_K = 0$ for the Kth amplifier OFF
p = smallest unit of input power

It follows that $p_K = m_K 2^{K-1} p$

$$P_{ci} = p\left[1 + \sum_{K=1}^{n} m_K 2^{K-1}\right]. \qquad (6)$$

Let V_{in} be the total binary weighted logic voltage gating the amplifiers and let V_ℓ be the actual logic level for a logic one. Then

$$V_{in} = V_1 + V_2 + \cdots V_K + \cdots V_{n-1} + V_n \qquad (7)$$

OR

$$V_{in} = m_1 2^0 V_\ell + m_2 2^1 V_\ell + \cdots m_K 2^{K-1} V_\ell + \cdots m_{n-1} 2^{n-2} V_\ell + m_n 2^{n-1} V_\ell \qquad (8)$$

where $m_K = 1$ for the Kth bit = logic 1 or the Kth amplifier ON
$m_K = 0$ for the Kth bit = logic 0 or the Kth amplifier OFF
$V_K = m_K 2^{K-1} V_\ell$.

Likewise, it follows that

$$V_{in} = V_\ell\left[1 + \sum_{K=1}^{n} m_K 2^{K-1}\right]. \qquad (9)$$

It must be remembered that the actual logic level gating each of the n amplifiers is V_ℓ. All V_1 through V_n are binary weighted voltages assigned by position in the binary word and corresponding power level of the associated amplifier.
Let P_{out} = total power out of the combiner/ modulator according to which $m_K=0$ and $m_K=1$. Combiner interstage power levels may be found according to the following series of equations derived from (2):

$$p_2' = \left(\sqrt{\frac{p_1}{2}} + \sqrt{\frac{p_1}{2}}\right)^2 \qquad p_n' = \left(\sqrt{\frac{p_{n-1}'}{2}} + \sqrt{\frac{p_{n-1}}{2}}\right)^2$$

$$p_3' = \left(\sqrt{\frac{p_2'}{2}} + \sqrt{\frac{p_2}{2}}\right)^2 \qquad P_{out} = \left(\sqrt{\frac{p_n'}{2}} + \sqrt{\frac{p_n}{2}}\right)^2$$

$$p_K' = \left(\sqrt{\frac{p_{K-1}'}{2}} + \sqrt{\frac{p_{K-1}}{2}}\right)^2 \qquad (10)$$

And that $P_{out\,max} = P_{out}$ for all $m_K=1$
Further, for the conditon that $p_n=p_n$ all $m_K=1$ since the sum of all amplifier output powers is necessary to add up to $p_n=p_n$. It follows that

$$P_{out\,max} = \left(\sqrt{\frac{p_n'}{2}} + \sqrt{\frac{p_n}{2}}\right)^2 = 2\left(\sqrt{\frac{p_n}{2}}\right)^2 = 2p_n,$$

Since $p_n = 2^{n-1} p$ (Fig 13) for $m_n = 1$ (11)

$$P_{out\,max} = 2p_n = 2(2^{n-1} p) = 2^n p. \qquad (12)$$

Shown another way for the same case, ie all $m_K=1$, no power is lost in the reject loads and

$$P_{out\,max} = P_{ci}, \quad \text{for all } m_K = 1, \qquad (13)$$

From (6)

$$P_{out\,max} = p\left[1 + \sum_{K=1}^{n} 2^{K-1}\right]. \qquad (14)$$

And from the identity

$$2^n = \left[1 + \sum_{K=1}^{n} 2^{K-1}\right], \qquad (15)$$

it again follows that

$$P_{out\,max} = 2^n p \qquad (16)$$

It was stated earlier and shown by example that the square of the input power to the combiner is numerically equal to the power output and that this square law is actually what provides the perfection of modulation linearity when voltages gate powers on and off. This leads to the following relationship (without presenting the messy mathematics) that

$$P_{out} = P_{ci}^2 / P_{out\ max} . \quad (18)$$

By combining (6) and (16), it follows that

$$P_{out} = \frac{p^2 \left[1 + \sum_{K=1}^{n} m_K 2^{K-1}\right]^2}{2^n p}$$

or

$$P_{out} = \frac{p \left[1 + \sum_{K=1}^{n} m_K 2^{K-1}\right]^2}{2^n} . \quad (19)$$

At this point in the general development of the modulator, expressions are known for P_{ci}, V_{IN} , P_{out} and $P_{out\ max}$ in like terms.

It becomes possible to determine the combiner/modulator system gain, G, from logic voltage level to output power, P_{out}. By definition

$$G = \frac{V_{out}}{V_{in}} , \quad G = \frac{\sqrt{P_{out} R_L}}{V_{in}}$$

where R_L = load resistance connected to P_{out}. (20)

Substituting (19) into (20) yields

$$G = \sqrt{\frac{p\left[1 + \sum_{K=1}^{n} m_K 2^{K-1}\right]^2}{2^n}} \; \frac{\sqrt{R_L}}{V_{in}}$$

or

$$G = \sqrt{\frac{p R_L}{2^n}} \left[1 + \sum_{K=1}^{n} m_K 2^{K-1}\right] \frac{1}{V_{in}} . \quad (21)$$

Substituting (9) into (21) provides

$$G = \sqrt{\frac{p R_L}{2^n}} \; \frac{\cancel{\left[1 + \sum_{K=1}^{n} m_K 2^{K-1}\right]}}{\cancel{\left[1 + \sum_{K=1}^{n} m_K 2^{K-1}\right]}} \; \frac{1}{V_\ell} ,$$

or

$$G = \frac{1}{V_\ell} \sqrt{\frac{p R_L}{2^n}} . \quad (22)$$

Further, from (16), $P_{out\ max} = 2^n p$ and substituting into (22) gives

$$G = \frac{p}{V_\ell} \sqrt{\frac{R_L}{P_{out\ max}}} . \quad (23)$$

G is seen to be a constant and therefore not a function of input power to the combiner or the output power. It is perfectly linear without taking into account the quantization error that takes place whenever n is less than an infinite number.

To find the efficiency of the combiner, it is known by previous numerical example that it is only theoretically perfect for the case when all amplifiers are on, ie. all m_K=1. Efficiency, η , is defined to be

$$\eta = \frac{P_{out}}{P_{ci}} .$$

From (19) and (6)

$$\eta = \frac{\dfrac{p\left[1 + \sum_{K=1}^{n} m_K 2^{K-1}\right]^2}{2^n}}{p\left[1 + \sum_{K=1}^{n} m_K 2^{K-1}\right]} = \frac{\left[1 + \sum_{K=1}^{n} m_K 2^{K-1}\right]}{2^n} \quad (24)$$

η is obviously a function of m_K and is less than unity except when all m_K=1 since power must go into the reject loads to maintain linearity.

For all m_K=1

$$\eta = \frac{\left[1 + \sum_{K=1}^{n} 2^{K-1}\right]}{2^n} ,$$

and from (15)

$$\eta = \frac{2^n}{2^n} = 1 .$$

For all m_K=0

$$\eta = \frac{1}{2^n} .$$

Although an infinitesimally small number for n large, the power actually being wasted is most of the power contained in p, the least significant power level. As an example, for a four bit combiner let the word be arranged according to"dcba" with "a" the least significant figure. Combiner/modulator efficiency is shown in Table 1 by using (24) and η subscripted according to the four bit word. When all inputs are off, p is still injected into the "carry" port of the combiner and becomes a source of quantization error of the modulator. p is representative of the modulator's inability to generate a power smaller than p if that power is what is called for by the analog waveform being digitized. Likewise, the analog to digital converter driving the modulator cannot resolve a waveform amplitude that is less than that represented by the least significant bit. To reduce quantization error, n is made large enough so that p may be made small enough to be imperceivable. For audio, n should be 12 or greater. The minimum value for n for video is unknown to the author, but it is anticipated that it must be at least 12.

dcba	dcba
1/16	0000
2/16	0001
3/16	0010
4/16	0011
5/16	0100
6/16	0101
7/16	0110
8/16	0111
9/16	1000
10/16	1001
11/16	1010
12/16	1011
13/16	1100
14/16	1101
15/16	1110
16/16	1111

Table 1. Example combiner/modulator efficiencies tabulated for a four bit modulator for all possible bit combinations.

Conclusion

A digital amplitude modulator has been described by general mathematical derivation. As technology moves toward digitization for signal processing and transmission, the modulator described allows for analog simulation of a digitized signal in a transmitted or broadcasted amplitude modulated system being received by ordinary radio or television receivers. Digital AM transmitters for the medium wave band already exist, but the technology that makes them possible is a step below what is presented here. Power combiners in those transmitters do not offer port to port isolation ie, no reject loads are associated with the power combiner which is simply a transformer with a single secondary and many primaries. Nor is an isolated combiner necessary at medium wave where switch-mode RF amplifiers make up the modulator RF source. This source impedance is either near zero when gated on or approaching infinity when gated off. Switch-mode amplifiers are not yet possible much above the medium wave frequencies, so something else must be done.

The modulator described here, because of its port to port isolation properties, maintains an impedance of R_L at all ports whether adjacent amplifiers are on or not, therefore, it is useful at all RF frequencies with any class of amplifier. Its speed is limited only by the ability to gate the amplifier in consonance with the analog to digital converter sampling rate. No doubt logic "glitches" may result as amplifiers are turned on and off, but with careful design, it is felt that this may be overcome.

If the digital modulator described were used for television visual transmission, vestigial sideband transmission is not yet possible. The technique for partial lower sideband "cancellation" is not developed as of this point in time nor is it known to exist, but it certainly deserves a try and is the subject of future work. Even if it doesn't exist, a high level vestigial filter could be constructed to pass the appropriate spectral components of the double sideband signal for NTSC television.

Because of the absolute linearity of the modulator, it is expected that the usual non linearities of television RF amplifiers would be non-existant. Non linearities such as differential phase, differential gain, group delay, low frequency, high frequency, etc. all seem to disappear. Use of this modulator would, indeed, be a reverse trend to high level modulation in TV transmission. It's more than a modulator, however. It is the transmitter with the modulator and RF power amplifier inherently are one.
The analog to digital circuitry with antialiasing and replicated spectra filters due to the sampling process is deliberately omitted from this paper because those circuits and processes are well known. They all apply in addition to the output bandpass filter to keep spurious signals from being radiated.

Handwritten Symbology

cos	is short for <u>cos</u>ine
ω	is Greek small omega,
t	is lower case of T
P	capital p
p	lower case of P. This is <u>not</u> the Greek ρ or rho.
m	lower case of M
k	lower case of K
n	lower case of N
V	upper case of v . This is not the Greek ν or nu.
ℓ	is a script lower case of L. It is used as a subscript of V meaning <u>V</u>oltage-<u>l</u>ogic
$'$	prime symbol when used as su<u>p</u>erscript as opposed to a superscript one, ^{1} which is not slanted
Σ	uppercase Greek sigma. Uppercase of σ not used here
G	uppercase of g
η	Greek Eta lower case. Not to be confused with lower case of N. Always carefully signified by the long tail.

DEVELOPMENT OF AN ALL SOLID STATE VIDEO RECORDER

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The development of the communications industry owes much of it's growth to the transistor. Certainly, advances in solid state technology have redefined the design, capabilities and function of almost every device in the communications environment.

In this paper I would like to report on an application of solid state technology in which the task of recording and playing back of full bandwidth video signals is accomplished through the exclusive use of solid state devices.

This task is further accomplished in the composite digital domain, and in a way that permits outstanding accessibility to the data for creative manipulation, and to provide highly satisfactory solutions to some long standing operational problems within the broadcast environment.

This paper will touch briefly on the advantages of solid state technology, will examine how these advantages have been used as design criteria for a new video recording device and finally, look at the device itself and give the reader a progress report on the current status of that device, the NEC VSR 10™ Solid State Video Recorder.

A short review of solid state technology reminds us that, generically, solid state devices possess a number of specific characteristics:

1. FLAT PERFORMANCE

We know that a solid state device is rated at a specific level of performance on manufacture and test. We further know that that performance is maintained at a consistent level throughout the life of the device.

2. HIGH DEPENDABILITY

The collective experience of the industry over the length of time solid state devices have been available, has confirmed that these devices tend to be very dependable. That is, their small size, low power consumption, low heat characteristics and non-mechanical nature contribute to a very long operational life. Mean Time Between Failure has become a very managable number in our operational planning.

3. EXCELLENT SERVICEABILITY

The use of solid state devices in the tools of our trade has redefined the approach we take to maintaining them. The sealed system nature of most of these devices, coupled with their extremely small size, now means that when problems occur we tend to replace entire modules, temporarily setting the problem aside for later repair, or more probably, return to the manufacturer for repair or exchange. The down side of thisis that our parts bins tend to call for a larger investment. On the positive side, solid state technology permits the use of CPU or micro processor based diagnostic techniques. Availability of memory analysis and similar aids, now lead the maintenance engineer directly to the problem area and even provide ways in which the problem component may be temporarily bypassed in the bit processing of the device. This means that the device tends to stay on line more often and can be off-line for a shorter period of time when under repair.

4. HIGH SPEED PROCESSING

The inherent speed of signal processing attainable with solid state devices supports the extremely large numbers of calculations required to process and manipulate the video signal. High speed processing additionally permits virtually simultaneous data handling, by the same component. This capability is of great importance and leads us into our final advantage of solid state:

5. MULTI-TASKING

Perhaps the most significant aspect of solid state technology is that it can be utilized in designs that process several different functions at the same time. This multi-tasking capability means that at any given time, devices based on solid state technology, can perform radically diverse operations, sharing the same basic data. This concept is central to the efficient utilization of solid state technology as it is applied to the multi-channel video recorder.

DESIGN CONSIDERATIONS IN THE VSR 10™

The VSR 10™ is a fully solid state digital video recorder. It incorporates Dynamic Random Access Memory (DRAM) for data storage and for high speed video signal processing. Design considerations for this system are as follows:

1. The VSR 10™ was developed to utilize the advantages of completely solid state technology and contains a100% Solid State Recording and Processing System.

2. The VSR 10™ has been specifically designed to perform a broad range of tasks related to the production, post production and on-air television environment. This is evident in it's internal architecture, interface and operational control.

3. The VSR 10™ can perform several tasks at the same time, offering simultaneous control of the system from several locations, or, control of several functions simultaneously from one location.

4. The VSR 10™ is designed so that it can be expanded in two significant directions. (1) Its capacity can be increased through the addition of memory units, and (2) it is designed to take advantage of future developments in solid state memory technology. This is important because significant progress is currently underway in the development of four Megabit memories. When these become available, they can be incorporated as additional shelves of memory, dramatically expanding the memory resoviour of the VSR 10™ system.

5. The VSR 10™ is highly maintainable. It provides internally generated test signals and routines which are invaluable in the diagnosis and repair of memory within the system. In conventional recording technologies, utilizing tape or disk, failure of the recording medium typically means loss of not only the program material but of the medium itself.

In the case of disk technology, a disk crash tends to permanently eliminate some portion of the memory from further use. Eventually, the memory capacity of the disk will become eroded to the point of required replacement. In the case of solid state memory, the failed memory section can be partitioned off, just as in disk technology; but it can then be recovered, through replacement of the the failed memory chip; thereby restoring the memory capacity of the system to it's original state. Becuase the VSR 10™ employs no moving parts in its physical design, it can be easily transported within a facility, between facilities or in a totally mobile application. The rugged nature of the VSR 10™ protects it from many of the hazards of accidental movement that may lead to head crash or other similar problems in disk based technology.

At this point we will look at the VSR 10™ system and examine it's physical form, control and function.

THE BASIC VSR 10™

The VSR 10™ is configured as a basic system with several areas of expansion available depending on application.

The VSR 10™ basic system is contained in one Analog Shelf, one Memory Shelf and a power supply (see fig. 1). The basic system occupies

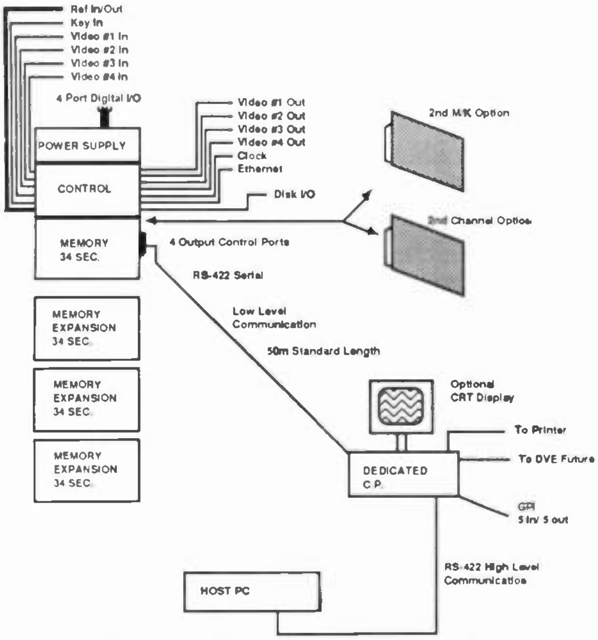

Fig. 1. VSR 10 Solid State Recorder.

about 25 vertical inches of rack space. The system is managed by a microprocessor based control panel. Up to four control panels can be connected to the mainframe via RS-422 Serial Interface.

The basic VSR 10™ provides 34 seconds of video recording time. Two input and two output ports are provided, either in NTSC Composite or in 4 X SC, 14.3 Megahertz digital. A very high quality mix/key Amplifier is integral to the system.

VRS 10™ SYSTEM OPTIONS

The VSR 10™ may be expanded in a number of ways depending on intended applications, and the way in which they are to be grouped in the typical operating environment.

Becuase the DRAM memory is volatile and not intended for archieval storage, the VSR 10™ is designed to interface with an external storage medium. This might be a high quality tape format recorder or hard disk storage unit. This interface is accomplished either in the composite digital or NTSC composite video domain.

The VSR 10™ memory resoviour may be expanded with the addition of three shelves, with 34 seconds of recording capacity each, totalling 1.536 GegaBytes or 136 seconds of recording time. Further expansion of the system includes adding a second AD/DA board which provides two additional external input and output ports. Two Read/Write ports out of digital memory are supplied in the standard configuration but this may be expanded to a total of four through the addition of a chroma inverter module. A second mix keyer can be added for simultaneous production multi-tasking.

Data backup in the digital domain is addressed through utilization of a tape streaming system. Using present technology, the entire digital data contents of the basic VSR 10™ memory can be transferred to a tape streamer in about two and one-half minutes. To facilitate the demands of tightly edited posting, a Q-tone scratch audio track reference can be added.

The basic VSR 10™ system draws about 1.5 KVa of electrical power. Uninterrupted Power Supply or UPS systems are currently available that will maintain the volatile DRAM in the VSR 10™ for a period of several minutes in the event of a power interruption.

Of the 17 boards contained in the Memory shelf of the VSR 10™, 12 are dedicated to memory itself. The memory consists of 384 MegaBytes, constructed from 1Megabit DRAM. The capacity of this memory unit is sufficient for 34 seconds of video recording which translates into 1020 frames.

The Analog Shelf is the point at which most interface of the VSR 10™ takes place. Four separated NTSC video input and output connectors are provided. In addition, two digital input and output ports are located here, as is an external key input, Hard disk I/O and Ethernet port for networking. Four RS 422 Serial control ports are provided at the analog shelf, utilizing low level communication to the external control computer.

Control panels for both sports and production work are currently available for the VSR 10™. The sports control panel is greatly simplified to permit full control of a limited range of functions. The "live" aspect of sports production dictates a very simplified fast moving control interface to support the on-air requirements of instant replay or live slow motion.

The Production control panel is designed to address the deeper range of control requirements called for in automated graphics rendering, multi-layering and precise field by field or frame by frame construction (see fig. 2).

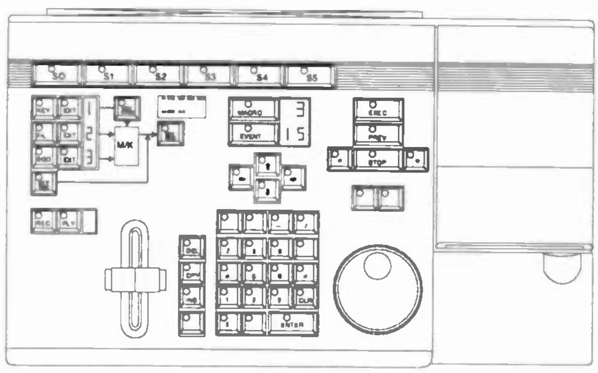

Fig. 2. Production Control Panel

The control panel is the point at which system management as well as creative control of the system are maintained. A mix of both dedicated and soft keys are provided, as is a fader bar for adjusting gain levels and a jog control for frame by frame playback of recorded material. A 3 1/2 inch floppy disk drive is utilized for operating system loading and off-line storage of programmed sequences.The control panel provides a BNC video jack for external CRT display. Four input and four output General Purpose Interface (GPI) contacts and outputs to an external printer are also located on the CP back panel.

Future development of the VSR 10™ calls for complete software control of an external DVE® system. A port is provided for that purpose.

Also located at the control panel is an RS-422 control port to access the VSR 10™ from an external device such as an edit controller. In this configuration, the VSR 10™ CP serves as a translator for the standard transport control codes originating from the edit controller. To the edit controller, the four ports of the VSR 10™ look like four tape transports. Commands, including PLAY, RECORD, STOP and REWIND are initiated at the edit controller and transferred via condensed ASCII string, through the VSR 10™ dedicated control panel to the mainframe of the system. It is important to note that control of each of the four ports of the VSR 10™ is completely independent of the control of the other ports, or it may be simultaneously linked to the other ports at the discression of the operator. Utilizing low level communications and full command level control protocol, an external computer may be directly connected to the VSR 10™ mainframe.

The control panel is divided into command sections that relate to the principal functions of the device. The control panel is designed to perform many of the functions that can be found on a typical edit controller as well as many functions dedicated specifically to the VSR 10™ system. The upper left section contains menu driven soft key control.

Dedicated buttons are provided for external or internal Key, Fill and Background input selection. Direct record and play commands are key assigned here as well.

Key gain is adjustable and input switching to the mix/keyer is controlled at this point. A key pad is provided for entry of numeric data which is used in the building of events and macros. Program preview and execution are controlled from this command cluster. The jog wheel permits manual viewing of a recorded sequence frame by frame.

Typical operation of the VSR 10™ begins with adjustment of phase and timing. This is accomplished from the control panel utilizing soft keys, fader and jog wheel. Further housekeeping for the VSR 10™ system is accomplished via assignment of memory resoviour and input/output ports to various control locations. The VSR 10™ uses a First In First Out control panel priority in which the first control panel to request system resources gets them. The next control panel has access to any un-used resources, i.e. ports, memory space, etc. An individual control panel must release it's allocated resources before they may be assigned to another control location. The memory resoviour of the VSR 10™ may be assigned in any value to each of a total of four ports, up to the combined maximum capacity of the memory. This assignment may be performed in increments as small as one field. Time division does not require an equal assignment of recording time to any of the four ports. Each port may be assigned the exact number of frames or seconds of reading or writing time needed to perform the function assigned to the port. We will look at time division in the VSR 10™ more closely in a following section.

A diagram of the internal signal path through the VSR 10™ mainframe helps us understand the logic and function of this device (see fig. 3)

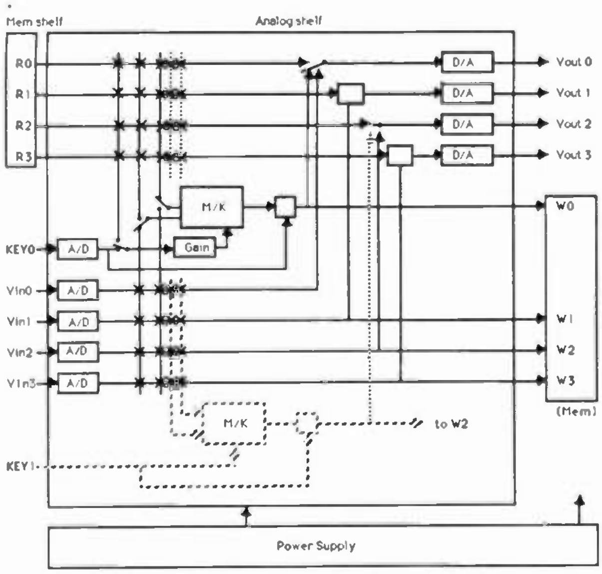

Fig. 3. Internal Signal Path Diagram

On the left side of the diagram, four external input ports are indicated. Two external digital input ports are also available but are not indicated in the diagram. A fifth port is provided for external key input. Each of the NTSC Composite input ports feeds an A to D for conversion into the digital domain. Four Read ports from memory are provided downstream of the A to D converters. Two high quality 4 X 1 input selection switching systems are standard in the VSR 10™ to facilitate signal routing in digital. Any combination of two internal, two external or one each internal and external signals may be applied to the digital mix/keyer within the system. A second digital mix/keyer is available for simultaneous mixing of two signals in two separate control locations. Any internal or external input may be assigned to memory, passed through the M/K Amplifier or read directly out of the VSR 10™ to any of four NTSC Composite or Digital outputs. In fact, a single input may be distributed to multiple memory or output ports at the same time. This signal path architecture is very valuable in the multi-tasked utilization of the system.

Now that we have a better understanding of the internal processing of the VSR 10™ Digital Recorder, I would like to examine several typical applications to which the recorder might be assigned (see fig. 4). All of the applications described can be processed simultaneously as well as individually. The only restraint is the available number of input and output ports and the memory capacity.

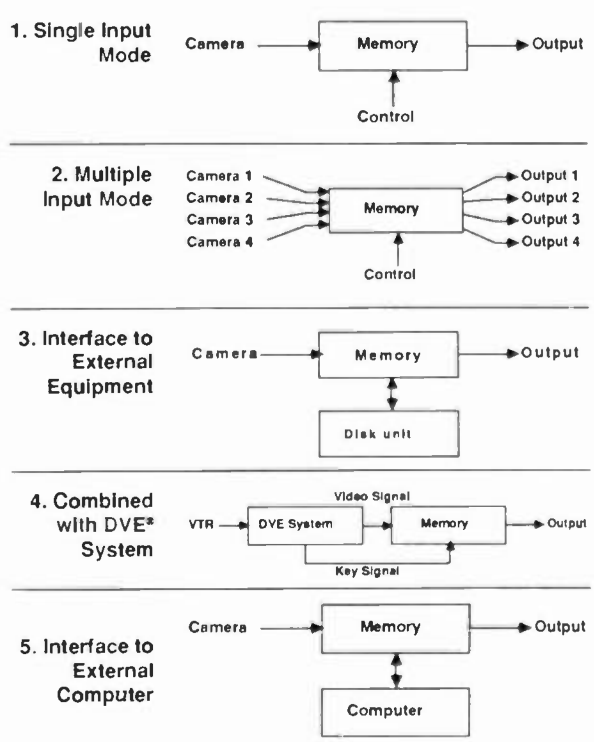

Fig. 4. Example Applications

SINGLE INPUT MODE

In this application, a single input is fed directly into the VSR 10™ mainframe, processed by the device and fed back out. Because record and playback of the same material can be accomplished in the digital memory, virtually simultaneously, with only one field of delay between record and play, playback of real-time events can be accomplished in variable time slow motion as they happen. The VSR 10™ permits variable delay of input and output signals passing through it's system. This leads to the creation of a fixed video delay function in which the VSR 10™ provides a highly reliable fully solid state method of delaying the transmission of live broadcast video signals. The technology can be combined with audio delay circuitry and placed on the output to the transmitter, giving us a "panic" button to use when the live video program gets out of hand. Individual field and frame identification and access mean that sequences read into the memory of the VSR 10™ 10 may be read out in any random order and in real time.

MULTI INPUT MODE

Because the VSR 10™ provides up to four NTSC composite video input ports, multiple cameras may be connected to the system for rapid and throughly controlled record and slow motion playback. This can be done simultaneously with playback of each input controlled at a different playback speed. Multiple views of the same action such as in sports isolation and motion study can be simultaneously recorded and played back individually.

INTERFACE TO EXTERNAL EQUIPMENT

The VSR 10™ may be used as a buffer for still pictures when connected with an external memory resoviour, such as a hard disk. These images may be taken out instantaneously as both still and dynamic sequences. In addition, the VSR 10™ can hold these images in buffer and play them out in any desired combination. The VSR 10™ offers Frame, Field and Color Frame modes for maximum flexibility.

COMBINED WITH DVE® SYSTEM

The VSR 10™ is designed to connect in the composite NTSC or digital domain with a DVE® effects system. Complicated layered pictures can be produced without picture overlap and with no picture quality degredation through many hundreds of generations. Since a built-in mix keyer is incorporated, wipes, fades, disolves and other transition effects can be easily created. Virtually any effect that can be created on a multiple channel DVE® can be easily created on a combined VSR 10™ single channel DVE® system.

INTERFACE TO AN EXTERNAL COMPUTER

The VSR 10™ provides several points at which an external computer may be used to access the video data and/or control the system. Increasingly, we find a large central computer, such as a VAX unit, acting as a central control point for all devices within a system. Such a computer supporting low level communication may be directly connected at the mainframe of the VSR 10™. An Ethernet™ access port is provided at the mainframe of the VSR 10™ for more extensive networking. And, of course, the edit controller may be directly connected to the system via the RS 422 port on the dedicated control panel of the VSR 10™.

MULTI-TASKING ON THE VSR 10™

You may recall that at the heart of the VSR 10™ design is centered the concept of multi-tasking. The speed of processing available in the solid state system combined with the multiple porting of the VSR 10™ open the door to the simultaneous performance of many tasks, basic to the video environment. In order to demonstrate this point, imagine a video environment in which an on-air operation is being maintained, post production work is scheduled around the clock and a rendering engine is assembling the output of the graphic artist into an animated sequence. All of these tasks call for utilization of some function of the VSR 10™. In figure 5 we can see how each of these separate tasks can, in fact, be addressed at the same time within the archi-

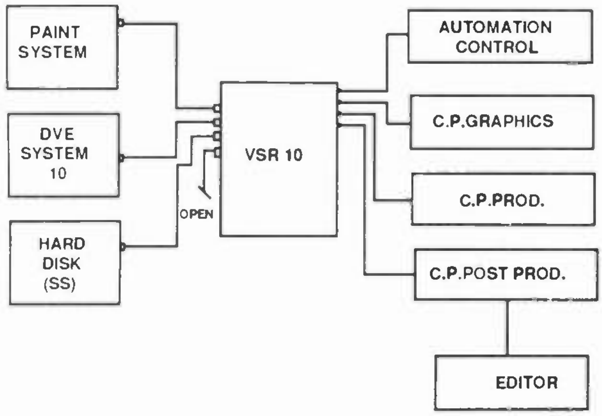

Fig. 5. Multi-Tasking Examples

tecture of the Solid State Recorder.

In this example, the graphic artist has set up 90 frames of animation. The graphics rendering computer is interconnected to the VSR 10™ mainframe via composite digital or NTSC input at port (channel) One. The production control panel of the VSR 10™ has been programmed to interact with the control system of the graphics generator to record a rendered frame as it is completed by the graphics computer, and then step ahead to be ready to record the next frame on it's completion.

Control data originates in a programmed macro within the VSR 10™ which triggers GPI pulses to the graphics computer signaling READY TO RECORD. The macro also looks for external GPI pulses from the graphics computer which signal back to the VSR 10™, RECORD and STEP. This macro function automatically runs to completion of the graphic sequence, a process that can take many hours for a few seconds of completed animation.

In our post production suite, a DVE® system is digitally connected to the VSR 10™. The editor is working with one-inch source material which is feeding the input to the single channel DVE® and being digitally manipulated and recorded on the VSR 10™ via ports Three and Four. The editor is utilizing the mix/keyer within the solid state recorder and layering back and forth between the two channels. While she is using two ports of the VSR 10™ to perform the layering function, it is important to remember that the VSR 10™ can essentially playback and record the same material on a single port. This means that a sequence can be built up, combined and mixed with new inputs from external sources or from Read sections of memeory on a single port of the solid state recorder. This is tremendously valuable when you wish to conserve memory within the VSR 10™ system. In this example the editor is operating the VSR 10™ via edit controller. The sequence on which she is working runs fifteen seconds.

In master control, the on-air operation needs an active library of thirty slides over the next three hours to support automated station control. The automation computer has been connected directly to the VSR 10™ mainframe via low level RS-422 control. The VSR 10™ has selected the stills it needs from the connected hard disk unit and stored the stills in 30 frames assigned through port Two of the VSR 10™. These stills are held in the solid state memory of the VSR 10™, acting as a buffer for the automation system. On demand, the automation system calls up the appropriate still frame from the VSR 10™ memory, which the VSR 10™ plays as a one frame still image as long as it is required. The image is output via NTSC to the station router for display.

Channel #	Function	Memory Allocation 34 sec. (1020 fr)
1	Paint System	3 seconds
2	Stills	1 second
3	Post Production (edit system play)	15 seconds
4	Post Production (edit system record)	15 seconds

Fig. 6. Multi-Tasking Example Time Allocations per Channel

Figure 6 shows how the time allocations per channel have been set up within the capacity of the 34 second memory of the basic VSR 10™ system. Port One has allocated 3 seconds or 90 frames to accept the rendering output of the graphics computer. Port Two has been filled with 30 frames or one seconds worth of stills from the hard disk. And ports Three and Four have been assigned fifteen seconds each for a total of 34 seconds, or 1020 frames. While these examples have been arbitrarily selected, you can complete your own task list. It might include the need for rendering from several graphic systems at one time.

You might need a three or four second video delay to support a live feed from a sports stadium or from the scene of a breaking news story. Or you might have two editors needing access to the recorder at the same time to meet a deadline. In each case, the architecture of the VSR 10™ gives you a real solution to a tough scheduling problem.

CONCLUSION

In conclusion, the VSR 10™ draws on the advantages of solid state technology throughout it's entire design. These include dependability, processing speed, multi-tasking and ease of diagnostics and service.

The VSR 10™ addresses a wide range of multiple functions and applications that make it an integral tool for the video environment. The Simultaneous Multi-Tasking and Multiple User capability of the VSR 10™ system make it highly cost effective, and facilitate it's interface with the other devices commonly found in the production setting.

The NEC VSR 10™ Solid State Recorder represents a practical application of solid state devices to the video recording medium. As advances in solid state memory technology are made, the solid state recorder will find an increasingly important place in our industry.

DIGITAL INTELLIGENCE IN PROFESSIONAL BROADCAST VIDEO MONITORS

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A few years ago the first use of microprocessors was introduced in professional broadcast video monitors. The purpose of the microprocessor was limited to perform some kind of automatic alignment, in an attempt to respond to the increasing demand of automation in professional broadcast equipment.
There is no reason however to limit the application of the microprocessor to mere automatic alignment. Most of the control functions in a monitor can be performed by the microprocessor, and with much greater accuracy than conventional analog circuitry. In addition the microprocessor will allow for new features, that were never dreamed of before.
Using the microprocessor in such an application results in a total new technology, design and concept of the professional broadcast monitor.

This paper describes this new concept and the major differences between the traditional and the intelligent monitor.

CHARACTERISTICS OF THE TRADITIONAL MONITOR

A traditional monitor is built up from different blocks.
- The CRT
- Power circuitry, including
  - power supply
  - deflection circuitry
  - high voltage circuitry
- Signal processing circuitry, including
  - input selection circuitry
  - decoder units
  - RGB video amplifiers
- Control circuitry, located on a front panel and inside a drawer.
- A limited remote control circuitry.

In the traditional monitor the design of these blocks is 100% hardware based, the interconnections between the different blocks is done by cables. As a result the monitor is designed for one fixed configuration and changing the configuration or adding new options is difficult, if at all possible.

Typically a monitor has over 50 controls that determine the display parameters. All of these controls have "preset" or "calibrated" positions. The set-up of a traditional monitor is to be performed with the controls in the calibrated positions. Repeatability of non-calibrated positions of these controls on different monitors is limited because of a limited status information. The user will be forced to "read out " all front panel- and drawer conrols right at the monitor.

Critical settings like color balance, peak white level and the saturation and hue calibration are not protected from unauthorized access by non-maintenance personnel.

CHARACTERISTICS OF THE INTELLIGENT MONITOR

To allow for the optimal use of the features of the microprocessor the concept and design of the intelligent monitor must be completely different from the traditional monitor. The design must allow for maximum flexibility, reliability, repeatability and stability.

Flexibility

The maximum flexibility is obtained by having a minimum fixed configuration. This fixed configuration is limited to the power circuitry and the CRT. The signal processing part of the monitor is composed of different plug-in modules. To cbtain an absolute flexibilty, the signal distribution inside the monitor is completely different from a traditional monitor. All the signals available inside the monitor are present on a digital and analog bus structure (fig.1)

The digital bus carries all the control signals in a sequential format. The analog bus controls all the analog signals and the different components of the video signal.

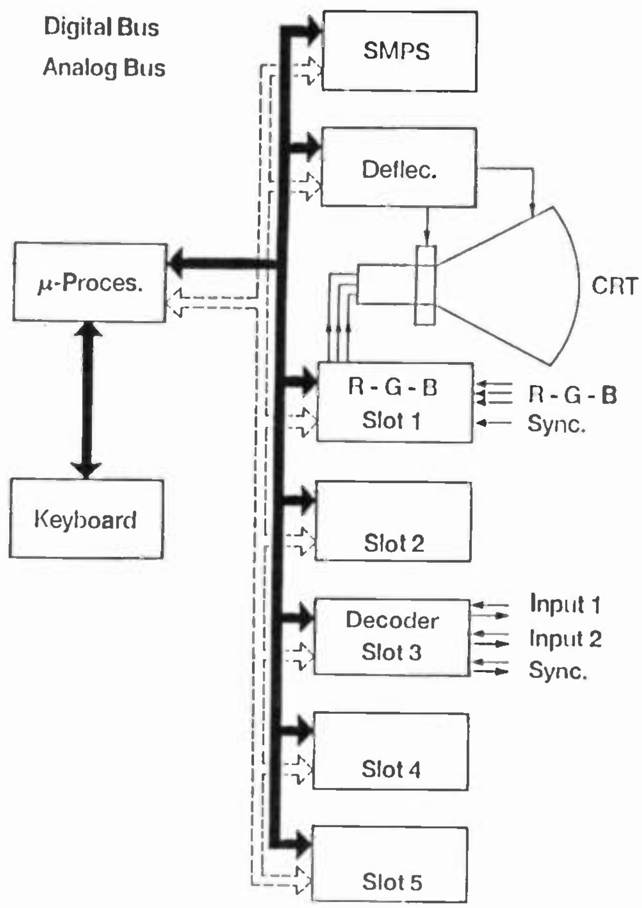

fig. 1

Such a signal distribution has two very big advantages.

First, the interconnections between the fixed configuration and the different plug-in modules is done by means of a motherboard.

Secondly, each of the plug-in modules has all of the digital and analog signals available on its connectors. As a result each of the plug-in modules (existing and future) can be built as stand alone units, adding such a module to an existing configuration can be done without any hardware changes inside the monitor.

All the controls of the monitor are performed by the microprocessor. Setting the different control parameters is done from a multi-purpose keyboard, and adding new features to the monitor will only require some software changes.

Conversion of the digital information into an analog voltage is performed using a digital to analog converter. The output of the D/A converter is demultiplexed because the bytes for every control function are converted in sequence.
Every DC control voltage is stored across a corresponding hold capacitor. The hold capacitors need to be 'refreshed'.
The refresh circuitry is built around a workram and an independent counter. During refresh, the independent counter generates the addresses of the workram and the bytes of the control functions are converted in turn.
The refresh routine is interrupted during a memory update and the corresponding byte is revised. (fig.2).

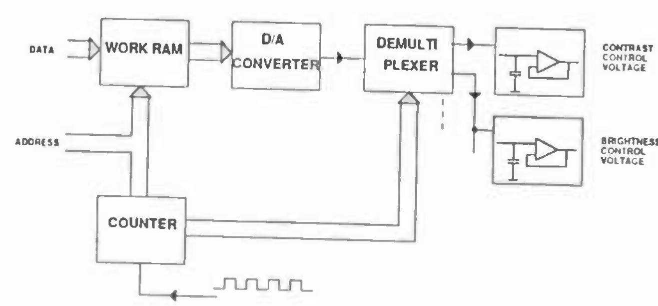

fig. 2

The remote capability of such a monitor is total by means of a serial port of the microprocessor. This allows for the use of different types of remote controls designed by the manufacturer, or the use of any other device having a serial output port.

Reliability

Maximum reliability is obtained because:
- the total number of components is reduced.
- the absence of moving mechanical parts.
- the total absence of cables and a reduced number of contacts.

Repeatability

Repeatability of the different control parameters is assured by the digital nature of the control functions in combination with the automatic alignment features of the monitor. Repeatability between different monitors is made possible by the enhanced status information. The status information of the different control parameters can be displayed on the screen at any moment and copied on another monitor.
In addition all of these control parameters can be returned to their factory calibrated positions at the touch of one button.

Automatic alignment

The purpose of the automatic alignment is to match color reproduction on different monitors.

In relation to color matching there are 3 important remarks that are to be considered:

- The automatic alignment is performed by setting the white balance of the monitor (lowlights and highlights).

- Color matching of different monitors is only possible between CRT's having identical phosphor ccordinates.

- The automatic alignment is performed in the center of the screen, the luminance and color shading of the CRT will influence the degree of matching across the whole surface of the screen.

To perform the automatic alignment different types of sensor devices can be used:

-Color analyzers
A color analyzer is a device that can analyze white light into the tristimuli values.
Two types of color analyzers are available, depending on the type of filter curves that are used.

-Narrow band filter color analyzers
Such a device has narrow band filters for red, green and blue. The device must be calibrated to match the phosphors of the CRT and for one particular color temperature. Although mostly referred to as color analyzers these devices are really color comparators, and can never be used as a stand alone source of calibration.

-Wide band filter color analyzers
Such a device has wide band filters matching the standard observer CIE sensitivity curves of the human eye. This device is a real color analyzer and needs no calibration to match phosphors and can be used as a stand-alone source for calibration.
Using a color analyzer as sensor for automatic alignment allows to set the monitor to any value for color temperature previously stored in the memory of the monitor. (fig. 3)

-The probe
A probe is a device that is limited to measure luminance. With the necessary intelligence built in to the probe this device can align a monitor to preset standards or it can copy any customized settings from one monitor to another.

Considering the price difference between color analyzers and probes, and the flexibility of such a probe, the probe will be used almost exclusively as a sensor device for automatic alignment in the future.
The flow chart of the automatic alignment is shown in figure 4.

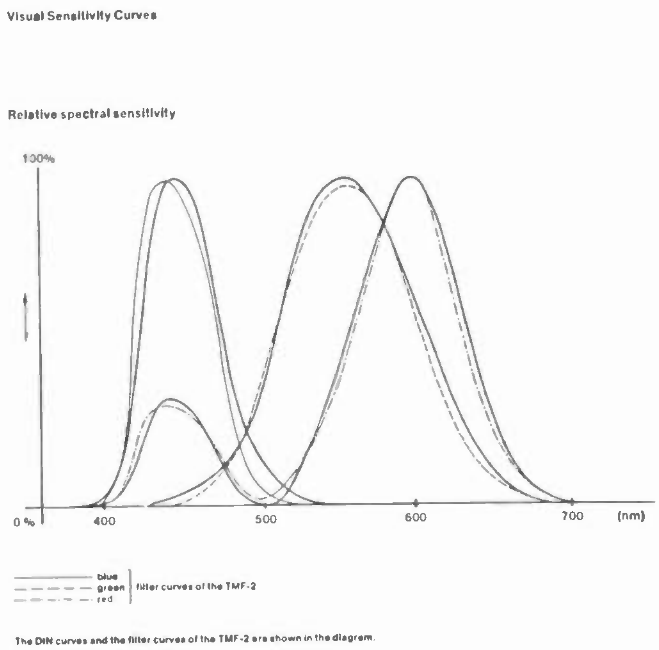

fig. 3

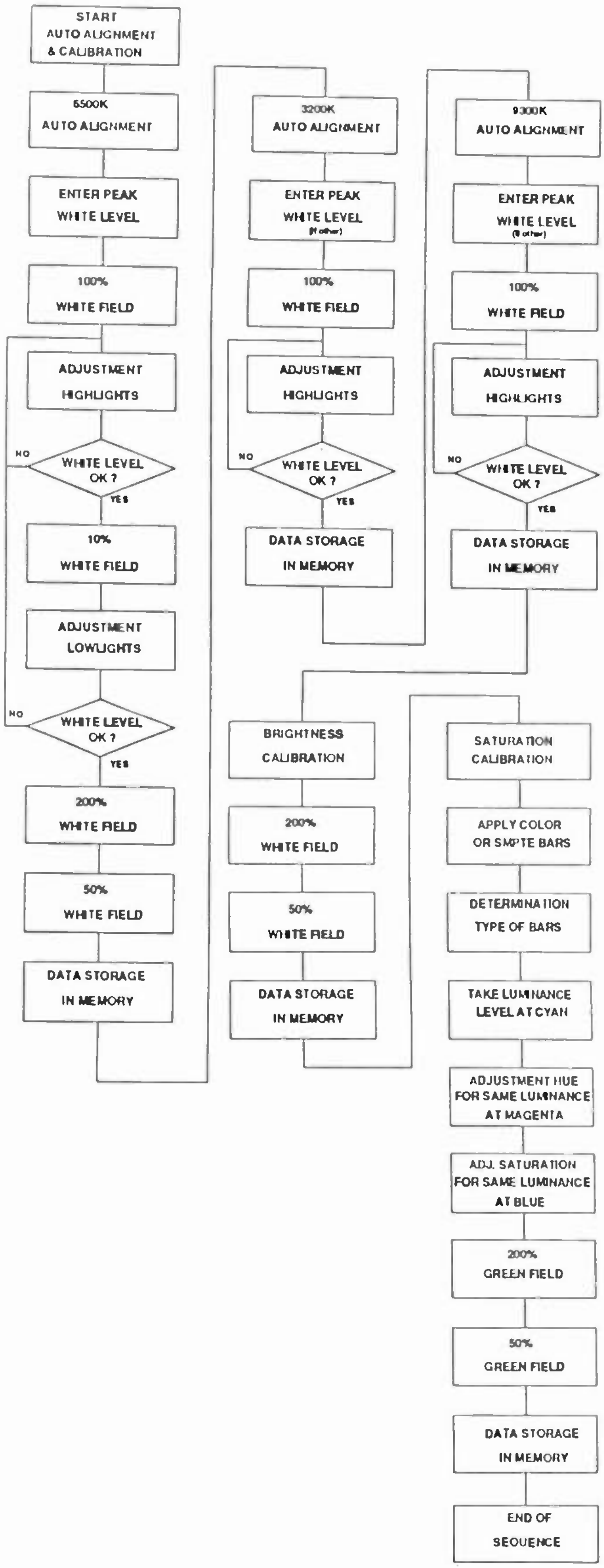

fig. 4

The block diagram of the automatic alignment is shown in fig. 5.

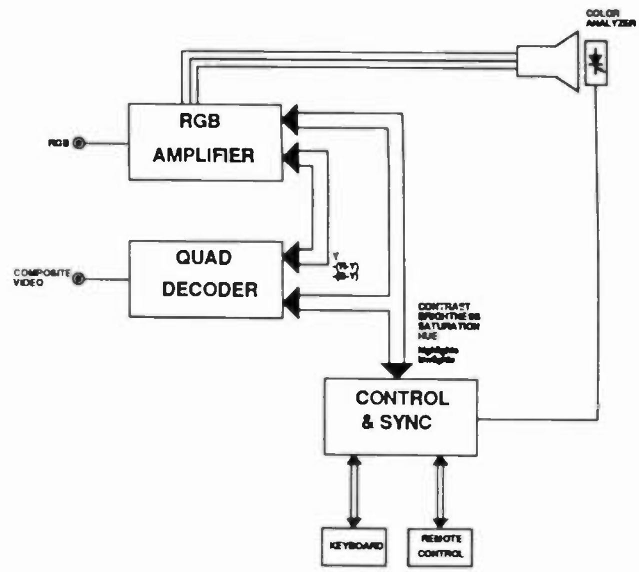

fig. 5

The reason to use the automatic alignment is twofold.

First, there is the economic aspect. Automatic alignment improves the consistency and the speed of monitor set-up. Consistency of color reproduction means that program material produced at different locations will be viewed exactly the same. This will reduce the quantity of material rejected and returned because "the colors are off". Also, by storing different preset color balances in the monitor allows to get an approximate idea how the video material will be viewed on displays other than class A monitors. As an example, a color balance at 9300 K will be very close to the image on an imported consumer set. The increased speed of alignment being performed at (10-20 times faster) and the alignment being performed at the operational location of the monitor, reduces the total down time for alignment. In addition, labor cost for maintenance is sharply reduced, as the procedure is faster and can be performed by non-technical personnel.

Secondly, there is the total absence of human factors in the set-up of the monitor. More and more studies indicate that the variations in color perception between different observers can be very big. Also, the color perception fo one particular observer will vary depending on fysiological and psychological factors. With the currently available optical metrology equipment, automatic alignment is far more accurate than a manual adjustment.

Stability

All of the control parameters being digital, the stability of these settings is assured by the control of the one voltage level applied to the digital to analog converters.

The short term thermal drift and the aging of the cathodes is compensated by a sophisticated current feed back system also known as the " automatic kinescope biasing " circuitry.

The eventual drift caused by the aging of some of the electronic components and the phosphors of the CRT is compensated by the automatic alignment procedure, resulting in a true optical feedback system.

Stability being one of the most important features of this type of monitor the automatic kinescope biasing (AKB) system deserves a closer look.

AKB

AKB is a process by which the DC-level of the R, G and B cathode signals are automatically maintained (automatic cut-off) within specified limits by comparing the seperate leakage current to an internal reference signal and thereby supplying correcting information to the cathode drive signals for accomplishing compensation for the minor variation of certain phenonema within the CRT. (fig. 6)

The hardware system consits of an electronic circuit that adds consecutive measuring pulses onto the red, green and blue cathode drive signals during the vertical interval.
The common leakage current of the 3 cathodes is converted to a voltage (VI) and is sampled by the AKB system before adding the measuring pulses.
During the 22nd horizontal line, a sample is taken which consists of common leakage increased with the leakage due to the red measuring pulse (VI+Vr).
During the 23rd horizontal line, a sample is taken which consists of common leakage increased with the leakage due to the green measuring pulse (VI+Vg).
During the 24th horizontal line, a sample is taken which consists of common leakage increased with the leakage due to the blue measuring pulse (VI+Vb).

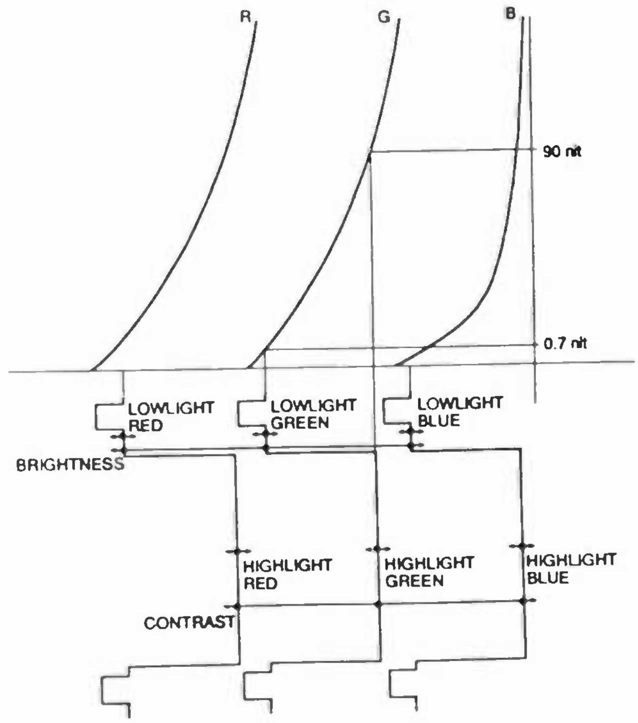

fig. 6

After the sampling sequence, the AKB system subtracts VI (common leakage) from VI+Vr, VI+Vg, VI+Vb. This results in 3 voltages Vr, Vg and Vb which are in direct relation with the leakage at their corresponding cathode.

Because of the comparison between an internal reference voltage and consecutive Vr, Vg and Vb, the DC-level of the red green and blue cathode drive signal is modified in order to compensate CRT leakage and to obtain an optimal black level stability.

Fig. 7 shows the bloc schematic of the AKB system.

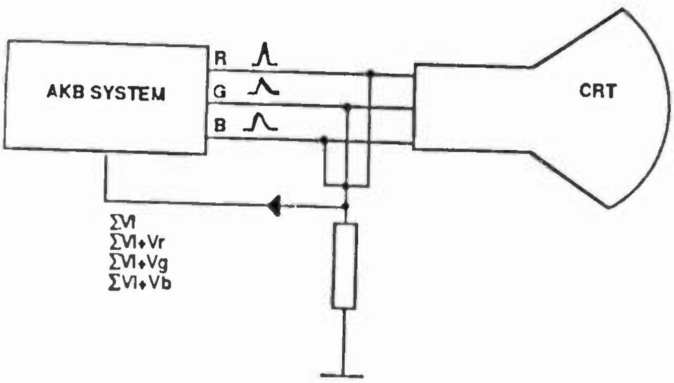

fig. 7

ADDITIONAL FEATURES

Using the features of the microprocessor at its fullest provides the following additional features.

Built-in test patterns

As the on-screen display of the different control functions requires a character generator, the same generator can be used to display different test patterns on the screen for a fast control of the operational quality of the monitor. These test patterns can include: color bars for evaluating the video amplifiers, a white field for white uniformity evaluation and for adjusting it, a crosshatch for control of the convergence and a geometry pattern for alignment of the scan size and the linearity of the displayed image.

Keyboard Extension

In order to minimize the key strokes on the frontal keyboard an additional "soft keyboard" extension is added to address over 50 settings inside the monitor.

Input Related Presets

The preset menu permits access to any of the monitors controls and settings which are used to select various operating features, permits resetting or clearing certain areas, and allows the operator to save and recall the preset menu of every available video input. The menu contains the functions of the frontal keyboard and these available in the keyboard extension menu. The features can be programmed. Pressing the frontal preset key causes adaption of the entire monitor to the preset conditions that were entered for the selected input. This results in an operators programmed picture, without changing the calibration of the monitor, as the calibrated settings can be recalled at any moment at the touch of one button.

Configuration Menu

The built-in flexibility of the intelligent monitor allows for a fast reconfiguration by adding plug-in modules to the standard minimum configuration. At power-up the microprocessor scans all the optional slots and recognizes the plug-in modules that are installed, and decides which part of the available software to use. This menu also allows to customize the input selection numbers, and displays the relationship between input numbers and the physical location of the input connectors at the rear of the monitor.

Status Menu

This menu allows for a fast overview of the status of all control functions and settings of the monitor for the selected input. It can also be used to create a preset direct form operational picture mode.

Service Page

Similar to the status menu it gives an overview of the actual calibration settings of the monitor.

Help Menu

A quick reference guide of the operation manual is available on screen, to help the user familiarize with the operation of the monitor.

GENERAL CONCLUSIONS

For a very long time the professional broadcast video monitor has been the weak link in the video chain. Traditional monitors were limited by the fixed configuration, the intensive maintenance needed and the very poor results of true color reproduction and stability.

All of this has changed with the introduction of the intelligent monitor.

In this age of fast changing broadcast formats, maximum flexibility is a must. Many times the broadcaster is forced into purchases of expensive equipment that becomes obsolete very fast. Equipping the monitor with a minimum fixed configuration is restricted to the CRT and the circuitry to drive it, the currently available intelligent monitor will not become obsolete as long as the CRT is the best display method available.

Handling new broadcast formats is just a matter of installing a new option in the monitor. The combination of bus structures and a multipurpose keyboard avoids the need for hardware changes when installing a new option. Also, option boards are not limited to one dedicated monitor and therefore the broadcaster can limit purchases of less frequently used option boards to an absolute minimum.

The automatic alignment feature of such a monitor will reduce maintenance costs considerably. Not only because the alignment is much faster, but also because it can be performed by personnel having less technical knowledge.

True color matching and long term stability will reduce costs of rejected program material. The operational personnel will regain confidence in the fidelity of the displayed image, and therefore detect errors in other video equipment much faster.

Too often engineers develop new technologies and products with amazing technical capabilities but with poor economic features.

The intelligent monitor provides these technical capabilities in an economical solution.

VIDEO MEASUREMENTS—A COMPREHENSIVE SOLUTION

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A new computer-based video measurement set is described. It features a high resolution raster-scan display and functions as a combination waveform monitor, vectorscope, noise meter and automatic measurement set. The instrument has a uniquely versatile and friendly user-interface.

INTRODUCTION

Computer and digital technologies have revolutionized the operational aspects of television production and distribution, but they have made slow progress in video test and measurement. The waveform monitor and vectorscope still play a major role in the quality monitoring of the analog television signal. The accuracy required for video measurements and the complexity of the signal itself have provided significant technical challenges to the application of computer technology. However, the need to automate the quality monitoring of the television signal is increasing, as it becomes increasingly difficult to staff television facilities with technical personnel having a good knowledge of video signal attributes.

BACKGROUND

In 1977, Tektronix engineers began to investigate the application of digitizer and microprocessor-based technology to the analysis of the analog video signal. By 1981, this effort culminated in the introduction of the Tektronix 1980 (ANSWER) Measurement Set. Although well suited for unattended transmitter monitoring, this instrument is constrained by the processor technology available when it was designed and it executes the measurement computations too slowly for effective use in an interactive adjustment environment.

Recent rapid advances in software technology, coupled with the availability of 32-bit microprocessors and 10-bit precision, video-rate, analog-to-digital

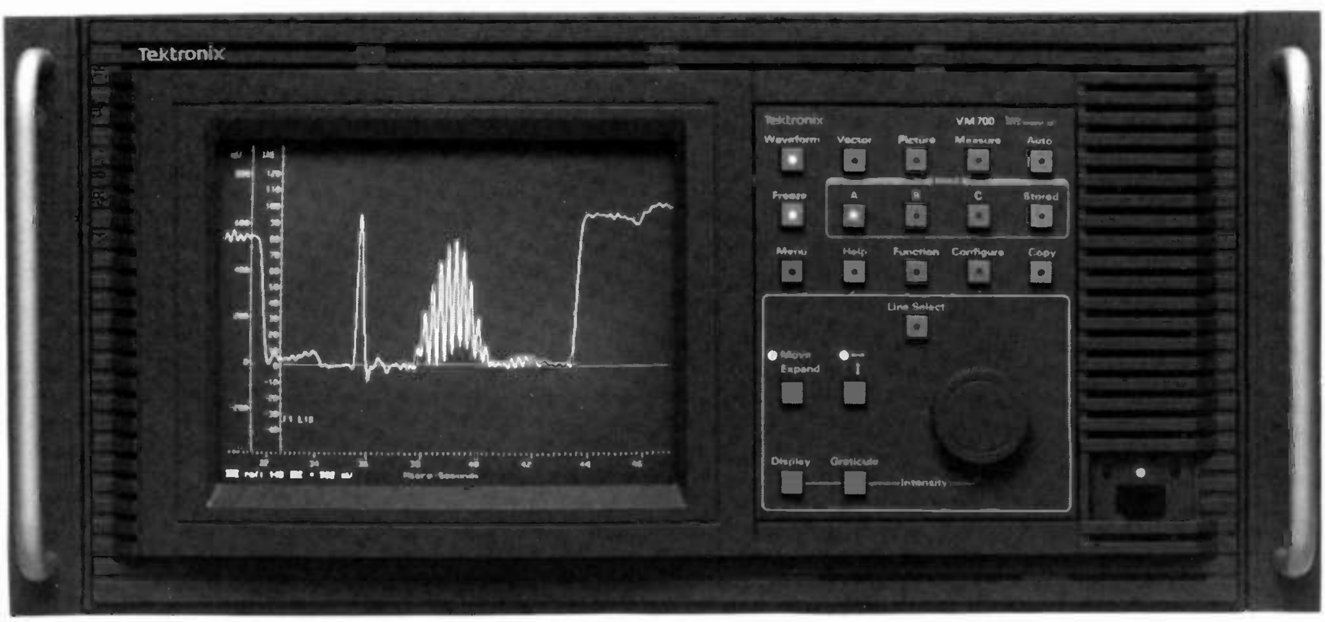


Fig.1 The VM700 Video Measurement Set

converters encouraged Tektronix engineers to take a new look at the application of computer technology to video measurements. The instrument described in this paper, the Tektronix VM700 Video Measurement Set is the result of these efforts.

GENERAL DESCRIPTION

The initial objective for the VM700 was to provide a single instrument capable of comprehensive video monitoring and analysis, by using the raster-scan display and on-board computer to accurately replicate the key attributes of a waveform monitor, vectorscope, automatic measurement set and noise measurement set. Not only was this objective achieved, but a simple and consistent user-interface resulted, and the instrument is generally easier to operate in each of its key modes than the corresponding analog instrument.

The VM700 is pictured in Fig.1. It has been designed as an easy-to-use video measurement instrument with full analytical capability. A front panel with a few buttons, a multi-use knob and a touch-screen raster-scan CRT make this possible. Display resolution is 640 pixels horizontal by 480 vertical and the display is fully graphics-capable with 2 bit-planes. One bit-plane is used for electronic graticules while the other is used to display the signal. Each has independently adjustable brightness using the knob in conjunction with dedicated push-buttons.

USER INTERFACE

The main functions of the VM700 are accessed from a 15-button key-pad. Secondary functions are available as soft-key selections via the touch-screen. Only one variable control is provided, a large knob internally connected to a rotary encoder. The obvious danger in this approach is operator-confusion as to the exact function of the knob in a particular operating mode. Careful thought was given to ensure intuitive coupling of the knob function to the chosen operating mode. Favorable user- reaction indicates that the integration of a single knob into the design of the VM700 has been highly successful.

OPERATING MODES

The VM700 has five principal operating modes: WAVEFORM, VECTOR, PICTURE, AUTO and MEASURE.

Waveform

The default waveform display (Fig 2) is one line per frame, but the horizontal expansion is adjustable to display from several hundred nanoseconds up to 11 video lines. Electronic graticules are provided which continuously track the waveform expansion in both horizontal and vertical direction. The waveform display is derived by interpolation of 10-bit sampled data acquired by an A/D from the input waveform. Sampling is approximately 4x subcarrier frequency. Computation/display speed is such that the display is real-time for frame-rate samples of one line or less. Since the displays are digitally derived and displayed on a raster-scan CRT, the vertical-interval test signals are viewed with full screen brightness. In the 'horizontal' mode, the knob scrolls the waveform continuously across the display. Field and Line numbers are 'attached' to the displayed sync pulses and move with the waveform. A SELECT LINE button is available on the main keypad. Operation of this button while in 'horizontal' mode results in time steps of one video line, permitting fast access to any line in the frame.

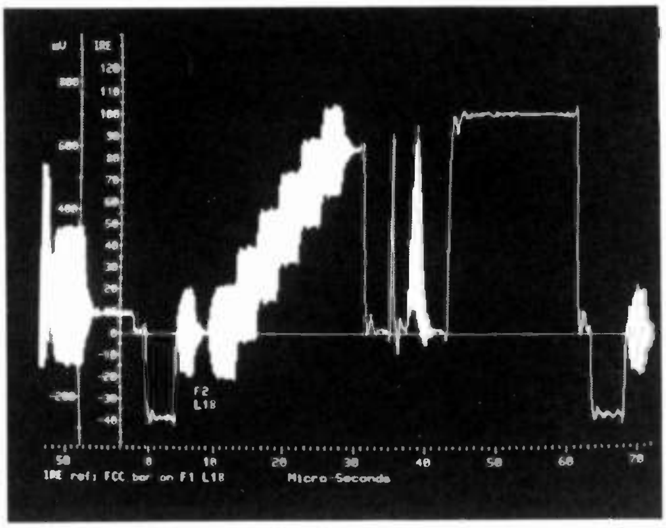


Fig.2 Waveform Mode, 1H display

A unique innovation in this instrument is the electronic graticules in Waveform mode. The horizontal time graticule is calibrated in microseconds, with its zero reference locked to the leading edge of line sync. The vertical IRE graticule zero point is referenced to signal blanking level. Both graticules continuously track the signal during position and expansion adjustments, and may be readily used for quick visual checks of signal amplitude and timing. For specific amplitude and timing measurements, a cursor mode is available via a soft-key selection.

It should be noted here that measurements in WAVEFORM mode are normally confined to non-standard signals or test configurations. Standard video measurements are normally performed with much greater accuracy and speed in the MEASURE or AUTO modes.

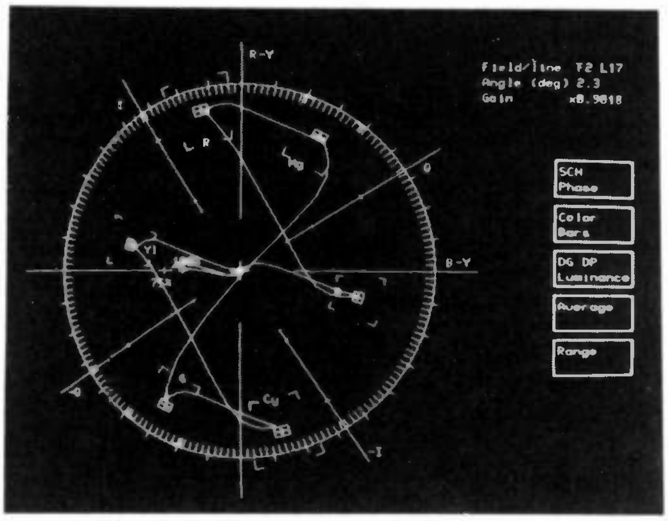


Fig.3 Vector Mode, color bars

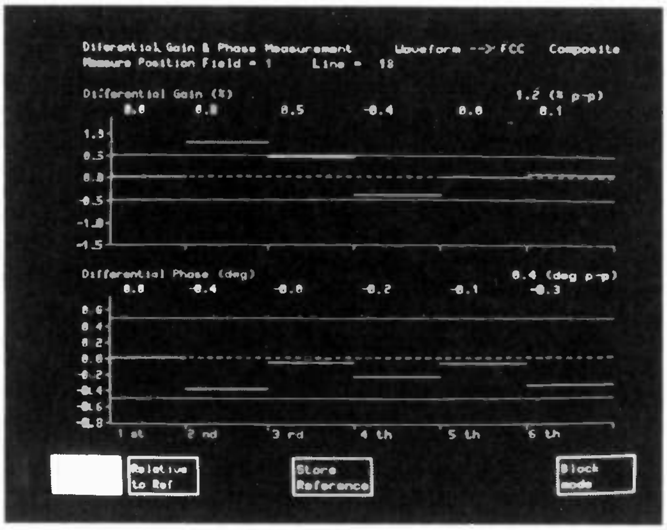


Fig.4 Diff.Gain/ Diff.Phase

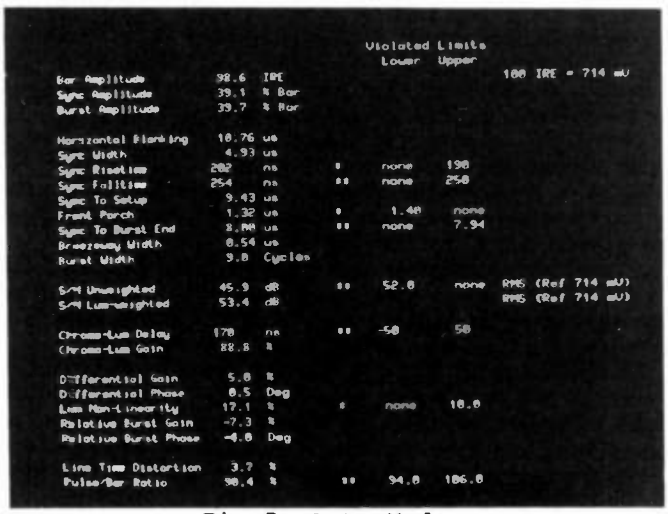


Fig.5 Auto Mode

Vector

A standard Vector display is presented in Vector Mode (Fig.3). In the 'move' mode, the knob functions as a digital goniometer. In the alternate 'expand' mode, the knob functions as a calibrated vector gain control. The Field and Line number of the selected Color Bar are also displayed. Ancillary Vector mode functions are Color Bar search and a Differential Gain/Differential Phase display (Fig.4).

Picture

A low-quality picture for source verification is provided in PICTURE mode. It has a 1 second update rate and a pseudo grey-scale is provided by algorithmic manipulation of the display and graticule bit-planes. The picture display always corresponds with the currently selected channel and a bright-up line is provided, corresponding with the current line selection in WAVEFORM mode.

Auto

In AUTO mode, most standard measurements can be made quickly and accurately with the touch of a single button. FCC, EIA-250C and RS-170 measurements are provided. The user can configure his own groupings from the available measurements. He can also generate his own 'caution' and 'alarm' limit files and associate them in a flexible way with named input signals. Printer support, both Postscript (for laser-printers) and Epson, is provided and hard-copies may be triggered on specific out-of-limits conditions and/or requested at specific times of the day. All user-defined parameters are stored in non-volatile memory. Lithium cells are used for a 10-year retention life.

A typical AUTO screen is shown in Fig.5. From left to right are the Measurement name, Value (in appropriate units), Caution (*) or Alarm (**) Flag, Violated Limits, and comments. The displayed measurement values are updated in place. The full measurement list normally occupies a space equivalent to several screens of text and the knob is used to smoothly scroll through the list.

Typical measurement accuracies are 0.1% Bar Amplitude, 0.25% Diff. Gain and 0.25 deg. Diff. Phase. Continuous auto-calibration of the analog signal path, using a 12-bit DAC, maintains both gain and DC stability. The VM700 is also capable of very fast measurement execution. For example, signal-to-noise (using a 512-point FFT) takes less than 1 second.

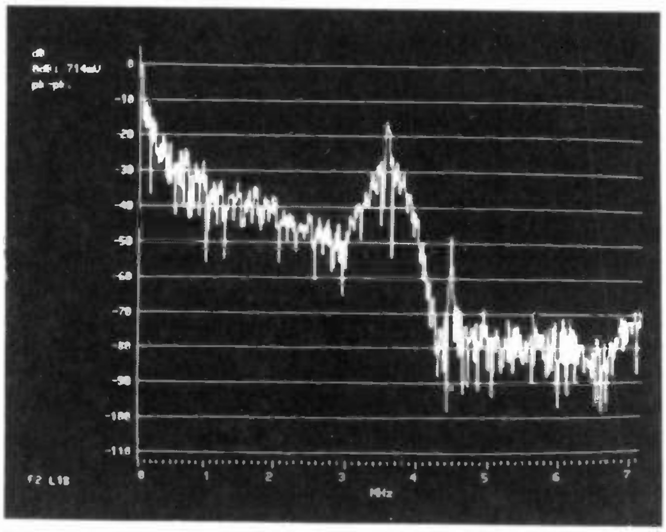


Fig.6 Measure mode, spectral display

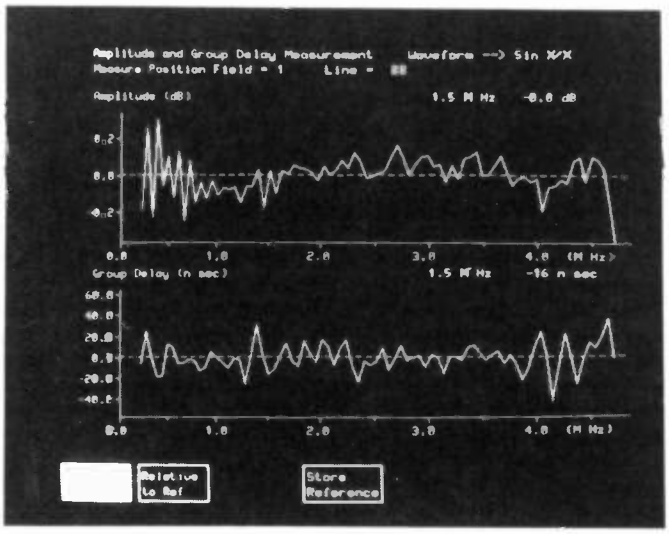


Fig.7 Freq. Response/Group Delay

Measure

In the MEASURE mode, the AUTO measurements are available for manual interaction by the user. Specific measurements may be grouped together and looped while the user performs equipment or system-related adjustments.

The MEASURE mode also incorporates several additional graphical displays, including a spectral display of a user-selected line (Fig.6). This is particularly useful for inspecting the noise spectrum on the 'quiet' line and can help trace the sources of noise-related system or equipment problems. A frequency-response/ group delay display (using the sinx/x test signal) is also provided (Fig.7).

Miscellaneous Features

The VM700 is equipped with two RS-232 ports which can be interfaced with modems, printers or external computers.

Function Key 'macros' are selectable from a soft-key menu. A function key is configured by putting it in a 'learn' mode whereby it records the steps taken by a user through a series of operations.

Three uncommitted option slots are available on the VM700. A GPIB option is planned. Further computer interfaces, such as Ethernet, will be made available as the need arises.

HARDWARE

The VM700 overall block diagram is shown in Fig.8.

Processor

The main CPU is a 16.7 Mhz 68020 32-bit microprocessor. Floating point math support is provided by a 68881 co-processor. The bus architecture is optimized for speed and minimum memory latency. System memory is 2 MBytes. Display resolution is 640 by 480, non-interlaced, with 60Hz refresh. Display data clock is 25MHz and data is supplied to the display by video RAMs appropriately multiplexed for the 2 bit-planes.

User-Interface

Control of the keypad, knob and touch-panel hardware is handled by a 68008 microprocessor interfacing with the 68020 via an interrupt and shared memory.

The keypad consists of momentary-action push-button switches with integral LEDs.

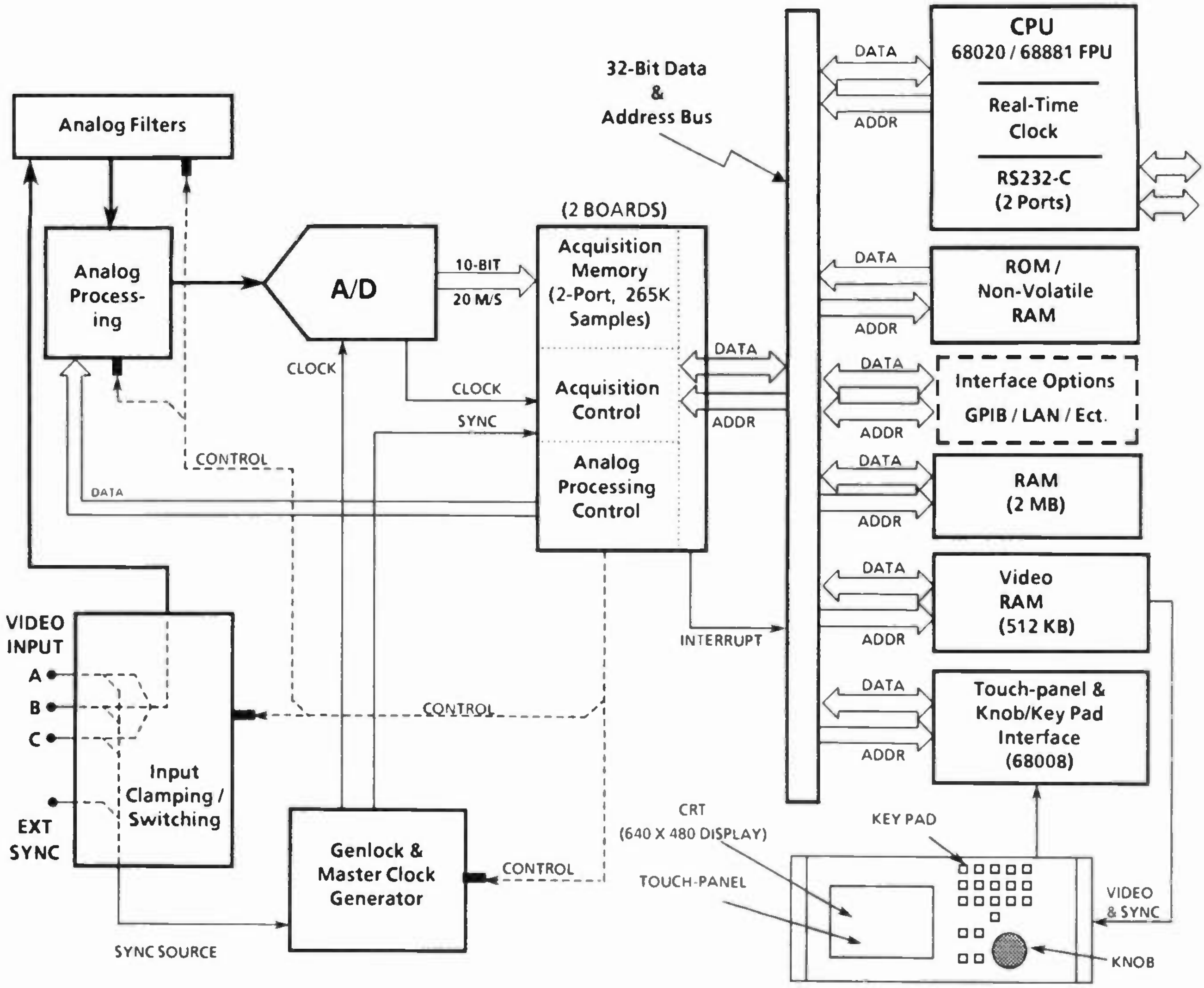


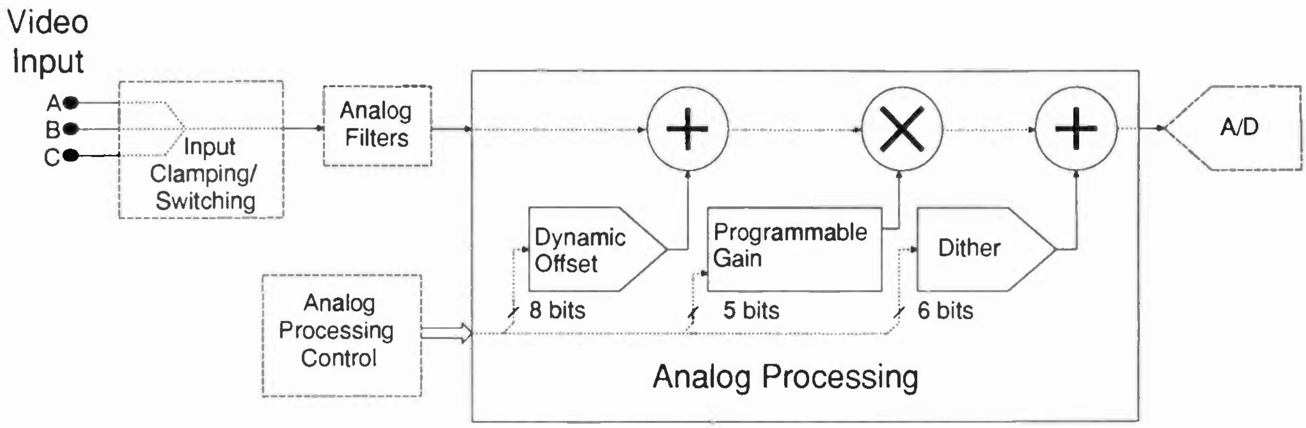
Fig.8 VM700 Overall Block Diagram

The touch-panel is designed and manufactured by Tektronix. It is analog-capacitive, and consists of a hard oxide coating deposited on a glass substrate. It is very damage-resistant and may be cleaned using normal CRT glass cleaners. It has excellent electrical linearity. The associated electronic circuitry provides X, Y data together with touch status to the 68008.

Analog Video Circuits

The VM700 has three video inputs. An external sync input is also provided. All inputs are high-impedance loop-through. For broadcast-quality signals, the 10-bit A/D and data acquisition clocks are genlocked to the incoming video. Sampling clock rate is 910 samples/line (4x subcarrier). The genlock source may be selected from any of the four inputs. A signal combiner/switcher is downstream of the inputs and its output is passed to the Analog Filter block, which incorporates software-selectable analog filters.

From the Analog Filters, the signal is passed to the Analog Processing module (Fig.9). In this module, the signal is manipulated under software control to maximize the utilization of the A/D dynamic range and minimize its cut-point errors. Typical steps in the operation of the module are as follows: An initial acquisition of the test signal is made with zero Dynamic Offset and a low value of Programmable Gain. The computer analyses the data and determines for each data sample the optimum offset and gain to comfortably fill the A/D dynamic range. It then loads a state-machine controlling the Analog Processing module with the offset and gain information. The computer then synchronizes the state-machine with the incoming video and resamples the signal. The offset and gain values are later backed out in the measurement computation. Dither



Dynamic Offset: -1.28 V to +1.27 V, in 10 mV increments
Programmable Gain: 0X to 7.75X in 0.25X gain increments
Dither: 1/8 LSB to 8 LSB

Fig.9 VM700 Analog Processing

is added as a small analog signal offset once per repeated signal acquisition. Dither increments available are 1/8 LSB with a maximum range of 8 LSBs. Using 1/8 LSB steps, the resolution of a 10-bit A/D, after averaging 8 repeated acquisitions, and assuming zero cut-point errors, is effectively 13 bits. Averaging data after applying dither amplitudes greater than 1 LSB tends to average out adjacent cut-point errors. The effective digitizing accuracy of a typical VM700 A/D, after 32 repeated signal acquisitions with an overall dither range of 4 LSB in 1/8 LSB increments, is approx. 12-bits (0.025%) at low frequencies and 10-bits (0.1%) at subcarrier.

Analog-Digital Converter

The VM700 A/D is a two-stage 10-bit flash A/D of proprietary design. Maximum clock rate is 22MHz. It has a removable anti-alias filter with a flat (0.1db) bandwidth to 5.2MHz and very carefully controlled group-delay characteristics (<2ns error at 3.58 and 4.43 Mhz). The A/D clock and data output signals are ECL logic levels, for minimum injection of digital noise into the analog path.

Data Acquisition and Control

Data acquisition is performed by a 2-port memory under control of a state machine clocked at sample rate. One port of the memory interfaces with the A/D output, while the other interfaces with the CPU bus. Memory size is 256K samples (approx. 1 NTSC field at 910 samples per line). The state machine is loaded with the required sampling pattern and acquisition is initialized after a predefined stimulus is received. Typical stimuli are frame and line sync.

FIRMWARE

All normal operations of the VM700 are controlled by the firmware. The unique user-interface, the various innovative display modes and the comprehensive set of measurements are all attributable to very clever software engineering. One of the most difficult challenges in the design of a complex firmware-based instrument is the preservation of real-time response from the instrument controls at all times. The VM700 excels in this regard.

The firmware architecture of the VM700 is based on a modern real-time operating system. The programming language is 'C' with minimal assembly-language optimization.

CONCLUSION

The VM700 is the first of a new family of computer-based test and measurement products for the television industry, incorporating advanced measurement and display techniques and unparalleled user-interfaces. It is the first instrument to successfully combine the functionality of a waveform monitor, vectorscope, automatic measurement set and noise measurement set in one convenient, economical package and may truly be viewed as a comprehensive solution to baseband video measurement needs.

ACKNOWLEDGMENTS

The author expresses his appreciation to the VM700 engineering team for their outstandingly successful implementation of brilliantly original concepts and to Tektronix management for their consistent support throughout this development program.

NEW HIGH RESOLUTION CCD IMAGER

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At NAB '88 Sony introduces the new BVP-7 broadcast CCD camera and the new DXCM7 professional video CCD camera. Both utilize a new high resolution CCD imager with more than 380,000 pixels. The paper will describe the attributes of a new HADS sensor - specially developed to increase sensitivity, substantially reduce dark current, improve spectral characteristics, and incorporate novel variable speed electronic shutter technique.

The new imager extends the portfolio of Sony CCD cameras to now include many EFP applications.

INTRODUCTION

At NAB '86 Sony introduced our first broadcast CCD camera - the BVP-5. This camera employs three CCD imagers in a classic RGB optical configuration. This CCD device - our ICX-018 - has some 268,000 pixels structured in a 2-dimensional geometric format of 510 discrete horizontal elements and 493 rows of these. This device is of the genre known as Interline Transfer (IT). Details of the ICX-018 have been described elsewhere.(1,2,3)

Every CCD imager has a distinct electronic personality which is very dependent upon the many separate design decisions made by the semiconductor physicists. It should be appreciated that CCD's are highly complex semiconductors. The final pixel design represents a myriad of separate ingenuous solutions to a wide complex of interacting mechanisms. It should also be noted that the CCD imagers is unequivocally an analog device, and just like its pickup tube counterpart, is fraught with many individual characteristics - some highly desirable - others less so. Just as decades old sustained struggle continues in the refinement of Plumbicon and Saticon tubes - so too, will the same struggle be a vital and never-ending element of the future evolution of the solid state imager.

At the SMPTE '87 Fall Conference, Sony introduced a second broadcast CCD camera - the BVP-50. This camera is physically almost identical to the BVP-5, but it employs another CCD imager - the ICX-028. This device represents a significant departure from the ICX-018 - primarily in the methodology employed to transfer the discrete charge packets. The ICX-028 uses a hybrid transfer mechanism commonly called Frame-Interline Transfer (FIT). Details of this device have been described in our paper read at the SMPTE '87 Conference.(4)

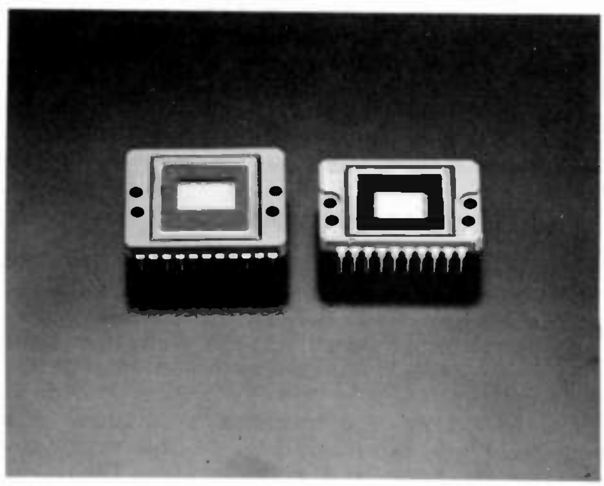
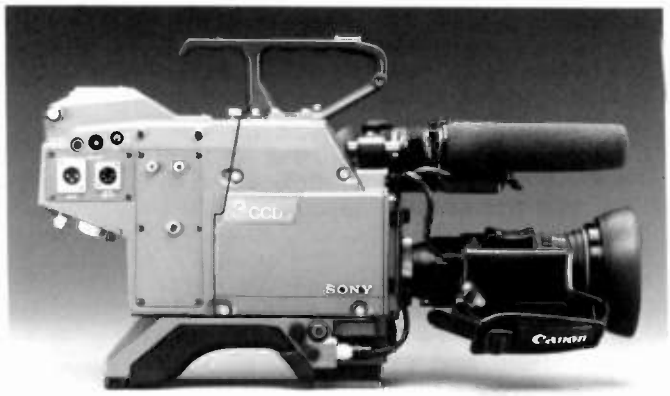


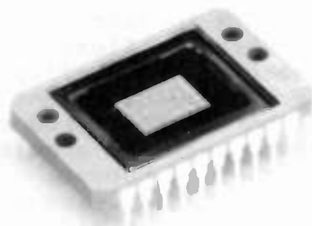
FIGURE 1 - PHOTOGRAPHS OF ICX-018 510" IT CCD AND ICX-028.

510 FIT CCD

The ICX-028 CCD offers two striking embellishments over its sister ICX-018 device: first, an elimination of the unwanted vertical smear artifact of the latter which becomes visible when it encounters a severe highlight (more than 4 f-stops above normal exposure); and second, it includes an integral electronic shutter mechanism. In all other respects, namely the individual pixel design, and the number of pixels - these two devices are very similar. Some refinements in sensor design did improve the S/N performance of the later device.



At this NAB '88 Sony introduces its third CCD device - ICX-032 - which represents some very significant departures from the two just described. The pixel count has been substantially elevated - to almost 380,000 - representing a total of 768 horizontal elements while retaining about the same 493 horizontal rows of the other devices. The individual pixel design itself represents a radical departure from that of the ICX-018/018 devices. This paper will describe the technical aspects of the new CCD design and their implication on the performance of the our new broadcast camera - the BVP-7 and the new professional video camera - the DXC-M7.



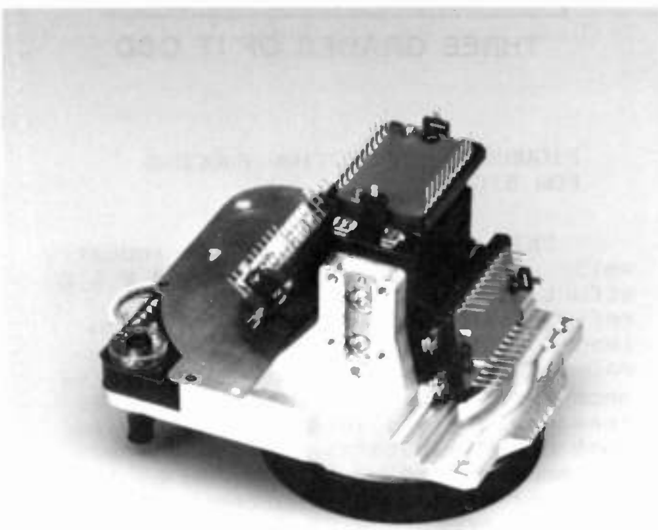


FIGURE 2. NEW ICX-032 CCD AND NEW BVP-7 AND DXC-M7 CAMERAS

A MULTIPLE APPROACH TO CCD DEVELOPMENT

But first - a word on the rationale underlying our offering a number of different CCD device designs in our Broadcast cameras.

In formulating an overall approach to our CCD development program, we recognized that both broadcast and professional video production has splintered into a wide variety of applications and needs. At one extreme, these embrace picture capture within the unpredictable and fast paced environment of electronic news gathering, while at another level they reach to the highest form of disciplined quality production required for commercial production, drama, etc. Just as a variety of professional VTR formats have been developed to offer a flexibility in optimally meeting a plurality of need - so too, are there different camera needs. The total video production industry also speaks to a quite wide spectrum in capital budget capabilities - no small issue. Anticipating that this wide disparity in needs would be an ever present dynamic, Sony has mapped out a long term blue print in CCD imager development.

Our major CCD development programs are most heavily driven by the development of devices that will service the extremely large volume of the consumer camera marketplace. Typically these cameras employ a single imager (pickup tube or CCD), and color stripe filter technology, to implement a cost effective reasonable performance criteria. The imager must have a resepctable inherent resolution capability to adequately service the requisite luminance and chroma needs of such cameras. It became a central part of our strategy to develop each generation of CCD with a resolution capability that would simultaneously meet this requirement -- while, at the same time meeting full professional video requirements when three of the devices were employed in such cameras. Further, this mass produced device would be based upon Interline-Transfer technology primarily because of the relative ease of manufacturing such devices with a respectable yield.

A small number of key design refinements are made to the basic IT CCD thus developed. These produce a variation on the generic design of the CCD which yields imagers admirably applicable to a professional video cameras.

Recognizing also that CCD's have a tolerance range on many performance parameters (a function of manufacturing tolerance control) a highly sophisticated testing system was developed to allow a device selection program to be implemented for these special devices. This would cull out two performance categories of CCD's emanating from these basic same production line - a high grade for professional video cameras and a superior grade for broadcast cameras. (Figure 3).

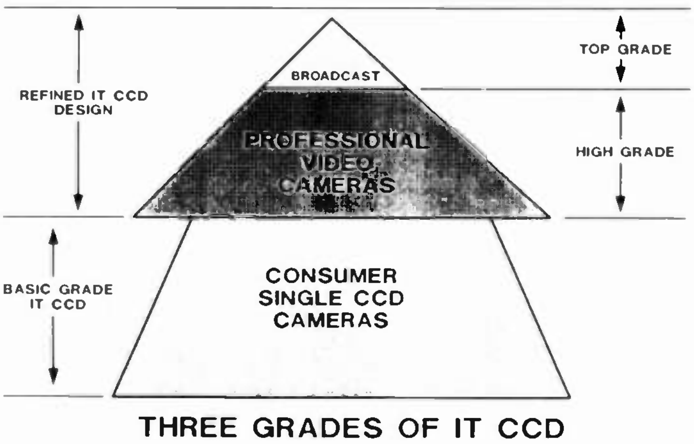


FIGURE 3 - SELECTION PROCESS FOR 510" IT CCD

This approach serves the industry well. A major and more focused R & D effort can be brought to bear on refining an inherently superior CCD imager - which is then produced in volume. This has two direct benefits once manufacturing achieves a reasonable yield level: one, device cost drops substantially; two, the enormous volume base provided by the consumer outlet ensures that the much smaller percentage yield of very high performance devices still adequately meets the more modest quantity requirements of professional video and broadcast.

Once this device successfully leaves the R&D lab we follow with another, more substantial, variation on the same basic design - namely, a hybrid Frame-Interline Transfer (FIT) device - tailored exclusively for the high end requirements of certain broadcast applications. (Figure 4) Of necessity, this volume base is substantially lower than that of the parent IT design. Device Cost is significantly higher - but a very superior device in terms of highlight handling is produced. The FIT CCD also allows easy incorporation of a multiple speed electronic shutter control.

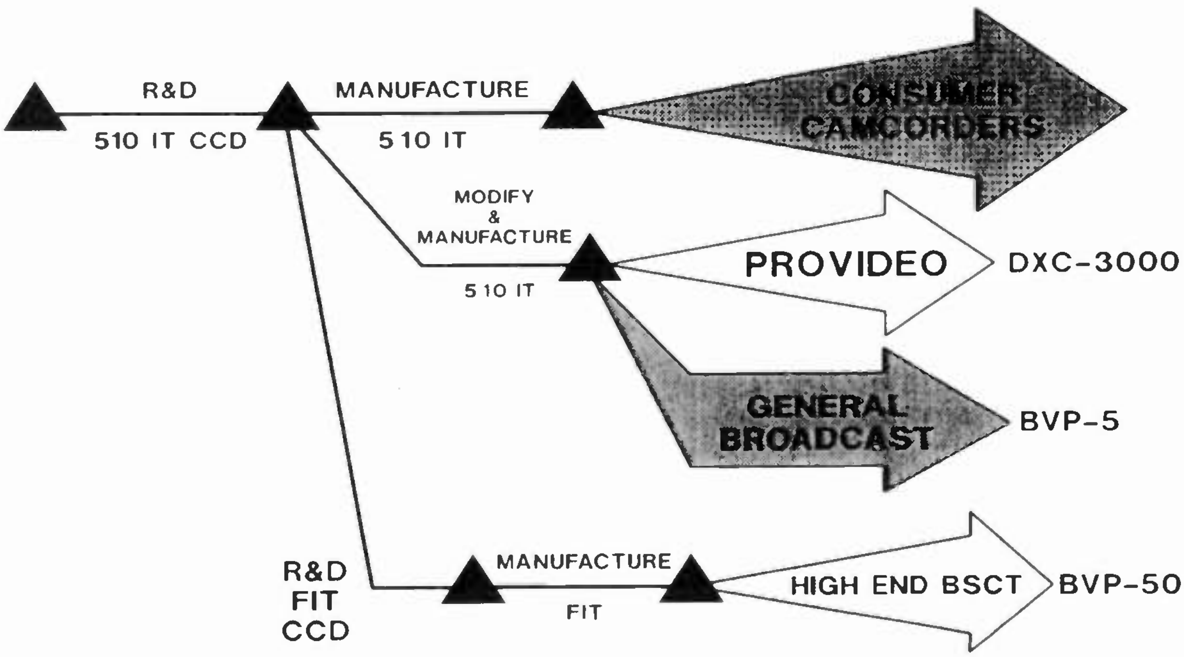


FIGURE 4 - SEQUENTIAL CCD R&D AND PRODUCTION CYCLES

This approach has worked with eminent success on our current BVP-5 and BVP-50 cameras. The successor FIT device is not at all intended to obsolete its IT predecessor. Two cameras each of intrinsically high picture performance, are offered as members of a family. They are, however, separated by some unique features - and by cost.

The device to be described in this paper carries this approach forward to the next step. A lengthy R&D project has now produced a CCD ICX-032 which exploits some quite different semiconductor technologies than the 510 element ICX-018/028. It is being simultaneously unveiled at this NAB '88 in the new DXC-M7 professional video camera and in the new BVP-7 broadcast camera. An R&D project to develop a future FIT version (more costly) is currently underway. We anticipate offering all four devices in a range of professional cameras for quite some years to come.

The New ICX-032 CCD Imager

Table 1 summarizes the salient parameters of this new 768 element CCD as compared to the 510 element CCD:

Table 1

		768 IT	510 IT	510 FIT
DEVICE	NTSC	ICX-032AL	ICX-018L	ICX-028L
	PAL	ICX-034AL	ICX-021L	ICX-029L
PIXEL NUMBER	NTSC	768^{H} x 493^{V}	510^{H} x 492^{V}	510^{H} x 492^{V}
	PAL	786^{H} x 581^{V}	500^{H} x 582^{V}	500^{H} x 582^{V}
UNIT CELL SIZE (µ METERS)	NTSC	11^{H} x 13^{V}	17^{H} x 13^{V}	17^{H} x 13^{V}
	PAL	11^{H} x 11^{V}	17^{H} x 11^{V}	17^{H} x 11^{V}

THE PIXEL DESIGN

The ICX-018/028 510 element CCD's now in quite widespread use within the broadcast and professional video industries are distinguished for their unique pixel design. Of special note is the MOS diode sensor employed as opposed to the more conventional photodiode employed in most other IT CCD's.

This sensor has been described elsewhere.(5) It strikes a state-of-the-art compromise between sensitivity, dynamic range, noise and lag - the latter, in particular, being superior to that of the conventional diode.

The new ICX-032 is a further step forward in individual sensor design. A totally new sensor technology, which we call the Hole Accumulator Diode Sensor (HADS), is a unique new sensor structure developed by the Sony Semiconductor Division. Figure 5 shows a simplified cross section of a single pixel and is indicative of the complexity of the composite semiconductor design:

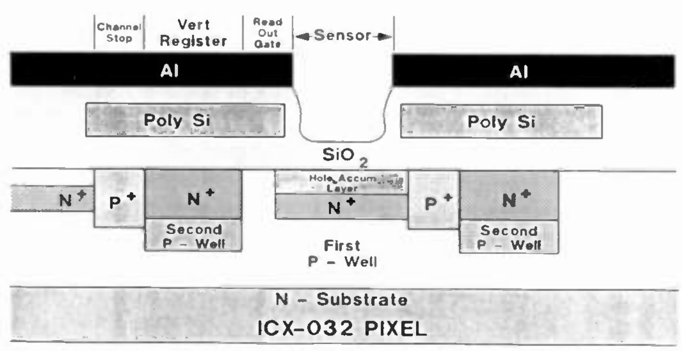


FIGURE 5 - CROSS SECTION OF SINGLE PIXEL

The HADS sensor offers five distinct advantages over the conventional diode or MOS diode sensor:

- Very low dark current
- Improved Dynamic Range
- Virtually lag free

- Improved spectral characteristic
- Electronic shutter unique new - built-in mechanism within each pixel

As each of these characteristics make an important contribution to overall picture quality we will look closely at each:

Low Dark Current

As discussed in some detail in reference 7, sensor dark current represents one of the very fundamental assessment criteria of CCD performance. The dynamic range of the imager is limited at the lower end primarily by the dark current characteristics. Dark current effects are exacerbated by the fact that its magnitude rises with temperature (doubling in level for every $10^{\circ}C$ temp. rise), and by the more troublesome aspect that its level variation from pixel to pixel can be a random distribution. This latter random level distribution is the primary contributor to the Fixed Pattern Hoise (FPN) behaviour of CCD's - which, for some device designs can seriously limit their imaging capabilities in low light scenes. **Figure 6** describes in simplistic manner the effect of random dark current levels on FPN:

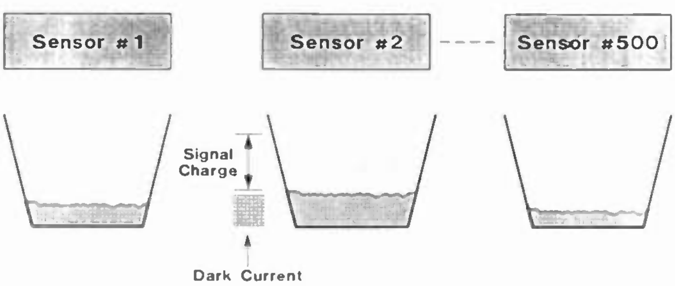


FIGURE 6 - MECHANISM OF FIXED PATTERN NOISE

The HADS sensor exhibits a dark current level some ten times less than that of our 510 element MOS diode sensor. This is a very significant step forward - the noise "floor" has been dramatically lowered.

Highlight Handling

At the other extreme, that is, under conditions of high light level inputs the sensor has also been endowed with some improvement.

The sensor design is such that a deep potential well is produced with a capacity that insures a 600% overload capability. So highlights up to this limit are handled in a perfectly linear fashion with no charge spills - and thus zero blooming effects. When a highlight exceeds this limit an overflow barrier comes into play that diverts all excess charge generation into an adjacent overflow drain of enormous capacity. Thus a clean "limiting" action takes place which ensures even the most severe of highlights cause no blooming artifacts whatever.

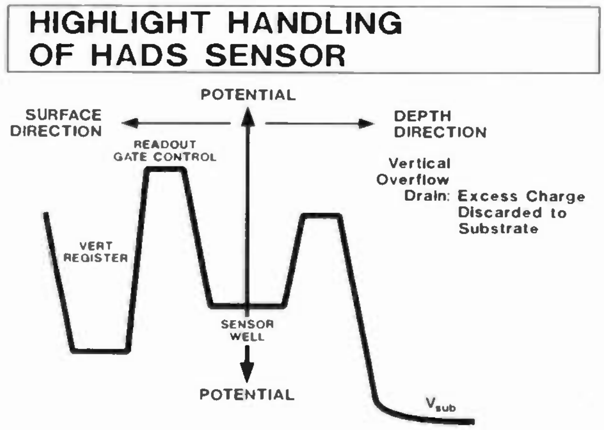


FIGURE 7 - HIGHLIGHT HANDLING MECHANISM OF ICX-032 CCD

Dynamic Range

Figure 8 shows the two extreme limitations which form the boundaries to the sensor signal generating capabilities - thus defining its dynamic range. At the upper boundary we are curtailed by the inherent capacity of the sensor "well" to store accumulated charge. The new HADS sensor can linearly handle light levels up to 600% of normal exposure. When this level is exceeded a very effective overflow mechanism comes into play which totally absorbs all excess charge and thus precludes any blooming artifacts.

At the lower boundary we have the "floor" defined by the dark current, thermal and 1/f noise, and the reset noise.

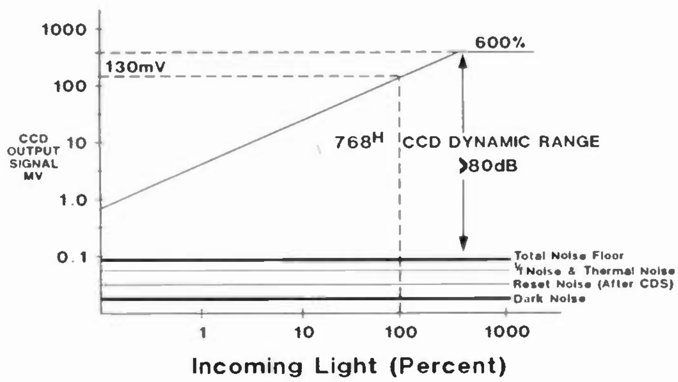


FIGURE 8 - DYNAMIC RANGE OF ICX-032

The extension of both boundaries of the HADS sensor has produced a dynamic range typically greater than 80 dB. This is far in excess of the best pickup tube available, and is about 10 dB beyond the capabilities of most contemporary CCD's. This greatly enhanced dynamic range will constitute a major boon to night time ENG shooting in allowing a quiet, lag free, image to be extracted from very poorly lit scenes while simultaneously handling background highlights without any blooming. In production work it will allow a very close approximation to the contrast range handling capabilities of modern film - thus offering a new video production flexibility hitherto denied the video camera.

Sensitivity

The spectral sensitivity of the HADS has been improved over that in our ICX-018/28 510 element CCD's. In particular, blue sensitivity exhibits a 50% improvement. The sensor Opening Ratio of the new ICX-032 is 34% in contrast to the 25% of the 510 element CCD (Figure 9).

One of the thin film layers - the poly-silicon - used on both of the 510 element CCD's has been removed in this new CCD, thus raising the sensitivity in the region of the shorter blue wavelengths.

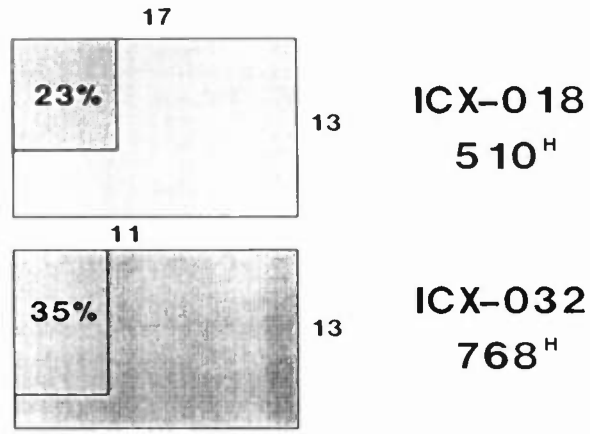


FIGURE 9 - OPENING RATIO

This larger opening ratio directly improves the light capture and while the intrinsic sensitivity of the new sensor is essentially the same as that of the ICX-018 - the effective sensitivity is directly enhanced by the amount of increase in Opening Ratio. The effective signal current which exits from the final output diode into the floating diffusion is almost 1.5 times greater. This allows the gain system of the camera to be setup to effect both a low light (high gain wide open lens sensitivity) enhancement and a basic luminance signal to noise improvement. The BVP-7 camera performance is as follows:

Nominal Sensitivity:	f5.6 for 2000 Lux Incident 3200°K Light on an 89.9% Reflectance White
Low Light Sensitivity:	15 Lux required to produce full Luminance level output with f1.4 lens wide open and gain set to +18 dB.

Signal to Noise: 62 dB Luminance unweighted pp signal to rms noise (4.2MHz)

Colorimetry

The improved spectral characteristic of the imager has a beneficial impact on the camera colorimetry as a whole. The BVP-5 and BVP-50 cameras have two primary curtailments to their ability to achieve a colorimetry approximating that of today's pickup tubes: at the blue end of the spectrum the response falls off more sharply than a PbO or Sat, while at the red end the response is somewhat impaired by the IR filter employed. This filter is more severe than it should be because of our desire to strike a compromise between an adequate red response and an attenuation of the red wavelength dependence of the ICX-018/028 CCD's on vertical smear. This problem does not exist in the new sensor.

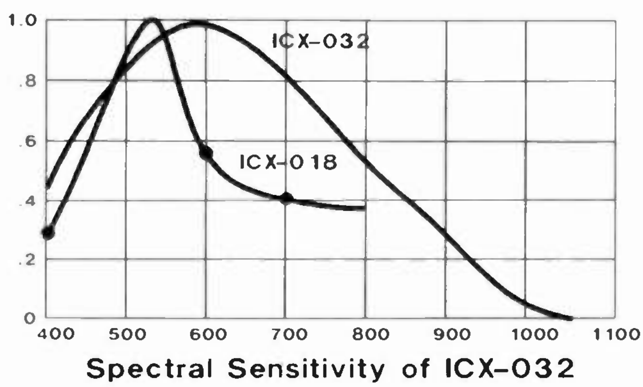


FIGURE 10. SPECTRAL CHARACTERISTIC OF NEW ICX-032 CCD VERSUS THAT OF ICX-018.

Figure 10 contrasts the spectral characteristic of the new ICX-032 imager with that of the ICX-018. Clearly the blue response is much improved and a broader spectral curve allows a better optimization of the prism taking characteristic. By carefully controlling the spectral taking characteristics of the beam splitting block, and utilizing an appropriately tailored linear matrix, the new BVP-7 camera exhibits an excellent color reproduction very closely approximating that of a modern Plumbicon camera.

ELECTRONIC SHUTTER

The new HADS pixel design brings a major evolution to Interline Transfer (IT) CCD design - namely, the incorporation of an electronic shutter mechanism. To date the various IT CCD designs have been without an electronic shutter. It has conventionally required the addition of a Frame-Interline Transfer (FIT) or a semi-FIT mode of operation to successfully add a variable speed electronic shutter control.(4)

In the FIT CCD the shutter action is achieved following the first high speed transfer - the Interline transfer - which transfers the sensor's accumulated charge packets into the adjacent registers. The shutter action involves a dual vertical transfer process - whereby wanted charges are transferred down into the frame storage array (for subsequent horizontal transfer out) and unwanted (or shuttered) charges are transferred UP into a vertical overflow drain.

The new HADS IT CCD, however, takes an entirely different approach by accomplishing the charge "separation" within each individual pixel itself. Figure 11 is a simplified rendition of the semiconductor action which separates the wanted charge information from that which is shuttered and discarded.

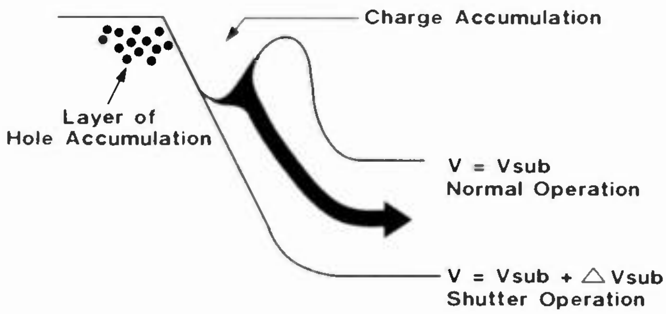


FIGURE 11 - ELECTRONIC SHUTTER ACTION OF HADS SENSOR.

When the shutter is not actuated the sub potential is held high and normal Interline Transfer action takes place. That is, the sensors integrate charge during the active field period - and a readout pulse which occurs in the succeeding vertical blanking period rapidly empties all sensors simultaneously into the adjacent registers (from where they are clocked out during the next active field period). Figure 12. When the shutter is actuated a series of pulses (occurring during each horizontal/blanking period) are applied to the substrate. These break the potential barrier of the well - and all of the accumulated charge is swept off into the substrate. This effectively "dumps" the charge accumulated during the horizontal active period into a large reservoir overflow drain.

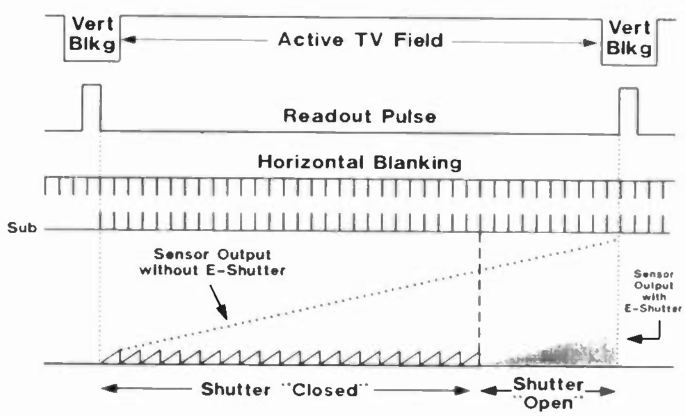


Figure 12 represents a timing diagram indicating the operation of the control pulses which actuate the useful charge accumulation and the "shuttered" charge removal.

In effect these pulses occur during the shuttered or ineffective period and serve to discard the unwanted charges. When the camera is unshuttered - at some time during the active field - the pulses switch off and normal Interline Transfer action resumes.

The BVP-7 has an operator control on this shutter timing position - and hence on shutter "speed". This is switchable from 1/60 to 1/100, 1/250, 1/500, 1/1000, 1/2000.

VERTICAL SMEAR

As was discussed in some detail in reference (4) the highlight associated artifact of vertical smear which can be generated from a severe highlight stimulus is indigenous to all CCD imagers. It can only be solved in the case of the Frame-Transfer (FT) CCD by utilizing a mechanical shutter interposed between the beam splitter input and the lens final port, and carefully synchronized to totally cut off light input during the vertical blanking when the actual frame transfer occurs. In the case of the Interline-Transfer (IT) CCD, the artifact is generated by different means - primarily involving an unwanted contamination (originated by the severe highlight stimulus) of the charge packets passing vertically through the registers while they are clocking during the extended time period of the active field period. This contamination can be reduced to a low level by specific semiconductor design precautions (2) - but not eliminated.

The Frame-Interline Transfer (FIT) CCD, on the other hand, can virtually eliminate the artifact by the expedient of a dual high speed transfer.(4) This elimination, being totally electronic, requires no employment of mechanical shutters and thus represents the optimum solution to the problem. For this reason, the FIT CCD is expected to represent the ultimate imager of choice for the higher performing cameras which can fully exploit all of the striking advantages of solid state imaging while remaining unhampered by this vexing limitation of vertical smear.

In our earlier ICX-018 device, our physicists had successfully curtailed the level of vertical smear to a point where only severe highlights caused a visible artifact. In the new ICX-032 CCD we continued our efforts to attenuate the stray carriers reaching the vertical registers. The utilization of an N substrate absorbs ~~the primary source of contamination - the stray carriers. This totally~~ removes the wavelength dependent mechanism which causes the subjectively unpleasant red tinge to the conventional IT CCD artifact. However, this now exposes the remaining mechanism of contamination (generally masked by the electronic) - namely, an optical "sneak" path - which stimulates a much lower level vertical smear which becomes visible on only the most severe of highlights.

Figure 13 indicates the mechanism by which a stray optical path is found which allows direct optical stimulation of the vertical register. Only the most intense of highlights will activate the generation of stray carriers in the vertical register by this means. Thus the vertical smear level overall is lower than that of the ICX-018 CCD - and it's also monochromatic and thus has a less objectionable appearance.

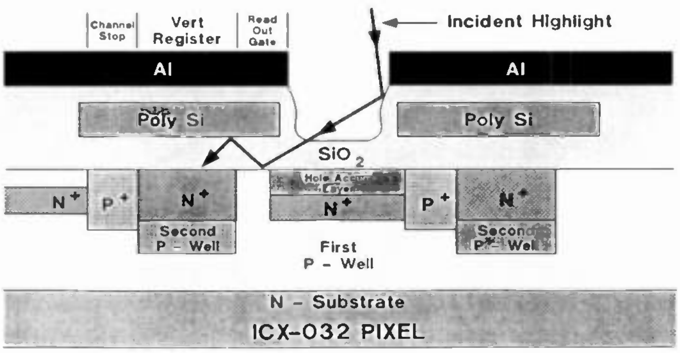


FIGURE 13 - MECHANISM OF VERTICAL SMEAR GENERATION IN NEW ICX-032 CCD.

RESOLUTION

The subject of resolution constitutes today the most oft-quoted yardstick of a given CCD's overall performance.(1,3,6) It certainly represents a primary parameter of assessment, and from the viewpoint of overall picture quality it undoubtedly is of considerable significance. It also speaks to the greatest struggle on the part of the CCD imager physicists and manufacturers. Increasing the number of pixels to raise resolution rapidly escalates the manufacturing tasks. In particular, it poses challenges to achieve an adequate yield of working devices while at the same time maintaining a high level of process control on the individual pixels to ensure reasonable performance tolerance limits on their many analog parameters - all of which impact picture quality.

CCD resolution is a major new topic in TV imaging studies it has introduced into TV cameras a new variable not encountered in 50 years of pickup tube cameras - namely, horizontal aliasing. This manifests itself as an in-band interference resulting from the viewing of high frequency scene information via the geometric sampling structure of the CCD imager.(3,6)

The final horizontal resolution of a broadcast CCD is a complex function of the number of horizontal pixel elements, the width of an individual pixel, the employment (or not) of the technique of spatial offset, the characteristics of an optical low pass filter (assuming this too is employed) and the characteristics of the electrical low pass filter which follows the output sample and hold circuit.

Figure 14 shows the response of each of these contributing elements in the new BVP-7 camera.

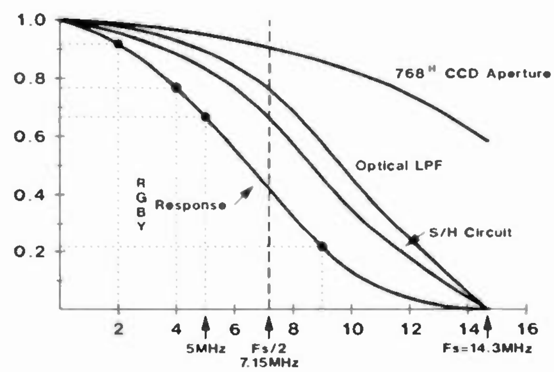


FIGURE 14 - TYPICAL MTF OF 3 CHIP 768 CCD CAMERA

In terms of Luminance frequency response it is clear than an excellent compromise has been achieved. The 768 horizontal elements in the new ICX-032 CCD clearly represents a number that superbly addresses the resolution requirements of the 525 NTSC system. A depth of modulation, superior to any present day pickup tube (of any image format size) is achieved. Without aperture correction we realize 66% MTF at 400 TVL/ph (5 MHz) and a 76% MTF at NTSC band edge of about 4 MHz. These are numbers which do not include the lens MTF.

Depending, therefore, on the quality of lens chosen very excellent MTF can be achieved in the useful 4.2 MHz NTSC frequency band - and beyond.

However, there still remains the issue of aliasing - always present to some degree despite the increase in pixel count. As clearly indicated in Fiugure 11, the luminance response extends beyond the critical FS/2 frequency. Beyond 7.15 MHz, therefore, on aliasing interference will be generated if the camera views high frequency information in the region 7.15 MHz.

Spatial offset offers a powerful means of combating Luminance aliasing. It does so by effecting a considerable reduction in the first order sideband alias signal amplitude. This comes about because of the reversal in phase of the Red and Blue first order sideband relative to that of Green which results from of the spatial offset. When the three signals are linearly summed to form a luminance signal in the encoder a cancellation effect occurs because of this phase reversal. The cancellation is not complete, however, because of the particular coefficients of R, G and B already structured into the NTSC Luminance equation. Nevertheless, only a very small residual alias remains and its distribution is shown in Figure 15. It should also be pointed out that the chance of scene information containing frequencies higher than 7MHz (of any real energy) is a rare statistical occurence indeed.

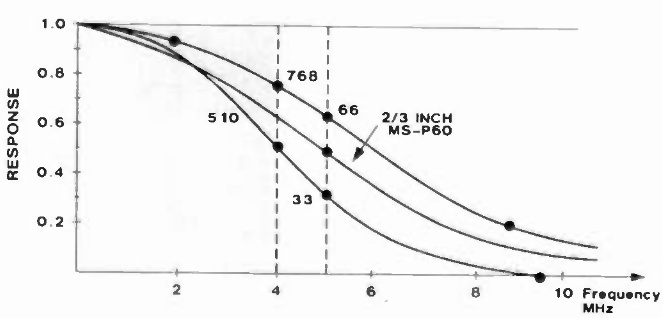


FIGURE 15 - MTF CURVE OF 768^H CCD COMPARED WITH 2/3 INCH PBO TUBE.

To put a perspective on the performance of the BVP-7 from a horizontal resolution viewpoint, we compare its MTF curve with that of our present 510 element BVP-5/50 cameras - and also with a high performance 2/3 inch Mixed Field PbO tube (as used in our BVP-350 camera).

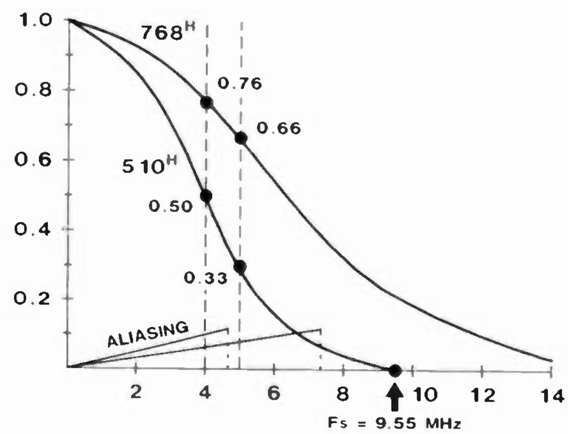


FIGURE 16 – COMPARISON OF 768^H CCD AND 510^H CCD

In summary, the introduction of a 768 element CCD has moved camera horizontal resolution beyond the capabilities of the best of pickup tubes.

CONCLUSION

The new 768^H element Interline Transfer CCD represents an important step forward in the paced evolution of the CCD imager. The increase in pixel count has dramatically elevated the horizontal resolution. A new sensor element has been described which narrows the gap significantly between video and film image capture by virtue of a dramatic increase in dynamic range, virtual removal of lag, and an improved spectral response. Vertical smear, indigenous to the Interline Transfer technology employed, while significantly improved remains, however, as a limitation to be considered for certain types of difficult shooting. This has been improved considerably: its inherent level is lower than our existing 510 IT CCD, and more importantly its subjective appearance is monochrome rather than the more disturbing red of the earlier CCD.

We have described the attributes of our 510 element CCD in some detail in previous publications - and pointed out that in summary the positive attributes of this device were admirably suited to the needs of ENG where flexible picture capture generally supercedes the need for highest picture quality. The new CCD described indisputably has much to offer high quality program production - as long as the still resident limitations are understood and worked around. In terms of overall picture quality the 2/3 mixed field PbO still remains the ultimately superior device. Yet the BVP-7 will offer important advantages even here - for certain types of program production.

The gap between the pickup tube and the CCD has narrowed. Each retains distinct advantages. It's with great difficulty that the true significance of the differences - on overall picture quality - can be described on paper. The end user needs to carefully evaluate each in light of the particular picture capture criteria sought - and then choose. Happily the choice exists.

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A NEW APPROACH TO EDITING EPISODIC AND MOVIE-OF-THE-WEEK TELEVISION PRESENTATIONS ORIGINATING ON FILM

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INTRODUCTION

During the past twenty years, there have been numerous attempts on the part of the videotape community to encourage television producers to originate their programs on videotape, and post-produce them electronically. Indeed, videotape production and post-production are ideally suited for certain types of programs, such as variety shows, situation comedies, soap operas and awards shows. All of these share common characteristics: much like theater productions, they take place basically in one setting, and they are essentially performed in one continuous run from beginning to end. The television programs are thoroughly rehearsed, and then shot (often before a live audience) with multiple cameras picking up a variety of camera angles simultaneously. The first cut of the program is essentially accomplished at the same time as the principal photography.

Dramatic presentations, however, have traditionally been, and continue to be, shot on film. Data supplied by Eastman Kodak indicate that 83 percent of all prime-time television programs this 1987-88 season are still originated on film. There are several reasons for this. One is flexibility in production. Most dramatic film production takes place in a variety of locations, and is shot over a period of several days, mainly using a single camera to shoot the entire program. Moreover, the film camera's wireless portability and reliability provide greater flexibility and freedom of movement during the production process.

Additionally, the much desired "film look" in dramatic programs denotes quality and production values which add prestige to the show. Lastly, the recent preoccupation with high-definition television has convinced many television producers that the higher resolution provided by film will assure their product long life in syndication and in foreign markets, no matter what HDTV standards are adopted anywhere in the world.

Up until recently, programs that were shot on film were also post-produced and edited on film using the same techniques that have served film makers so well over the past 60 years. The resultant film program was deliverd to the network for transfer to videotape and broadcast.

ELECTRONIC POST: LINEAR AND NON-LINEAR EDITING

In the mid 1970s two technological developments changed the situation dramatically. One was the adoption of a universal SMPTE timecode that could provide absolute reference for film and tape regardless of frame-rate. The second was the development of film-to-tape transfer systems (such as the Rank Cintel Flying Spot Scanners and others) that allowed the transfer of original negative directly to videotape. Film producers were now presented with what appeared to be a golden opportunity. The ability to shoot on film, post-produce on videotape, and be able to save time as well as money in the process.

The initial expectations, however, that television programs originated on film could be edited and post-produced quickly and cost-effectively on videotape failed to materialize, primarily because computerized videotape-based editing systems were essentially unsuited for single-camera dramatic shows because of the differences between linear and non-linear editing.

Videotape editing systems operate in a linear fashion: source tapes are played, edit decisions are made, and the selected segments are recorded in sequential order onto a new roll of blank videotape. Lengthening or shortening the assembled edit requires either repeating all the edits before or after the change, or taking large sections of the program down several generations. These techniques, that work so well for programs that are shot on videotape using several cameras simultaneously, do not work well for the film editor. The essence of the film editor's art is a constant process of re-editing and making changes. The first cut is always long or short. Through a series of changes and re-edits, the cut program is trimmed and polished until director and producer are satisfied with its final state. These changes are accomplished by the film editor, who is accustomed to working in a non-linear fashion, with ready access to any frame of film in a production, using scissors and splicing tape.

The Montage Picture Processor, introduced at NAB 1984, was the first commercial, viable, non-linear electronic editing system. It literally revolutionized the industry by providing the critical component that made all the elements in the electronic post-production come together.

THE MONTAGE PICTURE PROCESSOR

The Montage Picture Processor combines modern videotape and advanced microprocessing techniques to create a powerful, menu-driven, non-linear film-style "electronic flatbed" editing system. It is a cost-effective tool for organizing, storing, randomly accessing and retrieving vast quantities of picture and sound information -- greatly facilitating the post-production of theatrical features, made-for-TV movies, episodic television series, commercials, and other single-camera projects shot on film or videotape.

The Montage Picture Processor offers the opportunity to evaluate as many different versions of a scene as desired for immediate comparisons. Edits are automatically tracked and stored in the computer's memory. Entire previous cuts can be stored and accessed for reevaluation at any time, without rerecording or requiring additional lab prints.

Closely immitating the way film editors work, Montage is a picture-oriented editing system that enhances the creative, intuitive manner in which editors cut film -- editing and re-editing -- pacing the action for dramatic impact. User-friendly controls are not numeric, so the editor can concentrate on the pictures to edit without ever worrying about the underlying computer equipment. Indeed, Montage adds a new dimension to television and film post-production that simply cannot be matched by conventional film editing (nor by competeing electronic editing systems).

Hardware

The systen consists of a picture processing console, a C.P.U cabinet, two cabinets housing 17 half-inch VCRs (which act as source memory modules), and a terminal with a custom keyboard and stand. A laser "storyboard" printer (and stand) is optional.

The Picture Processing Console features: fourteen 4" monochrome monitors, four 5" monochrome monitors, one 13" color monitor; picture processing control wheels (or knobs), levers, and keys; electronic stylus "grease pencil", adjustable operator console, audio level and balance control, and a pair of compact monitor speakers and stands.

The C.P.U Cabinet features: a Motorola 68000 central processor as the main C.P.U., two memory boards with a total of 3 megabytes of random access memory, an 80-megabyte Winchester hard disk drive, removable 10-megabyte cartridge disk drive, an 8" floppy disk (to provide compatible edit decision lists in the industry's standard format), power supply, central AC system power distribution and filtering, 3/4" editing videotape recorder, 90-watt stereo amplifier (45 watts per channel), proprietary picture and sound digitizers and smooth scrolling picture display processors.

The two "source memory" cabinets feature two racks housing a total of 17 SuperBeta Hi-Fi 1/2" VCRs with Montage proprietary interfaces and time code readers, a video and audio switching matrix, system master timecode generator (longitudinal and/or VITC), a Montage dissolve/wipe special effects generator with approximately 20 standard wipes, dissolves, and transitional effects, as well as two digital time base correctors, including master sync generator, color black generator, and genlock capability. All are slaved to the main computer. These VCRs, which store 17 identical copies of the same material, feature a load capacity of 4.5 hours worth of video and stereo audio source material for editing. The two time base correctors, master sync generator, yield a quality, stable video image.

THE MONTAGE POST-PRODUCTION PROCESS

The post-production process begins with a transfer from the original negative to videotape. It is then synchronized electronically. A 3/4" copy is recorded simultaneously which will serve as the source material to be loaded into the Montage via the 5850 VCR playback. The 3/4" VCR will participate later in the assembling of the finished worktape, as an editor/recorder.

The editor is provided with seven "electronic workbins" (preassigned storage areas in computer memory) -- including Source Bin, Pull Bin, Discard Bin, and four Work Bins -- for organizing the program material as the workprint is built. Pictorial material can be clipped into segments of any length, stored in the various "workbins" until needed, and then brought out at random for viewing.

The Montage Processor also has a unique capability to create what amounts to an "electronic trimbin", which allows the editor to clearly identify each clip by the digitized picture labels that represent the head and tail frames of each clip. At least 2,500 different clips of varying lengths can be stored in memory and easily accessed for viewing at any one time. A unique visual display features seven pairs of B&W screens, showing the editor the picture labels for head/tail frames of seven (out of the 2,500) potential clips that can be seen simultaneously. Clips can be accessed by scrolling, with control wheels that look much like the knobs on the KEM flatbed transport. Selected clips can be displayed on four larger B&W monitors. Segments, selections, or an entire edit session can be viewed on a large 13-inch color monitor.

The editor can move clips between the bins in the system with single keystrokes, insert them between two other clips, rearrange and try out as many different versions as desired. Computer-controlled efficient playback procedures, which incorporate a "look-ahead" capability, make it possible for the editor to electronically "splice" a large number of clips, and the "play" them back in full color with their attendant sound, exactly as they were arranged, without waiting for the machine to search for the next edit. He or she can then make all the desired changes, and immediatly play the cut again, or try it several different ways and play them back in immediate succession for instant comparison. All this is easily accomplished with unprecedented flexibility and speed.

A special "writing tablet" located in the center of the Montage console allows editors to identify clips and mark takes, using an electronic stylus the way they would use a grease pencil, noting where special effects, moves or cropping may be needed. These markings are then visible on specific clips when viewed on the systems clip display monitor. Indicating the desired number of frames wanted for the transition, the editor can easily add fades, dissolves, soft cuts and wipes between scenes, all at the touch of a button. The built-in special effects generator features some twenty standard transition effects that can actually be seen during preview "playback."

A unique "storyboard" capability allows Montage to print out the contents of any or all of the workbins of the system in pictorial form, with the picture labels for each head and tail frame of the selected clips. The printer will also produce lists of cuts, and other desired user information.

Instant sound overlaps are easily accomplished with Montage. Matching sound to action, the editor can separatly trim picture to sound frame-by-frame or can even lay a temporary music or effects track, or record a narration.

Designed to edit two sound tracks and one picture track, the Montage Picture Processor will track all three independently. No time code or any other numbers need ever appear until the end of the editing process, after all the re-editing has been done.

For motion picture theatrical release, the Montage computer will then generate a frame-accurate cut list for conforming the answer print and, finally, the original film negative. The hard copy printout provides edge numbers and frames, feet and frames, total frame count, and sound code numbers. The printout automatically indicates two-frame "hot splice" proximity edits, as well as duplicate use of dailies.

For television release, a single, clean edit decision list is generated for electronic auto-assembly of the 1" videotape master. The EDL can be printed out or recorded on industry standard format 8" floppy disks (for use in on-line assembly).

THE "SCORECARD"

The success of any system is determined by use and growth. Since it was introduced at NAB in 1984, the Montage Picture Processor continued to grow and gain more adherents, scoring significant breakthoughs in the electronic editing of feature films and television programs originating on film, revolutionizing the film editing process by providing film editors and directors with unprecedented flexibility in the creative process. The impact on the electronic post-production process has been enormous. As a result, 50% of prime-time TV this season is now post-produced electronically, using the Montage and other systems.

Throughout these past years, the Montage system has been used extensively to edit commercials, music videos, episodic television series, full lenth documentaries, as well as theatrical motion pictures films. To date, the system has already been used in the editing of six major motion picture releases: Sidney Lumet's POWER (1985), Alan Alda's SWEET LIBERTY (1986), Susan Seidelman's MAKING MR. RIGHT (1986), and Stanley Kubrick's FULL METAL JACKET (1987), also a new feature film starring Tom Waits in concert, to be released in 1988, and Susan Seidelman's COOKIE, starring Peter Falk, a new feature produced, photographed and edited in New York this spring.

Montage is currently being used on major network television series, such as MacGYVER for ABC, BEAUTY AND THE BEAST for CBS, THE NEW ADVENTURES OF BEANS BAXTER and WERE-WOLF for Fox Broadcasting: THE NEW MONKEES and WAR STORIES for HBO, as well as THE HIGHWAYMAN, SUPERCARRIER, THE NEW SEA HUNT and STAR TREK: THE NEXT GENERATION. Recent network specials and made-for-TV movies include KIDS LIKE THESE, THE DAY MY KID WENT PUNK, THE ANN JILLIAN STORY, DENNIS THE MENACE, EMINENT DOMAIN and UN-HOLY MATRIMONY.

FUTURE APPLICATIONS

With the adoption of HDTV as a viable production standard for the television industry, we believe that single-camera film-style projects will be increasingly produced on HDTV videotape. Montage is ready, willing and able to serve as a sophisticated editing system for precisely that type of application. Montage can fulfill its potential as we continue to update and develop new software for a wide range of applications.

Montage is also delving into various editorial functions such as archival storage and retrieval of news and sports material for the broadcast user. We also intend to interface with small format ENG equipment to provide "instant" editorial access.

ON SUMMARY

Montage research and development is experimenting with developing various types of storage devices including videodisc. We are also trying to develop interfaces that will enable our system to relate directly with sophisticated production switchers, downstream audio and video devices, as well as on-line recorders and editors. It is our mandate to remain in the forefront of post-production and non-linear editing applications in the television and motion picture industries.

FACTORS AFFECTING ON-AIR RELIABILITY OF SOLID STATE TRANSMITTERS

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ABSTRACT

Solid state technology in and of itself is not inherently the answer to every broadcasters prayer for a long life, low maintenance transmitter. As with any technology there are numerous factors influencing operational lifetime, maintenance costs and ultimately the on-air time of high power radio and TV broadcast transmitters.

This paper will examine factors such as design philosophy, device technology, power supplies, cooling, cabinet design, control systems design as well as system architectures that lend themselves to increased or shortened lifetimes of solid state transmitters. This paper hopes to inform broadcasters who, armed with this information, will be in a better position to judge for themselves the merits of one transmitter design over that of another.

INTRODUCTION

Solid state technology has been around for nearly 40 years. During those 40 years obviously many improvements in the technology have occurred. Most early applications were centered around the use of diodes or transistors as substitutes for applications utilizing tubes. Early applications, especially those of an analog nature, were naturals for this substitution. As a result, the operational amplifier was developed in the early sixties.

Improvements were being made in parallel on the digital side bringing us first resistor transistor logic (RTL), diode transistor logic (DTL) and finally transistor transistor logic (TTL). This accelerated the electronics industry into what today is known as the "age of the computer". While these developments were occurring on the analog and digital side, not nearly as rapid development occurred in the area of solid state power devices. Solid state technology at radio frequencies is in the infancy stages when compared to the use of solid state in logic and control.

The first uses of solid state RF power amplifiers were medium wave AM radio frequencies. Applications at FM, VHF and UHF were initially as tube driver amplifiers. Consequently, not as much is known by the broadcaster today of the pitfalls of the solid state transmitter designs as certainly is known of tube transmitter designs. We are comparing the knowledge gained in tube transmitter design over perhaps a 75 year time span to the knowledge gained in solid state transmitter design over perhaps a 5 to 10 year period.

What then should a broadcaster look for in a solid state transmitter or what questions should one ask as curiosity or need drives them down the road to solid state? This paper will examine many factors affecting the end goal of a solid state transmitter, namely on-air reliability. Although the paper is of a survey nature, it is hoped that it will stimulate the thinking that broadcasters should go through in deciding what product best services their needs. No technology, regardless of its technical merit, is useful until it serves a customer's need.

DESIGN PHILOSOPHIES

Lets first examine what design factors affect on-air reliability of a transmitter. Many of these factors, of course, can apply to both tube or solid state technologies. However, as mentioned above, because we are in the infancy of solid state design, special attention should again be paid to these factors because a technology in and of itself does not insure a good design. You, the user, and the transmitter design engineer are the ones that determine the success or failure of a design.

Protection Against Extreme Load Events

Failures occur when a part or subassembly is subjected to loads that exceed the design rating of that part or subassembly. If the load excursion above the safe operating limit is considerable, failure is almost instantaneous. Typically a component operated significantly below its rated limit will have a lower failure rate. Protection against load excursions above the safe operating limits therefore is crucial in designing a transmitter for the highest on-air reliability. Device derating is a must. Derating should include the effects of temperature, thermal shock, current, voltage, vibration, altitude, power, VSWR and input levels to name just a few. Transmitter system designs should be laid out to minimize heat input from other members of the system.

Transient voltage protection on incoming power lines and circuits and where solid state devices interface with either capacitive or inductive loads is also good design practice. Even passive devices such as resistors can sustain damage from transients if the energy levels of the transient are high enough. Electrostatic discharge protection is also key to providing long life to metal oxide semiconductors commonly used as solid state RF amplifiers today. Static discharge protection using shielding and good grounding practices is essential for long life and high reliability.

System reliability is also improved by eliminating system failures caused by the effect of combined loads. Although difficult to analyze, it is extremely important to protect against impact of a combined loading on a device or subassembly. For example: a good design will take into account combinations of voltage, current, and temperature when developing a cooling system. How many times have you read a spec sheet and wondered if the device or transmitter could operate with all the specifications at their maximum simultaneously?

Another affect of combined loads is the combination of a high frequency stress that maybe superimposed on a low frequency stress. Thermal shock is an example of high frequency stress that may occur on top of a low frequency stress such as variations in the ambient temperature.

System reliability is also improved by eliminating system failure caused by sneak faults. Sneak faults occur when current flow is along an unexpected path, current doesn't flow along the expected route, current flow is at the incorrect time (poor timing) or doesn't flow at the correct time (also poor timing).

The presence of false or ambiguous indications or false or ambiguous labels on controls or indicators also compromises system reliability. If you are unaware of an over-stressed condition, you will continue to operate the transmitter without vital knowledge that may dictate replacement of a module or a subassembly. System reliability diminishes when protection circuitry fails and that failure is not self-revealing. This phenomenon is called "the enabling event phenomenon."

The enabling event phenomenon are events which while not causing a failure in and of themselves will set up or enable a second event that could lead to a transmitter failure. This type of phenomenon in a design is insidious especially if the enabling event is not self-revealing. Examples of enabling events are warning systems that have failed or have been disabled for maintenance or because they create spurious warnings and therefore are not dependable. Incorrectly set controls that give false readouts are another example of enabling events. Redundant elements that are out of action for repair or maintenance are yet another example of enabling events. You should be prompted to ask what the consequences of a warning device failure, incorrectly set controls or routine maintenance will be in a design. Your transmitter manufacturer should be able to show you how the design indeed was subjected to testing of these phenomenon and resulted in a predictable safe outcome.

Common Mode Failures

A common mode failure is a failure which can lead to failure of all paths in a redundant configuration. In the design of redundant systems, therefore, it is important to identify and eliminate sources of common mode failures or to increase their reliability to at least an order of magnitude above the reliability of the redundant system. Examples of common mode failure points in a design are switching circuits that activate standby or redundant systems, sensors that detect failure of a path, indicator systems to alert personnel of path failure, operating actions common to different paths, software which is common to all paths or software timing problems between parallel processors.

System Design

Poor system design practices can degrade transmitters having the best components. If two events, A and B, are independent, the probability of A occurring is independent of the probability of B occurring. The reliability of an independent event, A, can be expressed as Equation #1:

$$P(A)=EXP(-\lambda T)$$

where T represents the amount of operation time in hours and LAMBDA represents the failure rate expressed in failures for unit of time of that event. The calculated reliability of a non-redundant (independent) system is calculated by summing the failure rates of each independent component in a subassembly and then calculating the probability of failure over some period of time (T) using Equation #1. This technique is commonly referred to as "piece part reliability prediction". Intuitively one can see that if the number of individual piece parts is high a higher failure rate will occur than a subassembly with fewer parts of the same type. Hence the expression, "keep it simple" has meaning when it comes to subassembly or system reliability.

A method to improve reliability in systems with independent parts is by using the concept of redundancy. In a redundant system consisting of two identical independent elements, A and B, operating in parallel the probability of failure is increased because only one of the two elements is required for system operation. Mathematically this probability is expressed as Equation #2:

$$P(A+B)=P(A)+P(B)-P(A)xP(B)$$

If the probability of failure is .5 for subassembly A and the probability of failure is .5 for subassembly B, then the probability of a redundant system consisting of subassembly A or B is improved to .75. Clearly in the example, by adding redundancy one can improve the reliability of a system.There is, however, a dichotomy in that redundancy protection circuitry is inconsistent with fewer parts and less complexity.

Another area related to MTBF that is perhaps more important than MTBF is system availability. System availability is the actual time that the transmitter is in service or could be in service. System availability is defined as Equation #3:

$$Availability=MTBF/(MTBF+MTTR+MPMT)$$

MTBF equals Mean Time Between Failure measured in hours. MTTR equals Mean Time To Repair also expressed in hours. MPMT equals Mean Preventative Maintenance Time also expressed in hours.

From Equation #3, one can see that system availability is dependent upon Mean Time To Repair. Mean Time To Repair includes not only the time actually to repair the failure but also the preparation time prior to doing the job to handle any logistics delay such as waiting for spare parts. Preventative maintenance directly affects system availability. If a system has to be taken off line for preventative maintenance that amount of time reduces the time that the system is available for its intended function. Another subtle issue related to mean preventative maintenance time is whether or not the preventative maintenance is a planned or unplanned activity. For example: a transmitter design that is so unforgiving that it requires maintenance in the middle of a Superbowl is certainly not desirable. Preventative maintenance should be performed when it is convenient to your operation and not necessarily convenient to the equipment itself. When considering the impact of maintenance on the reliability of a transmitter one should consider whether a part that is replaced in a maintenance mode has a decreasing, increasing, or constant failure rate over time. If the part has a constant failure rate over time, then replacing it will make no difference to the failure probability. If on the other hand, the part has an increasing failure rate with time, then replacing it before failure will improve the reliability of the system. Replacing a part having a decreasing failure rate with time can actually decrease the reliability of the product by allowing one to insert in its place a part with a higher failure probability.

Built in Test

Built in test functions can be very effective in increasing system availability by increasing our confidence in the well being of the system. Usually built in test adds additional circuitry to the system. If not kept simple, a decrease in system reliability can occur due to false indications as a function of the failure of the built in test system.

In considering built in test, a logical question that one might ask is "what is the level of test coverage?" Figures of merit should exist to show which possible failure modes can be detected and diagnosed. High test coverage generally means quick and easy repair which consequently leads to higher system availability.

A another aspect in system design is what consideration has been given to the human reliability factor? By human reliability factor I mean, what consideration has been taken for the interface between the transmitter and the user? A question one might ask is does the design make it easy during operation, maintenance or repair to do the right thing and difficult to do the wrong thing. Another question that one might ask is how much of the design's operational performance or reliability depends on the human factor for correct adjustment in order to have proper operation. Finally, transmitters that have high levels of part interchangeability reduces the need from many types of spares and reduces probable error due to misrepair by substitution of the wrong spare.

SYSTEM ELEMENTS

Now that we have talked about system design philosophy in a general sense, lets turn to specific system elements and comment on questions that a broadcaster might ask of a transmitter vendor.

Solid State Devices

Broadcast transmitters today cover the frequency spectrum from 100 kHz up to 1000 MHz. Operation is linear for TV but not required for FM and AM service. To cover this broad frequency range with good gain and frequency response, two types of semiconductor technology exist today that have wide usage. These two technology types are the Bipolar Transistor and the Field Effect Transistor (FET). Although both technologies exist in the market today, the tendency as of late has been for the market to migrate toward a higher usage of FET's and away from the Bipolar Transistor. This in part is attributable to the higher amplification gain and supply voltage of the FET. Higher gain reduces the number of devices or drivers required in the amplifier chain while higher supply voltage reduces the current capacity requirement of the power supply. Other factors favoring FET technology are simpler for bias circuits, greater thermal stability of the device and lower noise factor than the corresponding Bipolar technology.

Closely related to the device technology is the method of packaging the device in terms of its contribution to device reliability and performance. Plastic packages are readily available and cheaper than hermetically sealed devices. However, plastic is more susceptible to moisture ingress especially if the temperatures are cycled from hot to cold. The moisture provides a medium for electrolytic corrosion at the interfaces of the conductor tracks and wire bonds or of the conductor tracks through any holes in the glassivation layer of the semiconductor. One method of hermetically sealing plastic devices has been the use of glass seals that enable the chip to be hermetically sealed and still realize the cost savings of plastic over ceramic packages. As discussed earlier, perhaps the largest contributor to reliable semiconductor operation is its ability to handle heat dissipation. As power level requirements for a single semiconductor device increase today, increases in package volume are also necessitated. This leads to difficulty in thermal management within the semiconductor package to prevent junction temperatures from reaching levels that would catastrophically affect the device's life. Good thermal conductivity within the package is essential to eliminate hot spots in the die which can cause catastrophic failure of the device.

Next to thermally related device failures, perhaps the second major cause of failure within a semiconductor is poor die bond wire attachment causing open circuit conditions to occur. These open circuit conditions are usually latent in their occurrence and precipitated by excesses in temperature or by mechanical vibration. Transmitter vendors who use parts supplied by semiconductor manufacturers having good military programs in house increases the likelihood of good quality control methods. Standards such as U.S. Mil Standard MIl-M-38510, the equivalent European CECC specification or British Standard 9400 outline good quality control practices. A second method of establishing high quality is to work with a supplier who insists on screening semiconductor devices prior to their installation in modules. The screening can eliminate many thermal or mechanical bonding problems before the fact.

Control and Monitoring

The requirements for a control system for a solid state transmitter should be less demanding than those for a tube type design. For one thing, there are fewer voltages to control. Secondly, on better transmitter designs there are inherently fewer operator adjustments. Modules are self-protecting in most designs requiring no external control circuits to protect them. The use of more redundant elements and the soft failure inherently possible from the use of many modules likewise make the control function simpler. Status monitoring and remote control are areas for potential improvement in future transmitters. Even these areas will become much simpler in solid state transmitter. Module failures will not tend to gradual degradations over time like tubes. The tendency will be for no degradation prior to failure. Metering on each module may yield little useful information and may add only to unneeded system complexity or worse, operator confusion. On the other hand, as discussed earlier, built in test diagnostics may provide welcome tools to a smaller population of technical service people faced with servicing a new (to them) technology.

The concept of software reliability in control and monitoring has limited meaning in that a good program will always run (and copies of that program will always run). On the other hand, a program with an error will always fail and so will copies of the program fail. Software, unlike hardware, possesses an attribute that it will not degrade over time. Additionally, software, unlike hardware, can not find improvement in reliability through redundancy if the software in the parallel path program is identical to the primary path. The principle means to increase reliability of software is through the development of a comprehensive specification of what its intended function is before the fact. It has been proven that more than half the errors recorded during software development originate in misunderstanding of the specification of requirements. A good specification should describe more than just programming requirements. It should describe the structure to be used, the program requirements, the program test requirements, documentation needed during design and development, as well as the clearly defined input and output (boundary conditions) down to voltage levels and timing.

Since most transmitters operate without built in delays the software must operate in real time. Real time systems are ones in which the structure and software operate as dictated by the speed of the system input and outputs. It is critical to include system and event timing interrelationships in the specifications of a real time system. Timing errors are a common cause of failure in real time systems and often difficult to detect, particularly by simple code inspection.

Another cause of software reliability is data error or corruption. Data corruption may occur due to transmission errors, memory cell failure, or self corruption due to spurious transmitter noise. Improved reliability may occur through use of error detection and correction schemes. Such schemes are readily covered in basic communication theory courses.

COOLING

If the solid state module constitutes the heart of the transmitter, then the cooling system is not unlike the circulatory system of the human being. A poorly functioning circulatory system can only lead to heart or in this case, module failure.

Several approaches to cooling exist. They are central, distributed, and hybrid cooling systems. Distributed cooling systems were once in vogue owing to the belief that many fans lessen the likelihood of a catastrophic failure should one or more fans fail. However, current motor/fan technology has matured to the point that actually a central fan can be more reliable than many "muffin" fans.

Cooling of the die is very critical in keeping the thermal stress within device safe operating limits as discussed above. The MTBF of a silicon device essentially doubles for every 10 degrees centigrade drop in the device junction temperature. There are many techniques employed to cool solid state devices today. Some transmitter vendors employ copper plates to spread the heat from the solid state package and then employ fins of either aluminum or copper to radiate the heat. Of course, the density of copper is approximately 3 times that of aluminum so an all copper heatsink would be very heavy compared to one constructed strictly of aluminum. For modules in the 600 watt to 1500 watt category, this means a package with a potential of 40 pounds or more.

Another area to look at is the interface of dissimilar metals in the heatsink design. Such an interface might occur between the solid state device flange, a copper spreader plate, and an aluminum heatsink. If the interfaces are not machined perfectly flat with good thermal contact and transfer characteristics, there can actually be a degradation in heat transfer owing to the thermal resistance of the interface. The key is to look for designs with minimum levels of interfaces between the silicon and the heatsink.

The method of heat removal is also key. Generally one can utilize air, liquid such as water glycol, or high technology heat pipes to remove heat from the module area. An air system is perhaps the cheapest and simplest when compared to water glycol or heat pipes. Air systems are not subject to leakage or freeze as is a water base system. Additionally, an air system is typically more efficient from a power requirement view point than a water based cooling system.

If one chooses an air cool system, the trade offs are with the cooling system design. These trade offs are in terms of cooling efficiency in watts per unit of power consumed to run a cooling fan as well as the noise levels associated with air pressures and volume requirements. Generally a noise level of 65 to 70 dBa is tolerable for a human required to work in the vicinity of a transmitter. Higher noise levels although perhaps more efficient in cooling, may necessitate locating the technical staff remotely from the transmitter. Noise generation in air plenums typically occurs when using sharp edges in the air system or small orifices and systems with high air pressures (typically greater than one inch of water).

Service of the cooling system is also important. The frequency of filter, orifice, duct or fin cleaning is important for proper maintenance of a solid state system. Questions that one should ask are: how is the cooling system affected if one removes a module for cleaning? Does a large amount of air spillage occur, and subsequently poorer cooling of the remaining modules occur? Will the system clog owing to the use of small orifices.

The semiconductor is not the only element of a transmitter than requires cooling. Other elements of the system that may require cooling are reject loads, circulators, power supplies, control systems and displays. There should be consideration given by the transmitter vendor to the heat removal from these elements. Recall that the reliability of a transmitter can be compromised as a result of an undetected failure (such as in a reject load) the failure while undetected may not cause any degradation under normal operating conditions but could cause catastrophic failure in the transmitter when the failed element is called upon to protect the transmitter.

POWER SUPPLIES

Good power supply design is also a key to the long life of a solid state transmitter. Power supply MTBF is dramatically increased if good thermal design management is used just as silicon life is extended when good thermal design management is used.

Power supply complexity is also a factor in higher reliability systems. Switching power supplies are compact and efficient. However, reliability is generally a factor of 2 or 3 less in a switching power supply than in a non-switching type design such as linears, ferroresonent or SCR controlled power supplies. Line transients and surges also need to be considered in a power supply design. In areas of high lightening or poor power line conditions, ferroresonent supplies offer better inherent transient protection than linears or switchers. Add on zener diodes and transorb protection is essential in linear and switching type supplies. Likewise, good system grounding practices and equalization of potentials with arresters is also important. In transmitters providing video, supply voltage regulation tolerance is more important than other types of service such as FM or AM. Therefore, FM transmitter power supply designs are generally simpler and more rugged than UHF or VHF video power designs.

Another consideration in the purchase of a solid state transmitter is how the system power is distributed. Consider now that the problem is not of distributing high voltage, low current, but just the opposite of distributing low voltage (5 to 300 volts DC) and high current (upwards of 100 to 300 amperes depending on the volt ampere rating required of the power supply). This shift of voltage and current brings with it a whole new set of problems related to safety. Consider the effect of dropping a screwdriver across a 24 volt 200 ampere line to ground. Additionally, the effect of loose wires, physical size of wires needed to distribute efficiently low voltage high current, is also something for consideration. Last but not least the physical size of components such as filters to handle the large currents is also a consideration in a solid state low voltage high current power supply.

CABINET DESIGN

Good engineering practice for higher reliability and long life considers cabinet design. This means that cabinets are not too large to fit into an existing transmitter control room nor too small to service. All subassemblies should have ease of access for routine preventative maintenance inspections and/or service. Lift out, removable doors allow for installation near walls without the clearance needed for swinging doors. However, removable doors must not compromise operator safety.

Good cabinet designs are modular not only in the amplifier section but also in other aspects. The choice of a single or dual exciter system should pose no problem if the customer decides to add that function in the field at a later date. Likewise, if the customer decides to purchase one power level now with the potential to upgrade later in the field to higher power as a result of a rule change or decision to change the antenna system in the future, the design should easily accommodate this. Removal of major system elements such as power supplies, combiners, cooling fan motors, exciters, driver modules and control systems for upgrade, repair or maintenance must be easily accommodated by the system cabinet design.

SYSTEM COMBINER CONSIDERATIONS

The last area of discussion is the system power combiner. Because we are adding the power developed by many modules the combiner system plays an important role in terms of its efficiency and continuity of operation in the event of a fault in either the combiner or the solid state module. The function of the combiner is to combine lower power levels with proper phasing into higher levels of power. Additionally, combiners served to provide isolation between modules in the event of a failure.

Combiners can be designed using N-way combiner technology that utilizes impedance transformers and ferrites, hybrid couplers, Wilkinson couplers or combinations of hybrid couplers and N-way combiner impedance matching systems. Combiner cost, ease of repair, and heat dissipation are areas to pay attention to when reviewing a transmitter design. In the event of a failure can a combiner be easily and inexpensively repaired in the field. How is the heat load handled should a module fail or be removed for routine cleaning maintenance or inspection? What loss in output power occurs in the event that one or more modules are removed from the system? From the efficiency viewpoint, what are the combiner loses experienced at the point of highest system power (the output)? Remember that a fraction of a dB more here can translate to the requirement of more drivers or modules in the RF chain.

SUMMARY

This paper has attempted to give the broadcasters a wide view of elements to ponder when considering the purchase of a solid state transmitter. It is hoped that this paper has provided some basis for further thought stimulation and provided some insight into differences between a tube and a solid state transmitter design.

For the transmitter manufacturer it is hoped that the topics discussed in this paper provides stimulus and impetus to further improve the product delivered to the industry.

SECOND GENERATION ANALOG COMPONENT VTR'S—A USER'S PERSPECTIVE

Karl Renwanz
WNEV-TV
Boston, Massachusetts

At WNEV-TV, Boston, our component analog video (CAV) experience began nearly six years ago with the purchase of M-Format equipment in the summer of 1982. It soon became obvious that to obtain the best multi-generation video quality, it was necessary to interconnect the VTR's in component form. These holes in the CAV highway included no component switcher, timebase corrector, matrix converter, test signal generators, waveform monitors, and video monitors.

Working with manufacturers and in parallel with other broadcast organizations, it has been a great challenge to convince manufacturers to fill these holes with products. In 1982, many companies were not eager to enter the new hardware design fray for the sake of a few maverick broadcasters and production companies. The video CAV switcher was the key to opening up system design including multi-machine edit bays and component graphics systems. After numerous attempts in 1982 to talk manufacturers into developing a three-channel switching system with effects (many manufacturers said there was not going to be a market), we were successful. This type of persistence earns one the title of "pioneer". It is only later you learn that pioneers are easy to recognize--they are the ones with arrows in their backs.

After five and one-half years of component experience with M-Format, we were faced with the decision of which hardware to choose to replace aging U-Matic and M-Format equipment. The evaluation process lasted six months and based on a combination of performance, features and reliability, Betacam SP was chosen.

The rapidly changing technological advances in today's broadcast community make users shy of a format change when we are told that digital in the field may be only five years off. While digital component and composite machines are here in early form at this convention, we are far away from having lightweight, portable Digital VTR's in the field. As a user of ENG/EFP component formats, we project five or six years of useful life because of wear and tear as well as technological advances.

Specification Testing

Because our evaluations were based on prototype and pre-production hardware, we felt the need for careful check-out of the final production run equipment when delivered.

We received our first shipment of BVW 75s, BVW 5s, and BVW 35s in September, 1987. One of the most important steps in implementing new equipment, and especially a new format, is to thoroughly test each piece of hardware to ensure all meet specifications. These tests, if performed properly, can take many hours for each machine. At WNEV-TV, our proof-of-performance tests averaged five hours for each machine.

Because the VTR's process and record the video in component form and the resulting distortions are different from those encountered in NTSC, typical NTSC measurement techniques need to be modified and/or eliminated. In some cases new tests and signals must be created. Test gear is different as well. A number of new component generators are available that allow programmable flexibility. Here is the list of test equipment we typically use in our measurement procedure: Sound Technology 1510A, Magni Systems 2015-33 and 1515B, Shibasoku 925 D/1 video noise meter, and Shibasoku SV-11A video sweep generator, Tektronix 520A, 1480, 1570, and 1410, and Leitch DTG-1000.

As these measurements are being made, they should be logged into a data base according to model and serial number. This allows you to 1) prove initial acceptance, 2) have a "new" condition specification reference for which to compare as each machine ages, 3) get your maintenance staff firmly established in the total documentation concept. Don't be too surprised when some machines are rejected for not meeting specifications.

Edit Suite Configurations

At WNEV-TV we utilize 20 edit suites, most of which are two-machine suites used for news editing. The two-machine edit suite utilizes a Sony BSBX 100A as a changeover switch for full control of our archival U-Matic/M-Format machine. The BSBX 100A allows you to switch inputs into the composite input of the record machine. A number of three-machine suites are used for news series productions as well as our total Programming and Public Affairs Departments.

All two-machine edit suites are operated in the component or compressed time division multiplex (CTDM) dub mode whenever possible. The CTDM mode gives you the highest video quality but requires a 5 second pre-roll because it is not timebase corrected. The CTDM mode does not allow you to use the DMC (slo motion) edit feature. The second highest quality choice--the component output--requires only a 4 second pre-roll because it is timebase corrected. Unfortunately, when using the DMC edit feature, you have to use the component mode and the operators can forget to switch back to CTDM. We wish the mode switcher was automatic, transparent to the operator. In the fast-paced and often hurried environment of news editing, a short pre-roll is very desirable.

The Changeover

Training, one of the largest hurdles, must next be faced. Getting a few pieces of hardware in advance can allow an early start for operators. For ENG use, a Beta playback VTR was first added to each two-machine U-Matic editing room. This allowed us to accommodate a mix of U-Matic and Beta SP recording in the field while still editing to U-Matic masters. Several weeks later when the field crews were completely Beta SP and the editors fully trained, the remaining U-Matic VTR's were replaced with Beta SP. This changeover took place in one night. We have left one machine from our previous format in each suite for archival input. In one year, we expect to keep only three edit rooms capable of U-Matic playback.

Utilizing all of the features that are available in the BVW 75s is not as easy as meets the eye. Four channels of audio are something to rave about but we are still struggling with how to use them to their fullest. We have found the longitudinal audio track automatic gain control (AGC) to be quite good, much improved over previous small formats. The AGC is not susceptible to "pumping" or "ducking" with which we have become familiar. This frees photographers from the need for constant level adjustments. The dynamic range and headroom in the two FM tracks can offer some creative flexibility. However, if you find yourself cueing occasionally to audio, the FM tracks are not audible in shuttle due to their positioning in the video tracks. The first time this occurs, it is a bit of a surprise to the operator and generally results in a late re-cue.

As soon as we finish upgrading the audio facilities in our edit rooms to handle four channels of audio, we intend to use the FM audio tracks to record stereo natural sound in the field and combine them with reporter and voice over sound recorded on the longitudinal track. The phasing between the longitudinal tracks is held to within a few degrees on an interchange basis, as is the phase between the FM tracks. However, the phase relationship between the longitudinal and FM tracks is unpredictable and can be substantial enough to cause partial cancellation. Recording similar sound on these tracks and subsequently combining them in the editing process must be avoided.

Since the FM tracks cannot be inserted separately from the video, some editing situations which are currently done in mono with the two longitudinal tracks cannot be handled in stereo without forcing an extra editing generation. While the extra editing time might be available in our EFP operation, it is not possible in a competitive news operation.

We believe "stereo ENG" can only be successful if it imposes no operational restraints or extra time penalties in either field recording or editing.

Moving from a control track to a time code based editing system allows the reporter to come back to the studio or bureau and expedite the viewing and editing process. This has overcome years of bias by many photographers and editors that time code slows down the edit process.

There are certain time code issues however that must be addressed. Field tapes have Vertical Interval Time Code (VITC) and longitudinal recorded time code on them. Operators must remember to start both time code generators with the same numbers. When feeding field tapes via microwave or satellite, new VITC and longitudinal time code must be recorded or the BVW 75 will become confused during the edit process because it is constantly switching from VITC to longitudinal depending on tape speed.

SP Performance Measurement

Typical performance measurements for our Beta SP machines are as follows:
Video Measurements:

Lum	s/n	52db
Chrom	s/n am	56db
	pm	57db
Y/C Delay		<10ns
K-factor		<2%

Freq. Response

Y	10Khz to 4.2Mhz	+0 -.5db
	to 4.5Mhz	-3db
R-Y, B-Y	10Khz to 1.5Mhz	+0 -1.0db

Audio Measurements:

	Longitudinal	FM
Freq. Resp.	<3db @ 18Khz	<3db @ 24Khz
S/N (Dyn. Range)	64db	87db
Dist. @ 0 vu	.1%	.04%
Xtalk @ 1Khz	<65db	<70db
Flutter	.05%	not measurable
Phase Dif. @ 15Khz	<5°	<5°

Other Considerations

As in any new format, we have encountered some problems. The ninety minute cassette required a software and hardware change to keep the tape in perfect position within the tape path.

One operational consideration is the increased time a longer length large cassette takes to thread up, versus that of a small cassette, compared to a BVU 800. The slowdown is due to slow thread-up time during the first half of the threading rotation. This may be critical in areas you rapidly cut up tapes for on-air playback.

The 5X shuttle speed certainly gives a recognizable picture but the next shuttle speed of 24X does not lend good quality viewing. An interim shuttle speed would be a welcome addition and has been proposed by us. When the BVW 75 is controlled by a BVU 800, there is a 10X shuttle speed available. The 35X shuttle speed offers little value during the edit process due to a non-viewable picture.

If your station rotates courtroom coverage with fellow broadcasters, as occurs in many cities, using the Betacam SP one-piece package or two-piece configuration will leave you with the inability to dub directly from U-Matic to an SP field recorder.

Your typical bulk tape eraser will be of no value with metal particle (MP) tape. In fact, if you attempt to bulk erase an MP tape on your typical bulk eraser, it will look as though you haven't even plugged it in. There are new bulk erasers available that claim to be capable of erasing MP tape, but we rely on the full-track erasure in the machines themselves and have had no problems to date.

Metal Particle Tape

One of the more controversial areas regarding new half-inch formats is metal particle (MP) tape. Betacam SP utilizes fifteen micron MP tape that is key to the format success. The question of overall shelf life of the MP product is not field tested but projected to be good. Initial comparisons of the cost of 3/4" videotape to MP tape could convince you that the extra quality is not worth the expense.

A deeper analysis, which includes the number of recycles each field tape can withstand, may swing the bottom line value toward the quality of MP tape. At WNEV-TV we have gone from four passes for each field tape to more than ten with Beta SP. Additionally, the Beta SP timebase correction is superior to that of U-Matic and M-Formats, therefore allowing better dropout concealment and contributing to more passes. Betacam SP machines also replace many of the program-length recording positions typically handled by 1" and MP tape costs less than 1" tape.

The balance of all these comparisons means very similar costs in the end.

In designing second generation component analog VTR's, it is clear that the manufacturers have addressed most of the shortcomings of first generation hardware. The new machines have program-length capability, full on-board time code facilities, dynamic motion control (slo motion), and much improved NTSC encoding and decoding. In fact, they appear to deliver most of the features of the 1" VTR at a lower cost in a cassette-based format, but they still suffer from subtle multi-generational distortions which prevents them from replacing the 1" VTR in certain high-performance applications.

TELETEXT: A UNIQUE APPLICATION FOR ELECTION NIGHT RESULTS

J. Talmage Ball
KSL Television
Salt Lake City, Utah

BRIEF HISTORY:

When Johannes Gutenberg, and his associates were perfecting their type mold, there were probably those who said it would never amount to much of a printing press. In 1895, Guglielmo Marconi sent his first radio waves through the air in Italy. Engineers began experimenting with crystals and vacuum tubes in the early 1900's and the newspaper people probably didn't pay much attention to them either. Radio Station KDKA in Pittsburgh began their first transmissions in 1916. It is interesting, for the purpose of this report, that the KDKA broadcast of the 1920 presidential ELECTION RESULTS on November 2, 1920, is generally considered the beginning of professional broadcasting.

For the most part, in this country, television grew out of radio engineering. In 1936, RCA installed television receivers in 150 homes in the New York City area. An experimental station then broadcast its first television program, a cartoon of "Felix the Cat" to these receivers. At KSL Radio some of the engineers purchased equipment and were becoming excited about sending pictures through the air while management shook its head and said that it would never provide the broadcast promises (profit) to replace or even compete with radio. Now we are in the 1980's and there are those who would dare to suggest that TELETEXT doesn't have a chance to become profitable with all of the other modern forms of communication. We don't agree with that idea in Salt Lake City, and we are determined to make TELETEXT a major innovation in modern communications.

KSL has been pioneering TELETEXT since 1977. When our past president of Bonneville International Corporation, brought the idea back from England, we studied his findings and programmed a General Automation computer to build the TELETEXT pages on the British system format.

On June 15, 1978, KSL received permission from the FCC to broadcast the service on vertical interval lines - 15 and 16. TELETEXT was updated each day to include sports information, news headlines, weather data, etc. That sounds pretty comprehensive for a beginning but the fact is, that just one engineer programmed the keyboard each day during this meager beginning. Sometimes it would be updated and sometimes it would not. We also commissioned a series of Tandy Model III computers connected to phone modems to serve computer enthusiasts who would phone in and draw out the TELETEXT data. This system is still in operation today and still receives about 1500 calls per day for updates on the stock market, weather, sports scores, headlines, etc.

In 1982, CBS announced Extravision. This provided a new challenge for us because the British and Extravision systems were totally incompatible. We had to decide between broadcasting our own local system or completely abandoning it and changing over to the CBS system. The decision was made, when we moved into our new building in the Triad Center, to also make the switch to CBS Extravision. In order to continue our local TELETEXT service KSL purchased it's own TELETEXT encoding system and changed our local service to the Extravision Format. The NABTS TELETEXT software is comprised of several magazines, starting with magazine "0". We asserted ourselves and recirculated CBS Extravision from the first magazine "0" over to Magazine "1" and opened a hundred pages of our own on Magazine 0. The TELETEXT office grew from the one part-time engineer to a staff of four. A manager and three programmers. The TELETEXT office has been connected to the Channel 5 "News Specialist" Newstar Computer and soon news information will be automated right into the magazine.

With all of this there were still only six TELETEXT decoders in the Salt Lake Valley able to decode the information. These decoders were set up in shopping malls and high traffic areas to demonstrate the new service. In the Summer of 1986, Samsung Electronics of Korea shipped several hundred new decoders to Salt Lake City and shortly sold out of the first shipment. The interest was really beginning to build. Sometimes it is necessary to shut TELETEXT down and take it off of the air temporarily for minor adjustments and updates. The telephones are not exactly jammed with complaints as yet, but that day will also come. The challenge now was to find sponsors to return our investment.

PRELUDE TO ELECTION 1987

Three days prior to the November, 1987 elections, a discussion took place concerning the possible use of TELETEXT in our local election coverage. We wanted to get the results on the air before our competitors. The best way to do this was with our TELETEXT system, but our present local TELETEXT system required nearly 20 minutes of leafing through graphics design menus to get a single line of information changed on the air. KSL had purchased the local encoder and software from Norpak Corporation, in Canada. We called Norpak to see if there was a way to get election results on TELETEXT and on-the-air in a timely manner. It was as though they were anticipating our next move with TELETEXT. Their engineers were in the process of developing a new card that could take data from any computer over a phone modem and with the proper programming place the

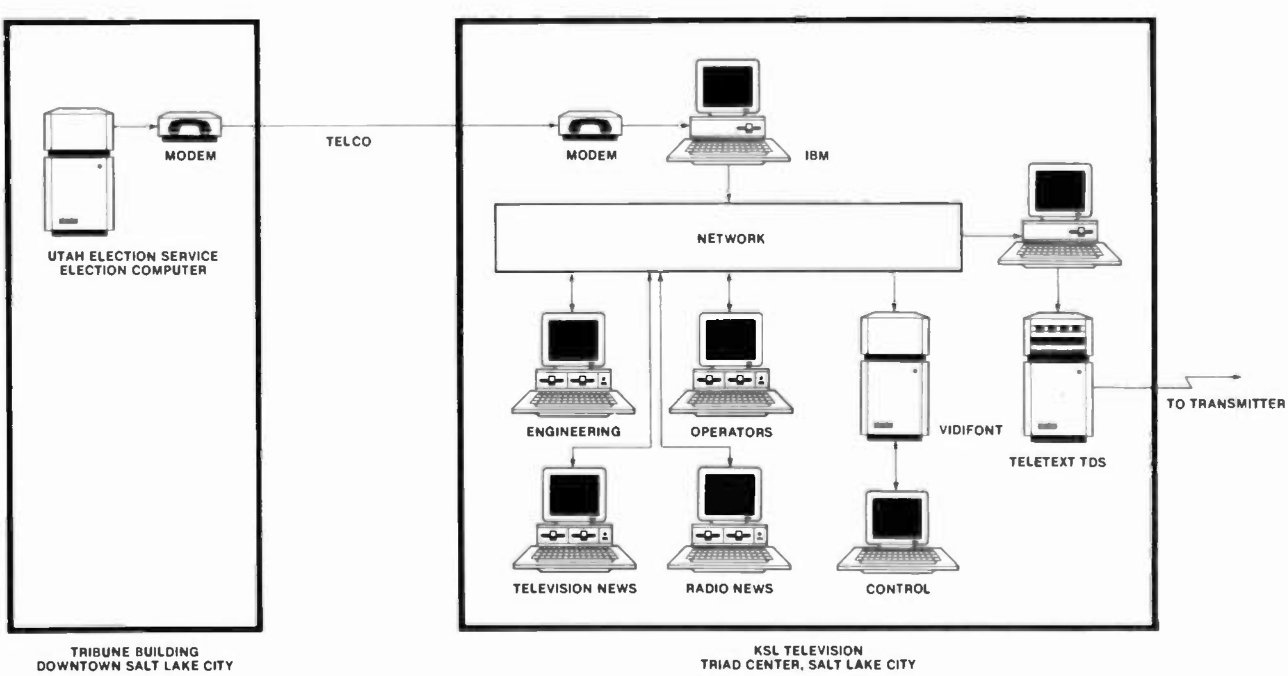

Fig. 1

information on the air. We pressed to have them send us their R&D prototype card. They felt that this would be a good test for the new card and shipped it on the plane that same day. Norpak also FAXED us the coding instructions, protocol for the card, and some examples of things they had programmed it to do. We received the card the Friday before election night and immediately began programming our computers to interface with it.

MORE BACKGROUND:

Two years ago the TV News Department requested that we help them speed election results from the State's downtown "Utah Election Service" (UES) computer to our own Telemation Character Generator (now replaced with the VIDIFONT, VIDIVOTE system). For years election results had been coming in over the telephone from someone stationed at UES. With hundreds of races, it was just too slow to receive the information this way and type it into the Character Generator. Owen Smoot, our Engineering Supervisor, developed a program using Borland's Turbo Pascal, (very popular here at KSL-TV Engineering) to receive direct raw election data from UES, place it in a buffer and check for any updates. The program then interpolates the updates and loads them into the station Novell Network for station personnel to see the progress of the election. We wrote an additional Pascal program that provides the VIDIFONT character generator access to the Network so that it is continuously being updated for "on air" broadcasting of election returns via the VIDIVOTE.

Engineer, Mark Fenton, developed a display program in the station's own Novell computer network system. Election results were buffered, interpolated, and fed into this system. This provides Television News, Radio News, and telephone operating personnel comprehensive information on every race in the election. (See Fig. 1) Telephone operators watch the terminals and answer information to anxious callers concerning the results. 50,000 watt AM KSL Radio's personalities use their display terminal on the same network to give half hour "over-the-air" updates on the races.

BACK TO TELETEXT ON ELECTION NIGHT:

With just three days to go, before election night, we programmed an IBM to take the organized election data out of the Novell network, interpolate it for the new TELETEXT display system prototype card, and put it out on-the-air.

Now Let's Get Technical About It:

Our Pascal program instructs the card in the TELETEXT Display System (TDS) to create and color the pages for receiving election data information. It recognizes a series of numbers coming from the control computer and responds with the construction of new pages. In our application each page contains 16 lines of text with a possible number of 32 characters on a line.

We selected, for the election, the RED, WHITE, and BLUE colors. Numbers #71,#74 are sent to the card and repeated three times to provided the top three lines with Red background and white letters. The fourth line is encoded with #64, #71 giving the page a white stripe with black lettering. The following twelve lines are encoded blue with white lettering by the characters #71,#72 repeated twelve times. After the background colors are ordered and the page lines are set up this procedure writes "TELETEXT 5 ELECTION '87'" in the top red section for a heading. It prepares the white stripe to receive "Precincts Reporting" information with this heading and the 12 line blue field is left for election data. (See Fig. 2)

The card was programmed and connected to the Station Network system. It was fun to see the data come up on the TELETEXT system and out on the air almost as fast as it was available in other areas on the Network. This made it possible for TELETEXT to bring results to viewers seconds after it arrived into the computer. In fact, at 8:22PM, the first precinct reported in and we had results on these races on the air through TELETEXT at 8:23PM. TELETEXT WAS THE FIRST MEDIA SERVICE OF ANY KIND IN SALT LAKE CITY TO PROVIDE THESE FIRST ELECTION RESULTS TO THE PUBLIC.

THE NOV 87 TELETEXT SERVICE PROVIDED:

All 224 races were programmed on pages in TELETEXT Magazine 2. This first experiment contained seven total pages. The 224 races were divided up and multiple races would share a single page. Someone interested in a specific race was able to look up their race in the index and locate it on one of the seven pages. Each race would rotate around and

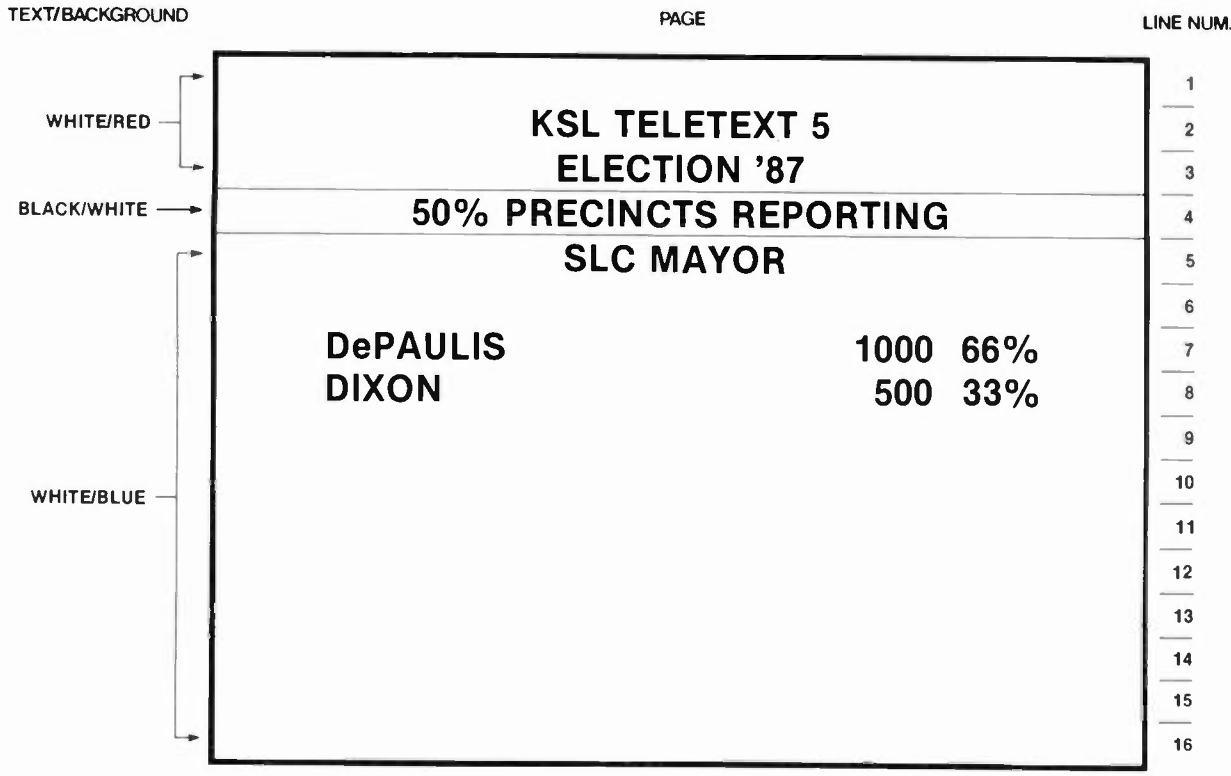

Fig. 2

sequentially appear on the same page every few minutes. The races would always be automatically updated with "up-to-the-minute" action in the race. Data available and displayed on each race included:

1. Percent of precincts reporting
2. Candidate's names
3. Vote totals
4. Percent of vote for each candidate

The widespread KSL translator network (121 total) made "up to the minute" results available to every household in Utah, parts of Idaho and Wyoming.

PRELUDE TO ELECTION 1988

THE FUN BEGINS:

Several weeks after the 1987 November election a recent corporate meeting was held wherein election races using TELETEXT were demonstrated. Our TV News Director and Radio's News Director were asked to comment on the future of TELETEXT in the 1988 elections. Both responded with: "It's a real competitor for attention on election night and could draw numbers from our audiences." However, an important point to consider is that if their audiences are watching KSL TELETEXT, they would still have background TV audio and in our case, radio audio on the SAP channel. Viewers could tune in during our election coverage news and interviews at 6:00PM and 10:00PM. IF SALT LAKE CITY VIEWERS ARE WATCHING OUR TELETEXT THEY ARE NOT WATCHING THE COMPETITION!

This coming year we will have more than just three days to program the system and get the election data on TELETEXT.

In November 1988 important election results, such as the Presidential race, Governor's race, and key Senate and Legislative races will each be programmed onto a page of their own. This will provide viewers, Candidates, and

Republican and Democratic Headquarters the fun of watching the numbers roll as the votes come in. There may be as many as 20 or 30 pages set up with one race per page. Other races may still share pages as they did in the 1987 election.

KSL Television and Radio have long been the standard source for information on election night in Salt Lake City. The introduction of TELETEXT has broadened this base to give viewers the ability to study a particular race minute by minute all evening long. TELETEXT is the only service today that can come close to providing this kind of service for our viewers.

ANOTHER PROFITABLE USE FOR TELETEXT!

Our discovery of this systems' capabilities while working on the election program gave rise to another service that we could provide with TELETEXT. We have now connected the same prototype card (Norpak didn't get it back after the election) directly to our Bonneville Telecommunications Service (BTS) financial computer. We are now providing our TELETEXT viewers with 500 individual Stock Market listings, Commodities, Currency, Precious Metals, and Money Market reports that follow Wall Street only 15 minutes behind "real" time. The 15 minute delay is written into our contract to protect those who are paying for the BTS service. This service went into effect in December 1987 and is now fully operational. Our own engineering department wrote and produced a video tape explaining this new Stock Market service with other services on teletext. The sales department began using this tape in sales presentations and in just two weeks they sold a $5,000 dollar sponsorship on TELETEXT. The client is now sponsoring the Stock Market pages.

SUMMARY

KSL has been broadcasting TELETEXT to a limited number of decoders in the Salt Lake Valley for nearly 10 years. Even though the profit outlook has been slow in coming, KSL management has continued to support the research and development to someday see TELETEXT become a major force in the communications industry. The elections held in November 1987, provided a perfect opportunity to advance the idea that TELETEXT can provide valuable services to the public. It was just the boost it needed to broaden its interest and establish its unique value in the broadcast industry. The technology we learned in the November Election also helped us to develop the "automated" stock market report which TELETEXT is currently providing.

DESIGNING BROADCAST FACILITIES FOR COMPOSITE AND COMPONENT DIGITAL VIDEO TECHNOLOGIES

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ABSTRACT

This paper will outline system considerations when interfacing 4:2:2 Component and Composite Digital based technologies into existing analog, hybrid or all digital teleproduction facilities. A cursory review of relevant protocols and formats pertaining to interfacing hardware to conform with the given standards will be discussed. Also described are technical considerations and solutions when integrating digital component and composite based products into existing or future facilities. The remaining part of the paper will discuss application considerations when implementing digital component and composite technologies into analog and digital environments. This may imply a divergence from operating methodologies of today as well as peripheral support equipment functionality.

1. THE DIGITAL COMPONENT AND COMPOSITE SOLUTION

A universal tape format that meets the needs of the broadcast and teleproduction industries would be ideal. Such a device would contain the following attributes:

1. High quality video and audio performance
2. Transparent Multigeneration capability
3. 2 to 4 hour record and play time
4. Low media cost with environment protected tape housing
5. Built in editing and mixing functions
6. Low cost, power consumption, and light weight
7. Compatibility with present and future formats
8. High level of reliability and serviceability

At present, due to different standards, economics and requirements for each application in the professional, industrial and consumer marketplaces, many tape formats proliferate. This is due to the fact that each application requires a different set of attributes from the recording device. In looking at tape machines for post production versus electronic news gathering, this can be clearly seen. Endusers ultimately will have to decide on whether to use a single format device for all applications and accept the compromises in quality and performance or match each application to a format that was designed for that particular need, and create a cost effective interformat environment. At least at present, a single universal tape format that meets all requirements worldwide seems impractical.

After many years of research, two digital video formats have emerged. The D-1 Component Digital format and the proposed D-2 Composite Digital format. The D-1 format was created by the cumulative efforts of many committees to meet the requirements of CCIR Recommendation 601 where performance and functionality were of concern. The proposed D-2 format is a result of user group requirements based upon functionality and economics of a VTR that is compatible with existing analog standards. The marriage of digital video and audio technologies brings about a new revelation in the recording, manipulation, reproduction and transmission of audio and video signals. Digital processing technologies promises the elimination of multi-generation signal degradations of analog VTRs and an increase in reliability, performance and built-in intelligence for diagnostics.

2. REVIEW OF RELEVANT FORMATS AND PROTOCOLS

The standardization of a digital format is essential for progressive efforts towards the realization of the digital studio. The standardized D-1 format and the proposed D-2 format will allow a facility to use their current technology base and also meet the future requirements of high picture and sound quality reproduction. The proposed Composite Digital format is compatible with the existing composite facilities and equipment worldwide. This allows facilities to gain the benefits of digital recording without obsoleting existing equipment or having to purchase signal handling and routing equipment to support the format. The 4:2:2 Component Digital format is a worldwide format which is independent of the coding schemes of NTSC, PAL and SECAM composite transmission systems. The format is unique in two respects: First, it allows for the design of the mechanisms and signal processing systems to be used worldwide for any digital television signals that conform to CCIR Recommendation 601. Second, the designer of a D-1 tape transport mechanism can use one of several different combinations of tape scanner diameters and data head arrangements. Both formats permit different design choices for different applications. However, implementation of the hardware must be in accordance with the guidelines set forth by the respective formats and protocols to allow compatibility between systems. The following is a short review of the relevant format parameters and protocols pertaining to the interfacing of hardware to the D-1 and D-2 formats.

TABLE 1

D-1 AND D-2 SAMPLING PARAMETERS

VIDEO	D-1 FORMAT	D-2 FORMAT
SAMPLING RATE	13.5 + 6.75 + 6.75 = 27MHz	4Fsc = 14.31818MHz
RESOLUTION	8 BITS	8 BITS
NUMBER OF TV LINES/FILED	250 (525/60) / 300 (625/50)	255
AUDIO		
SAMPLING RATE	48kHz	48kHz
RESOLUTION	16-20 BITS	16-20 BITS
ANCILLARY DATE	DEFINED BY AES SERIAL INTERFACE	

TABLE 2

COMPARISON BETWEEN TYPE D-1 AND D-2 FORMATS

APPLICATION	D-1 FORMAT		D-2 FORMAT
	LAYERING MULTI-EFFECTS EDITING MASTERING/REPLICATION FILM TO TAPE DATA STORAGE/ARCHIVE		PRESENT TYPE "C" APPLICATIONS
VIDEO I/O	RGB/Y, R-Y/B-Y/BETACAM/ RP-125/EBU TECH-3246-E		COMPOSITE ANALOG RP-125X
AUDIO I/O	4 CHANNEL (AES/EBU)		4 CHANNEL (AES/EBU)
CASSETTE	19mm CASSETTE - S,M,L		19mm CASSETTE - S,M,L
PLAYTIME	16um	13um (TAPE)	13um TAPE
SMALL	11 min.	13 min.	32 min.
MEDIUM	34 min.	41 min.	94 min.
LARCE	76 min.	94 min.	208 min.
TAPE COATING	850 Oe Metal Oxide		1500 Oe Metal Particle
TAPE SPEED	286 mm/sec.		131.7 mm/sec.

2a. CCIR Recommendation 601

This recommendation specifies the basic parameter values for the 4:2:2 Component Digital standard. The format specifies that the Y, R-Y and B-Y signal components are to be formed separately and encoded using the internationally agreed upon sampling rates based upon a 4:2:2 ratio. Hence, the Y channel is sampled at 13.5MHz and each of the color difference components (R-Y, B-Y) are sampled at 6.75MHz.

2b. Proposed Recommendation 601X

This recommendation specifies the sampling rate and precision for the digital encoding of composite video signals. It specifies the relationship between the sampling phase and the color subcarrier as well as encoded levels of peak white, blanking and sync tip.

2c. SMPTE RP-125

This practice describes a bit parallel unidirectional digital interface for component video signals, meeting the requirements of CCIR Recommendation 601. The interface is applicable for 525/60 and 625/50 Systems M digital television equipment.

The video signal is transmitted in a parallel arrangement using eight conductor pairs. Each pair carries a multiplexed stream of bits of each of the Y/R-Y/B-Y signals. A ninth conductor pair carries a clock signal at 27MHz. The signals on the interface are transmitted using balanced conductor pairs for a distance of up to 50m without equalization and up to 300m with appropriate equalization.

2d. SMPTE RP-125X

This proposed practice describes a bit-parallel digital interface for composite video signals, meeting the requirements of Proposed Recommendation 601X, Encoding Parameters of Composite Digital Television for Studios.

The bits of the digital words that describe the video signal are transmitted in a parallel arrangement using ten conductor pairs. An eleventh conductor pair carries a clock signal at 4Fsc.

2e. EBU TECH 3246-E

This specification for a bit parallel unidirectional, nine pair interface is functionally equivalent to the SMPTE RP-125 document. However, the specification is for systems operating in the 625/50 environment and conforming to CCIR Recommendation 601. The only addition is the proposal to allocate two lines explicitly for the transmission of auxiliary signals. The data signals are time-multiplexed and transferred as an NRZ code. The signals consist of video data, timing reference, ancillary and identification signals.

2f. ANSI S4.40 - 1985

This document describes a serial digital interface for the transmission of digital audio signals between digital audio systems. The interface is designed for the transmission of one, two or four channels of digital audio over either a pair of wires, or an optical fiber, in a serial format. In addition to digital audio channels, the interface also permits the transmission of information related to the channels. These are user-definable data, information on the interface itself, error protection and additional digital audio channels.

2g. EBU TECH 3247-E

This interface scheme allows for the serial transmission of video data between systems through either coaxial cable or optical fiber. The encoding scheme is based upon an 8B-9B bit mapped block encoding technique with a transmission rate of 243 Mb/s.

3. <u>TECHNICAL CONSIDERATIONS</u>

The era of the all digital facility has begun, but is many years away from its full implementation. As such, changing from the present technology to a full digital environment is both a time consuming and expensive proposition. Therefore, a means by which a facility can use their current technology base and also meet the requirements of high picture and sound quality should be investigated.

Before looking at some technical considerations for implementing the two formats, let us first look at an overview of their respective system topologies. Figure 1 shows a typical D-1 processing device outline. The D-1 device can accept various inputs ranging from composite analog to analog component to a digital video parallel input. Analog signals are converted to a digital format before processing and vice versa for output. Utilizing 4:2:2 based processing equipment, limitless multi-effects layering can be accomplished by bypassing the analog chain and output via the digital I/O. Most DVTRs will have a high quality decoder to convert an analog signal to digital and an encoder to do the opposite. Figure 2 is a simplified block depiction of the proposed composite format DVTR. The main difference between the two formats is that the D-2 format doesn't need a seperate encoder/decoder combination for the conversion of signals and the 4Fsc processing block for the encoding and decoding of data. Notice also that the parallel digital video I/O is different as explained earlier. In order to properly integrate these two formats into existing plants, several considerations must be taken into account despite the increase in reliability and transparency.

As with analog technology, various parameters must be monitored, tested and adjusted to conform to given constraints. This holds true in digital technology as well. With the implementation of a hybrid or all digital studio, these guidelines should be adhered to, thus minimizing any timing, amplitude or phase discrepancies between the integration of various format systems.

3a. Timing and Reference Signals

When implementing digital based hardware, the same timing and reference considerations should be taken into account as with analog. At present, discussions are underway regarding reference signals for the evolving marketplace. As can be seen in Table 3 and 4, existing reference signals such as mixed sync and black burst are widely accepted. However, with the onset of digital based systems, digital signals with inherent timing references need to be taken into account when designing a system for the near future. System designers must consider the timing relationships between input reference for overall system timing and its relationship to digital interface clocks for the transferring of data.

Implementation of digital video hardware to existing analog facilities have already begun. Designers are paying careful attention to timing relationships between the analog, hybrid or all digital environment. Within the equipment, timing adjustments for critical processing blocks that relate to the outside world are important. However, the use of digital technology applied to circuit parameters pertaining to timing, level, differentail gain and differential phase have minimized or deleted these routine adjustments. Concerns as to where and how the timing should take place and the relationship to the reference or input signal must be addressed. The adjustments will either adjust the analog or digital parameters independently or together. If the adjustments are global in nature, care must be taken in planning both the analog and digital paths independently. This is because the timing for the digital interconnects between sources may not be the same for their analog counterpart.

Audio timing relationships plays an equally important role in the facility. With the introduction of four digital audio channels and the availability of digital audio support equipment, timing and phase relationships need to be monitored closely. Care in the distribution and handling of signals as well as conformance to specific sampling rates will become a concern. Pitch changes relates to variances in the sampling rate and the need for sampling rate and pitch converters will also be needed. Present digital video recorder technology allows some variance in the timing of the audio edit point. This technology may allow an edit point resolution to be moved in 6.6ms intervals for further accuracy. Additionally, due to extra read and write heads, sound on sound editing is possible to coincide with the video data. Last, the audio chain and its associated timecode reference must relate to the reference sync. This implies that throughout the production, care must be taken to allow for synchronization of all processing systems.

3b. Level and Phase Matching/Color Correction

Table 5 outlines concerns regarding level and phase when integrating composite and component systems. In interformat systems, conversion of signals between composite and component based systems may result in mismatched levels and phase disrepancies. This is valid if each recording or processing device is not adjusted properly. Care must be taken to monitor each signal before and after encoding or decoding. If the signal is to be digitally encoded, levels and timing relationships must be matched and corrected prior to digitizing. If it becomes necessary to color correct a source signal after digital encoding, then there are two alternatives. One is to convert back to analog, do the correction and convert back to digital. The other alternative will be addressed by the introduction of a digital color corrector, thus avoiding generation loss through the A/D and D/A chain. This would be useful in film to tape applications as well as multi-effects layering and compositing.

3c. Monitoring/Test/Adjustment

The widespread acceptance of digital video recorders into the marketplace will take many years. As such, most of the monitoring and testing of signals will be in the analog domain. With the proliferation of 1/2" component formats and the introduction of the 4:2:2 component digital VTR, test equipment manufacturers are now offering composite and component based test equipment. Table 6 outlines monitoring, test and adjustment considerations for both analog and digital formats. To consolidate functionality, many test devices have dual format functions. Additionally, test generators can now output test signals via interfaces conforming to various digital protocols such as RP-125. In addition, digital test devices will be available to monitor and test the DVTR. This may encompass an error rate checker and monitor to measure a DVTR's performance as well as other devices to check processing blocks. Other devices will check the performance outputs of digital serial feeds as well as monitoring timing, phase, and level relationships of signals. An important benefit of this new technology is the equal advancement in the use of internal diagnostics. This technology allows for internal diagnostics and adjustment of processing integrity, error rate, servo ballistics, and head record/playback attributes. Connection to an external computer via a interface bus can minimize downtime and serve as an in-circuit emulator. With expanding knowledge in digital technology, adjustments and hardware failure will be minimized due to the inherent stability and advantages of digital processing techniques and VLSI chip integration.

3d. Analog/Digital Interfaces

Table 7 outlines analog and digital interfaces needed to integrate both formats effectively. Depending upon the application, the signal will be converted and distributed accordingly. Many manufacturers are now offering digital video I/Os to minimize degradation losses. With the availability of serializers and deserializers, extended coax or fiber optic signal runs are now possible. Advancements in design topologies have also reduced the adjustment time for distribution black boxes such that auto phasing or synchronization capabilities are an integral part of the device. In addition, new advancements in VLSI will reduce the distribution black box to a single chip set for integration into existing equipment. This may result in enhanced capabilities in routing switchers, matrices and switchers, and peripheral support devices.

3e. Video System Interfaces

Figure 3 categorizes the interface applications in a teleproduction environment. The solid arrows represent connection alternatives between protocols. Due to the expense in parallel digital component hardware, this category is limited to production where cable runs of 50m or less is viable. With the increased use of optical fiber technology for the transmission of signals, channel capacity will increase with a marked reduction in distribution costs.

4. APPLICATIONS OF COMPONENT AND COMPOSITE DIGITAL TECHNOLOGIES

As discussed in Table 2, each format has distinct advantages in application. The D-1 format will be ideally suited to high end production whereas the proposed D-2 format will find application in existing composite based facilities. Figures 4 and 5 show the Component DVTR in a typical application. Projects requiring transfer from film to DVTR tape from a 4:2:2 based telecine and input to an editing system will achieve a high degree of transparency. Using the D-1 DVTR in conjunction with an Ultimatte for compositing can give very impressive results. Using present 525/625 line graphic and video effects systems, projects requiring multilayering and rendering can use the DVTR as an ideal transparent storage medium. This is also true with animation computers conforming to the 4:2:2 standard. In this instance, the DVTR can be either controlled by the computer through the RS-422 interface or via a vtr animation controller which allows for housekeeping and control of still frame recording. Figure 6 shows the D-1 DVTR in a multi-effects layering system utilizing a composite based editing system. A high quality decoder and encoder is used for converting RGB to composite and back. Upon completion of the project, the recorder DVTR can output component analog directly to 1/2" format or via conversion from an encoder to Type C for subsequent on-air broadcast. In considering an all digital component environment, let us first look at Figure 7 which depicts a typical analog component editing system. Two major concerns that plaque most edit environments are related to timing and levels. Referring to Figure 7, timing problems can be reduced by the insertion of a source sync generator. The main TSG supplies reference signals to the complement of equipment. A source synchronizing generator which locks to the master TSG and drives the video source, allows easy timing adjustments for the source with respect to the system. Normally, the device with the longest path length is taken as the reference device and delays are then introduced in other devices through coax or lumped delay lines. In addition, other attributable delays are introduced from coax length, D.A.s, and the equipment itself. Automatic delay D.A.s or isophasing amplifiers can correct some of the timing problems. However, the best solution is to plan carefully, accounting for all timing problems in the initial design stage.

In a component island, level matching between the three video components is critical. A variety of component based test equipment is emerging to allow us to monitor both levels and timing relationships between systems. As a result, a new generation of test signals are evolving to meet the accuracy needed to test the new 1/2" format vtrs.

The evolution of 4:2:2 based technology will bring about many benefits in the production and post production arenas. In considering the "Digital Production Suite" as depicted in Figure 8, many benefits will result.

In using a component based digital system, the recording of components permits multi-generation processing without any degradation of image quality. The heart of the system is a digital switcher/mixer. The digital interface between sub-systems can be either the standardized parallel interface conforming to RP-125 or the serial interface specified by the EBU TECH-3247-E. With the availability of digital serializers, interconnection between systems will become easily attainable. The monitoring of signals would continue to be made in the analog domain; in component form. Within the near future, due to VLSI design, it may be possible to implement digital parallel or serial inputs/outputs to existing high quality monitors. The digital color corrector will also play an important role in the post environment. A color corrector may be required for a playback DVTR and preferably one should be assigned to the switcher so that independent assignment to any source is available. The color corrector should correct for black and white levels as well as gamma and operated in RGB or its digital equivalent. Suitable digital D.A.s and an assignment matrix would be required. It is also conceivable that the architecture of the switcher and the use of the DVTRs will change in the digital environment, due to the ability of the system to make recursive effects/ edits without picture impairment. Acquisition of material from the composite domain should be kept to a minimum and, when necessary, a high quality decoder used as depicted in Figure 8. If a direct analog camera source is brought into the area, an RGB to RP-125 converter box would be necessary. However, given the advances in technology, it's conceivable that this may be built into the switcher or other devices. Another alternative is that the camera can be fed into the RGB input of the DVTR initially since the DVTR's video bandwidth will not limit the picture quality. Paint box and DVE devices can also be fed via an PR-125 interface. It's apparent that more equipment in the high end production arena are now using digital component based systems to create a high quality end product. The end goal is to bypass the analog interfaces for the I/O and only use digital interfaces. The real benefits of multi-layered effects, clean chroma keys, and minimal signal degradation using equipment such as computer graphics, paint systems, digital slide stores, and effects processors can then be realized.

Another area of concern is the use of the four available digital audio channels in the D-1 format. With the progression of MTS broadcasting and the proliferation of stereo based software being made available, production companies are paying more attention to the field of audio. Presently, it is envisioned that the audio sweetening area will continue to be a separate function, which has the responsibility of sweetening and conforming the audio tracks to the final end product. However, with digital audio technology available today for production, it is equally important to consider the significance of this medium in how it carries a product. The following are some considerations in applying digital audio technology to the hybrid or all digital facility. In the production stages of the shoot, if digital audio is to be used, it should be noted that the 48kHz sampling rate is recommended for recording and subsequent data transfers. In both cases, whether analog or digital equipment is used, the systems should be referenced to master sync. During predubs or playback, the systems should be locked via a audio/video synchronizer that is referenced to house sync. The worldwide acceptance of the AES/EBU serial data bus will play an integral part in the digital facility. The standard will allow for the transfer of digital audio data between systems without any signal degradation. The DVTR should have provisions for advance digital audio data to be available with adjustable delays. This is needed if the audio is to be sent to a digital audio mixer for sweetening or that the audio is to be distributed over long runs and channeled through extra processing.

The proposed Composite D-2 format is compatible with the existing composite facilities worldwide. This allows broadcasters and production houses to gain the benefits of digital recording without obsoleting existing equipment or having to purchase additional support devices as explained earlier. The Composite DVTR will be lower in price than its Component DVTR cousin. This allows the D-2 DVTR to be implemented in applications as depicted in Figures 9-12. The same considerations in the component system for sync, phase and level should be adhered to. However, one will notice the absence of any transcoding devices. Signal distribution can be either in analog or digital. With analog distribution and processing, signal degradations will also occur. Should Composite Digital based switchers and effects processors become available, multi-layered productions similar to the D-1 applications will be a reality. Coupled with the introduction of digital serializers and deserializers, D-2 based editing environments will become commonplace.

5. FUTURE POSSIBILITIES UTILIZING THE D-1 AND D-2 FORMATS

The digital D-1 and D-2 VTR, due to its ability to store and reproduce transparent images, will become a cornerstone in the foundation of a new era in broadcasting and production. With the flexibility of digital signal processing implementation, editing islands of the future may instill more creative freedom. This may imply a divergence of operating methodology from today as well as peripheral support equipment. Also, new standards may affect the user-friendliness of equipment as well as their communication both internally and to the outside world. The improvement in quality of the end product due to digital implementation is impressive. Digital production techniques also saves steps in the production chain as compared to an analog equivalent. The following are examples of what may change when using digital production techniques.

- Various operations can be performed without loss of quality: multilayering, processing, and recording.

- Stage or on-location shooting will be simplified since it will be easy to modify the image to correct for lighting and colorimetry. This may imply fewer cameras and increased use of robotic camera controllers.

- Downstream chroma keying off recorded material from a DVTR will be commonplace.

- Effects generation with no loss of quality during compositing.

- Automatic phasing of source and processing signals.

- 2-D and 3-D image synthesis systems with direct digital I/Os.

- Composition of complex images with various sources in which frame and perspective transitions can be controlled with minimal effort.

- Possibilities for greater artistic freedom in all stages of production.

In addition, picture resolution and quality will increase. This will be complemented by increased intelligence for internal and external diagnostics. The future digital studio may change both the design concept of the studio as well as its intended use. Since each DVTR can transparently store and reproduce images in the digital domain, their applications will become more diverse. Digital switchers will include digital chroma keying, color correction and picture manipulation systems as an integral of the unit. New forms of graphics and animation outboard equipment will enhance the post production environment. In addition, editing systems will become more user friendly with expanded memory capacity for the storage of programmed effects. It is feasible that many of these systems will become more integrated into the post environment or become a separate fully self-contained production turnkey system.

6. CONCLUSIONS

The agreement and standardization of a format allows manufacturers to competitively design and manufacture a new breed of equipment technology. The now standardized D-1 format and the proposed D-2 format will find many applications in the marketplace. However, interfacing considerations must always be taken into account. This paper has outlined some of the technical considerations in interfacing these two technologies into existing analog hybrid or all digital facilities. In addition, comments were given to the perceived problems in implementing the formats as well as application examples given to show the advantages of utilizing these two new formats.

The Component Digital format and the proposed Composite Digital format launches a new era in video evolution. The marriage of digital video and audio technologies brings about a new revelation in the recording, manipulation, reproduction and transmission of audio and video signals. As a result, the broadcasters, production houses, manufacturers, and endusers can now consider the implications and possibilities of the all digital production facility.

7. REFERENCES

1. "Integrating Digital Component Systems Into The Analog And Hybrid Broadcast Plant", Curtis Chan, Ian Collis, NAB Conference, 1987

FIG 1

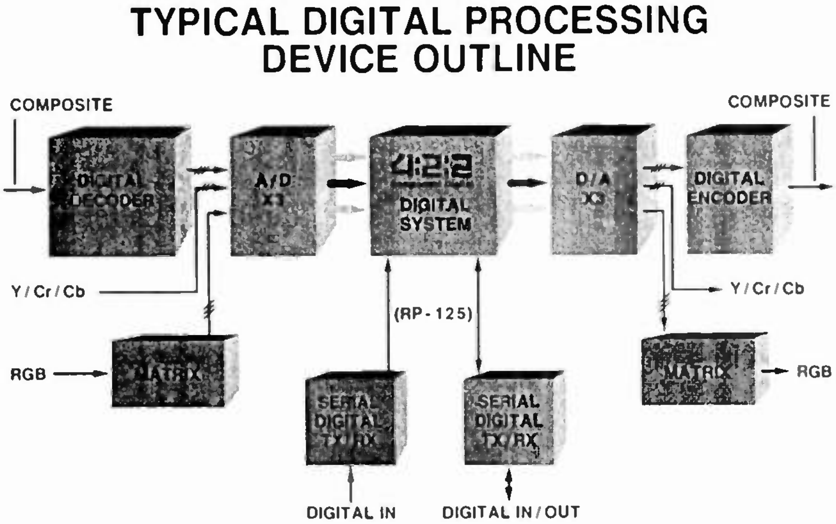


FIG 2

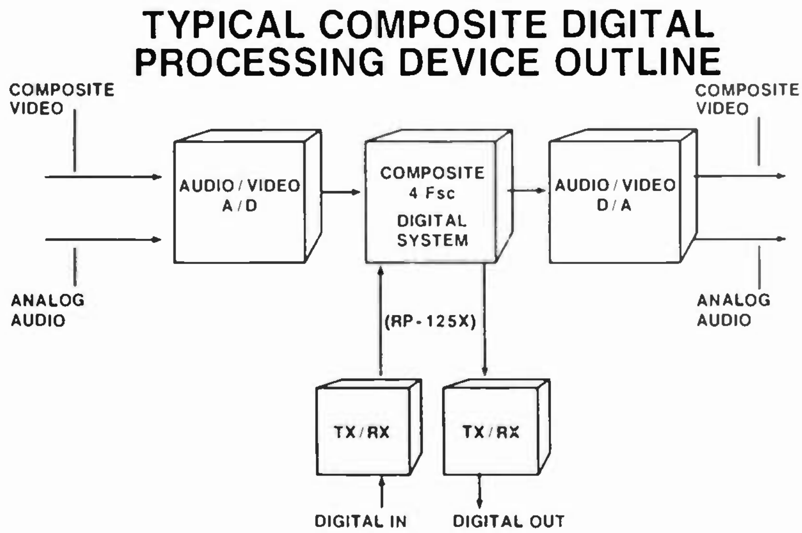
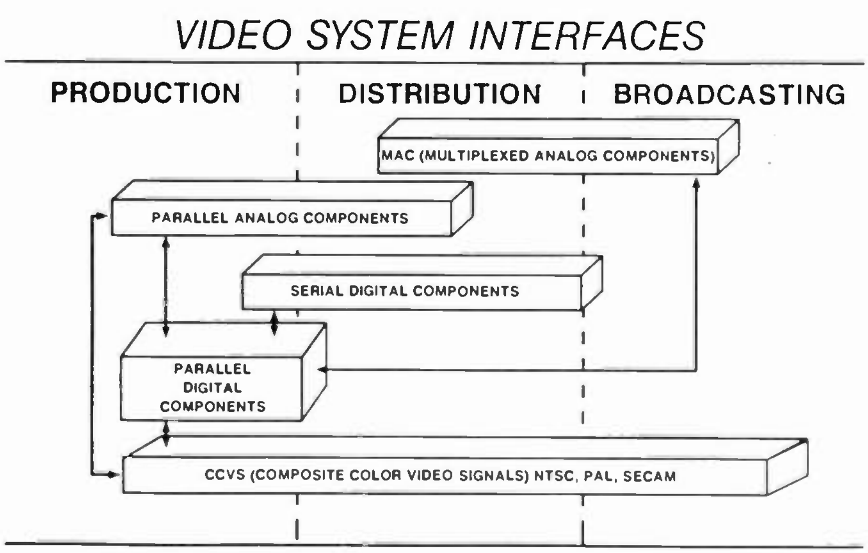


FIG 3



DVTR IN FILM HOUSES

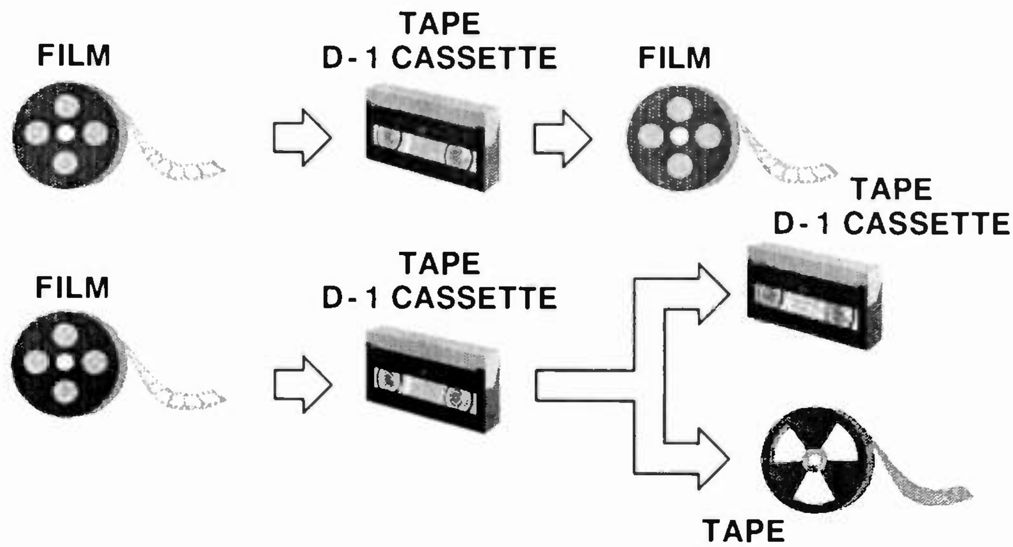


FIG 4

DVTR IN ANIMATION & PRODUCTION EFFECTS EDITING / ANIMATION STORAGE

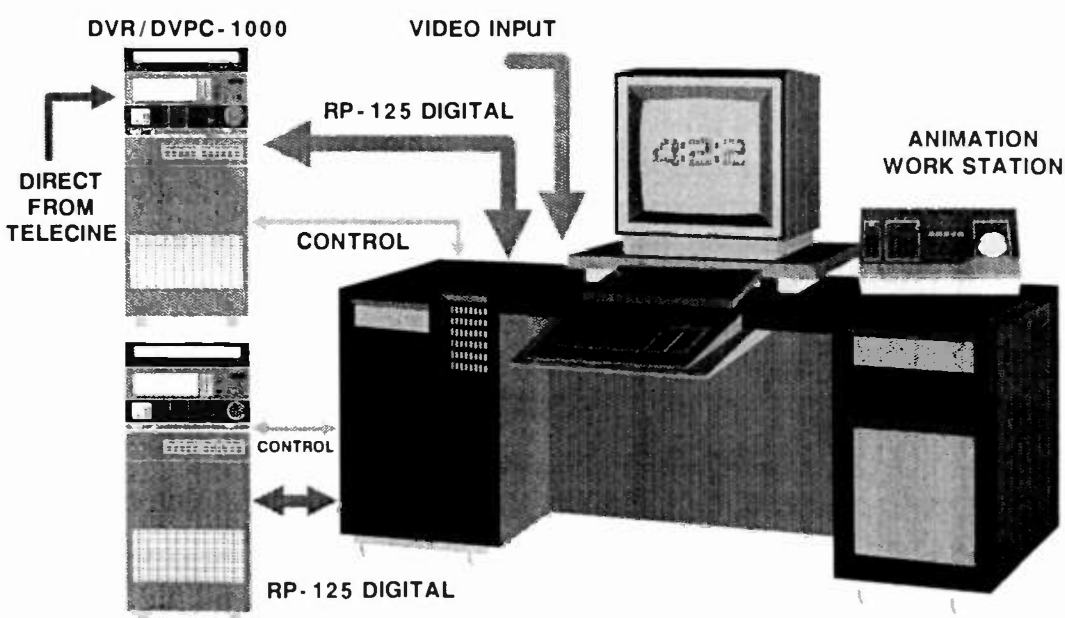


FIG 5

DVTR IN VIDEO PRODUCTION LAYERING / MASTERING

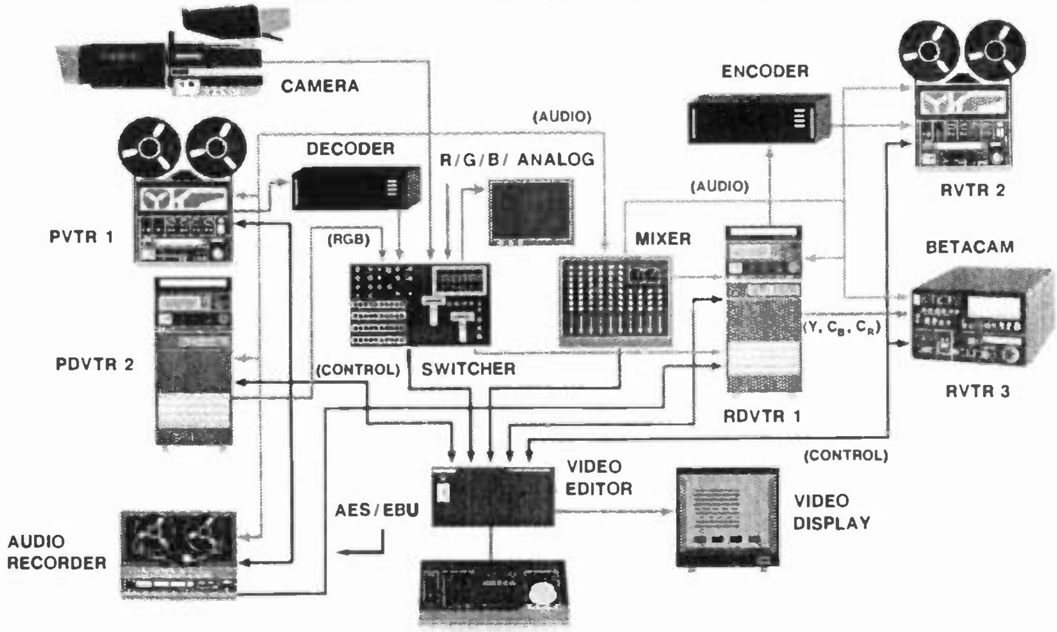


FIG 6

ANALOG COMPONENT EDITING

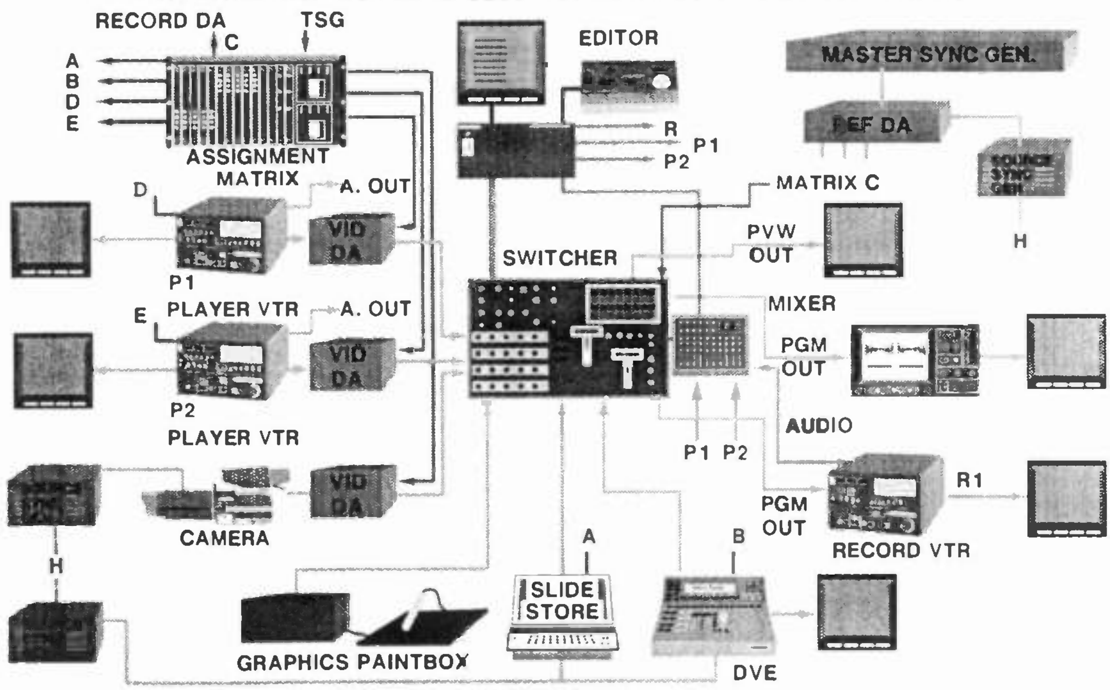


FIG 7

COMPONENT DIGITAL EDITING SYSTEM

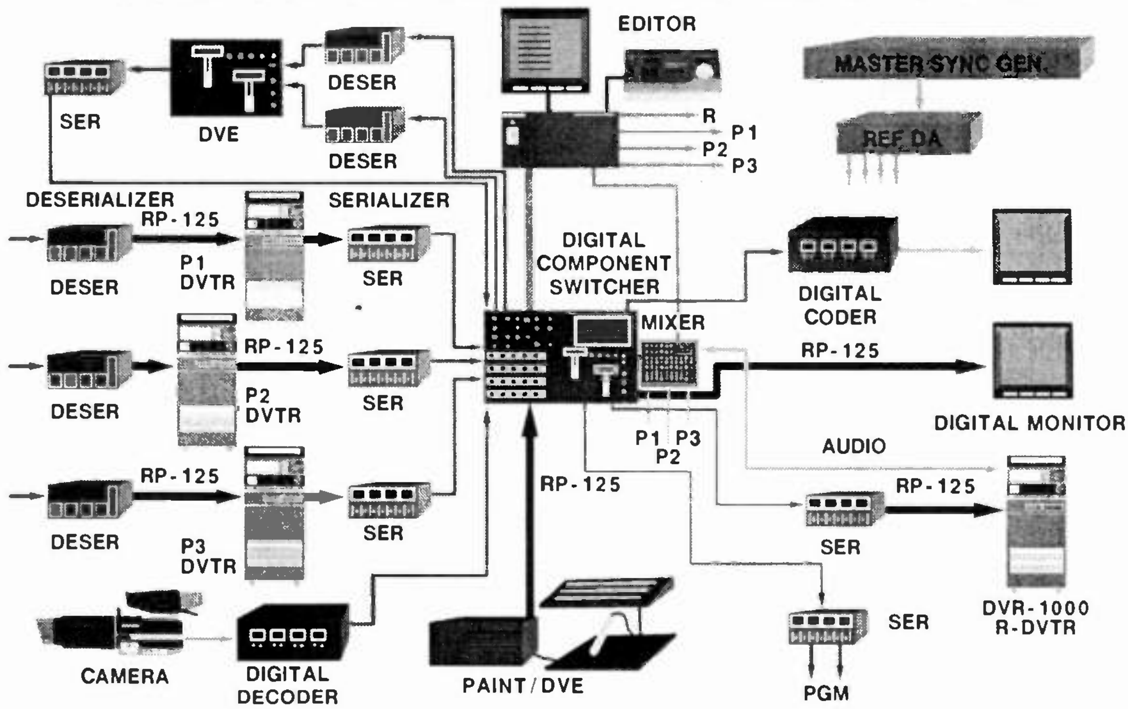


FIG 8

COMPOSITE DIGITAL/ANALOG EDITING

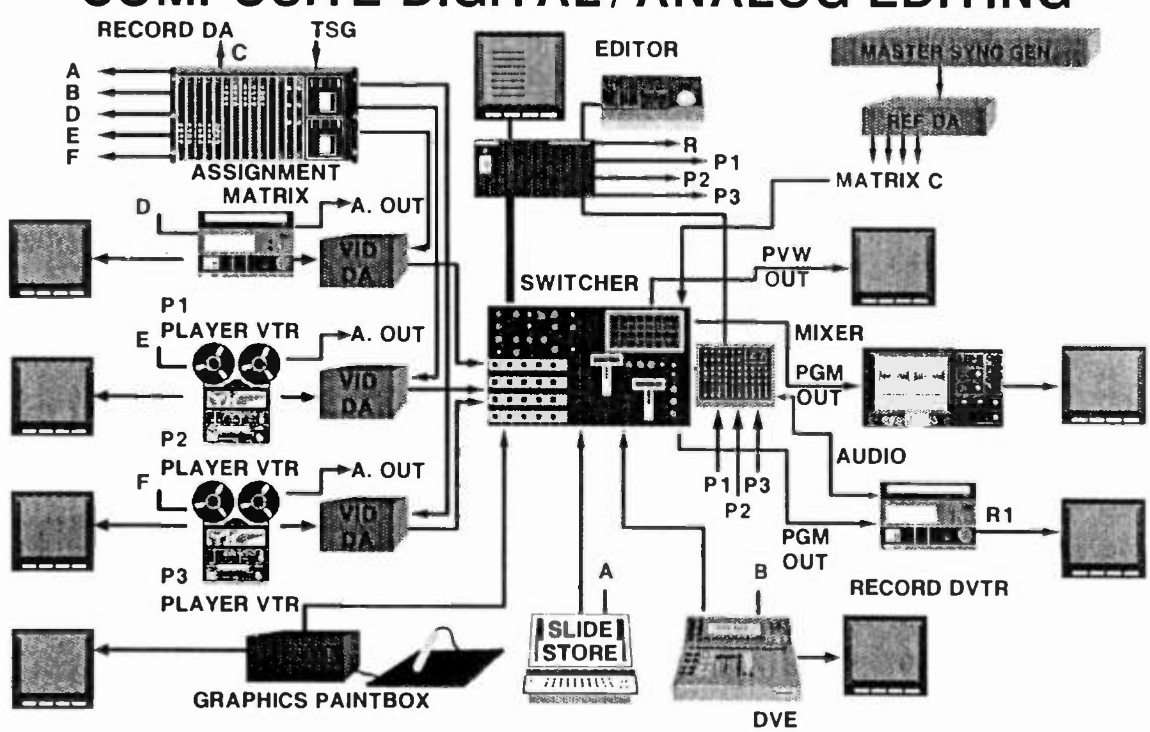


FIG 9

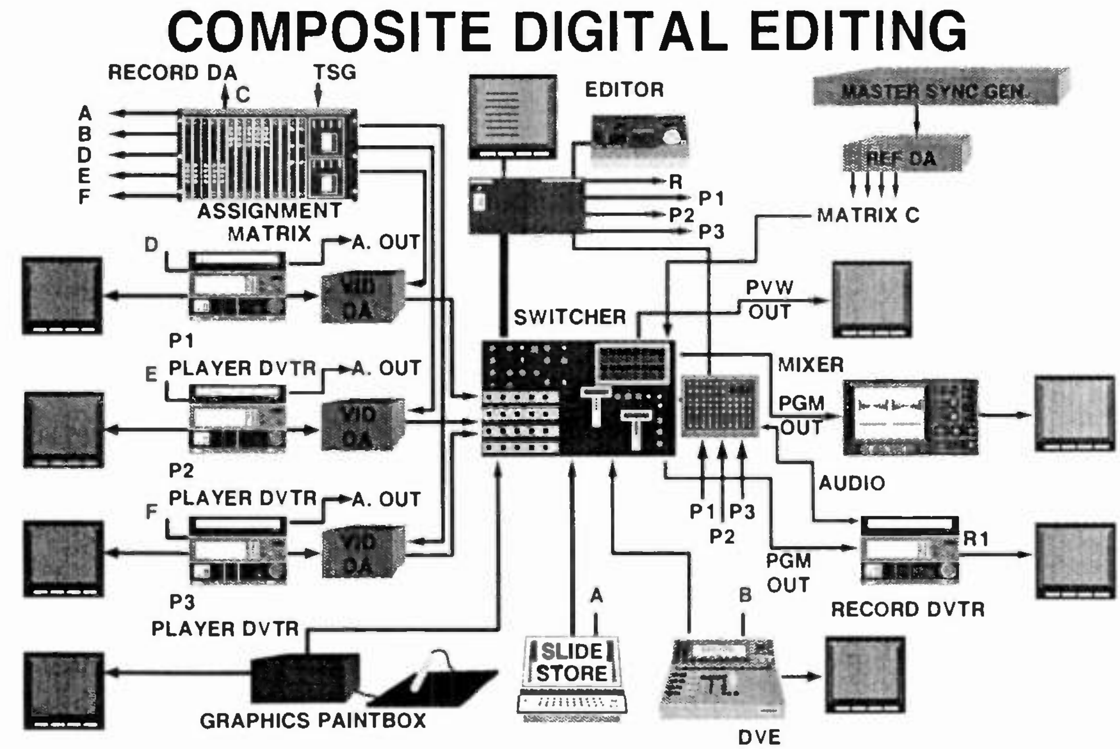

COMPOSITE DIGITAL EDITING
RECORD DA
TSG
C
A
B
D
E
F
ASSIGNMENT
MATRIX
EDITOR
MASTER SYNC GEN.
REF DA
R
P1
P2
P3
MATRIX C
D
A. OUT
VID
DA
SWITCHER
PVW
OUT
P1
PLAYER DVTR
E
A. OUT
MIXER
PGM
OUT
P2
PLAYER DVTR
F
A. OUT
AUDIO
P1
P3
P2
R1
PGM
OUT
P3
PLAYER DVTR
A
B
RECORD DVTR
SLIDE
STORE
GRAPHICS PAINTBOX
DVE

FIG 10

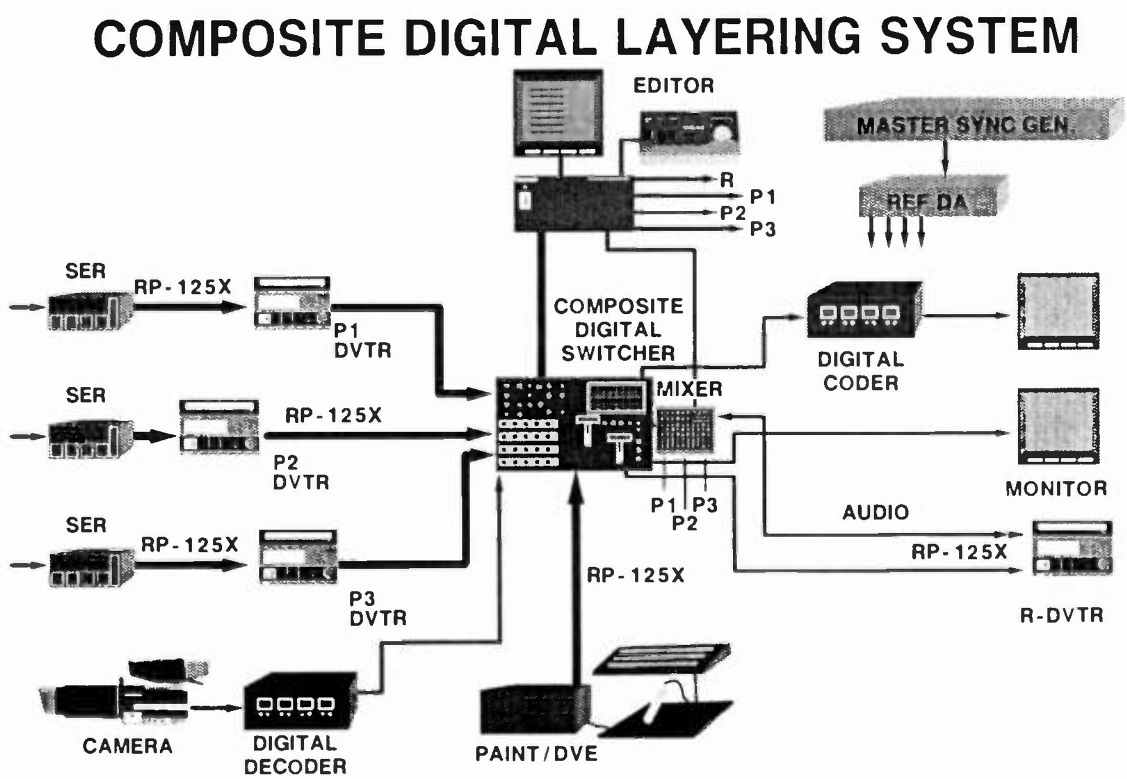

COMPOSITE DIGITAL LAYERING SYSTEM
EDITOR
MASTER SYNC GEN.
REF DA
R
P1
P2
P3
SER
RP-125X
P1
DVTR
COMPOSITE
DIGITAL
SWITCHER
MIXER
DIGITAL
CODER
SER
RP-125X
P2
DVTR
MONITOR
P1
P3
P2
AUDIO
SER
RP-125X
RP-125X
P3
DVTR
RP-125X
R-DVTR
CAMERA
DIGITAL
DECODER
PAINT/DVE

FIG 11

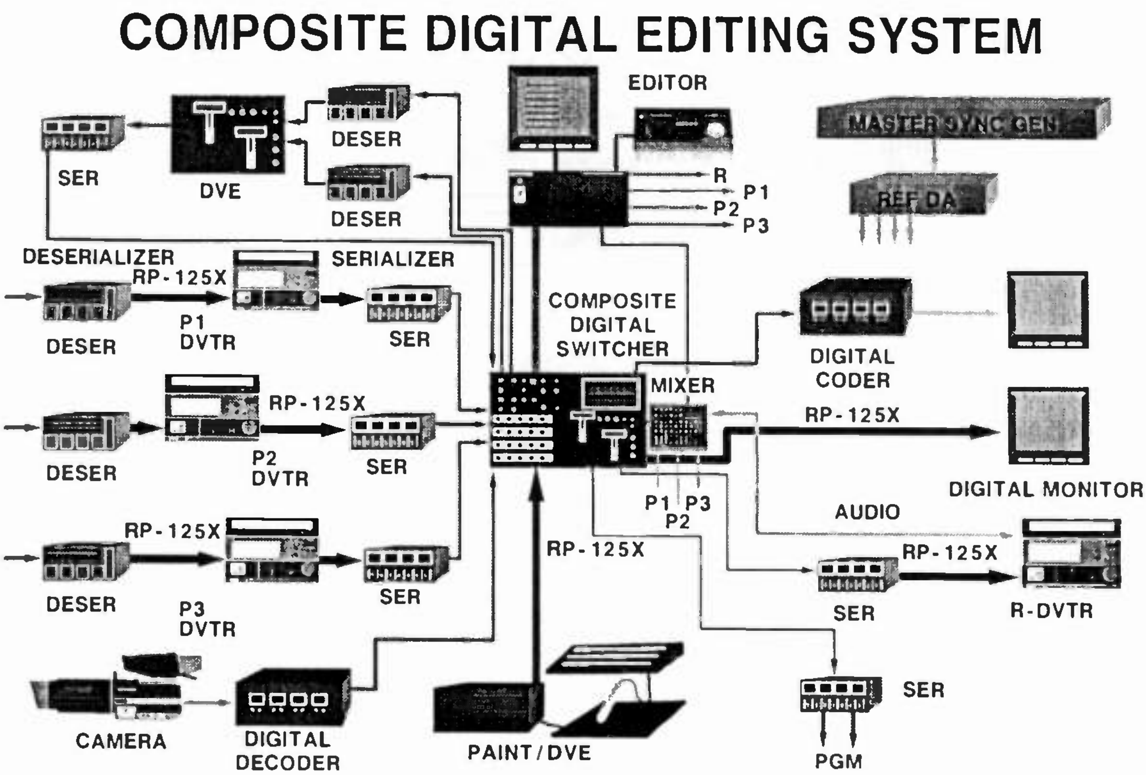


FIG 12

TECHNICAL CONSIDERATIONS

TABLE 3

REFERENCE SIGNALS
- MIXED SYNC
- BLACK BURST
- SMPTE RP-125/RP-125X

INTERFACES
- PARALLEL INTERFACE
- SMPTE RP-125/RP-125X
- EBU TECH 3246-E

SERIAL INTERFACE
- EBU TECH 3247-E

TECHNICAL CONSIDERATIONS

TABLE 4

TIMING CONSIDERATIONS

RELATIONSHIP BETWEEN:
- ANALOG
- HYBRID
- DIGITAL

AUDIO TIMING CONSIDERATIONS

RELATIONSHIP BETWEEN:
- SAMPLING RATE
- ANALOG/DIGITAL TIMING
- ADVANCE & DELAYED DIGITAL I/O
- SYNCHRONIZATION TO VIDEO

TABLE 5

TECHNICAL CONSIDERATIONS

LEVEL MATCHING

- MONITOR COMPOSITE LEVEL
- MONITOR DISCRETE LEVELS
- MATCH LEVELS BEFORE PROCESSING

PHASE MATCHING

- MONITOR COMPOSITE/COMPONENT PHASE
- MATCH PHASE RELATIONSHIPS BEFORE PROCESSING

COLOR CORRECTION

- ANALOG CORRECTION - D/A → CORRECTION → A/D
- DIGITAL COLOR CORRECTION - MINIMIZE SIGNAL DEGRADATION

TABLE 6

TECHNICAL CONSIDERATIONS

MONITORING/TEST/ADJUSTMENT

MONITORING:	ANALOG (COMPOSITE/COMPONENT) DIGITAL (COMPOSITE/COMPONENT)
TEST:	COMPOSITE/COMPONENT GENERATOR (ANALOG/DIGITAL) COMPOSITE/COMPONENT ANALYSIS (WAVEFORM/VECTOR)
HARDWARE:	INTERNAL/EXTERNAL DIAGNOSTIC TOOLS ERROR RATE/PROCESSING CHECKER LOGIC TIMING/STATE/SIGNATURE ANALYZER
SOFTWARE:	IN-CIRCUIT EMULATOR STATE/SIGNATURE ANALYZER SERVO SOFTWARE CHECKER RS-422 LINE TESTER
MECHANICAL:	ADJUSTMENT JIGS (TRANSPORT) DRUM/SCANNER ASSEMBLY JIGS

TABLE 7

ANALOG/DIGITAL INTERFACES

ANALOG TO ANALOG INTERFACES
- DECODER - COMPOSITE TO COMPONENT
- ENCODER - COMPONENT TO COMPOSITE

ANALOG TO DIGITAL INTERFACES
- RGB TO RP-125 D.A.
- COMPONENT TO RP-125 D.A.
- RGB TO SERIAL D.A.
- COMPOSITE TO RP-125X

DIGITAL TO ANALOG INTERFACES
- RP-125 TO RGB D.A.
- RP-125 TO COMPONENT D.A.
- SERIAL TO RGB D.A.
- RP-125X TO COMPOSITE

DIGITAL TO DIGITAL INTERFACES
- RP-125 TO SERIAL
- SERIAL TO RP-125
- SERIAL (TX) TO SERIAL (RX)
- RP-125/SERIAL SYNCHRONIZERS
- RP-125X TO SERIAL
- SERIAL TO RP-125X

METHODS OF PRODUCING HIGH LEVELS OF RF POWER FOR TEST PURPOSES

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INTRODUCTION

At some time, early in the life of every passive RF component, the designer must seclude himself away in a dimly lit room and consult the spirits of electromagnetic phenomena, seeking guidance in assigning a power rating to his new creation. Sometimes, the guidance received is good and the new product begins a life of trouble-free operation. At other times, the guidance is not so good, and troubles with failure plague the new component until its true limitations are discovered.

Because of the many "unpredictables" involved with a component that is to be used in uncontrolled environments, establishing a safe power handling capability for the component requires some degree of mysticism as well as a healthy dose of science. The scientific portion of the procedure involves determining a realistic breakdown point under known conditions such as temperature, pressure and humidity. The mystical portion comes into play when the breakdown under known conditions must be adjusted in order to account for the unknown conditions under which a customer will actually use the product. This paper is a short discussion of three techniques used at Shively Labs to perform the scientific portion of the procedure - determining where breakdown occurs under known conditions.

FAILURE MODES AND BREAKDOWN LEVELS

When we test a new component, two of the first questions that come to mind are, "How is it going to fail?" and, "How much power should it take?"

RF components can fail in two ways: high current failure or high voltage failure. High current failure occurs when the heat produced by I^2R losses in the component exceeds the heat that can be removed from the component while the component remains at a reasonable temperature. As power is dissipated in the component, its temperature rises, and the component rejects heat to its environment. If the dissipated power is too great, the component's temperature may rise high enough to cause detuning or even meltdown of the component. On the other hand, high voltage breakdown occurs when the voltage gradient or electric field (measured in volts per meter) between two points exceeds the breakdown strength of the insulating material separating the two points. The material ionizes, and an arc occurs.

Both of the above breakdown modes are very difficult to evaluate without experimentation. The ability of some arbitrary shape to reject heat to its surroundings is almost impossible to accurately predict. Add to this, uncertainties in actual conductance, the effects of oxides on conductor surfaces and other unknowns, and it is easily seen that the high current breakdown point of a component is not an easily determined parameter. Similarly, the high voltage breakdown point is no easier to predict. Although the voltage between two points may be calculated, the voltage gradient depends not only on the voltage, but also on the spacing and shape. So, in both cases, experimentation is the only sure way of answering "How is it going to fail?"

The second question, "How much power should it take?" is sometimes a bit easier to answer. If we are designing a component intended to be used with a particular size transmission line, for example, it shoud be capable of handling at least as much power as the transmission line. However, in some cases (depending on the component's use) the component's capabilities must be several times those of the transmission line.

Once we determine what power capability the component should have, we need to determine the power level up to which we want to test the component. This is the mystical part mentioned earlier. In the laboratory, we can create some conditions - rain, high temperature, etc., but we can't create all the possible conditions under which the component might be operated. Therefore, we need to include a safety factor that will account for the degradation occuring in the field. Since we cannot predict everything that might happen, we try to select a safety factor large enough to ensure safe operation under any foreseeable conditions and hope that we've created a component that also works even under some unthinkable ones.

This brings us to the meat of this paper - once we've determined that some component should be tested to 100 kW, for example, how do we create this rather considerable amount of power?

WHAT'S IN A WATT

I think that it's safe to assume that we all know the most basic definition of power:

(1a) P(in Watts) = V(in volts) I(in amps).

Using Ohm's Law, we can substitute for V and I above to get some of the other forms of expressions for power:

(1b) $$P = VI = (IR)I = I^2R$$

or

(1c) $$P = VI = V(V/R) = V^2/R$$

(The same expressions hold true if we use complex power and impedance, Z.)

Now, recall that the two breakdown modes are not strictly dependent on power; they occur either due to a high voltage or due to a high current. Furthermore, most of the components we design are built for use in 50 ohm transmission line systems; using the above expressions for power, we can write:

(2a) $$P = VI = I^2(50) = V^2/50$$

where R = 50 has been used in order to specialize the expressions to 50 ohm transmission systems. Therefore, when we speak of testing a component to 100 kW, we are implying that the 100 kW is being applied to a 50 ohm system. This means that

(2b) $$100 \times 10^3 \text{ Watts} = I^2(50)$$

$$\text{or } I = \sqrt{100 \times 10^3/50} = 44.72 \text{ amps},$$

and

(2c) $$100 \times 10^3 \text{ Watts} = V^2/50$$

$$\text{or } V = \sqrt{(50)(100 \times 10^3)} = 2{,}236.1 \text{ volts}.$$

In short, the above expressions state that 100,000 Watts will produce 44.72 amps and 2,236.1 volts on 50 ohms. The realization of this fact reveals a means of testing high power components without the high power; simply impress on the test component the same current or voltage as is produced by the desired power level on 50 ohms.

As an example, suppose we have a coupler in 4 1/16" coax transmission line that we'd like to test to 100 kW. The coupler consists of a very small loop inserted through the outer conductor of the line. First, we can determine which breakdown mode is more likely. Because the coupler carries only an insignificant amount of current, there is essentially no chance of high current breakdown, but, because of its sharp edges and position between the inner and outer coaxial conductors, there is some chance of high voltage breakdown. In order to completely test the coupler for high voltage breakdown at 100 kW, we need simply to apply 2,236.1 volts between the inner and outer conductors. The amperage and, hence, the power that we apply is unimportant as long as the voltage is 2,236.1 volts from inner to outer (and, to keep things exact, the frequency is in the range of the coupler's intended use). In a sense, we are causing a small amount of power to look like a large amount of power.

THE STANDING WAVE RESONATOR

(making a mountain out of a molehill)

I can just hear some people cynically saying, "So what? So now we only need some source of 2,000 plus volts!" Let's take a look at equation (2c) for a minute. This equation states that the square root of power, P, (100×10^3 in eq. 2c) times line impedance, Z , (50 ohms in eq. 2c) gives the voltage across the line's conductor. Generalizing this using P, Z and V leads to

(3a) $$V = \sqrt{ZP}.$$

If we can come up with any combination of Z and P that will yield 2,236.1 volts at the coupler, we can perform our test.

Now, let's switch tracks for a moment. Look at the Smith Chart shown in Figure 1. Using the Smith Chart as an impedance plot, the left edge of the chart indicates an ideal short circuit, zero ohms; the right edge of the chart is an ideal open circuit, infinite ohms. If we put a real short circuit on the end of a piece of coax line and measure the impedance, we will find a small real impedance, which can be plotted on the Smith Chart as a point on the horizontal axis just inside the circle; the distance inside the circle's edge is proportional to the resistance of the short circuit (an ideal short would have zero ohms and would be on the edge of the circle).

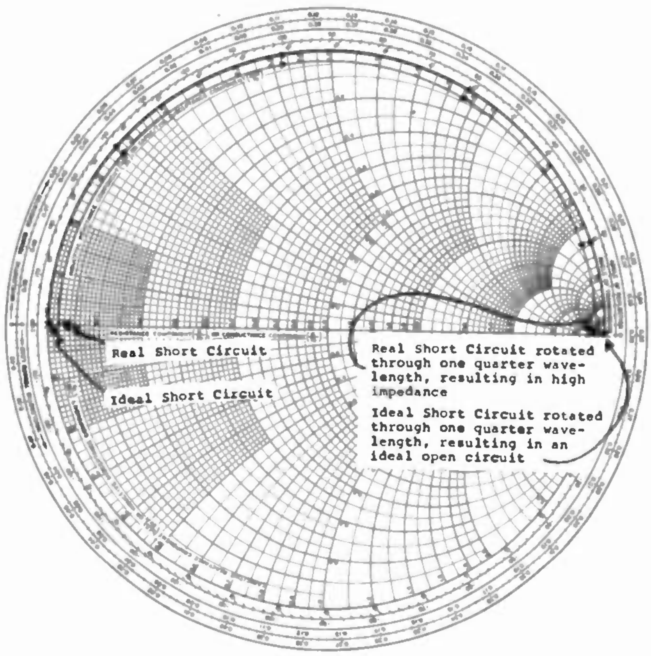


Figure 1. Smith Chart

Let's move along the coax line away from the short circuit; the impedance will travel along the Smith Chart at a constant radius in a clockwise direction. At a quarter of a wavelength, /4, from the short circuit, the resulting impedance will be found on the right hand side of the Smith Chart. The real short circuit will still be slightly inside of the circle, and the ideal short circuit will still be on the circle's edge. As just mentioned a bit ago, the right side of the Smith Chart is a near-open circuit in the real case; the ideal short circuit would result in an ideal open circuit. This shows us that, by starting with a real-world short circuit, we can rotate through a quarter wavelength of line and produce an open circuit.

How does this help us test our coupler? Let's place our coupler in this coax line a quarter wave from the short circuit. The coaxial impedance at the coupler location will not be a true open circuit because of losses in the short circuit and the coax line, however, it will be very high; for simplicity, let's assume that the impedance at the coupler is 1500 ohms. We now have 1500 ohms, and we know that we need 2,236.1 volts. Equation 3a gives us a way of relating the voltage we want, the impedance we have, and the power we need to produce the required voltage:

(3a) $$V = \sqrt{ZP}$$

$$\text{or } P = V^2/Z$$

so that, using 1500 ohms and 2,236.1 volts,

$$P = (2{,}236.1)^2/1500 = 3.33 \text{ kW}.$$

In other words, we can produce 100 kilowatts' worth of voltage effects on our coupler while using only slightly over 3 kilowatts of true power!

The next question some people I'm sure are asking is, "How do you get a transmitter to put 3 kW into a load like the one presented by the short circuited transmission line?" Elementary, my dear Watson! True, the impedance of the transmission line is far from 50 ohms, but nothing stops us from using a variable transformer to match our test circuit to a transmitter's 50 ohm (or any other impedance) output. Figure 2 shows a picture and schematic of the test arrangement.

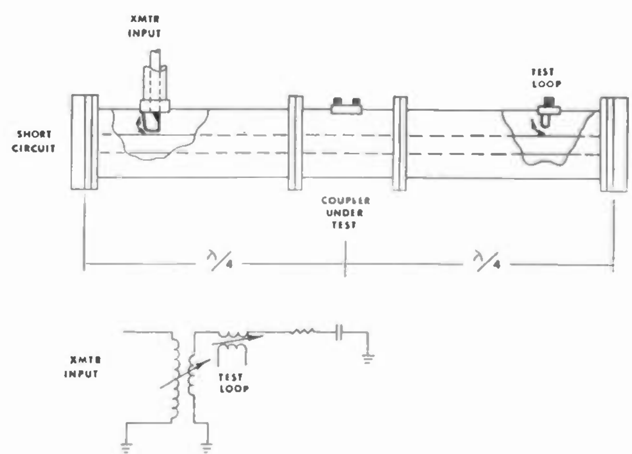


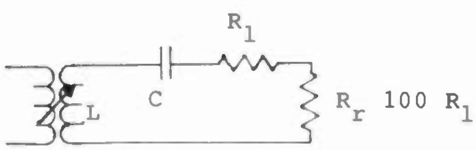
Figure 2. Coaxial Standing Wave Resonator

The transmitter input consists of a rotatable loop inserted through the coax outer conductor near the short circuit. Because the short circuit creates a very low impedance at this point, high currents - and high magnetic fields - will exist here, allowing a very strong coupling between the input loop and the fields in the coax. At the resonant frequency - that frequency at which the test circuit is a 1/2 wavelength long -the rotatable input loop acts as an impedance transformer, matching the small loss resistance of the short circuit and coax line (shown as R in the equivalent circuit) to the 50 ohm transmitter output. The test loop, located at the second short circuit, allows monitoring of the equivalent power level. It indicates the magnitude of the power that would produce the existing magnetic fields if the line were actually 50 ohms. In our example, if we were putting in the 3.33 kW, our test loop would indicate a power of 100 kW, because the currents at the short circuited ends of our test circuit (and the voltage at the coupler in the middle of our test circuit) are equivalent to those produced on a matched 50 ohm line by 100 kW. We therefore have a means of producing either a voltage or current equivalent to 100 kW while requiring only 3 kW of true power. The secret to this technique is that we produce the voltage or the current equivalent of 100 kW, but not both at the same time and place.

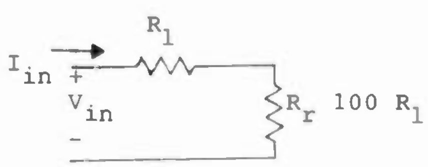
This is only the first of three resonant test circuits commonly used for high power testing. The two other circuits are described in the following sections.

ANTENNA TESTING

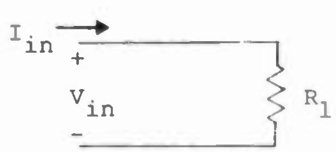
The standing wave resonator described above works well for almost any component that can be incorporated into a piece of transmission line. Antennas, however, cannot be tested using this arrangement. So, how do we test an antenna? Let's take a look at how an antenna works and see.



A. Circuit model of loop coupled antenna.



B. Circuit at resonance where $wL = \frac{1}{wC}$.



C. Circuit in metal room at resonance.

Figure 3. Circuit Model of Antenna.

An antenna, to a very close degree, can be modeled as a series resonant circuit consisting of an inductor, a capacitor and two resistors; this circuit is shown in Figure 3A. The illustration shows two resistors (rather than just a single resistor equal to the sum of the two) in order to emphasize the two very different sources of resistance.

R_l is a relatively small resistance caused by the actual conductor losses in the material of which the antenna is made. When power is put into an antenna, R_l is responsible for the slight heating of the antenna. Therefore, any power dissipated in R_l is not radiated into space, i.e., is lost power.

R_r is a much larger resistance (usually more than 100 times the resistance of R_l) and is called the radiation resistance of the antenna. The source of this resistance is the power radiated into space by the antenna, i.e., R_r is an imaginary resistance that represents the portion of input power that is transferred into electromagnetic waves in the air surrounding the antenna.

In Figure 3A, the transformer is shown to indicate that most of the antennas we build are loop coupled or externally fed. For the time being, we'll ignore the transformer and look at the series RLC circuit on the right side (i.e., from the transformer secondary to the right). At resonance, the impedances of the inductor and capacitor are equal in magnitude and opposite in polarity so that they cancel. We can therefore ignore L and C for the time being and look at the simple resistive circuit of Figure 3B.

This circuit is now a simple resistive divider. For the sake of simplicity, let's assume that $R_l + R_r = 50$ ohms. This means that R_l is approximately 0.5 ohm and R_r is approximately 49.5 ohms. It also means that we can feed the antenna directly with 50 ohm transmission line. Let's put 10 kW into the antenna at its resonant frequency and see where the power goes. Using our previous equations relating power, voltage, current, and impedance (resistance in this case), we can find V_{in} and I_{in} as shown in Figure 3B to be

$$V_{in} = \sqrt{(50)(10{,}000)} = 707.1 \text{ volts}$$

and

$$I_{in} = \sqrt{(10{,}000)/(50)} = 14.1 \text{ amps.}$$

Let's look at the power dissipated in each of the two resistors:

$$P_{Rl} = I^2R_l = (14.1)^2(.5) = 99.4 \text{ Watts}$$

and

$$P_{Rr} = I^2R_r = (14.1)^2(49.5) = 9.8411 \text{ kiloWatts,}$$

where each was rounded to the nearest tenth of a Watt. This shows that a vast majority of the input power (over 98%) is radiated by the radiation resistance into space, as it should be.

Now, let's take this antenna, still feeding it its diet of 10 kW, and put it inside a large, sealed, metal room. If our room were made well enough, none of the radiated power could escape from the room to the outside. But, we're still pumping 10 kW out of our transmitter into the antenna, which is inside the room. Where does the power go?

When the power radiated by the antenna encounters the metal walls of the room, it is reflected from the walls back to the antenna. A very small amount of the radiated power is turned into heat in the walls, but nearly all of the radiated power is returned to the antenna. Since there is almost no radiation, there is little or no radiation resistance in our antenna's equivalent circuit, so that, at resonance, the equivalent circuit of the antenna in the metal room is that shown in Figure 3C.

Now, we have 10 kW being dissipated by only the very small loss resistance of the antenna, R_1. Let's take a look at the currents needed to do this.

$$10{,}000 \text{ Watts} = I^2R_1 = I^2(.5 \text{ ohm})$$

therefore

$$I = \sqrt{(10{,}000)/(.5)} = 141.4 \text{ amps},$$

i.e., we've increased the current by 10 times. Remember that, even though we've ignored the impedance effects of L and C, they are still in the circuit, and we have increased the currents through and voltages across them also. These components represent the resonant elements of the actual antenna; hence, we have created voltages and currents on the antenna equivalent to much higher power levels than we could possibly achieve in real life. If the antenna were to have 141.4 amps flowing into it in free space, where the radiation resistance is not suppressed, the power input would be

$$P = I^2(R_1 + R_r) = I^2(50) = (141.4)^2(50) = 999.7 \text{ kW!}$$

Again, as in the coax resonant circuit, we need some way of matching the low impedance antenna in the metal room to the 50 ohm transmitter output. This is accomplished with the coupling loop, represented by the transformer of Figure 3A. The loop allows the antenna to be reasonably well matched both in free space and in the metal test room.

A little thought might lead to a question of measuring the currents and voltages existing on the antenna in the test room. Unlike the coax resonant circuit, we cannot actually make measurements under power. However, there is an accurate means of determining the relative equivalent power levels between free space and the test room.

Figure 4 is a sketch of a ring-style antenna used for FM broadcasting. The two vertical (one up, one down) arms of the antenna are high voltage points while the back side of the horizontal rings (where they meet with the rectangular "block") are high current points. Small probes (loops for current probes, electrically small monopoles for voltage probes) are inserted in these areas, as shown in the blow-up sketches. Cables from these probes are run through the tubing of the antenna (to provide shielding) and out of the back of the antenna through a shielded conduit. These probes allow current and voltage measurements to be made as described below.

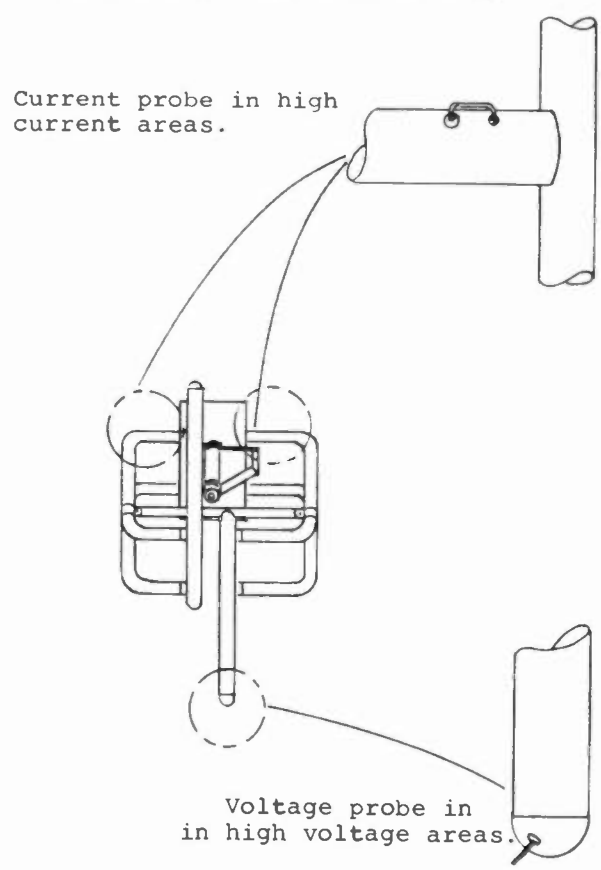


Figure 4. Antenna Test Probe Types and Locations.

A network analyzer (typically a HP8505 or HP8753A) is calibrated in reflection and transmission and connected to measure the return loss of the antenna on its feed line using one channel of the analyzer (as shown in Figure 5). The second analyzer channel is used to measure the transmission between the antenna and each of the small probes; these measurements are performed in free space and again in the test room. Although the measurements are not valid in an absolute sense (i.e., in reading the electric and magnetic fields in volts and amps per meter), because the probes are uncalibrated, they allow an accurate determination to be made of the relationship between the fields in the test room and the fields in free space. In other words, we can measure the difference in current and voltage between the antenna operating in the test room, where there is essentially no radiation resistance, and free space, where the radiation resistance is present. This allows us to determine an equivalence in power, such as we did in the example above. In that example, you'll remember, 10 kW into the antenna in the test room produced the same currents and voltages as nearly 1 MW would produce on the antenna in free space!

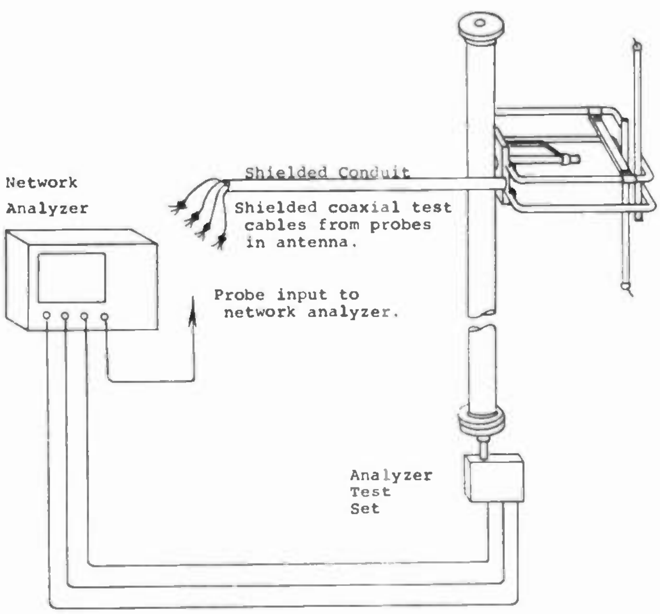


Figure 5. Antenna Test Configuration.

Armed with this knowledge, we place the antenna in the test room, which is actually constructed of wire mesh, and connect it to the transmitter. All probes are either removed or capped (to prevent their incineration!) and the power is turned on. Monitoring the transmitter output power tells us how much power is actually being dissipated in the loss resistance while the knowledge of our free space-to-test room readings tells us what equivalent power would produce the same currents and voltages in free space that are produced by our transmitter power in the test room. Using this technique, we have produced equivalent power levels of over 1.5 MW.

While power is being applied to the antenna, water can be sprayed through the screen of the test room in order to simulate rain collecting on the antenna. The power level at which the wet antenna arcs is considered the breakdown power level. This level is then derated by a safety factor to compensate for the effects of weathering, corrosion, exposure and other degradations that will occur when the antenna is used in the "real world."

The two techniques described so far for creating currents and voltages equivalent to high power levels are both standing wave resonator devices. Essentially, the impedance at various points is manipulated in order to produce high voltages or high currents, but not both at the same place at the same time (and, therefore, not high power). The third and last technique, the resonant ring, differs from the previous two in that an actual high power level is produced, rather than just an equivalent voltage or current.

THE RESONANT RING

Figure 6 shows a schematic of a simple resonant ring. It consists of an input coupler, a monitoring coupler, two sliding short circuits, the device under test (DUT), and enough transmission line to connect these components together in a closed loop that is an integral number of wavelengths long at the frequency of operation.

Power is provided from the transmitter to the input coupler. Some of this input power is coupled into the ring and creates a wave traveling in a counter-clockwise direction around the ring. If the ring is an integral number of wavelengths long at the frequency being used, the wave in the ring arrives back at the input coupler in phase with the transmitter input to the coupler and attenuated by the ring losses. Because the two waves (the one in the ring and the other, from the transmitter, in the coupler) are in phase, the voltages will add and will strengthen the wave in the ring. This buildup of ring power and energy continues until, in a perfectly adjusted ring, nearly all of the transmitter's power is coupled into the ring and is dissipated in the ring losses. This is somewhat analogous to one child pushing another on a playground swing. Initially, the child pushing cannot transfer enough energy to the swing to raise the riding child very far off the ground. Each time the standing child pushed the swing, more energy is added, and the swing rises higher; each time the swing falls, some energy is lost to friction due to air resistance. At some point, the energy lost due to friction is equal to the energy that the child can provide with each push, and the swing rises to the same height each time - or goes over the top of the bar! In the same way, the transmitter keeps adding energy to the ring until, at some point, the energy added by the transmitter equals the energy lost in the ring.

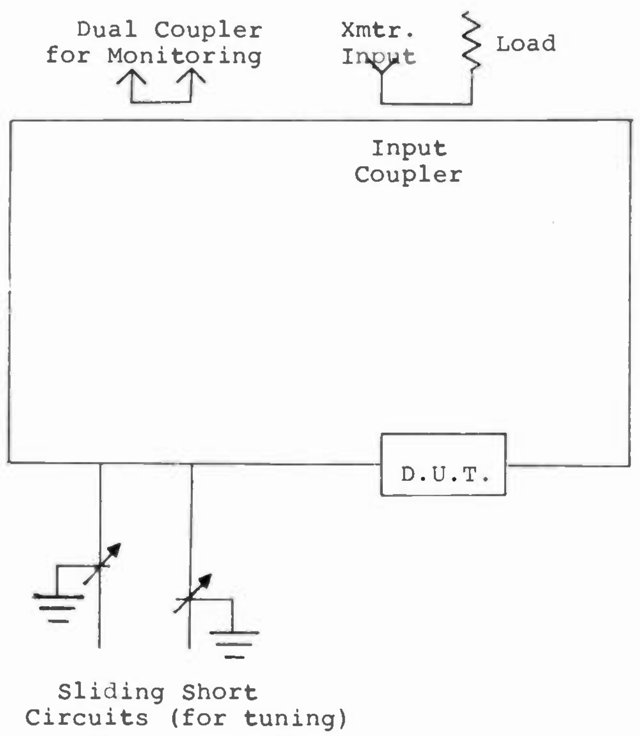


Figure 6. Resonant Ring.

The exact expression for ring gain can be found in "Microwave Filters, Impedance-Matching Networks, and Coupling Structures," authored by G.L. Matthaei, L. Young and E.M.T. Jones, and published by Artech House Books. This expression is

$$G = C^2/[1-10^{-(a/20)}(1-C^2)^{1/2}]^2$$

where G is the power gain in the ring, C is the voltage coupling ratio of the input coupler ($C = 10^{-g/20}$, where g is the coupling in dB) and a is the one trip attenuation in the ring. Knowing the ring gain is not absolutely necessary as long as the monitor coupler is included in the ring. Once calibrated for forward and reflected power readings, this coupler allows us to monitor the actual power in the ring during operation.

Using a 6-inch coaxial line resonant ring, we have produced actual ring power levels of nearly 200 kW, while using less than 10 kW of input power!

CONCLUSION

The preceding discussion has introduced three techniques for creating or simulating power levels significantly higher than those obtainable from standard transmitters. The use of one or more of these techniques in testing a component to breakdown, combined with the use of a reasonable safety margin, provides a means of rating a component's power handling capabilities that ensures many years of failure-free operation.

SECOND GENERATION ENG CAMCORDER

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Following close consultation with a broad segment of Electronic News-gathering (ENG) operators, Sony has developed a new high performance, ultra compact and lightweight, single-piece camcorder that dramatically enhances mobility and flexibility of picture capture. A 3 CCD high resolution camera has been integrated with a new miniature transport Betacam-SP VTR, using the most state-of-the-art components, electronic packaging techniques, and innovative mechanical design.

This paper will describe the many design choices that were examined in light of the pressing nature and demanding needs of ENG shooting. It will outline the techniques employed to implement a camcorder exclusively tailored to meeting those needs.

BACKGROUND

The mid 1970's gave birth to a new generation of single piece portable 2/3 inch TV cameras (shoulder mounted in the fashion of their 16mm film camera predecessors). Connected to the equally new portable 3/4 inch VTR's, they represented a dramatic step forward in rapid and convenient television image capture. Yet, the combination of the two equipments, inextricably coupled as they were by a few meters of unwieldy multicore camera cable, had indisputably taken a step backward in terms of physical convenience of a total shoot and capture system. While a dramatic new immediacy in TV daily news portrayal added a new dynamism to newsgathering, the home viewer remained impervious to the struggles of the hapless duo who daily shouldered 25 lb. cameras (or more) and 20 lb. VTR's - hampered by the unforgiving electronic umbilical cord which inextricably bound them to each other.

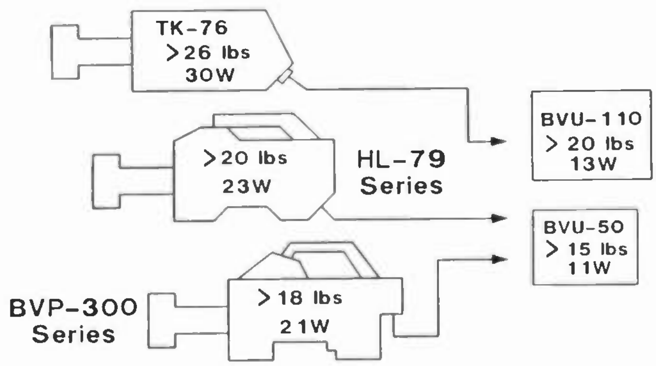


FIGURE 1. EARLY ENG EQUIPMENT (1976-1982)

By 1979 the ENG experience was widespread. And the message emanating from the field was clear - ENG cameras and VTRS were too heavy. Their voracious appetite for power only exacerbated this problem by adding weighty batteries to the overall load. The cable remained a severe restriction on the mobility and shooting flexibility demanded by the often hectic news scene environment.

The birth of the broadcast combo camcorder concept at NAB 1981 signaled the manufacturers' decisive first response to the accumulated experiences of this first era of ENG shooting. The concept was warmly greeted by the broadcast industry. Yet its initial market takeoff remained slow as the broadcaster struggled to absorb the significant implications of the new and incompatible 1/2 inch video recorder formats which were so central to the dramatic size and weight reduction of these ENG VTR's. There was a hesitancy, too, over the combined total size and weight of these combo camcorders which hovered in the neighborhood of 25 lbs. (fully loaded with lens, battery, etc.) Figure 2. In one sense, this seemed a retrograde step (certainly for the camera person) as standalone portable cameras by this time were lowering to the neighborhood of 15 lbs.

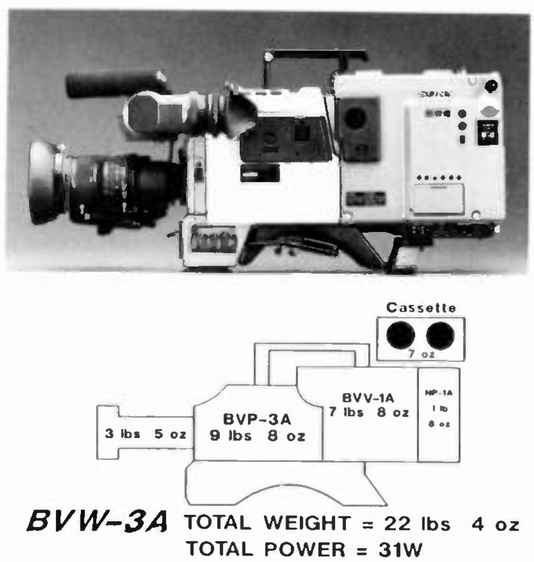


FIGURE 2. FIRST SONY 3-TUBE CAMCORDER BVW-3A (1983).

But it was clear that there was no turning back. The camcorder concept was indisputably in the right direction. The broadcasters growing acceptance of this new state-of-the-art acquisition system paralleled the manufacturers' continuing efforts to refine the total ergonomics of the package.

By 1986 Betacam was an internationally established ENG/EFP field acquisition format. The combo camcorder generally had been accepted by every major network and large numbers of local broadcasters in North America. But the four year experience was still clearly signaling for a substantial trimming of the total package size, weight and power consumption. Apparently - the ideal ENG acquisition system still eluded the industry.

At NAB 1986 Sony introduced the BVP-5 broadcast CCD camera to the industry. This camera, when married to the Betacam recorder took another important incremental step in the right direction. Now the total weight became 20 lbs., power consumption dropped to 20W (and shooting time became 50 minutes with a modest NP 1A battery). Of special significance was the speedy industry recognition of the special attributes of the CCD imagers within the hard pressed environment of unpredictable ENG shooting. By the end of 1987 the CCD was rapidly establishing itself as the ENG camera of choice. But also, by the close of 1987 it was clear that the industry would welcome a still further honing of the primary physical parameters of the combo camcorder. The search for the ideal ENG acquisition system continued.

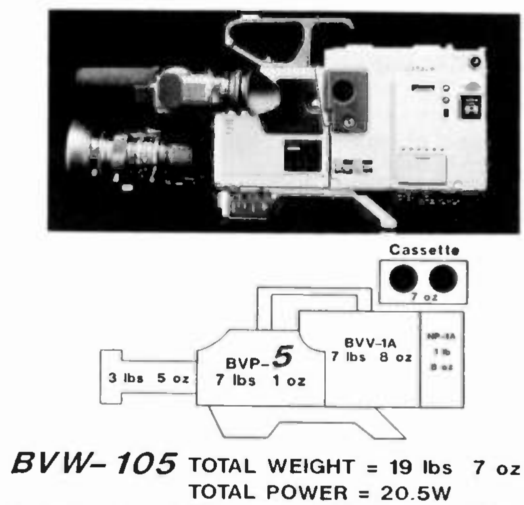


FIGURE 3. BVW-105 CAMCORDER (1986)

THE IDEAL ENG ACQUISITION PACKAGE

The five years of accumulated 1/2 inch camcorder experience did, however, serve to better focus the comments of the many camera operators to the manufacturers. Throughout Sony maintained a continuing dialogue with the broadcast community at large and also with the very rapidly growing video production community. During 1986 and 1987 many meetings were held with end users throughout the U.S.A. - and internationally - to discuss the possible next advances in camcorder design. A number were invited to our factory in this same era for direct discussions with our engineers and industrial designers. From the diverse experience and extensive exchange some basically agreed-to criteria emerged which served as important guidelines in our unrelenting search for optimum one-person ENG acquisition system.

For ENG the overiding considerations were clearly:

- o WEIGHT
- o SIZE AND SHAPE

- o BALANCE
- o POWER CONSUMPTION

A rigorous analysis of all conceivable design options - each measured against these primary requirements - was undertaken (Figure 4).

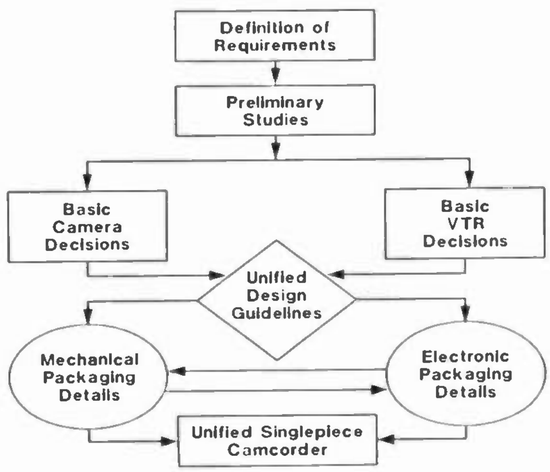


FIGURE 4. SYSTEMATIC ANALYSIS OF ALL DESIGN OPTIONS.

As our engineers evaluated these criteria against all available technologies and packaging techniques some basic decisions were posed:

TWO PIECE OR SINGLE PIECE? - there was clear indication that a unified single piece camcorder design would offer significant electronic packaging advantages - and an improvement in overall reliability.

VTR TRANSPORT - With the advent of the CCD solid state imager cameras had become smaller than 1/2 inch VTR's. This disparity would need to be decisively addressed. All indications were that the present Betacam SP transport would require a substantive reduction in size, weight, and power consumption.

CAMERA IMAGING SYSTEM - there was little dispute that the CCD represented the imager of choice - but which CCD? As our analysis progressed, the pickup device technology turned out to be a pivotal element in the overall cost of the total package. Recognition of the finality of the end users choice when investing in a camera that was inextricably tied (for all time) to a recorder lent considerable weight to the cost issue. We would use the most cost effective and popular CCD device that adequately met the criteria of Betacam SP based ENG imaging.

BASIC ERGONOMIC AND OPERATIONAL GUIDELINES

A vast amount of input was gleaned from the many international ENG crews currently using Betacam. This was assimilated with those very specific comments we received from the ENG camera personnel who directly met in planning sessions with our engineers and designers. From all of this there emerged the following broad consensus regarding the basic aspects of the ideal ENG shoulder mounting acquisition system:

Fundamental Ergonomic Requirements

o Target weight should be 15 **lbs. total** - this to include lens, viewfinder, battery and cassette.

o Power consumption should be minimized to allow use of smaller light weight batteries. A goal of **20W** seemed to reflect a general consensus.

o Single small battery run time should encompass **two** 20 minute cassettes minimum.

o Overall length of camcorder should be shorter than **15 inches** from lens interface to rear extremity.

o A **low profile** camcorder - whose height should allow clear visibility when shoulder mounted.

o There should be a shoulder **balance adjustment** to accommodate weight variations between the many lenses and batteries which would be employed.

Primary ENG Operational Requirements

o Picture quality should fulfill the capabilities of the Betacam SP recording format.

o Quick startup of system.

o Ergonomic displacement of all displays and controls.

o Viewfinder Playback.

o Viewfinder:

- Zebra Indicator
- Character Display of operational status and Diagnostics
- Audio level indication

o Fully automatic operation of camera.

o Frame accurate back space editing.

o Integral but detachable microphone and a phantom supply to power external microphone of choice.

BASIC ENGINEERING DESIGN GUIDELINES

Having assimilated a vast amount of end user input Sony examined a wide variety of packaging scenarios - some of which called for an investment in considerable engineering hardware exploration. It also tasked our industrial designers and mechanical engineers with considerable experimentation with a variety of mockup models. From all of this there emerged the following design decisions:

1. Highly compact CCD camera - preserving our standard 2/3 inch optical interface and employing 3 CCD's.

2. New VTR transport employing smaller diameter drum - but recording precisely to the Betacam SP format.

3. Single motherboard to embrace camera and VTR processing circuits.

4. Extensive employment of the most advanced in state-of-the-art electronic components and packaging techniques.

5. An overall mechanical design which embraced all of the important ergonomic considerations of single person ENG shooting.

6. A high degree of reliability would have to be ensured for the total system. If either the camera or the VTR "went down" - the entire camcorder system would be taken out of action.

7. Maintenance would now acquire a whole new dimension. All major subsystems, including especially the VTR deck, must be modular to facilitate quick and convenient replacement.

For the first time, camera engineers and VTR engineers would meet on the common ground of a single PC motherboard. An all-out onslaught on literally every element of the integrated unit was early recognized as being essential to our success in achieving a 15 lb. total weight. Every single component - mechanical, optical, every PC board, every semiconductor, resistor and capacitor, every operational control would be scrutinized. The very latest in component technology, in every single category, would be seized upon and used.

The degree to which we were successful in realizing a dramatically improved shoulder mounting system is apparent in Figure 5 - which introduces our new BVW-200. Now we will examine some of the engineering details required to achieve this important step forward.

FIGURE 5. NEW SINGLE PIECE BVW-200 CAMCORDER.

CHOICE OF CCD CAMERA

As outlined, the choice of imaging pickup device for this integrated camcorder was made with care. The criteria we established were chosen with full recognition of the irrevocable nature of the choice faced by the end user. The decision to weigh the outstanding new shooting mobility offered against the inescapable fact that the camera and recorder cannot ever be separated was critical to this choice. The camera choice would have to be lived with for the life of the product. Accordingly, our final criteria were tightly steered by real world ENG considerations:

1. Above all else - the BVW-200 is intended to be the most streamlined, lightest and smallest camcorder available - exclusively tailored to facilitate a more mobile and reliable ENG picture capture.

2. CCD imagers, while falling somewhat short of high performance 2/3 inch pickup tubes in total picture quality - far outstrip any known pickup tube in reliable picture capture under difficult scene lighting and awkward shooting environments.

3. CCD imagers drastically reduce camera size, weight and power.

4. The 510 IT CCD (Interline Transfer) developed by Sony has - by far - the industry's most superior track record in ENG acquisition. This device is now widely acclaimed by approximately 3,000 in actual use by broadcasters worldwide. So it represents a tried and true, - a known quantity = - no small issue when integrating a camera intimately to a VTR.

5. The 2/3 inch format 510 IT CCD is more sensitive than any 2/3 inch pickup tube.

6. The 510 IT CCD is - by a distinct margin - the most cost effective professional solid state imager produced by Sony. This key fact represented the final clinching decision.

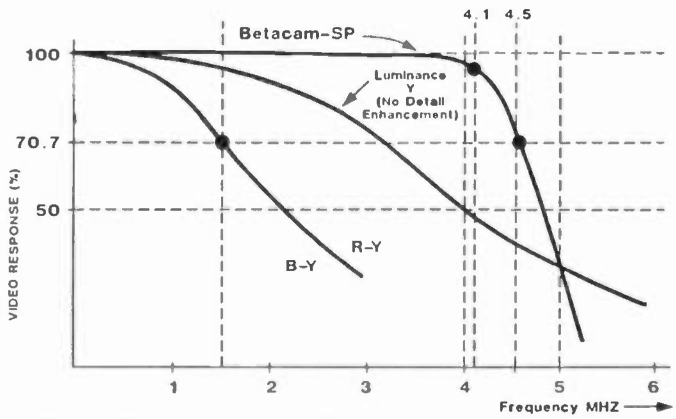
Until the radical concept (as viewed today) of the single piece camcorder establishes a firm track record, the industry would understandably hesitate to invest high sums in an unnecessarily high performance (and high cost) machine.

The final decision also centered about the relationship of the CCD imaging system's resolution capabilities in relationship with the VTR recording format capabilities. The Betacam SP portable VTR records the following video signals:

Luminance Y	4.1 MHz $^{+0}_{-0.5}$ dB
	4.5 MHz $^{+0}_{-3.0}$ dB
Two Color Difference Signals (R-Y and B-Y)	1.5 MHz $^{+0}_{-3.0}$ dB each

The Luminance signal is filtered to roll off rapidly after 4.5 MHz.

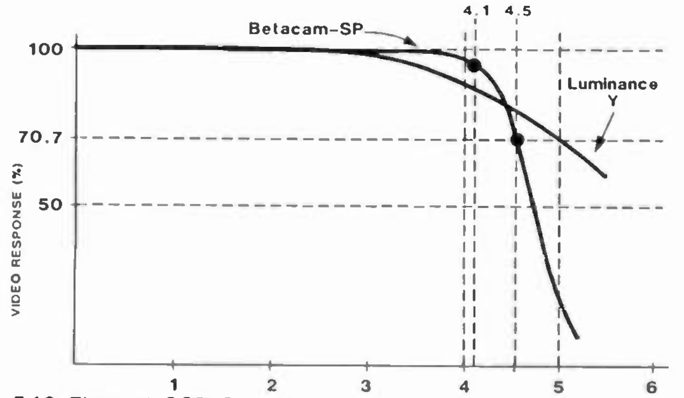
In examining the 510^H IT CCD camera Luminance response in relationship with the above VTR specifications:



510-Element CCD Resolution

FIGURE 6 510^H CCD LUMINANCE RESPONSE (WITH NO DETAIL CORRECTION) RELATIVE TO VTR RECORDING RESPONSE.

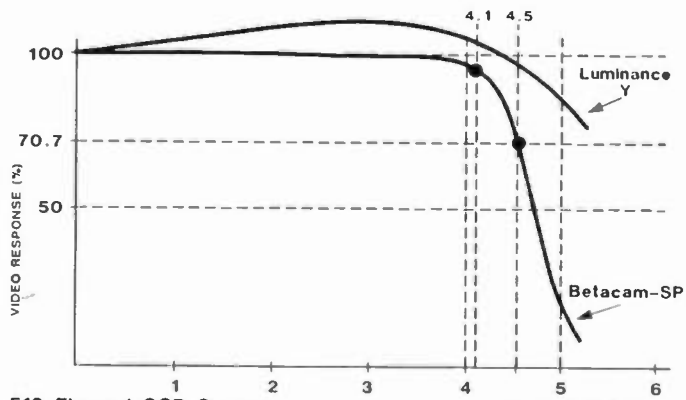
Now - if the camera detail enhancement is adjusted for optimal flatness of the Luminance response:



510-Element CCD Camera
with
Detail Enhancement Set for Maximal Flatness

FIGURE 7 510H CCD LUMINANCE RESPONSE (WITH DETAIL CORRECTION ADJUSTED FOR 100 MTF AT 3MHZ) IN RELATION TO BETACAM SP LUMINANCE RECORDING RESPONSE.

And finally, if the detail enhancement is adjusted to the more typical setting of a small amount of overpeaking we get:



510-Element CCD Camera
with
Normal Operational Setting of Detail Enhancement

FIGURE 8 510H LUMINANCE RESPONSE IN RELATIONSHIP TO VTR RECORDING RESPONSE (WITH DETAIL CORRECTION PEAKED).

An examination of Figures 6 through 8 clearly shows a superb match between the 510H CCD camera Luminance output and the Betacam SP VTR recording characteristics.

With all normal adjustments of the camera detail correction system, the camera is delivering **all** the resolution that the Betacam SP VTR can record. In the case of the two Color Difference Components R-Y and B-Y, it is clearly evident that both of these signals more than fill the available recording bandwidths (even with zero detail correction).

It should also be noted that the CCD camera utilizes the same precision spatial offset techniques of the sister BVP-5 camera[1] - which dramatically reduce the Luminance aliasing interference generated if the CCD camera views a scene containing high frequency information.

The camera performance specifications rank as true state-of-the-art - and provide superb ENG pictures.

Table 1

Nominal Sensitivity	2000 Lux F5.6 89.9% Reflectance White
Low Light Sensitivity	15 Lux F1.4 lens + 18dB gain
Dynamic Range	600%
Signal to Noise Ratio	58 dB
Horizontal Resolution	550 TVL/ph
Registration	0.05%
Warm Up Time (including viewfinder)	1.0 seconds
Viewfinder Resolution	550 TVL
Viewfinder Peaking	Variable

TAPE TRANSPORT SYSTEM

Clearly, the VTR section of this integrated camcorder posed the greatest challenge. It was decided that a decisive reduction of scanner size represented a pivotal first step in attacking the VTR size. The scanner was ultimately reduced to 2/3 the diameter of the standard Betacam-drum: It was essential, of course, that the standard Betacam SP tape footprint be accurately preserved - so the writing speed was restored by elevating the angular velocity of the smaller drum.

Writing Speed is given approximately by:

$$2 \quad fR$$

where R = Drum diameter in mm
f = Revolutions per sec

Assuming f_1 and R_1 are the parameters of the standard Betacam scanner, then we can quickly calculate the required drum speed of the smaller diameter scanner according to:

$$2 \quad f_1R_1 = 2 \quad f_2R_2$$

So the required new revolutions

$$f_2 = \frac{f_1R_1}{f2}$$

$$= 30 \times 74.487/49.643$$

$$= 45Hz$$

The track length was maintained by increasing the wrap angle from the 180^o of the conventional Betacam Scanner up to 270^o on the new smaller screen (Figure 9).

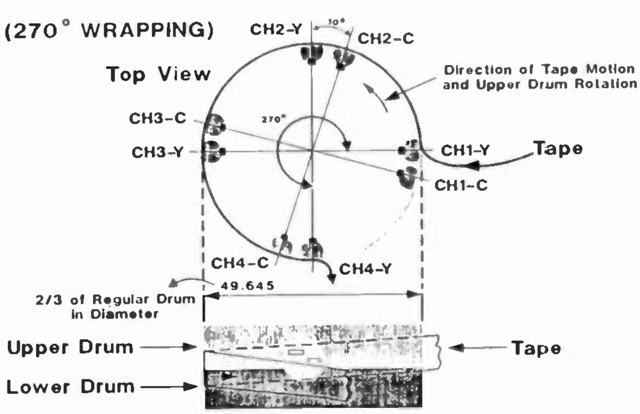


FIGURE 9. SMALL SCANNER.

The degree to which the smaller scanner allowed a reduction in overall VTR volume is shown in Figure 10. This compares the VTR section of the BVW-200 against the dimensions of the standard Betacam SP BVV-5 portable combo-VTR:

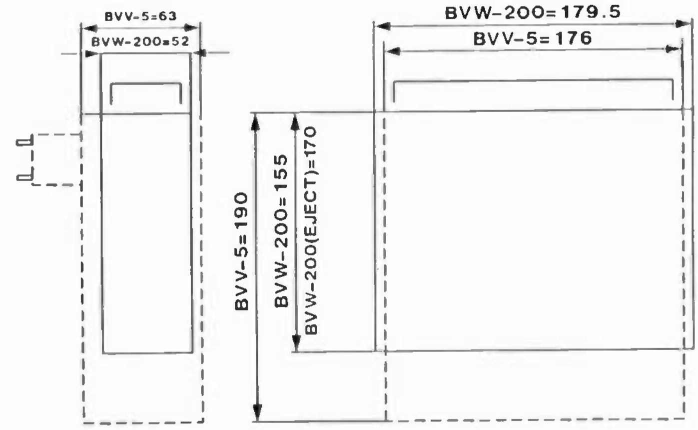


FIGURE 10. OUTSIDE DIMENSION COMPARISON OF BVV-5 DECK AND NEW BVW-200 DECK.

BVW-200 INTERNAL CIRCUITY

While a CCD camera and a miniaturized VTR did much to reduce the size and weight - analysis indicated that it would not be enough if we were to reach our 15 lb. target total weight. The only recourse remaining was to institute an all-out effort to reduce the weight of the basic "innards" of the machine.

This area represented a subject of the most intense study. Considerations of:

- o System partitioning
- o PC board breakdown and disposition
- o Motherboard
- o Weight and size of individual PC boards
- o Weight and size of electronic components
- o Power Consumption of each electronic subsystem
- o Wiring harnesses
- o Thermal distribution
- o Stablility and Reliability

All spoke to enormous study and analysis programs which wrestled with a host of interactive variables. Finally, aided by a considerable amount of laboratory investigation and computer simulation a series of key design decisions were settled upon.

The most contemporary of electronic components would be employed - wherever possible:

1. New lightweight PC boards would be developed.

2. Multilayer boards would be extensively employed.

3. Custom built integrated circuits - both digital and analog would be developed - where appropriate.

4. Hybrid integrated circuits would be developed where they fitted best.

5. A main cable harness would be replaced by an appropriate motherboard.

6. A bus-line interface would be employed between all system CPU's.

7. A software based micro Computer controlled VTR Servo system would be developed to reduce hardware.

SYSTEM PARTITIONING

An enormous effort went into the seeking of a compact system partioning that would yield an optimum packaging that recognized many conflicting requirements of:

- o Performance (camera and VTR)
- o Power disposition among boards
- o Crosstalk
- o Servicing
- o Alignment

The VTR offered the greatest challenge overall - the camera packaging followed a fairly conventional pattern (other than the miniaturization of the PC boards. Most of this total system was physically partitioned into 13 PC boards all of which are electrically interconnected via a single motherboard.

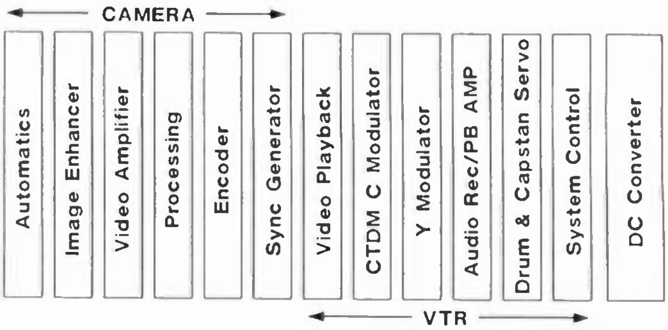


FIGURE 11. PC BOARD DISPOSITION INSIDE THE BVW-200.

Electronic Packaging

1. PC Boards: Recognizing that some 13 basic PC boards would be required to support the majority of the camera and VTR circuitry, a concerted effort was instigated to reduce the total weight of this central electronic system. A new epoxy-glass laminate material was employed in place of the conventional ceramic based PC boards themselves. Board thickness was reduced to 0.6mm compared to the 0.8mm currently employed in the Betacam BVV-5 portable recorder (and the more substantial 1.6 mm used in the studio VTR's). The new material is actually stronger than presently employed PC boards - but a significant 25% weight reduction was achieved. Figure 12 shows a photograph of some of the miniature PC boards.

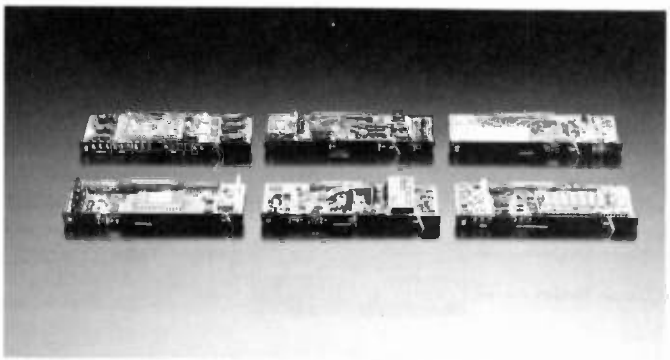


FIGURE 12. BVW-200 PC BOARDS.

2. Electronic Components: The retaining of all the essential processing and system facilities of a modern CCD camera and Betacam SP VTR dictated the employment of a very considerable amount of electronic circuitry. Again - weight, power consumption, and PC board area represented a major challenge that dictated an in-depth scrutiny of all electronic components. The system was closely studied to identify areas and subsystems which might yield to microcircuit packaging all conventional components - particulary the very large number of resistors and capacitors - were carefully studied from the viewpoints of voltage rating and power consumption.

The BVW-200 adopted the use of a newly developed chip resistor - type 1608 - which surface mount directly to the PC board. These represented a significant total volume/weight reduction as they are only 1.6x0.8mm in size compared to the 2.0x1.25mm Type 2125 more conventionally employed in our cameras and VTRs.

A new vertical surface mount capacitor (which have no lead wires) - called a Chemical Chip capacitor - was extensively used and also yielded a valuable size/weight reduction.

New lightweight surface mounting multipin connectors were employed on all PC boards - and made their incremental contribution to the size/weight trimming.

The increased component packing density, of necessity elevated the complexity of PC board layout and interconnection. Where necessary, resort was made to multilayer PC boards to alleviate this.

A new method for soldering all of these components (to both sides of the PC boards) was used. This is the Air Reflow method which simultaneously solder both sides of PC board. A simplified schematic of the process is shown in Figure 13.

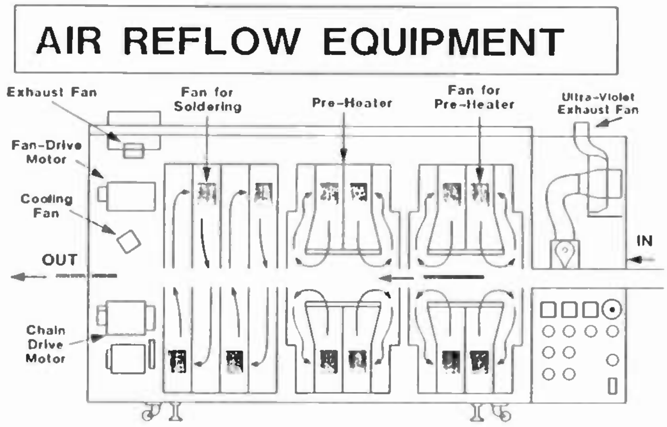


FIGURE 13. AIR REFLOW SOLDERING SYSTEM FOR DOUBLE SIDE COMPONENT MOUNTING.

3. New Microcircuits: A significant investment in custom large scale integration was made which contributed much to the dramatic reduction in total PC board area. The technology employed was basically of two types - monolithic LSI and hybrid LSI.

Monolithic LSI:

- Video Playback Timing Generator - CXD-1357.

Hybrid LSI:

Here 3 IC's were developed:

- Y/C FM modulator - SBX-1524
- Video Record/playback amplifier - SBX-1491
- CTDM Subsystem - SBX-1523

Figures 14 - 17 show in block diagram form the major new IC's that were developed for the BVW-200.

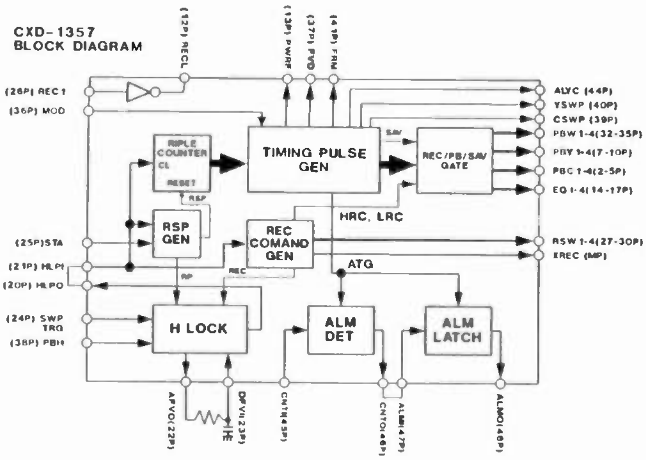


FIGURE 14 VIDEO R/P TIMING GENERATOR CXD-1357

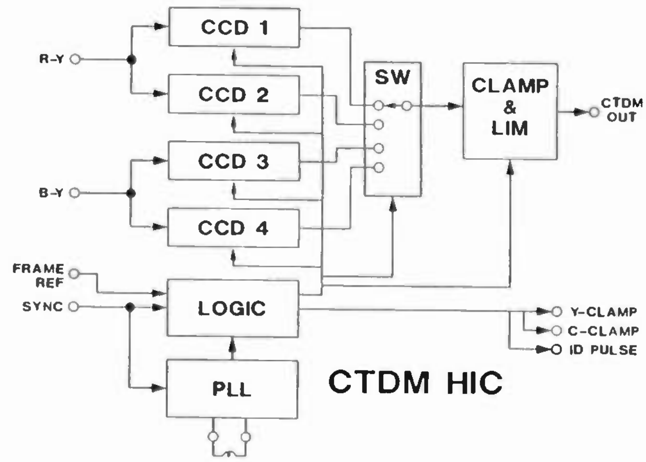


FIGURE 15 CDTM SUBSYSTEM SBX-1523

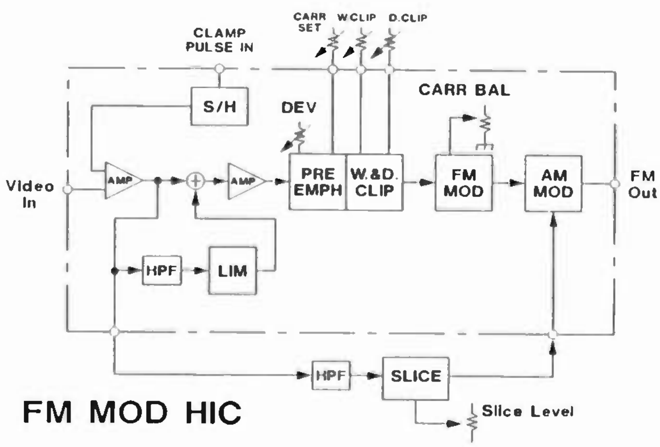


FIGURE 16 FM MODULATOR SBX-1524

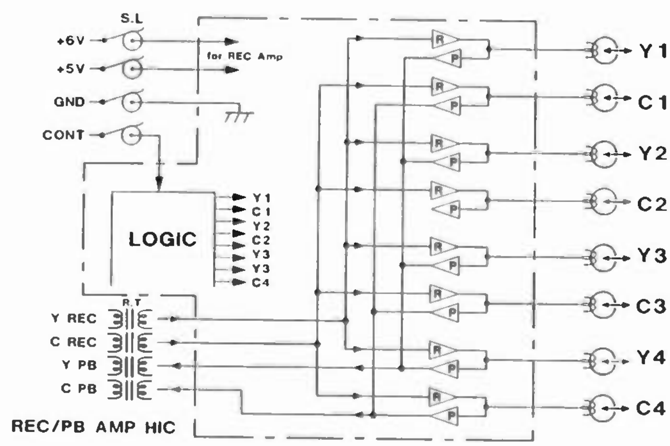


FIGURE 17 R/P AMPLIFIER SBX-1491

In addition to the newly developed integrated circuits a variety of standard custom built integrated circuits (for other VTR's) were employed:

Monolithic LSI:

SXD-1132 Time Code Generator/Reader
CXD-1133 Time Code Microcontrol System
CXP-80100 Drum/Capstan Servo System Control

Hybrid LSI:

SBX-1489 Dolby Audio Noise Reducer

The effectiveness of this major LSI investment is clearly evident from the block diagram shown in figure 16 which indicates the very substantial simplification achieved in the VTR section.

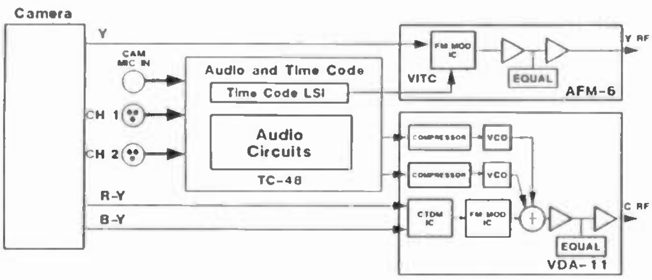


FIGURE 18. SIMPLIFIED BLOCK DIAGRAM OF VTR RECORD SECTION.

4. The Final Package - Figure 19 through 20 shows photographs of the BVW-200 and gives clear indication of the dramatic achievement realized in producing a new era of high performance ENG camcorder. The weight contribution of the major components are as follows:

ITEM:	WEIGHT (lbs)
o Mech. Casing and External Attachments	3.27
o CCD Block	0.98
o Camera PC Boards and Connectors	0.99
o Lens	3.12
o Viewfinder	1.42
o Microphone	0.32
o VTR Deck	1.67
o VTR PC Boards and Connectors	1.44
o NP-1A Battery	1.52
o Cassette	0.44
TOTAL:	15.2 lbs.

FIGURE 19. BVW-200 PHOTOGRAPH. SHOWING PRIMARY OPERATIONAL CONTROLS AND DISPLAY.

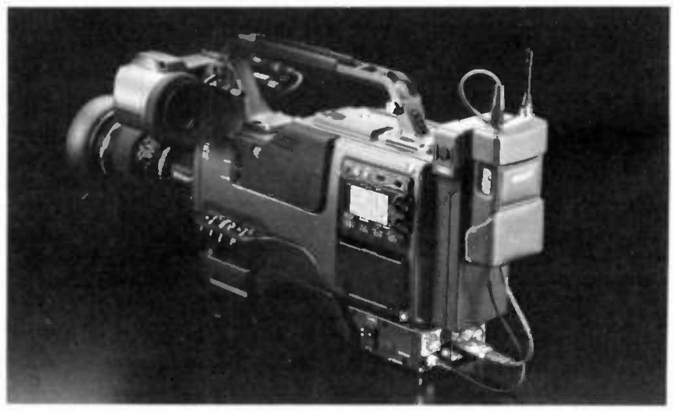


FIGURE 20. BVW-200 PHOTOGRAPH. SHOWING OPTIONAL WIRELESS MICROPHONE ATTACHED.

Maintenance

As mentioned earlier, a great deal of thought went into the issue of maintaining a single integrated camcorder. The plug in PC board packaging for both the camera and the VTR did a great deal to facilitate this. The VTR deck itself is a self-contained unit that also plugs into the main motherboard via 4 PC board connectors. Further design advances in the VTR section will contribute significantly to the improvement in maintenance activities:

- o No solenoids in the BVW-200 - they conventionally have represented a 1000 hour changeout - now totally eliminated o A new timing belt with a wire core extends the average life to 3000 hours compared to the more typical 1000 hour of a normal portable VTR.

Diagnostics

The BVW-200 utilizes 3 CPU within the total VTR system and a single CPU in the camera. All four of these are connected via a single data bus interface. This facilitates a software based centralized diagnostics system which interogates both VTR and camera sybsystems (Figure 21). The camera CPU outputs related diagnostic information to the character generator, which in turn delivers a video output displaying alpha numeric messages in the viewfinder. This diagnostic system can report on the status of the following:

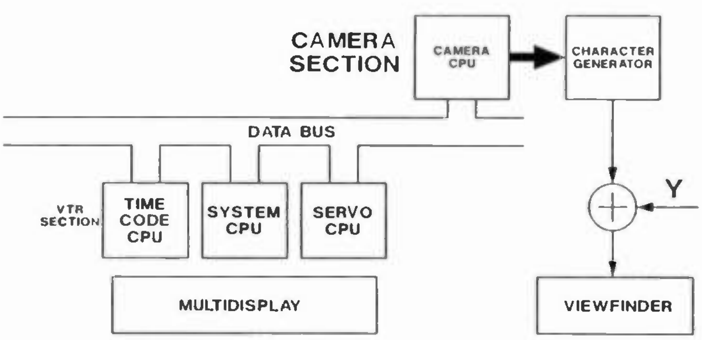


Figure 21. DIAGNOSTIC SYSTEM OF BVW-200.

Camera Check:

- o Voltage setting of battery (before end)
- o Switch interface
- o Auto white circuit check
- o Auto black circuit check

VTR Check:

- o I/O check of system CPU
- o EEPROM interface
- o Hour meter
- o Drum rotating time
- o Tape running time
- o VTR turning on time
- o Display check of multi display

CONCLUSION

An important new tool has arrived for ENG picture acquisition. A unified camera and VTR design, that was exclusively directed to the physical and operational requirements of newsgathering in the field, has produced an integrated camcorder weighing slightly in excess of 15 lbs. (fully loaded on the shoulder). A 3-CCD camera, of standard 2/3 inch image format size, offers a sensitivity and dynamic range that surpasses the best pickup tube and thus represents a new ability to capture images under difficult lighting conditions. The camera's resolution is fully commensurate with the recording capabilities of the Betacam SP VTR to which it is mated.

A system that is making full quality pictures one second after power switch on (including a viewfinder picture of unusually high resolution); a weight and balance that allows easy handheld or shoulder mount operation; a new small VTR deck that further reduces gyroscopic errors - all add up to a new freedom and mobility so vital to the fast pace of modern newsgathering.

A major investment in custom-built LSI; the extensive employment of miniature high technology surface mount components; a significant reduction in power consumption - all contribute to the realization of an elevation in system reliability, operating life, and a decisive curtailment in camcorder system costs.

A STUDY OF MAINTENANCE REQUIREMENTS FOR COMPONENT LEVEL DIAGNOSTICS IN DIGITAL EQUIPMENT

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Montreal, Quebec, Canada

ABSTRACT

Broadcast engineers have been forecasting the "digital revolution" as imminent for at least ten years but when the first D1 DTTRs were produced this year, a well defined benchmark for the "digital revolution" and the "all-digital studio" was at long last firmly established.

Yes, digital television is a reality — not a dream, but it could be the beginning of a high cost nightmare unless we ensure that local broadcast technicians are capable of maintaining and repairing this new and complex technology. Manufacturers in general espouse the view that broadcast maintenance staff are unable to troubleshoot and repair the next generation of digital equipment. It is the authors' belief that the best interests of broadcasters can only be served by adopting a position, as potential equipment purchasers, that unless manufacturers supply diagnostics, documentation and training sufficient to permit a competent maintenance technician to locate faults to the component level, equipment may not be purchased.

In order to address day to day equipment maintenance needs, technical management must direct the efforts of its maintenance staff into specific types of activity.

This paper will discuss a number of maintenance activities which must co-exist in today's analogue digital production facility if the established maintenance needs are to be met. Manufacturers must come to understand that most equipment owners have a need for not only emergency repairs but, as well On-Line, Off-Line and Preventive maintenance where faults are repaired to the component[1] level and equipment is maintained to specifications.

[1]*Component is defined as any sub-assembly, sub-sub-assembly or part that is used in the manufacture of a DTTR or its interfaces. Because some mechanical assemblies and electronic circuitry are extremely complex in function, the definition of a component does not always remain the same. More specifically, complex function chips such as a PLA or ASIC should be user replaceable if defective as should amps, MSI & LSI chips, regulators, drivers, decouplers, etc...*
In the signal system where several VLSI circuits may be found assembled as error encoders, decoders or multiple filters should, although they are sub-sub-assemblies, be considered as components and be replaced as units if defective. A mechanical assembly or sub-assembly such as a motor or shuttle control is considered a component unless there are user replaceable/alignable brushes, potentiometers,clutches, etc...

RESPONSIVE MAINTENANCE

The repair of failed equipment is an important aspect of maintenance activity. When the equipment is required for use, it becomes the first priority. This maintenance work is done in response to events, hence the term "responsive maintenance." The term "fire-fighting" is often used to describe this activity.

Generally speaking, if equipment which is necessary for the production or broadcast of programs fails suddenly and without warning, there must be a maintenance technician available to make emergency repairs. That person must have the skills to effect rapid repairs under pressure of time, and must be supplied with the necessary tools and spare parts.

It follows that maintenance shops must be staffed at all hours when these services are required. The decision as to the urgency of the service must be based on the potential impact of failing to provide the service.

It must be noted at the outset that there is always an impact due to failure, and the amount of reduction of this impact by prompt and efficient maintenance action may or may not be significant, depending upon the circumstances. For example, failure of a key component during a fifteen minute newscast might take ten or fifteen minutes to repair and the program could be lost. An equivalent failure during en editing session could be covered by a coffee break with minor inconvenience.

Restoration of service is speeded up by the availability of back-up equipment which can be plugged in to replace a failed device. Spare modules for equipment serve a similar purpose. The principle is to reduce the time spent identifying the source of a failure by enlarging the size of the blocks being examined, hence reducing the total number which need to be looked at to find the faulty one, and then to reduce the repair time by rapidly replacing a faulty block by one known to be good. The faulty item must, however, be repaired later, preferably off-line when there is no immediate urgency.

The back-up equipment or modules may be provided specifically for this purpose (equipment redundancy), or equipment which is not in use, or is being used by a lower priority program, may be appropriated.

If the blocks are large enough, operators themselves may be able to isolate the faulty block and replace it in the signal path, thus obviating the need for immediate maintenance. Subsequent repair of the faulty item by a maintenance technician remains necessary.

Emergency repairs are, by definition, unscheduled, and emergency maintenance activity is therefore not readily subjected to managerial control. Scheduling of staff in terms of time and numbers, and assignment of work must be done on a more-or-less statistical basis.

REACTIVE MAINTENANCE VS. THE JOB THAT NEEDS TO BE DONE

It is useful, at this point, to investigate how effective a program of emergency maintenance really is.

1. **Remain on the air**

If a line in the transmission path fails in service, the program is off the air. The goal of emergency maintenance is to restore service as quickly as possible. Emergency maintenance therefore concedes loss of service, and focusses on minimizing the duration.

Rapid repair is essential, but the damage in terms of lost audience is likely to be more severe than the duration of the loss of service. If operators or automatic changeover systems cannot very quickly substitute a functional signal path for the failed one, maintenance must take over. Invariably, times of several minutes or more will elapse,and portions of the audience will switch to other channels, where they may remain for the duration of the program period. Loss of continuity in the programming thus expands the impact of failure, and even very efficient maintenance cannot compensate.

In television, there may be a loss of commercial revenues, depending upon the time and duration of the failure. In this case, prompt restoration of service will minimize the loss.

2. **Provide High Technical Quality**

There are three prerequisites for the achievement of high technical quality:

a) The equipment must be inherently capable of delivering it.

b) The equipment must be operated so as to take maximum advantage of its capabilities.

c) The equipment must be adjusted and maintained to operate according to its specifications.

Of these, the first two are beyond the control of maintenance. However, maintenance technicians, as technical specialists are more knowledgeable of the internal function of equipment than operators, and are therefore, more aware of the implications of operational factors on technical quality.

The maintaining of equipment within its technical specifications is not effectively handled with a program of emergency maintenance, because emergency maintenance is only done in response to reported faults. The success of the program is therefore directly related to the ability of the users (operators and production) to detect the often subtle effects of almost-but-not-quite-within-spec operation. While it might be argued that effects so subtle that a trained operator does not detect them are unlikely to be significant to the home viewer or listener, there is a cascading effect: if all equipment in a signal path is operating just at or outside the limits of its specifications, the overall path will probably be significantly degraded, and yet a cursory examination of the various links in the chain will not reveal the source of the degradation. If all equipment is maintained well within its specs, then the source of any degradation appearing at the ends of the chain can be more readily tracked down.

It is therefore logical to argue that maintaining the specs is really not good enough; the equipment can usually be adjusted to be better than spec, and ought to be so adjusted to ensure maximum quality in complex signal path.

Emergency maintenance is often done under pressure, and the equipment is returned to service in working order. There is no opportunity for a thorough check in such circumstances, and any non-catastrophic conditions will not be repaired. If the faulty equipment is replaced and repaired off-line, there is an opportunity to do a complete check-out, and the chance should not be wasted.

In principle, the necessity for a catastrophic failure to occur before equipment is adjusted is not conducive to consistently optimum alignment. Equipment which does not fail and receive consequent attention from maintenance may operate for long periods of time in an out-of-spec condition before anyone notices and draws it to the attention of maintenance.

3. **Achieve a high degree of reliability in technical operations**

Reliability means freedom from failure. Emergency maintenance does not respond to this requirement because it does not occur until after the equipment has failed.

Emergency maintenance can only address the question of maximum repair efficiency to minimize the impact of failures on schedules and budgets.

4. **Maximize the value received in return for the investment in Equipment**

Emergency maintenance activity, as suggested above, does not focus on obtaining maximum quality from equipment. It is therefore usual for equipment operated in such a maintenance environment to tend to be in less than optimum operating condition. Its performance may well be degraded to that which can be obtained from lower-priced equipment, raising the question of how the purchase of the more expensive item can be justified.

The counter-argument is, of course, that this lower-priced equipment, operated in the same environment, would be even worse, and would probably require extra maintenance to maintain even that level of performance. It must be concluded that, in an environment of emergency-only maintenance, good quality equipment is a good investment, but that the broadcaster does not receive full value for its investment.

The minimal attention received by equipment in such an environment is not conducive to promoting an extended lifetime.

EMERGENCY MAINTENANCE

Emergency maintenance activity is not an efficient and effective means of addressing technical requirements. Yet, it is an essential activity which must take first priority in a maintenance technician's workload. This contradiction between the necessity for the service and the potentially limited benefits which accrue from doing it as well as possible indicate the scope of the problem faced by maintenance departments, and suggest the potential for friction and misunderstanding between maintenance and operations/production, even in situations where emergency maintenance is conducted in an exemplary manner.

Therefore, the strategic thrust in the area of emergency maintenance should be directed towards optimising the speed of repairs, or more accurately optimising the speed of restoration of normal operating conditions.

At first glance, replacement of a failed unit followed by off-line repair appears to be most effective strategy from an operational point of view. In practice, to implement this strategy, it is necessary to have functional spare machines within the plant, which can be used to substitute the failed unit. There is, therefore, a planning and prioritizing process and negotiations with production which must be undertaken to determine what, if any, blocks at that cost, can be substituted following failure and the source of alternative block units for replacement purpose. In-situ repair as a solution in general is the practice today for most broadcasters because:

1. The cost of a full set of modules is higher than that of a complete machine.

2. We cannot easily predict which modules to purchase if the budget only allows a few to be chosen. (Murphy would ensure the wrong choice).

3. It is too tempting to leave a spare module defective following a substitution.

4. Spare modules too frequently won't work on another machine or one that is not the same.

5. A spare machine for maintenance is a luxury that no one can afford in a busy production center.

On-Line vs Off-Line Maintenance

The term "on-line maintenance" refers to maintenance work being done to equipment which is booked to, and being used by production.

This work is therefore subject to severe time constraints. It must be done under pressure,and often under the eyes of an impatient producer or artist, in the knowledge that program schedules and budgets may be directly influenced by the efficiency of the individual maintenance technician.

The urgency to return the equipment to service may dictate that some side-effects of failure may not be able to be adjusted — new components may change circuit characteristics, making alignment desirable to optimize performance, but there is pressure to accept a "good-enough — let's get back to work" point of view. The slowdown may force production and maintenance to compromise some technical quality in order to minimize the impact of the failure on schedule or budget. Under such circumstances, maintenance would come back to the equipment later to finish the job properly.

The term "off-line maintenance" refers to maintenance work being done to equipment which is not booked to or in use by production.

The equipment may be under maintenance because it has been substituted for in the production area following catastrophic failure, as a follow-up to o-line maintenance which could not be completed due to production pressure, or to clear up soft faults which were previously identified but not repaired because of production pressures.

Off-line maintenance tends to be thorough, because there is no pressure to release the equipment immediately for production, although there may be a deadline due to impending production needs. Therefore, equipment subjected to off-line maintenance is likely to emerge from the process in better condition than if the process was on line.

Preventive Maintenance

Preventive maintenance is work which is done, not in response to reported faults, but in the expectation that doing it will reduce the number of faults which subsequently occur.

An examination of technical writings, and discussions with a broad sample of people in the television and radio business reveals that almost nobody is opposed to the concept of preventive maintenance. However, many who are in the position of administering technical plants feel that they are not able to implement a thorough preventive maintenance program because of time and staff constraints.

It is clear that the process of preventive maintenance consists of identifying potential in-service faults, and taking action to prevent their occurrence. Since, by definition, a "potential" fault has not yet occurred, any work which is defined for a preventive maintenance program must be based, not on current status, but on experience.

The preceding argument is generally true for catastrophic failures of components, but must be revised when considering "soft faults," since these usually develop over a period of time, and can be anticipated by looking at the current condition of the equipment in comparison to a reference standard. Such maintenance is certainly preventive, but it is more appropriate to refer to it as "routine," as it consists of routinely examining equipment and adjusting it, when necessary, to return it to peak operating condition.

Perceived Situation

It appears that manufacturers don't understand that emergency repair does not represent a significant part of the workload for a technician in a well maintained plant. The high level of skill is concentrated on preventing trouble.

Manufacturers of digital equipment appear to be espousing the view that digital video and audio equipment will be so complex that broadcasters' maintenance staffs will be unable to troubleshoot and repair them. This position is predicated on the idea that the majority of their customers are small private stations. They propose to deal with this situation by:

i) Providing on-board diagnostics to identify faults to module level.

ii) Providing a module exchange service so that component-level servicing is done in the factory rather than in the broadcasters' maintenance department.

Impact

The vendor benefits from this arrangement in several ways:

i) He can provide a lower level of documentation, since less maintenance is done by the user.

ii) He can provide less training, for the same reason.

iii) He can sell the module repair service.

iv) He can cut costs by providing minimal diagnostics.

The user is impacted by this arrangement in several ways:

i) The support which he would need to do component level maintenance is not provided, depriving him of the choice and obliging him to use the vendor's factory service while still maintaining the same level of staff: for technical systems knowledge.

ii) The cost and turn-around time of the contracted repair service is beyond his control and is in addition to current budget requirements.

iii) Many more spare modules must be kept in stock.

iv) At first glance there appears to be a long-term trend toward a lower workload in the maintenance department but further study shows that just as many staff are required to accommodate a changing workload.

v) The training requirements for staff on digital equipment will increase dramatically at first as they did when color was introduced.

vi) There will be a reduced need for new and specialized test equipment.

vii) Fewer components need be stocked.

viii) Emergency repair through replacement of modules is only the first level of maintenance but does not represent the major part of the maintenance workload, i.e. all problems are not solved by simply changing a module.

ix) The use of on-board diagnostics to module level improves restoration-to-service time by identifying faulty boards immediately (however, this level of diagnostics would also be provided in the case of in-house component-level maintenance).

ANALYSIS

Certain aspects of the above are worthy of closer scrutiny.

i) **Maintenance Workload**

While the module exchange process might suggest a potential staff reduction resulting from reduced workload, the short-term consequences are minimal, due to a need to continue servicing the existing analog plant, and the long term possibilities are tempered by the fact that there must be a maintenance establishment to deal with such tasks as:

- mechanical maintenance
- power supplies and other non-modular, non-digital units
- system problems encompassing several pieces of equipment
- problems which are not solved by module substitution
- maintenance of equipment for which no factory repair service is offered

ii) **Skill Level of Maintenance Staff**

While the removal and replacement of a module is flagged by a flashing red lamp on the panel is essentially an unskilled function, the list of other maintenance activities given above suggests a skill level not significantly lower than would be needed to do full component-level maintenance; in fact, the last item on the list is exactly that.

iii) **Training Requirements**

The training requirements for specific pieces of equipment will be a function of the nature of the maintenance needed. Thus, less will be required for equipment serviced by module exchange. However, a certain level of understanding of the equipment, and appropriate basic training digital electronics is needed in any case, following the arguments of i) and ii) above.

iv) **Spare Modules**

To achieve rapid restoration to service, spare modules are used, even in the case where module repair is done in-house. On the basis of experience with factory repairs, it is estimated that twice as many spares would be needed to provide adequate guarantees of equipment availability if no in-house repair service is available. The cost of a complete set of spare modules for a piece of equipment is estimated to be 1.4 times the cost of the equipment itself (given that all modules of the same type will function properly when interchanged from machine to machine). This suggests that the purchase of spare units is more economical than the purchase of sets of modules. The units must not be placed in-service, however, but must be reserved as maintenance spares.

v) **On-board Diagnostics**

The provision of on-board diagnostics to module level is essential in any digital maintenance scenario, as it is the only practical way of assuring rapid restoration to service through substitution. The choice of factory repair relieves the vendor of the requirement to supply the more extensive diagnostics necessary for efficient component-level maintenance.

vi) **Programmable Chips**

Frequent use of PLAs, EPROMs and PROMs, etc.. of the same manufacturer type is anticipated in the DTTR machine design. Without appropriate listings and the ability to program these devices, an unnecessarily high level of user spares will be necessitated, i.e. at least one spare for every chip of the same type but with a different function in the machine with the attendant increase in costs.

Certain other factors must be considered in making this decision:

vii) **Cost**

The potential cost impact of a move toward factory repair services cannot be accurately assessed, but some aspects of it can be discussed:

- **Cost of spare module** will rise significantly (see iv).

- **Cost of training** slightly lower, due to reduced manufacturer training, but some still required, and basic digital technology training is still necessary.

- **Cost of factory repairs**: currently available programs list prices of the order of 25% of the cost of a new board. Experience suggests that the actual cost may be significantly higher, due to shipping and customs expenses.

- **Capital cost of Equipment** may be lower if the manufacturer passes on his reduced costs (less training to provide, less comprehensive documentation, no need for component-level diagnostics).

viii) **Experience with Factory Repairs**

Experience with outside repair services has been generally unsatisfactory. Excessive turn-around time and cost, and inadequate repair quality have been noted. There is no reason to assume an improved situation in the future. In particular, turn-around time is affected by customs and shipping, which is beyond the control of both broadcaster and manufacturer.

ix) **Quality of Maintenance Staff**

The manufacturers argue that the broadcasters' maintenance staffs are not capable of dealing with the complexity of the new digital equipment. While that may be true in some cases, it is felt that the majority of technicians are skilled and competent, and able to perform the necessary maintenance if supported by adequate documentation, training, and on-board diagnostics — how else would we still be on the air?

xi) **EBU Digital Maintenance Studies**

The European Broadcasting Union (EBU) have prepared a series of reports dealing with the techniques of maintenance for digital equipment. Their recommendations suggest that maintenance to component level is a practical proposition when supported by the manufacturer with appropriate training, documentation and on-board diagnostics.

CONCLUSION

Taking account of the various arguments that have been presented in this paper, we conclude that the best interests of the broadcasters will be served by adopting the position that manufacturers are to be requested to supply diagnostics, documentation and training sufficient to permit a competent maintenance technician to locate faults to component level in an efficient and expeditious manner. The diagnostics to module level are essential; those to component level may be offered as an option. The preferred situation for machine repair to component level is a plug-in analyzer, optimally remote, which strikes a balance between the obligatory built-in first level diagnostics and the need to repair to component level.

An engineering Guideline for the D1-DTTR expressing this point of view has been proposed to the VRRT committee but is not likely to have a significant impact on manufacturers because first generation machine design work has already been completed and machines will be delivered soon.

It is noted that the provision of on-board diagnostics and a rapidly maintainable design are concepts which must be addressed by the manufacturer in the earliest stages of design. Consequently, the point of view presented here cannot have significant influence on equipment already in production if the manufacturer chooses to ignore the need for efficient "repairability" of digital equipment by the client.

Digital equipment is a reality today but an adequate level of diagnostics, documentation and training is not. They will only become a reality when and if you, the users, speak out loudly and clearly. Your statement will be heard only when you insist diagnostic and technical support requirements be included with equipment before signing a purchase order.

SIGNAL PROPAGATION AND INTERFERENCE STUDIES FOR A COMPATIBLE HDTV TRANSMISSION SYSTEM

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Abstract

In approximately two years, full-resolution HDTV will be available in Japan and in the United States. This dramatic large-screen television can be distributed by satellite, cable and disc or cassette. For terrestrial broadcasters to compete successfully for audience, compatible full-resolution (over 750 TV lines) HDTV is the only viable solution within the spectrum now available. All attempts to improve resolution compatibly within the existing single channel have resulted in less than 450-lines resolution. For full resolution, additional spectrum is required.

The NYIT compatible HDTV transmission system provides over 800-line resolution. A real-time transmitter encoder and receiver decoder has been constructed. Tests have been started to study effects of multipath echo, noise, and interference using this system. The HDTV augmentation signal has been designed to produce minimum interference on other stations and to have a high tolerance to interference from other stations. These studies are intended to determine new protection ratio criteria for these augmentation signals. With new criteria, existing spectrum can be used for HDTV augmentation that cannot be used for other purposes. Hopefully with this system, all existing terrestrial broadcast stations can be augmented to HDTV using existing spectrum allocations.

Introduction

The television industry is on the threshold of a revolution in technology. Large-screen high definition television (HDTV) will be available both in Japan and in the U.S. perhaps as early as 1990. Initially, distribution will be via media other than terrestrial broadcast. Neither satellite, cable, nor pre-recorded media suffer from bandwidth limitations to the same extent as does terrestrial broadcast. Thus, these media can more easily adapt themselves without FCC approval, to accomodate the broader bandwidth signals needed for HDTV distribution.

There will be a dramatic difference in television picture quality and size between full HDTV images and those now transmitted by terrestrial broadcast. HDTV makes possible large displays with remarkable clarity - better than 750 TV lines of resolution with a solid viewing angle of over five (5) times that of current NTSC television displays. For these very large displays, the obvious differences in size, with sharp resolution, are immediately apparent.

What will be the consequences if the terrestrial broadcast industry cannot broadcast HDTV?

In the fierce competition for audience share, the terrestrial broadcaster can expect to lose the competitive edge, as more and more HDTV is delivered into the home by other media. Within several years after its introduction, HDTV receivers will be affordable to many of the more affluent viewers, just as color television sets have become today. To be sure, terrestrial broadcasters will continue to serve the less well-to-do viewers. However, being on the lower half of a two-tiered video distribution system will make advertising time difficult to sell.

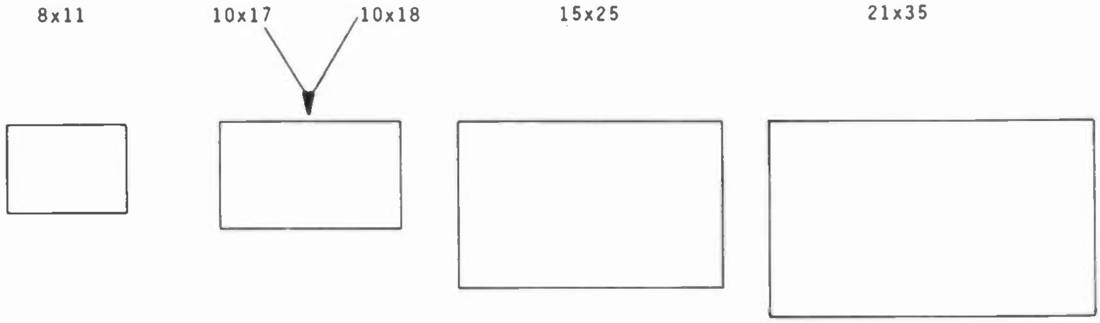
To date, several attempts have been made to improve resolution compatibly, within a single existing channel. These efforts have resulted in systems with resolution that is only marginally better than NTSC. Table 1 shows a comparison of NTSC with proposed advanced television transmission systems.

The single-channel systems are the result of clever schemes to pack additional information into existing holes within one TV channel (Fukinuki hole, vestgial sideband below the carrier, etc.). Although the signal can be derived from an HDTV camera and displayed on a monitorwith over 1000 scan lines, the resolution as seen by the viewer, on a standard EIA test chart, is only about 450 TV lines. For full HDTV (i.e., resolution of at least 750 lines) more information

TABLE 1. Comparative Performance of NTSC and Proposed Advanced Television Transmission Systems

TRANSMISSION SYSTEMS

PERFORMANCE CHARACTERISTIC	NTSC 1 channel	PHILIPS Compatible 2 Channel	ACTV (RCA) Compatible 1 Channel	MUSE (NHK) Incompatible 2 Channel (Adjacent)	VISTA (NYIT) Compatible 1½ or 2 Channel (Non Adjacent)
V Resolution	350	450	450	750	800
H Resolution	350	350	450	600	800
Video Bandwidth	4.2	8.4	4.2	8.1	7 or 9.5
Field of View (Degrees)	8x11	10x17	10x18	15x25	21x35
Relative size (Same Resolution on Viewer's Retina)					



must be transmitted and hence additional bandwidth is required. It has been possible to reduce the bandwidth for HDTV from 30 MHz to between 7 to 9.5 Mhz, and two transmission systems have been proposed to do this. Both systems depend on sending the detail in the image at a lower frame rate. The MUSE system, developed by NHK in Japan, requires 8.1 MHz bandwidth. Terrestrial broadcasting of this system has been demonstrated using two adjacent channels in the UHF spectrum.[1] However, since this system is incompatible, it would require three channels to broadcast both NTSC and a simulcast of HDTV in MUSE format. There is not enough spectrum available in UHF and VHF to support this concept.

Today, the additional spectrum needed for augmenting signals to HDTV is available and has been reserved by the FCC for purposes of television broadcast. This may not always be the case, as the land-mobile industry is making a concerted effort to obtain additional (TV-band) spectrum for their use. Unless the FCC is convinced that this spectrum space will be used for transmitting HDTV, they are likely to give it to the land-mobile industry. In effect, such loss of spectrum would be the death-knell for compatible terrestrial HDTV in the U.S..

At the Science and Technology Research Center of the New York Institute of Technology, an R and D program aimed at achieving NTSC-compatible full HDTV has been underway since 1981. Many people have suggested that we find a simple name for this system. We have decided to call it VISTA (visual system transmission algorithm), an acronym that reflects the system's dual conceptual origins: vision science and digital signal processing. VISTA has been demonstrated in closed-circuit form at the 1986 and 1987 NAB conventions. VISTA is a compatible bi-channel HDTV transmission system that digitally filters the video signal at the transmitter in accordance with well-known spatio-temporal properties of the human visual system. This forms the basis for the bandwidth compression algorithm that reduces spectrum requirements from 30 MHz to between 1.5 - 2 TV channels (7-9.5 MHz). To achieve both compatibility as well as good motion rendition, the baseband signal is kept the same as the local standard (NTSC, PAL, SECAM). The spatial resolution of the image is enhanced to HDTV by co-transmission of an "augmentation" or detail signal. As the visual system requires a longer time period to perceive high than low resolution information, the frame-rate for transmission of the detail information is lower than normal (7.5 - 15 frames/sec.). this saves the spectrum bandwidth.

Recombination of the detail and baseband signals is accomplished in the receiver, through the use of frame stores and separate synch signals. [2,3,4,5]

VISTA has two very strong points. First, because it is compatible with the present 525-line format, it provides a means for broadcasters to gradually upgrade their equipment to HDTV. It does not make obsolete much of today's broadcast equipment or the existing NTSC receivers. Secondly, because an additional 1/2 - 1 channel is used for a detail signal, the resulting picture, when displayed on a wide-screen 1125-line HDTV monitor, has exceptional clarity, in excess of 800 TV lines on a test chart both vertically and horizontally. VISTA also has a method for achieving aspect-ratio compatibility between 3 x 4 and 3 x 5 formats. [4]

Propogation and Interference Studies

Although it has come a long way, VISTA is still under development. In order to be adopted as the American standard for HDTV, more investigation is needed to address several specific issues. The detail signal has been tailored to make use of spectrum that is available, but currently unused both in the UHF and VHF bands. The signal may, but need not necessarily, be adjacent in frequency to the broadcaster's NTSC signal. The amount of spectrum that is available for HDTV depends on how many augmentation signals can be transmitted in the unused spectrum. In most cases, these channels are not used because a transmission at that frequency would interfere with another station in the same channel at another location, an adjacent channel, or another channel which must be protected because of a "taboo".

By proper design of the augmentation signal, much of this interference can be significantly reduced. The augmentation signal in VISTA has no main carrier (by the use of single side-band surpressed-carrier). It has no sound carrier or color carrier and its synch signals are at an amplitude lower than peak video. By this special signal design, new protection ratios can be established for VISTA augmentation signals. This will make possible the use of spectrum that cannot be used for standard NTSC transmissions or, in most cases, by land-mobile transmissions either. A study of the interference protection ratios of VISTA augmentation signals is vital to determine new spectrum allocation criteria for HDTV.

The converse question also needs study: namely, to what extent will the standard NTSC signal interfere with the detail signal on the VISTA augmentation channel? The detail signal in image enhancers successfully uses coring as a way of reducing noise in the image. Low level signals are simply "cored" out without noticeable image degredation. We expect that the use of coring of the VISTA detail signal will reduce the visibility of multi-path echos, noise and other low amplitude interfering signals from other stations. However, more studies are needed to determine protection ratios, and signal-to-noise ratios. We also need to determine the optimum power and weighting function needed for the detail signal. Previous spectrum analyses [6] have suggested that using 8 dB improvements in protection ratios, and a 1/2 channel augmentation signal, all existing U.S. broadcasters could augment to HDTV with spectrum space to spare. If protection ratios can be improved further, even more spectrum will be available for HDTV augmentation.

A new VISTA transmitter-encoder and receiver-decoder have been completed. These units include all of the signals necessary for actual transmission tests. Experiments have already started to determine the protection ratios, propogation condition, signal-to-noise ratios, power requirements and weighting curves for VISTA detail augmentation signals. The results of these tests should make possible a study to determine how much spectrum is available for HDTV augmentation, using these specialized signals.

The VISTA system that has been demonstrated uses 15 fps detail augmentation, which requires a full second channel. With a new set of protection ratios for interference from, and on, VISTA augmentation signals, it may be possible to augment all existing channels to HDTV at this frame rate. The highest priority for our program this coming year is to determine how much spectrum can be made available by reducing interference.

Previous tests of the system at 525-line resolution, that used using a 7.5 fps frame rate for the detail augmentation were relatively successful, at the sacrifice of some dynamic (but not static) resolution. By time-sharing a detail transmission on alternate frames between two stations, this lower frame rate could be used in areas where a full channel is not available for augmentation. There is no change in the receiver processing if the augmentation signal is transmitted at either 15 or 7.5 fps. A digital code in the main channel will program the receiver to load interval 1 or interval 2 of the detail signal into its store for 7.5 fps augmentation. If it loads both intervals, the augmentation is at 15 fps. Processing experiments to study 7.5 fps augmentation possibilities are planned after it has been determined how much spectrum can be made available by reducing interference.

Summary

VISTA represents a means by which broadcasters can join the HDTV revolution without sacrificing either present equipment or viewing audiences. It has been designed by an American laboratory for use in the U.S.. An R and D program to answer many of the spectrum allocation questions raised above is presently under way, at NYIT. Results to date are very encouraging. However, support is needed for this effort if terrestrial broadcasters and the viewing public are to share in the benefits of HDTV.

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BANDWIDTH—EFFICIENT ADVANCED TELEVISION SYSTEMS

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Abstract

A family of new television systems is described featuring improved utilization of analog channel capacity and a tailoring of the information transmitted to the characteristics of the human observer. Double-sideband quadrature modulation of a single carrier in the middle of the band is used together with division of the signal into components according to temporal and spatial frequency. One form of the system provides receiver compatibility and improved resolution on special receivers. The other, noncompatible form provides true HDTV performance on the same receiver. It is suitable for cable use and for the final stage of a 2-stage transition to HDTV for terrestrial broadcasting. Each fits into a single 6-MHz channel.

Introduction

The objectives of the family of advanced television systems (ATV) described here include the achievement of the highest feasible image quality within a prescribed analog bandwidth using receiver technology deemed to be practical in the near future. Bandwidth efficiency, i.e., the ratio of image quality to rf bandwidth, is very important in a period of growing pressure on limited spectrum resources. At the same time, the systems are to serve the most probable strategies that will be followed in the various broadcast media, in particular terrrestrial broadcasting and cable.

Terrestrial broadcasters must continue to serve their present audiences with NTSC transmissions. Although simulcasting is one way to do this,[1] American broadcasters overwhelmingly wish to use, instead, a receiver-compatible system, making the transition to 'true' HDTV at a later date, if at all. In such systems, existing receivers would display an image no more than slightly degraded, while new receivers would see improved images. Cable, on the other hand, does not really need receiver compatibility, since viewers are paying for service and could be induced to acquire new receivers by various methods. Although willing to consider systems that use more than 6 MHz, both media prefer channel compatibility,[2] which means that the ATV signal should fit within a single 6-MHz analog channel and perform adequately in the presence of the usual channel degradations. Needless to say, it is a challenge to achieve very high quality in 6 MHz by any method, and much harder if the burden of receiver compatibility is added. Receiver-compatible systems cannot achieve as high quality as equally clever noncompatible systems. It is generally agreed that true HDTV is not possible in 6-MHz compatible systems, so that only EDTV performance is generally claimed.

Other useful characteristics of a new ATV system include a friendly interface with 24-fps film, which now provides a very large share of programming. If additional bandwidth were every made available, it would be desirable to be able to make use of it to improve picture and sound quality without obsoleting receivers or studio equipment. It is also important that, as new knowledge about video signal processing developed[3] and, as improved signal-processing components became available, that these could be used to enhance performance of the system without obsolescence.

Two basic methods are used in these systems to achieve the desired performance. One is to use the channel more efficiently - i.e., to transmit more units of information per MHz - and the other is to select the information to be transmitted taking careful account of the properties of the human observer. The latter technique involves using more channel capacity for information that is most visually important and vice versa.

Scenarios for ATV Introduction

Noncompatible broadcasting systems. Systems such as MUSE [2], which are compatible neither with NTSC receivers nor with standard channels,[4] face two obstacles in the usual media. The 'chicken and egg' problem is one, and the realignment of channel assignments, a technical and regulatory nightmare, is the other. Therefore, this approach is not attractive to broadcasters. These systems could, however, be used for DBS, which is what MUSE was intended for, and conceivably for some other new wideband transmission system such as fiber. While these new systems would probably grow only slowly, they might take enough audience share from current systems to affect the latters' financial stability. Since free TV service in the US, unlike most other countries, depends on the economic health of the existing commercial broadcasters, any such development would have to be looked at very carefully even by a government committed to deregulation.

Noncompatible recorded media. It has been announced that Japanese HDTV tape and disk players, together with 1125-line monitors that will also display NTSC, will be introduced into the US in 1990. Although the high price of this equipment will probably cause consumer acceptance to be quite slow, even a small loss of audience share would be economically damaging to conventional broadcasters. The fear of this situation and the determination to compete with such recorded programs is the major force behind the current FCC Inquiry. [1]

Receiver-compatible ATV. One way to get to HDTV broadcasting is to introduce, at first, a receiver-compatible system that would not materially degrade reception on existing receivers, but would give much better quality on new receivers. Such systems would employ the basic NTSC signal structure and also transmit enhancement signals, either hidden in the NTSC signal or transmitted in an extra channel perhaps 2, 3, or 6 MHz wide. A second step would be taken, at a later date, to go to a final HDTV system. To make this scenario practical, the intermediate system would have to be a true technological 'bridge' as well as to provide images of intermediate quality. It would be highly desirable, for example, to use the same receivers, possibly with an adaptor or plug-in unit, for all three kinds of signals - NTSC, intermediate EDTV, and final HDTV.

Channel-compatible ATV. Freed of the constraint of receiver compatibility, we believe that it is possible to get quality at least as good as MUSE within the existing 6-MHz terrestrial or cable channels. It appears that such a system could also be designed to have good performance in the presence of normal channel degradations, and to have interference characteristics at least as good as NTSC. This kind of performance depends on a 'smart' sophisticated receiver having a substantial amount of storage and signal-processing capacity. The main cost of a receiver, especially one capable of displaying high-definition images, is due to the picture tube and associated circuitry, power supplies, cabinet, and front end. The incremental cost due to enhancing the signal-processing section will therefore be quite small.

An HDTV system of this kind would be immediately useful for cable, and could be used as the final stage in a terrestrial system, the intermediate stage being receiver-compatible. If, in fact, the higher picture quality of the EDTV system resulted in a growing population of new receivers, it would then become economically practical to switch over to the noncompatible 'true' HDTV system at a specific time, decided on and announced sufficiently in advance. The decision as to whether to provide any service at all to the older receivers when this switchover took place could be made on the basis of the number of old receivers then in existence.

Principles to be Used

Use of appropriate receiver technology. The simplest receiver is one that would scan in synchronism with the camera and do minimum signal processing. For HDTV, this implies parallel channels for red, green, and blue signals with a total of 30 to 50 MHz bandwidth. The most complex receiver would depend on all of the techniques developed for highly sophisticated video conferencing systems. This implies a transmission rate of, perhaps, 10 Mbits/sec, which could readily be accomodated in an analog channel of a few MHz. This tradeoff between receiver complexity and channel efficiency shows that the former has a profound influence on TV system design. A system to be deployed in the 90's and to be used well into the next century should use technology appropriate to its era, and not to times past. This means, at the very least, a good deal of computational capacity, digital signal processing, programmability, and generous use of memory.

Making space for the enhancement information. Both the compatible and noncompatible versions of the proposed systems construct the signal at the receiver from a number of transmitted components. In the compatible version, one of these is more-or-less 'pure' NTSC, and the rest are enhancement signals. The wideband NHK system has about five times the number of image samples/sec as NTSC. For enhancement signals to bring NTSC up to this level, the former would need four times the bandwidth! Even with some kind of compression, the amount of enhancement information needed is quite large. Hiding this in the NTSC signal itself, without harming NTSC performance, is rather difficult. We have therefore elected to make room for this additional information by usurping about 25% of the height of the NTSC frame. The remaining area has an aspect ratio of 16:9, so that both kinds of receivers would see precisely the same subject matter, but in different resolution. There are no side panels, and there is now enough space to do a good deal more enhancement than in any system that hides the extra information within the NTSC signals.[5] Of course, we could use those techniques as well for even further enhancement, but have chosen not to for two reasons. One is for improved receiver compatibility and better performance in degraded channels. The other is to preserve the family relationship with our noncompatible HDTV version. The visual effect of the usurped area is discussed below.

The principles that follow are used for all of the components in the noncompatible version, and for the enhancement components in the compatible version.

Better utilization of the analog channel. The 6-MHz analog channel should allow the transmission of 12 million active video and audio samples/sec, but NTSC allows little more than half this amount. To improve this performance, we eliminate the separate sound carrier, the retrace intervals, and the use of vestigial-sideband transmission. Instead, we use two 3-MHz baseband signals, quadrature modulated on a single carrier in the center of the band. The various components are time multiplexed, reserving about 1/12 of the time for audio and data in the noncompatible system.

Use of frame store with separate scanning standards for camera, channel, and display. A high line-rate progressively scanned camera and display permit very high vertical resolution without causing interline flicker. In the compatible version, some experimentation on this point is required to avoid excessive flicker on interlaced displays.

Higher spatial and lower temporal resolution. Film is the standard of quality for ATV, although the frame rate is only 24 fps vs 60 for TV. Today's 60 fps, chosen primarily on account of large-area flicker considerations, is a poor choice when a flicker-free display is used. For lay audiences, it is probably more important to improve the spatial resolution than the motion rendition. The biggest advantage, however, in using frame rates based on 24 fps, rather than 30 fps, is the much improved film interface. Finally, the use of motion compensation in the receiver will eventually permit very smooth motion rendition at these lower frame rates. In our laboratory, we have gotten good motion even from simulated 12-fps film. [9]

Reduced relative diagonal resolution. We are less sure that this technique really works, as our audience tests indicate no special preference for diagonal sampling. [10] However, even making the diagonal resolution equal to the vertical and horizontal resolution, rather than 41% higher, saves substantial bandwidth.

Relatively lower frame rate for high spatial frequencies. This principle is used in MUSE and the Glenn, Philips, and Sarnoff systems. It does seem to result in some blurring of objects moving slowly and steadily enough to be tracked, but not much more than caused by camera integration. Together with the previous principle, this implies that the spatiotemporal frequency response should be approximately diamond-shaped in cross section, as shown in Fig. 1.

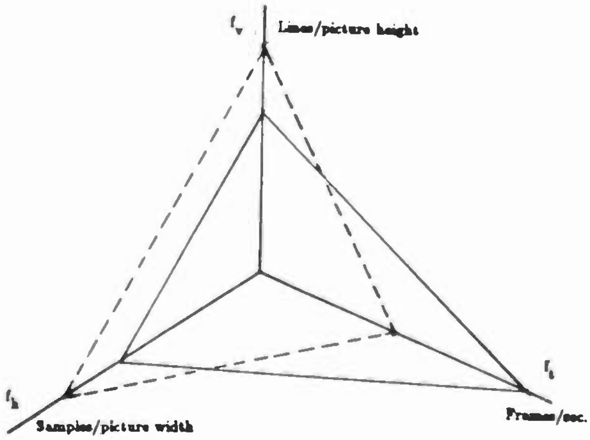


Fig. 1. Idealized Spatiotemporal Frequency Response
Psychophysical data indicate that the overall response should have approximately the shape shown here, for most effective use of channel capacity. The volume is proportional to the required channel capacity. We show two cases, using different tradeoffs between spatial and temporal resolution, as would be appropriate with different amounts of motion in the scene in front of the camera.

Scene-adaptive variable frame rate. Any particular frame rate represents a tradeoff between spatial and temporal resolution. Although we believe that 24 fps[6] is a good choice for fixed frame-rate systems, the smart receiver permits the selection, at the transmitter, of an optimum frame rate for each scene depending on the amount of motion. We have chosen to use a small set of rates that are multiples of 12 fps.

Relatively lower frame rate for chrominance. Psychophysical data indicate that the visual bandwidth for chrominance is about half that for luminance, and we have utilized this for additional savings.[7] This seems to be the one aspect in which NTSC is actually overdesigned; the temporal resolution of color information is higher than needed! In the noncompatible system, we have simply transmitted color information at 12 Hz. In the compatible version, we have added extra information to the chrominance channel in the space that would normally be devoted to chrominance high temporal frequencies. The added data can be used either to extend the horizontal chrominance resolution or for some other purpose. In one method, horizontal subsampling method is used, as in MUSE or Iredale's proposal [11], while in the other, extra data is added to chrominance using opposite polarity on successive frames. The resultant 15-Hz flicker is not seen on NTSC receivers because this rate is higher than the critical flicker frequency for color.[7] There is good precedent for this. Color NTSC itself has only one-half the alias-free temporal bandwith than its monochrome predecessor.

All of these improvements together result in a 5- or 6-fold increase in the perceived number of picture samples per frame in the stationary areas as compared with NTSC, giving a spatial resolution comparable to MUSE. For the receiver-compatible version, using 25% of the image area permits a 38% enhancement of the remaining 16:9 frame. Further improvements in the noncompatible version are obtained with the following two methods.

Mixed-highs color representation. An alternative to the luminance/chrominance (wideband/narrowband) representation used in virtually all existing and proposed systems is isotropic narrow-band RGB (color 'lows') plus luminance 'highs.' [12] Actually, most two-channel systems use some form of this scheme, which has several important advantages. The highs signal requires a lower SNR than the lows. Additional bandwidth can readily be used for enhancement. This facilitates using different frame rates for the different spatial-frequency components. Most important of all, it provides a way to suppress channel noise.

Improvement in signal-to-noise ratio. Although 'snow' is a TV defect widely recognized by viewers, and contributes to loss of quality in an obvious manner, no one, to our knowledge, has yet proposed improving image quality in the home by raising the SNR. It is quite clear that, at least in the US, the link between transmitter and receiver, in terrestrial broadcasting as well as cable, is one of the weakest elements in NTSC. We suspect that most lay viewers, shown noise-free pictures on a studio monitor, would call that 'HDTV.' It may well be that SNR is the most important factor in picture quality as perceived by lay viewers. The mixed-highs system permits the use of adaptive modulation, raising the signal level where it is small (in the blank areas where it is also most visible) and reducing it, along with the channel noise, to a corresponding degree at the receiver. Adaptation information is transmitted, being considered part of the signal description. Our experiments with both quantized AM [13] and with FM [14] indicate that a 10- to 12-db improvement in perceived SNR can be achieved in typical cases.

The Smart Receiver

A possible configuration of the smart receiver is shown in Fig. 2, where it is seen to consist of 3 sections. The input section comprises the tuner, RF and IF amplifiers (not shown), detector, analog/digital converter, and first frame store. This section is tunable, but not programmable. For a wide variety of input formats, it maintains in the store a digital version of a complete 'frame' as transmitted, but not in the form required for display. Interfaces are provided for rf and baseband analog signals from other signal sources.

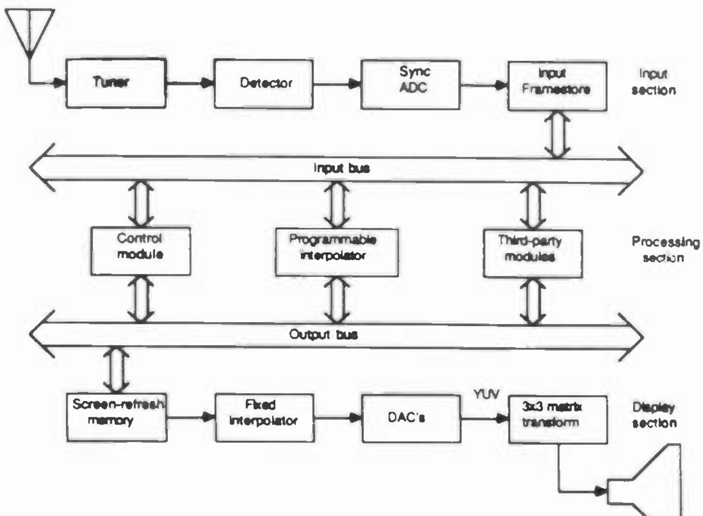


Fig. 2. The Smart Receiver
The input and display sections are fixed, while the processing section, organized like a personal computer, is programmable under the control of a small amount of digital data transmitted along with the signal. With an open architecture, this section could be upgraded by adding or exchanging modules, some of which could be offered by third parties. A more advanced concept would place the detector in the processing section, digitizing the input at some low intermediate frequency. This would permit programmable detection of signals with multiple carriers, such as used in the Philips or Sarnoff systems. Other configurations are possible in the display unit, which probably would use mixed highs or luminance/chrominance representation, rather than RGB.

The display section is likewise not programmable. It holds image data in the screen-refresh memory in correct geometrical arrangment, at a standard frame and line rate, (not necessarily that transmitted or displayed) and in some slightly compressed form, such as mixed highs or luminance/chrominance. Interpolation circuitry, a digital/analog converter, and some minor processing circuitry are used to get a high line- and frame-rate RGB signal for display. All receivers need not have the same display standard, but all kinds of signals received would be displayed at the same standard in any one receiver. A minimum would probably be 1200 lines, 60 fps progressively scanned, but higher would be better.

The center section does the digital signal processing. Well defined input and output busses provide the interfaces between the processor and the input and display sections, and also receive add-on third-party modules for enhancement or for interfacing with digital signal sources, such as fiber-optic transmission lines. A typical function of the processor is receiving information from the input frame store, then rearranging and interpolating it to the standard rates required in the display memory. It does this under supervision of the control module, which is programmed by a small amount of data transmitted along with the signal, or possible by manual viewer control when using other input sources.

While this arrangement may look more like a computer than a TV receiver, it should be noted that the input and display sections, which surely will account for an overwhelming portion of the cost, are required in almost all HDTV receivers such as those for MUSE or the Sarnoff system. The programmable processing section consists exclusively of low-level digital hardware which is bound to be very cheap in the quantities required for this application.[8] The ease (and cost) of programming such hardware, compared, for example, with changing the sweep rates on the display for PAL/NTSC adaptation, as required in today's dual-standard receivers, is quite low and getting lower.

Still greater flexibility would be provided by placing the A/D converter before the detector, and doing the detection digitally. This would facilitate handling various kinds of analog modulation schemes, such as those with multiple subcarriers, although it would not absolutely be needed for the system described in this paper.

A programmable receiver of this type would make it rather easy to adapt the decoding parameters to many different kinds of TV transmissions, including NTSC. It would permit improvements, such as motion-compensated temporal interpolation, echo cancellation, and noise reduction, to be added at a later date. The open architecture would also allow third parties to design plug-in software or hardware modules to give special capabilities, such as interactivity and connection to a wide variety of peripherals. These might include computers, cable, VCR's, video games, and, perhaps most important of all, electronic still photographic equipment.

An Example of a Noncompatible System

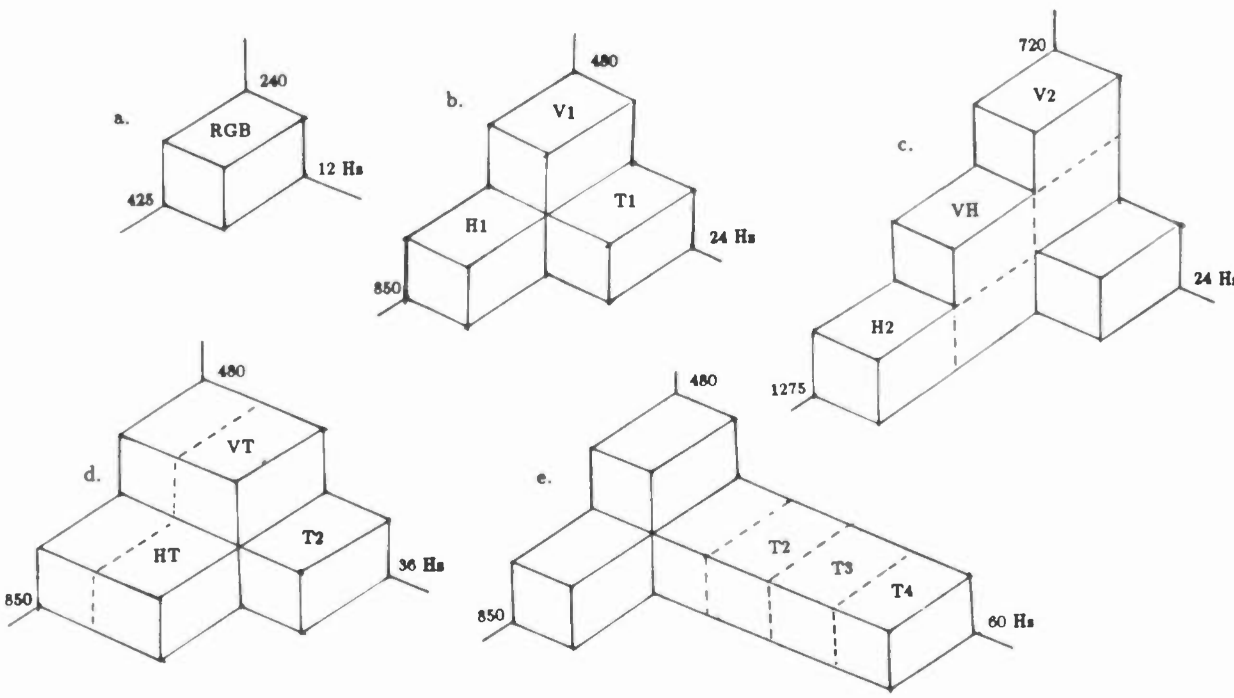
In this section, an illustrative example is given of a channel-compatible 6-MHz system designed in accordance with the principles discussed previously. It is not meant to be a final design - that must await further subjective testing with a variety of subject matter - but we believe it is a reasonable selection of parameters.

To simplify the calculation, we assume that each Hz of bandwidth is equivalent to two signal samples. Thus, the 6-MHz channel bandwidth is equivalent to 12x10^6 samples/sec., of which 1/12 are reserved for audio and data in any convenient format. The spatiotemporal frequency space, shown in Fig. 3, is divided into blocks of equal size, 240 pels/picture height high, 425 pels/picture width wide, and 12 samples/second along the temporal frequency axis. The blocks have spatial shapes of 16:9 with equal horizontal and vertical resolution, and nine blocks can be transmitted in 11x10^6 samples/sec.

In this system, there are three different selections of blocks transmitted, according to whether the motion is slow, medium, or fast. The choice is made manually at the studio and indicated to the receiver by a digital code word. In all cases, the effect is to give a coarse approximation to a diamond-shaped frequency response in all directions in 3-d frequency space.

Motion	Components Utilized
Slow	R, G, B, T1, V1, H1, V2, H2, VH
Medium	R, G, B, T1, V1, H1, VT, HT, T2
Fast	R, G, B, T1, V1, H1, T2, T3, T4

Note that in this arrangement, the RGB 'lows' signal is always 240x425, 12 fps. All the other components are luminance 'highs,' free of dc, and can be adaptively modulated for channel noise suppression.[9] (Alternatively, they can be reduced in amplitude and placed 'over' an extra digital signal.) For the slow-motion case, the stationary-area (0 to 12 Hz) spatial resolution is 720x1275, and the moving area (12 to 24 Hz) resolution is 240x425. For medium motion, the 0 to 24 Hz resolution is 480x850, and the 24 to 36 Hz resolution is 240x425. For the fastest motion, the 0 to 12 Hz resolution is 480x850, while 240x425 resolution is maintained up to 60 fps. For comparison, MUSE performance is shown in Fig. 4.



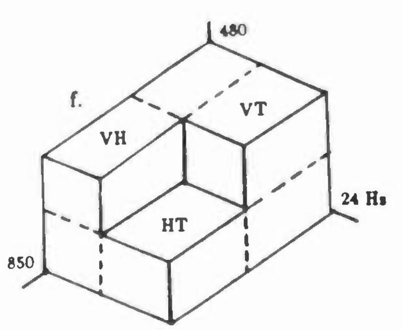


Fig. 3. Components of the Example System in Frequency Space
The components transmitted in the several modes of the system are shown. The three RGB "lows" signals in (a) are always transmitted, as are V1, H1, and T1, the first three spatiotemporal "highs" components, in (b). In scenes with little motion, the three spatial components V2, VH, and H2, are used, as in (c), while scenes with medium degrees of motion add both spatial and temporal components VT, HT, and T2, shown in (d). The fastest motion makes use of the temporal components T2, T3, and T4, shown in (e), to give information at 60 fps. (Note that to use this mode, the production system and the display must both operate at 60 fps or higher.) In the case of 24 fps film as input, the low-speed mode is (c) while the high-speed mode is (f). By this means, a superior overall performance is expected as compared with the use of a single set of components regardless of the motion.

A particularly simple and straightforward multiplexing scheme is shown in Fig. 5. The signal from the production system is digitized, divided into components by quadrature mirror filters, [15] and buffered. The nine components to be used are selected and divided into odd and even lines and time-multiplexed into two 3-MHz signals. When read out by a 6-MHz clock, the 425 samples per scan line require 70.83 microsec for transmission through each such channel. Multiplexing is on a line-sequential basis, with one line from each component read out at a time. After 11 blocks of 9 lines each, one block (425x9 or 3825 samples) of audio/data is transmitted. After 1/12 sec., 120 scan lines will have been transmitted in each block in each 3-MHz channel, or 240 lines for the entire signal. A very small amount of each line duration plus one or two lines per frame are devoted to synchronization signals.[10]

In this arrangement, errors in quadrature demodulation due to carrier phase errors cause only a small loss in vertical resolution, since the two signals in quadrature relate to successive lines in the picture. The effect of multipath transmission is minimized by using the slowest possible horizontal scan rate of the transmitted signal, which places any echoes as close as possible to the main signal.

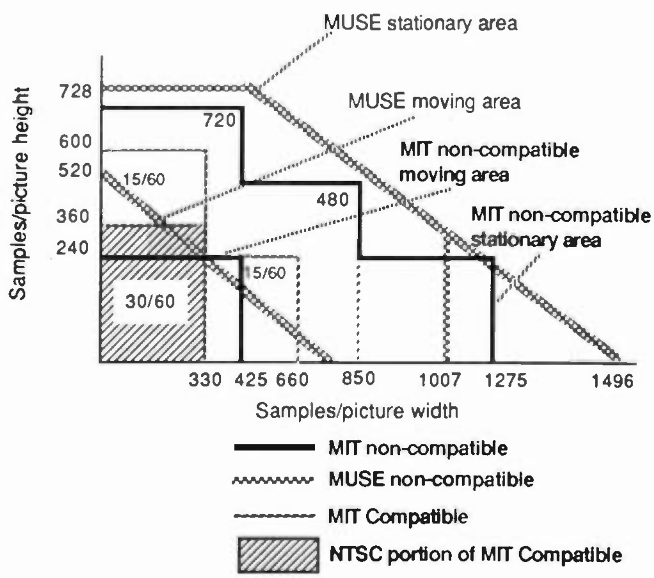


Fig. 4. The Frequency Responses Compared with MUSE
For comparison, the luminance frequency responses of both systems are shown in comparison with MUSE. 'Moving area' information is rendered at 30 Hz, while 'fixed area' information is transmitted at 15 Hz in MUSE and the compatible system. The rates are 24 Hz and 12 Hz, respectively, in the noncompatible system. Offset sampling of the original wideband signal, preceded by a diamond-shaped prefilter, sets the diagonal response, while the bandwidth sets the horizontal response. The use of an 1125-line interlaced display limits the perceived vertical resolution of stationary-area luminance to about 728 lines, but this does not affect the vertical response of the other components. There is some uncertainly about these calculations because MUSE signal processing has not been completely described in the various publications. Note that the response shown for the noncompatible system is that of the mode used with the least motion. The other modes have higher temporal and lower spatial bandwidths.

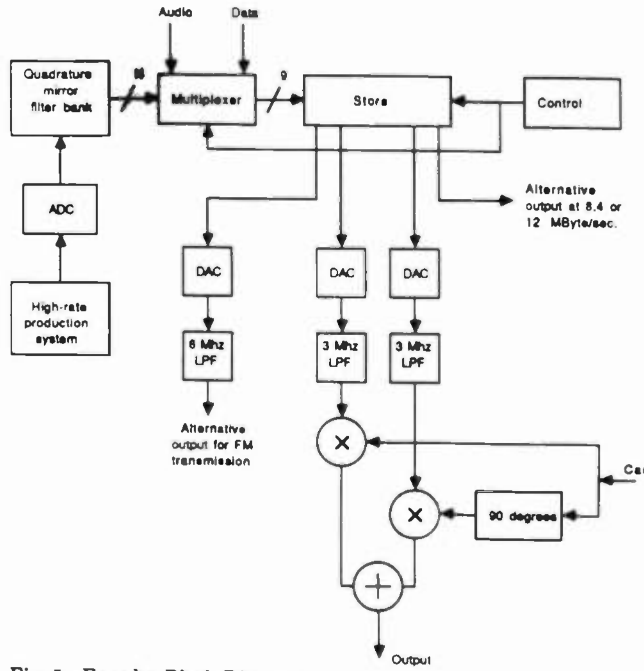


Fig. 5. Encoder Block Diagram
An example of how an encoder could be implemented is shown here. The output of a high-rate production system is digitized and divided into N (more than 9) components by a quadrature-mirror filter bank. The 9 desired components for the mode in use are selected and multiplexed, along with digital audio and data, and then buffered. Data is read out of storage in two streams, corresponding to odd and even lines, converted to analog form, filtered, and quadrature-modulated onto the carrier. The q-m filter bank is the most complex element. For digital transmission at 8.4 or 12 Mbytes/sec, a digital output is taken directly from the store. For FM transmission, a single 6-MHz bandwidth analog signal is derived (the same as the 12 Mbytes/sec. output converted to analog form) and applied to the adaptive frequency modulator.

An Example of a Receiver-Compatible 6-MHz System

Usurping 25% of the image height and replacing the normal video with enhancement information results in substantial improvement in spatial resolution. (This is given by way of example only; naturally a smaller usurpation could be made with less enhancement.) Displaying the result on an upconverted progressively scanned CRT gives further improvement. In this interval, the extra information is transmitted in the same manner as in the noncompatible example, in order to preserve the family relationship between the two versions. For that reason, the receiver here can be thought of as decoding NTSC for 75% of the time and HDTV for 25%, while the receiver for the noncompatible case displays HDTV 100% of the time. It is the same receiver, and, of course, it can be used for normal NTSC as well.

Since the sound carrier cannot be interrupted during the top and bottom bars, only about 5.5 MHz is left for enhancement information, providing a nominal channel capacity of 11 million samples/sec for about 60 microsec each line - a total data rate of 330x240 elements per 1/30 sec. Using fourth-order interlace to reduce the frame rate of enhancement information to 15 fps, we can have two 330x240 enhancement blocks per frame, which are used for increasing the vertical and horizontal resolution.

To simplify adding the extra information, NTSC chrominance and luminance are filtered one-dimensionally to 63 and 330 samples per line respectively. The luminance resolution in the ATV receiver is thus extended to 660 samples per line horizontally for the lower vertical frequencies and to 600 lines per frame for the lower horizontal frequencies, as shown in Fig. 4. The horizontal chrominance resolution is doubled by the subsampling method to 126 samples per line. Note that the vertical chrominance resolution is 360 lines per frame, which is higher than needed. Conceivably some additional technique could be used to exchange this excess resolution for some more useful information.

The Appearance of the Bars

It should be borne in mind that two TV systems using different aspect ratios are inherently incompatible. There is *no* way to display a wide-screen image on a 4:3 crt without some damage. In some cases, especially where the director has thought about it from the beginning, the edge areas can be left mostly empty, so that the reframing does not cause a visible problem. Of course, in that case, he cannot take advantage of the full visual effect of the original shape. Top and bottom bars are routinely used in most countries other than the US and UK when widescreen movies are shown on TV, and this has proven quite acceptable. For the *Cinemascope* 2.25:1 aspect ratio, these bars are 40% of the nominal picture height as shown in Fig. 6. In this country, the screen is usually filled, requiring the a substantial fraction of the image width to be discarded. The 'pan and scan' method is often used to minimize the damage to the

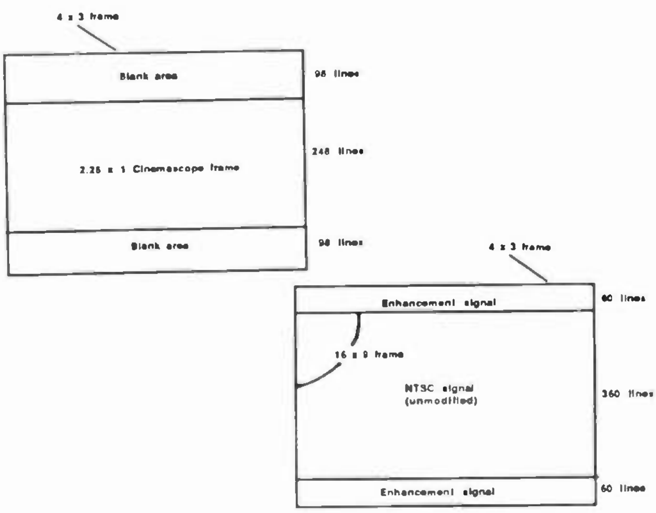


Fig. 6. The Appearance of the Screen on the NTSC Receiver
In the receiver-compatible system, 60 lines are used at both the top and bottom of the picture to make room for enhancement information. This leaves a 16x9 area for the image, which therefore has the same aspect ratio as that on the special receiver. For comparison, the appearance of the screen is shown when widescreen movies (2.25x1) are used on TV in most countries other that the US and UK.

directors' intention.[11] Our initial work with focus groups has shown that the audience resents the loss of the side panels, if they know about it, more than they are bother by the presence of the bars, if the latter are black. In that case, the bars were black, whereas in this case, we have to deal with data in the bar area.

For receivers with keyed clamps, the bars can be made black by using a white pulse on the back porch. For new NTSC receivers, the bars can be made black at negligible cost with simple circuitry. For a few dollars, the viewer could be provided the capability of expanding the sweep to fill the screen, if he wanted to. Receivers connected to an external tuner could be provided a low-cost adaptor. Finally, the signal structure can be organized in such a way that the bars contain fine-grain random patterns, for the most part.

There is no perfect solution to this problem. The one we have adopted does cause a visible change to the appearance of the screen, but in the image area, at least, all receivers would work perfectly. We have eliminated the necessity of cropping, since the two systems have the same aspect ratio, and we have thereby provided enough space for a large degree of enhancement. With appropriate marketing techniques, the bars might even give an incentive to buy new receivers without actually depriving anyone of the use of his unmodified set.

Conclusions

We have presented a family of ATV systems featuring improved channel efficiency through the use of double-sideband quadrature modulation and a selection of image information more in accord with human perception. We have given two examples of channel-compatible systems that can be implemented with these methods. One is receiver-compatible EDTV, offering substantially improved spatial resolution on special receivers, while old receivers would show a 16:9 image with top and bottom bars. The other is a true HDTV system, with image quality similar to MUSE. The noncompatible version is suitable for cable or other controlled-access medium immediately. It also features much better performance than any existing television system when the channel SNR is low. The same 'smart' receiver would be used for both systems, thus supporting the 2-step strategy to move to HDTV in terrestrial broadcasting.

Acknowledgement

The work reported here was supported by the members of the Center for Advanced Television Studies, and contributed to by many students and colleagues. Their help is gratefully acknowledged.

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[1] This was the method used in France and Britain when PAL was introduced. Some service was provided (by government stations) to old receivers for about 20 years.

[2] The very useful terms 'channel compatibility' and 'receiver compatibility' were coined by the FCC staff in connection with the current Inquiry. [1] In this paper, the single word 'compatible' means 'receiver compatible.'

[3] There is a lot we do not know about motion rendition, for example.

[4] Recently, NHK has announced compatible versions of MUSE, but, as of this date, they have not been described or demonstrated.

[5] The Sarnoff Laboratories' ACTV system [3] is of this type, as are the Matsushita [4] and Fukinuki [5] systems. In the Glenn [6], Philips [7], and AT&T systems [8], the enhancement information is in a second channel.

[6] Actually, that rate was chosen for the sake of the *sound* quality when audio was added to film!

[7] This rate is *below* the critical flicker frequency for luminance; using either of these methods in that case causes a very perceptible artifact.

[8] If such receivers become common, they will provide the largest market for many kinds of chips, including memory and digital signal processing circuits. Nearly any chip, even a very complicated one, made in the *millions*, would become cheap.

[9] Some additional noise reduction is possible by dividing the 'lows' signal into a very low-frequency component plus the balance, which then becomes a 'highs' signal that can also be adaptively modulated.

[10] Special signals, now using one or more lines of the vertical retrace interval, such as SMPTE time code and closed captions, could be incorporated into the data channel.

[11] For directors who care, such as Woody Allen, this surgery in anathema.

FINAL REPORT: THE MULTI DEPRESSED COLLECTOR KLYSTRON PROJECT

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INTRODUCTION

This program, involving the incorporation of depressed collector technology into UHF-TV klystrons, was initiated in June 1984. The objective of this program is to develop new klystron technology which will significantly improve transmitter efficiency. Support for the program has been provided by a cooperative group, including NASA, NAB, PBS, transmitter manufacturers, and Varian Associates, while interim reports have described the progress for this program [1-5]. This past year Varian evaluated an experimental model, using a 60 kW internal cavity klystron as a test vehicle. This klystron performance met expectations, and the program objectives have been accomplished. This paper describes the performance achieved and also the follow-on efforts at Varian to implement depressed collector technology for the UHF-TV community.

PROGRAM ACCOMPLISHMENTS

The Varian VKP-7555S klystron with five internal cavities was selected as the test vehicle for the multistage depressed collector (MSDC) performance evaluation. This klystron provides 60 kW peak sync power at the high band TV frequencies of 700 to 850 MHz. Figure 1 is a photograph of the completed tube. The collector, at the top of the

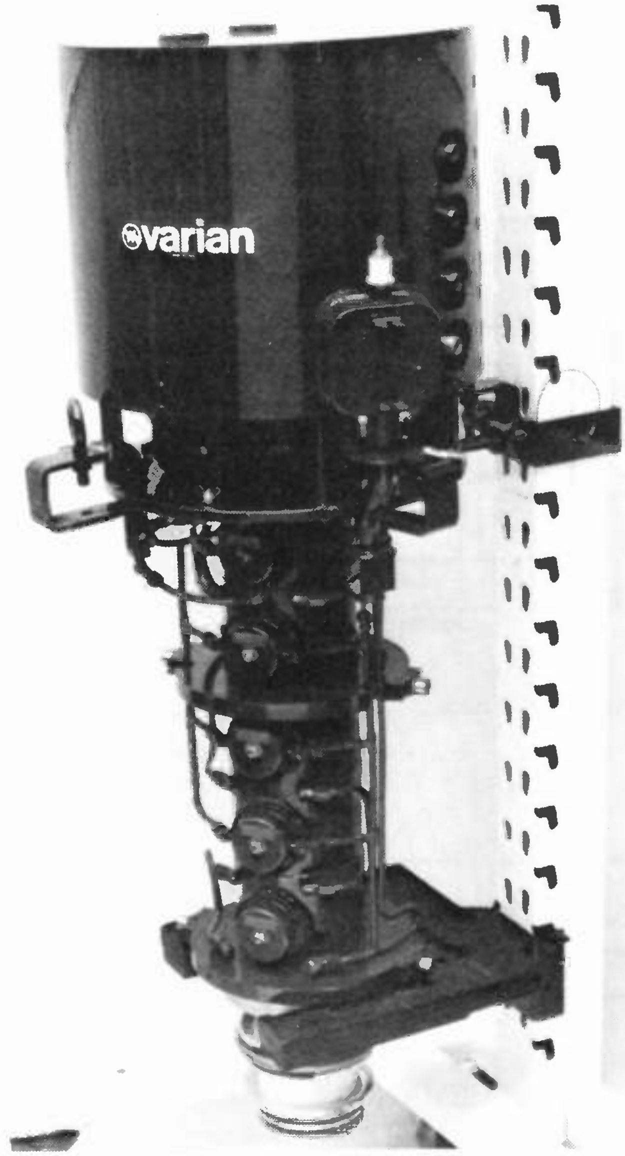

Fig. 1. DEPRESSED COLLECTOR KLYSTRON

picture, is enclosed in a metal shield to minimize rf interference. It is composed of five elements mounted between ceramic rings for electrical insulation. Each electrode contains passages for water cooling; the coolant connectors are visible, protruding through the collector shield. Inserted between the klystron and the collector is a refocusing electromagnet that adjusts the electron beam as it enters the collector.

Testing was accomplished in a pulsed mode to allow an evaluation over a wide range of operating conditions. Power supplies to operate this klystron can use either a parallel or series configuration as indicated in Figure 2. Although the testing was conducted in a parallel arrangement, the results are presented for both configurations.

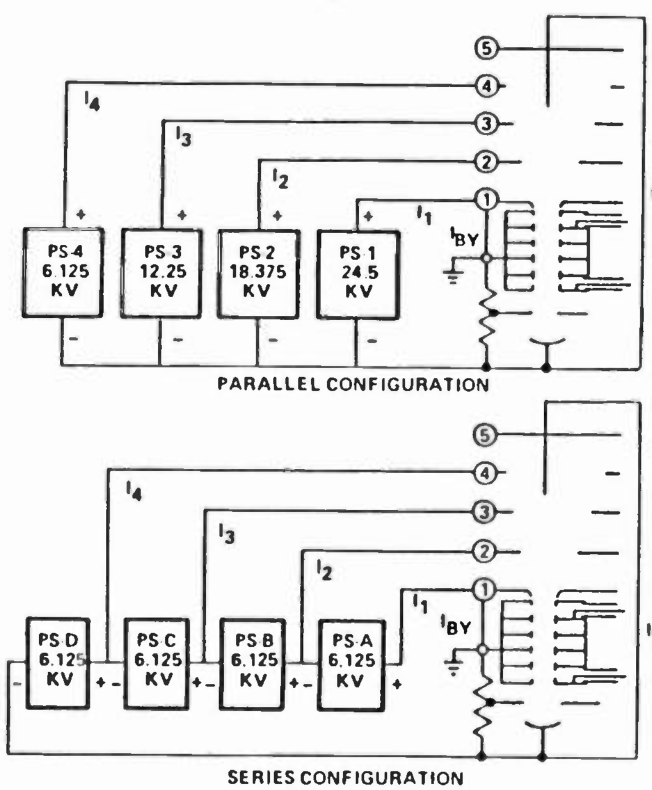

Figure 2. MSDC POWER SUPPLY SCHEMATICS

The tube performance evaluation began with determining the characteristics of the beam refocusing magnet. Adjustment of the refocusing coil current optimized the power recovery. The actual performance was very close to the expected value. The coil had only a half ohm resistance, so less than 20 watts of power was required for the normal operating current in the range of 4 to 6 amperes.

Measurements of current distribution to the collector elements were performed as a function of rf drive power. The results for the normal operating condi-

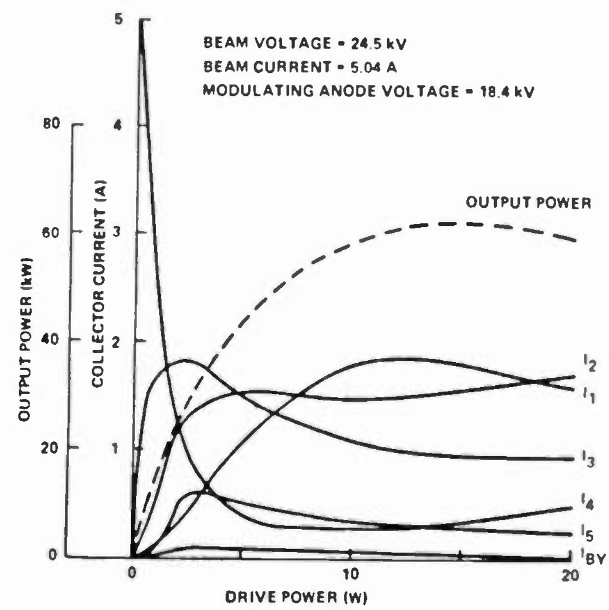

Figure 3. MSDC CURRENT DISTRIBUTION

tions are shown in Figure 3. For no rf drive, essentially all the beam current goes to electrode 4, but as the drive increases, I4 drops rapidly as I_3 increases, followed by I_2, then I_1. It is interesting to note that up to 10% of the current goes to element 5, which is at cathode potential and, consequently, does not involve power supply power.

Particular attention was given to determine if the MSDC had degraded the klystron performance in any way. Bandpass curves were measured, as in Figure 4, to represent normal operating conditions.

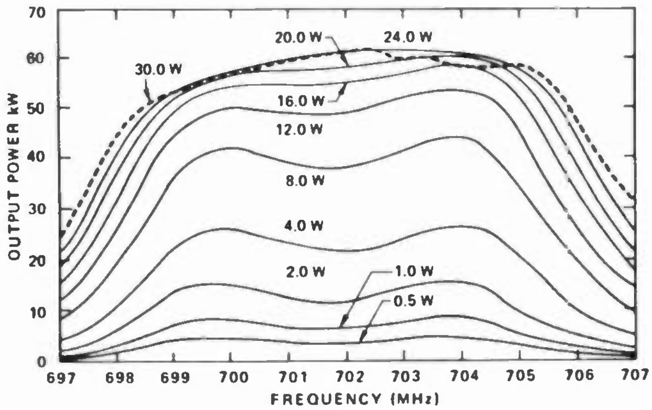

Figure 4. BANDPASS CHARACTERISTICS

Performance was normal with no disturbance to gain and power.

From the measured performance characteristics, the klystron efficiency was determined and plotted as a function of rf output power in Figure 5. The measured performance compares closely with previously calculated values, supporting the accuracy of our computer modelling technique.

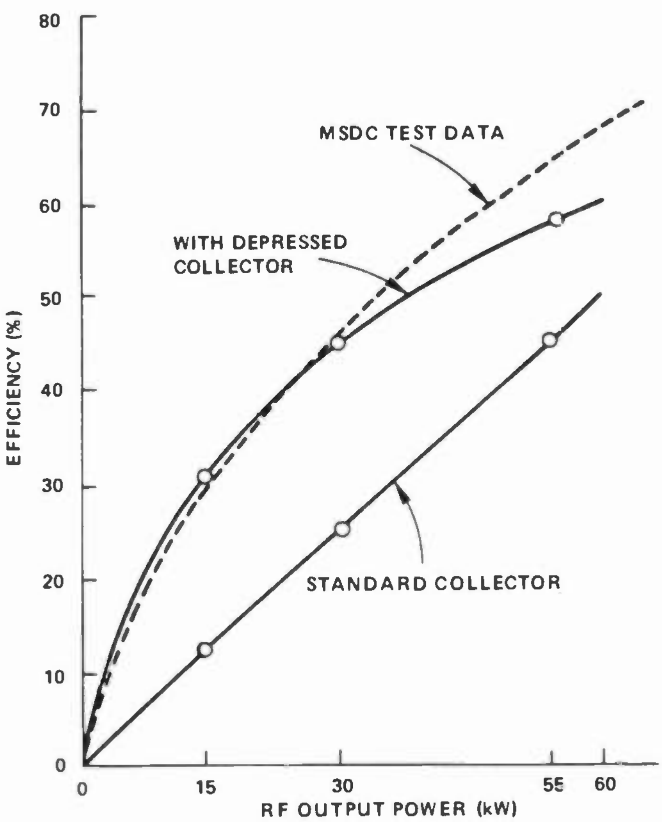

Figure 5. OVERALL KLYSTRON EFFICIENCY

Performance measurements were also taken for reduced beam current operation to determine the performance for beam pulsed operation. The current distribution for the reduced beam current case, shown in Figure 6 shows little difference from the full current case of Figure 3. These data, however, allow us to calculate the current and power requirements for the power supply configurations of Figure 2. For beam pulsed operation, the current and power values for each power supply are listed in Table 1. Note that the total input power for an average (gray) picture is 41.79 kW, which represents a 1.316 Figure of Merit (the ratio of peak sync rf output power to dc input power).

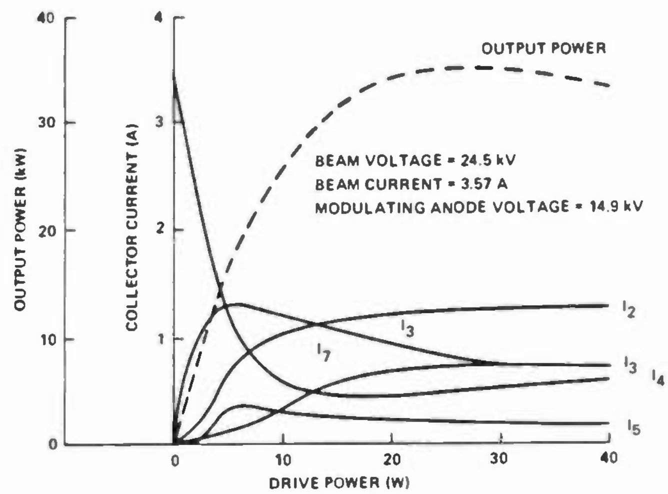

Figure 6. CURRENT DISTRIBUTIONFOR REDUCED BEAM CURRENT

Successful demonstration of MSDC performance marked the completion of the development program.

ADDITIONAL MSDC ACCOMPLISHMENTS

Varian has applied the MSDC technology to other UHF-TV klystrons. A klystron with four external cavities (similar to the TEV PT-5090) was selected

PARALLEL CASE

POWER SUPPLY			FOR TV SERVICE: BLACK PICTURE	AVERAGE PICTURE	WHITE PICTURE	
1	24.5 KV	CURRENT	0.573	0.331	0.279	A
		POWER	14.020	8.100	6.840	kW
2	18.375 kV	CURRENT	1.144	0.565	0.367	A
		POWER	21.030	10.390	6.750	kW
3	12.25 kV	CURRENT	1.150	1.121	0.482	A
		POWER	14.090	13.730	5.910	kW
4	6.125 kV	CURRENT	0.532	1.558	2.496	A
		POWER	3.260	9.540	15.290	kW
TOTAL INPUT POWER			52.400	41.760	34.790	kW

SERIES CASE

POWER SUPPLY			FOR TV SERVICE: BLACK PICTURE	AVERAGE PICTURE	WHITE PICTURE	
A	6.125 kV	CURRENT	0.573	0.331	0.279	A
		POWER	3.51	2.03	1.71	kW
B	6.125 kV	CURRENT	1.718	0.898	0.648	A
		POWER	10.52	5.50	3.97	kW
C	6.125 kV	CURRENT	2.869	2.018	1.131	A
		POWER	17.57	12.36	6.93	kW
D	6.125 kV	CURRENT	3.401	3.576	3.628	A
		POWER	20.83	21.90	22.22	kW
TOTAL INPUT POWER			52.43	41.79	34.83	kW

FIGURE OF MERIT = 55 / 41.76 = 1.316

Table 1: PERFORMANCE WITH BEAM PULSING

for the depressed collector technology. This design has been designated the VKP-7990. The first tube was completed and evaluated, and a second tube is in construction. Performance of the VKP-7990 compares well with the data for the experimental klystron described above. The external cavity klystron has somewhat lower interaction efficiency (45% versus 51%) while the collector power recovery is nearly identical. Consequently, the resulting Figure of Merit for the beam pulsing operating mode is slightly lower (1.250 versus 1.316).

CONCLUSIONS

Test results for the MSDC klystron are very encouraging. Power recovery compares closely with the predicted values, and overall performance meets our expectations. Developmental models are available to transmitter manufacturers for evaluation in TV service. Successful performance in television transmitter service will lead to new klystron transmitters with efficiency values well beyond what is presently available.

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UPDATING OLDER GENERATION UHF TELEVISION TRANSMITTERS TO CURRENT PERFORMANCE STANDARDS

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ABSTRACT

A broadcaster may readily upgrade his transmitter to increase performance, reliability, and transmitter output power by planning and implementing a staged, systematic, modernization schedule. This staged process will benefit broadcasters who are not able to replace their older UHF transmitter because of budget constraints.

This paper discusses modifications which have been accomplished to successfully update older generation UHF television transmitters to current state-of-the-art performance.

The positive and negative aspects of each area of modification and their relative complexity of implementation are discussed. Costs are analyzed to determine when it makes good engineering sense to update older transmission systems.

INTRODUCTION

The UHF television transmitter is one of the most costly components making up a UHF television station. Not only is the initial capital investment for transmission equipment high, but the UHF broadcaster must also pay increased operating and maintenance costs as compared to a VHF station. For instance, the cost of a klystron replacement can have an impact on an operating budget that has no parallel in a VHF station. Lost air time due to equipment failure is a major problem to many UHF broadcasters as older transmitters did not incorporate redundant systems.

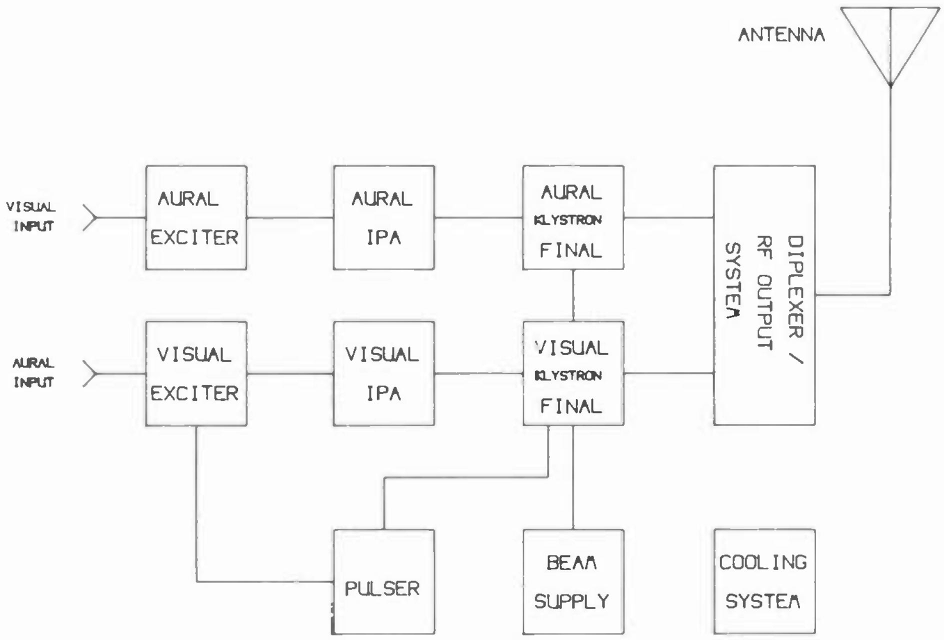
Newer generation UHF transmitters incorporate many features as standard, or as low cost options, which were not available in the early 70's design. Unitized high voltage power supplies and advances in cooling systems, klystrons, RF output, and exciter systems have greatly improved UHF transmitter reliability and efficiency.

The older generation transmission systems (pre-'75) all tend to have undersized cooling systems, beam power supplies using utility type power transformers, unstable exciters, troublesome RF combiners, weak antenna systems, and low output power (30 kW). These items cause the systems to be unreliable, inefficient, and/or present safety hazards. As power costs were low at that time, transmitter efficiency was not a major concern. The visual klystrons were limited to an output power of 30 kW. As a result, most UHF transmitters of that era were configured as single-ended 30 kW stations. A station desiring to run higher power used two 30 kW amplifiers in parallel.

North Carolina Public Television (NCPT) operates 8 UHF and 2 VHF television transmitter sites throughout the state. Much of the transmission network is well over 20 years old. The first transmitter, a VHF, went on the air in 1954 and the oldest UHF transmitter went into operation in 1967. Both of the transmitters are still in daily use. The majority of the other UHF stations were constructed in the early 70's. In the 80's it became very apparent that major changes had to occur to upgrade or replace our transmission facilities if we were to continue our broadcast service.

Our UHF transmitters proved to be the most troublesome to maintain. Age had taken its toll and the technology was outdated. However, serious budget constraints prohibited us from replacing these transmitters. We started to look for ways to extend the life of the transmitters by increasing their reliability.

By upgrading specific subsystems in the transmitter, NCPT was able to drastically improve the overall system performance without needing to replace the complete



TYPICAL UHF TELEVISION TRANSMITTER BLOCK DIAGRAM

transmitter. This saved money and extended the useful life of each transmitter. NCPT has budgeted to continue this upgrade program on all of our older UHF transmitters.

As we upgraded the various subsystems, we have found that it is relatively easy for the older transmitters to approach the performance standards found in today's new transmitters. In fact, we believe our upgraded transmitters to be as good as many of the new models on the market today.

Although most of our work relates specifically to General Electric external cavity, water cooled, UHF transmitters, the similarity of subsystems of other transmitters make these modifications generic to many older transmitters.

REASONS FOR UPGRADING

Today's new transmitters incorporate many refinements not available in earlier designs. Redundancy, reliability and efficiency are the current buzz words which describe any new transmitter. Transmitters are also becoming more space efficient thanks to a general advancement in overall technology. The only tubes used in the modern UHF transmitter are the high power klystrons in the final output stage. Even that mainstay is being challenged by the klystrode and other developing technologies. The next generation of UHF transmitters will offer advances in efficiency and overall performance that the broadcaster never before thought possible. Because the new technology is still developing, it may be advisable for the broadcaster to delay the purchase of a new transmitter until the technology stabilizes.

Most UHF stations today fall into one of two distinct categories:

- First, the large, wide area stations who are directly competing with the maximum service VHF broadcasters. These UHF stations usually are operating FCC/FAA maximum facilities for their location, (maximum antenna height and effective radiated power) to gain the very best advantage in the market. Transmitter power is typically 120 kW with 240 kW becoming the norm. These stations usually have made the commitment to a large transmission plant and would not be especially interested in upgrading older transmitters, but many of the techniques discussed may be applicable to large modern transmission plants.

- Secondly, the group of UHF stations operating in a more limited environment. They are generally independent stations or public television stations. Capital expenditures are usually limited and large dollar purchases are few and far between. Many of these stations operate older transmission facilities dating back to the late 60's. Transmitter output power is typically 30 kW and they operate from a less than optimal antenna location and height. Effective radiated power is usually less than 1 megawatt for omnidirectional stations. Most of these stations are struggling to reach their audience. Cash flow can support keeping the transmitter on the air, however, large expenditures for new transmitters are not a station priority unless the transmitter is suffering from severe reliability problems which have caused a significant amount of lost air time. These are the stations who would tend to benefit most from a UHF transmitter retrofit program.

FCC CONSIDERATIONS

Recent deregulation has made it easier to improve an older television transmitter without being overburdened with paper work.

Exciters, RF combining, and klystron changes used to be a real problem due to a once strict FCC policy on maintaining the integrity of the original transmitter. Present regulations allow almost any type of modification to the transmitter as long as certain criteria are met. (Ref. 1.) The only time the actual license will have to be changed is when you wish to increase transmitter power output.

Upgraded sections of an older transmitter can operate reliably in an emergency multiplex condition with little degradation in performance. For operation in a full time, single tube, multiplex configuration, RF filters may be needed to reduce out of band emissions at +9 MHz and -4.5 MHz, (referenced to visual carrier) which are generated in this configuration due to nonlinear mixing in the klystron. Additional linearization in the exciter may also be required to obtain the highest video performance.

STEPS IN UPGRADING

High Voltage Contactor Replacement

Many older transmitters were designed around the General Electric AK2-25 mechanical contactor/trip device. This contactor switches the primary voltage to the klystron high voltage power supply. The most serious problem with these units is the failure to engage or disengage reliably and being prone to false overcurrent trips.

Many stations have been off-the-air frequently and for extended periods of time due to failures of this type of contactor. Unfortunately, one of the more common failure modes can also cause klystron failure.

A change to a new generation contactor is straight forward. A plug-in replacement for the AK2-25 contactor is available. Other transmitter manufacturers may supply a contactor that is not a direct replacement, and requires some modifications for for installation, but results in the same improvement in reliability and klystron protection. The contactor should be rated to carry the load required for any future transmitter power increase.

The change can usually be made with minimal or no loss of air time. The end result is greatly improved reliability and much fewer maintenance problems.

Benefits:

.Improved klystron protection
.Increased reliability
.Reduced maintenance problems
(Mechanically simpler)
.Increased current capacity
(Can be sized for 60 kW)

Estimated Cost:

Vacuum Contactor	$7,500
Service entrance upgrades	0 - 10,000

Heat Exchanger Replacement

The heat exchanger is one of the easiest items to replace and solves many problems. The original water pumps and sump tank can be used, or newer more efficient pumps and stainless steel tanks can be installed outdoors adjacent to the heat exchanger.

It was common for transmitters to utilize a single, large (15 hp) blower motor to force air through the transmitter heat exchanger system. Newer systems typically use up to four, 1 hp direct drive staged fans which provide redundancy, efficiency (reduced power consumption), and temperature stability, thus improving transmitter stability.

Newer heat exchanger systems are designed to be mounted outside the transmitter building on a concrete pad. This provides two major advantages. First, it removes a major source of noise and heat load from the building. Secondly, it clears a significant amount of floor space for other uses.

The low cost of this type of cooling system makes it practical to size them for higher power operation and even to install two complete systems for redundancy or a separate system for each klystron.

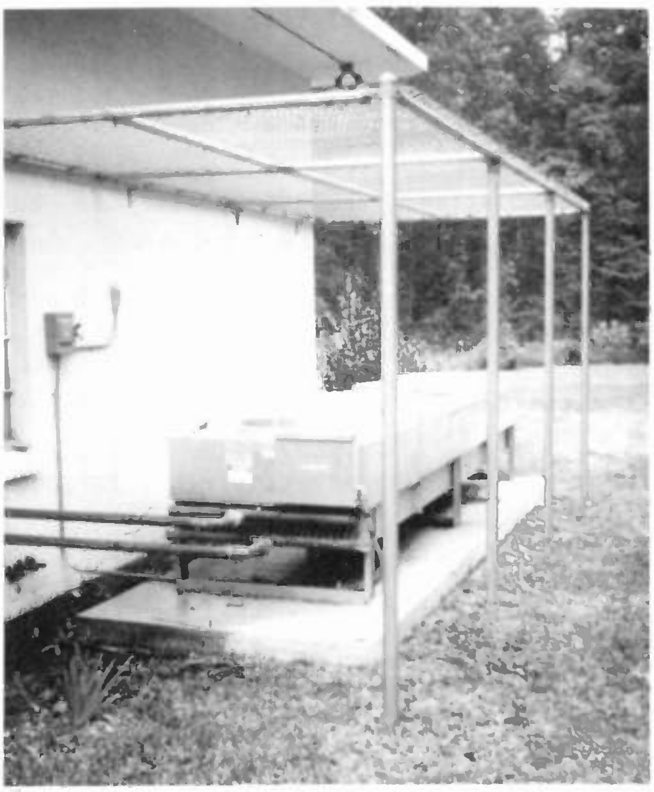
Benefits:

- .Increased reliability (staged multiple fans)
- .Cooler, quieter operation (mounted outdoors on concrete pad)
- .Reduced maintenance (small, direct drive blower)
- .Increased plant efficiency
- .Increased building space
- .Redundancy (multiple systems)

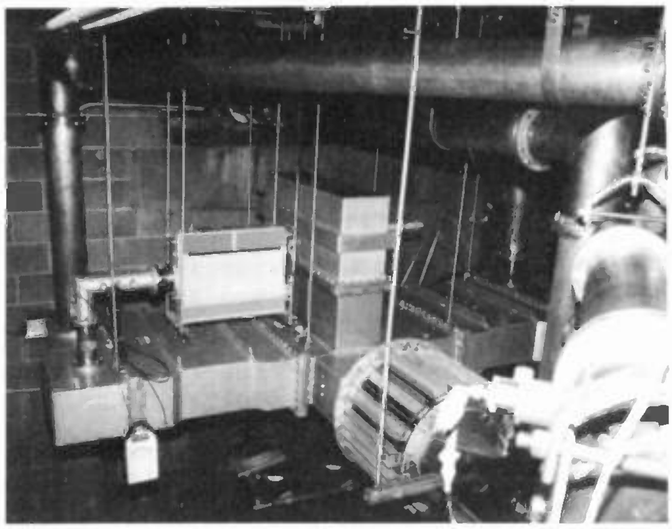
Estimated Cost:

4-stage outdoor heat exchanger	$5,000
Concrete pad	250 - 1,000
Plumbing supplies	250 - 500
Electrical supplies	250

Optional-	
Replace water pump 7-10 hp	1,000
sump tank, strainers, etc.	1,000



Pad mounted heat exchanger.



60 KW ceiling mounted diplexer

Filterplexer Replacement

Most older transmitters were fitted with hybrid filterplexers or traplexers. Due to their design, these units normally run very warm and the electrical performance suffers with thermal changes. In most cases peak visual power was limited to 30 kW. Today's modern waveguide diplexers are virtually trouble free and easy to install. They exhibit negligible loss and can be specified with multiple aural notch cavities for improved MTS performance. A power rating of 60 kW is generally standard. The system should be designed to transition to 6 1/8 inch line as quickly as possible after the klystron to reduce heat problems. Installation of an RF patch panel makes it possible to operate in multiplex mode through either the visual or the aural klystron. This feature gives a significant amount of operational flexibility to stay on the air should any klystron fail.

Most waveguide diplexer systems are larger than the old filterplexers and require more space in the transmitter building. Once the original heat exchanger is replaced with a outdoor unit, that space can be reclaimed to house the upgraded diplexer system.

Benefits:

- .Improved reliability/performance
- .Lower insertion loss
- .Can be easily up-graded to 60 kW
- .Capability for emergency multiplex
- .Improved MTS BTSC performance

Estimated Cost:

Waveguide diplexer	$30,000
6" Harmonic filters	5,000
Patch panel and transmission line	15,000
Directional couplers/peak detectors	5,000
Miscellaneous parts	2,000

Exciter/IPA Replacement

Over the years many UHF broadcasters have routinely replaced or upgraded their original low power exciter system as a means of upgrading transmitter performance. Exciter upgrades are relatively simple to accomplish and the results can be seen immediately in improved on-air quality as well as improved system reliability. Prices for exciters have decreased over recent years as technology has improved. A totally solid state, IF modulated exciter with a vestigial sideband SAW filter can now be purchased for less than $25,000.

Specific costs can vary depending upon output power requirements of the exciter. Increased klystron drive level and pre-correction are needed if newer generation tubes, or pulsing techniques are used or transmitter power is increased.

When replacing the exciter, it is essential to plan for the adequate drive requirements of newer generation klystrons. These higher efficiency tubes require much greater drive than the original tubes. As much as 6 dB in sync stretch may be required when operating in a non-pulsed mode. Some pulsing conditions may require sync reduction. Typical visual RF drive requirements can be as high as 85 watts on channel 14. Drive requirements are reduced as the operating channel increases. At channel 69, 10 watts should be sufficient. These drive requirements are much greater than the 5-10 watts which older visual klystrons require. The higher efficiency klystrons have a trade-off of reduced gain which must be considered. The exciter output amplifiers must be capable of these output levels. Increased aural drive requirements should also be considered although drive requirements for aural service are much less than for visual service. The additional amplification required is relatively inexpensive when purchased as part of the exciter system.

Most exciters now available have ample pre-correction necessary for the high standards of Multichannel Television Sound (MTS) in most transmitter configurations.

Benefits:

.Easy to upgrade
.Improved reliability
.Lower maintenance cost (no tubes)
.Improved performance
.Elimination of high level VSB filters
.IF modulation
.Pulser more easily interfaced
.Wider pre-correction range of video
.MTS BTSC compatible
.Space efficient
.Multiplexing capability

Estimated Cost:

Solid state exciter	$25,000 - 50,000

BCD/Mod Anode Pulser

Most klystrons can be fitted to operate with modulated anode pulsers. Klystron manufacturers can supply tubes with both modulated anode and beam control device (BCD) electrodes. BCD pulser costs are now comparable to older style modulated anode pulsers. Pulsers need to be interfaced with an IF modulated exciter which has ample pre-correction for ICPM and the ability to supply necessary drive signals to the pulser. Increased low frequency linearity and ICPM correction are required in order to pulse at the levels required for maximum efficiency.

Installation of BCD pulsers are usually easier than modulated anode pulsers. The BCD electrode requires a lower voltage pulse relative to the cathode than is required for modulated anode pulsing. This advantage allows the BCD pulser to be smaller and more reliable.

Benefits:

.Decreased transmitter power consumption

Estimated Cost:

Pulser, with ICPM correction (requires IF modulated exciter)	$22,000

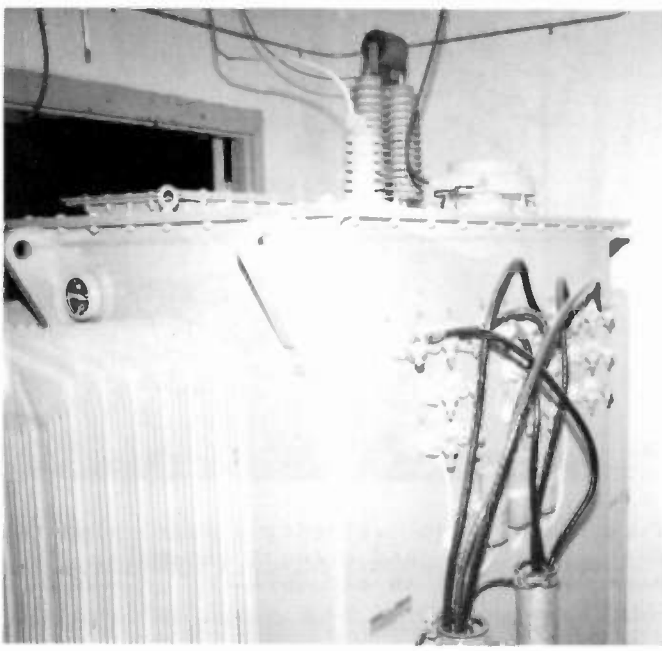
Beam Supply Replacement

Many high voltage beam supplies contain transformers and capacitors filled with oil containing PCB's. Although possession of these units is not illegal, they are subject to EPA regulations. (Ref. 2.) nits containing PCB's of any amount must be identified and properly managed, until they are removed from your facility.

Todays modern high voltage supplies are rugged and reliable and do not contain PCB's. A separate integrated high voltage beam supply can be mounted outside the building to reduce building heat load. These units require little maintenance. Many new unitized supplies are fitted with secondary voltage taps to obtain an easy reduction in beam voltage required for initial klystron tuning purposes. Typical power supplies are rated for 235 kVA with output voltages up to 27 kV DC at 8.5 amps. This is adequate capacity to operate a 60 kW transmitter with high efficiency type klystrons. The higher operating voltage is necessary for improved klystron efficiency.

Special care is required in the selection of the power supply. Proper installation of the new supply is very important. Minor modifications to the transmitter are necessary to allow for higher beam voltages. Unitized beam supplies exhibit a greater reliability and efficiently than separate transformer rectifier type power supplies. The location of the rectifiers in the oil bath of the unitized beam supply also reduces building heat load.

Most older generation transmitters were designed to operate with 3 phase, 208 VAC primary power. It is now common to specify 3 phase, 480 VAC service. The increased primary voltage is usually more efficient and will allow smaller gauge wire, more efficient motors, pumps, and power supplies.



Unitized beam power supply

For some operators it may be advantageous to change to 480 VAC. This is especially true if a station plans to completely upgrade the transmitter. Today, most pumps and motors are available with dual taps for both 480 and 208 VAC. It is worthwhile to specify these dual voltage taps so that you may convert to the higher voltage in stages. However, the premium cost for dual voltage power supplies is prohibitive and should be installed initially as a 408 VAC device.

Benefits:

.Elimination of PCB problem
.Increased reliability
.Increased beam voltage
.Higher efficiency
.Can be sized for increased RF power
.Cooler transmitter operation
.More room in transmitter cabinet for other transmitter modifications

Estimated Cost:

Unitized outdoor beam supply	$25,000
AC service entrance rework to 480 volts	15,000
Equipment pad (can be same as for heat exchanger)	500
Crane and/or fork lift rental (To unload and position)	250
Transmitter modifications	2,000

Klystron Upgrades

The major advances in klystrons are in the areas of efficiency, longevity, and power output capability. In the United States new transmitters are now configured with klystrons which are typically rated at 60 kW peak sync instead of 30 kW.

Older transmitters can usually enjoy the benefits of this new klystron technology with a relatively small increase in capital costs.

The time to think about upgrading or changing to a newer generation klystron is not when a klystron is failing or when the transmitter is off the air. It should be planned out in advance to insure that all of the details are in order. For instance, newer generation klystrons may require different filament voltages or more input drive power. The older 30 kW (Varian 4KM-100) tubes have high efficiency direct replacements designed to operate at the 55 kW level.

The newest wide-band 60 kW tubes require completely new circuit assemblies (magnetic housings and cavities). Corrosion in the water tubes which cool the klystron focus coils are a source of problems. Partial or complete magnet failure may occur resulting in serious klystron damage. Older klystron cavities also exhibit such problems as erratic tuning. Replacing the entire magnetic housing may be more cost effective than trying to repair the current circuit assemblies. The new wideband tubes and circuit assemblies do not require water for cooling the magnets or the body of the tube instead, a high volume forced air supply is used. Blowers for this purpose are commonly available and easy to install.

It is important to check with the klystron manufacturer to ascertain if your transmitter requires any additional specific modifications to accept newer tubes. Klystron manufacturers generally provide very helpful information and may provide on site assistance if required.

Benefits:

.Higher efficiency
.Longer life
.55 kW capability
(60 kW with wideband tube)
.Easier to tune
.BCD electrode

Estimated Costs:

Variable, depending on tube configuration.	$30,000

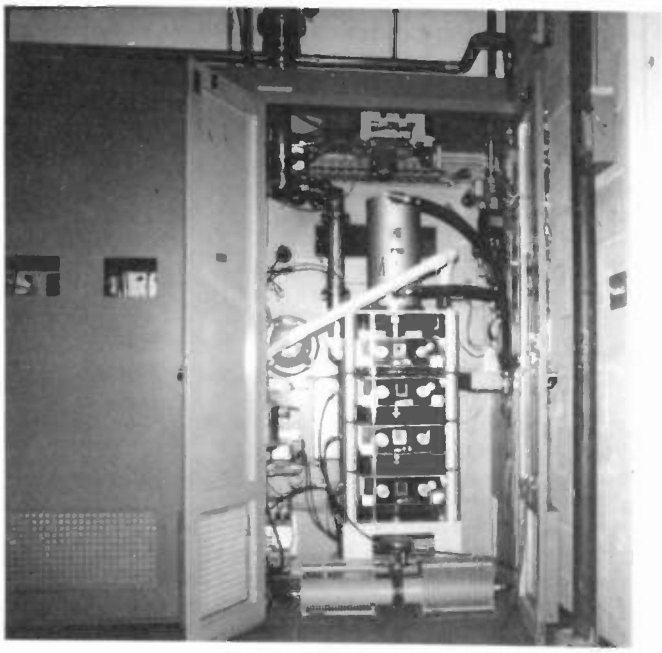
Broadband 60 KW tubes require new magnets, cavities, and other transmitter modifications. This can run an additional $30,000 over the basic cost of the klystron.

MODIFYING TO INCREASE TRANSMITTER POWER

Once the exciter, power supply, cooling system, klystron, and diplexer are replaced it becomes relatively easy to increase the transmitter power from 30 kW to 60 kW. For some time NCPT has routinely replaced 30 kW klystrons with 55 kW klystrons in anticipation of increasing transmitter power. These klystrons are more efficient than their older counterparts and fit directly into the older magnetic housings. Tube life has been excellent. In fact, the NCPT has yet to experience a normal end-of-life failure in any of these tubes which were placed in service since 1979.

Last year a successful retrofit was made to our 1967 General Electric TT-57 30 kW transmitter at WUNE-TV Linville, NC using an EEV K3672BCD high efficiency wideband 60 kW klystron . We are now in the process of increasing the output power from 30 kW to 60 kW.

60 kW High Efficiency Wideband Klystron Installed In GE TT-57 Transmitter



Note water connections at the top and the high voltage connections at the bottom of the tube. Also note the extra cabinet space.

View shows high velocity blower used to cool klystron and circuit assembly. PVC pipe is used as an air duct.

Increasing the transmitter efficiency of older transmitters makes it possible to increase power to the 50-60 kW level without significantly increasing transmitter AC power consumption.

Replacement Transmitting Antenna

Although not specifically a part of a transmitter, replacement of the transmitting antenna can have the greatest effect on the station's overall coverage. Antenna technology is continuing to evolve. Today, antenna design is much more of a science than it was in the past. Many older generation antennas (especially those of helical design) are still in service. These antennas are usually not radiating effectively or as predicted.

Aeronautical and/or field studies of all the UHF antennas in our statewide system have been conducted. These studies have proven that our older antennas of helical design are performing poorly even though they appear to check out well mechanically and electrically.

An example of this phenomenon took place at our WUNE-TV channel 17 transmitter site. Last year, a tower inspection revealed a crack developing around the main mounting flange of the old General Electric antenna. The antenna appeared to show a good electrical match and was thought to be one of the better performing GE antennas in the NCPT system.

Because of the pending mechanical failure, a small (20 ft.), low gain, side-mounted antenna was selected as a temporary replacement. The large (69 ft.), top-mounted GE TY-160B antenna gain provided 1.6 megawatts ERP in a cardioid pattern. The new Andrew ATW-8-G-HSS provided only 450 kW ERP in a similar pattern, side mounted to the tower leg. In both cases the antenna input power was approximately 28 kW peak.

Our survey truck made a series of measurements at 56 different locations immediately before and after the antenna change. Measurements were made from 1 to 60 miles from the transmitter site. In all cases, we found the received signal after the installation of the temporary antenna was greater than that measured from the old GE helical. The average increase was 6 dBmV. (margin of error = $\pm$1 dB). We had suspected that the GE antennas were performing poorly, but the results of the antenna change were quite surprising. Favorable viewer response has confirmed our findings.

This low cost option has proved to be a good alternative for the network. Not including installation, the cost of the temporary antenna was less than $30,000. Tower mounting brackets were fabricated locally and the installation of this small antenna was not complex or costly. By making this change, we now have time to plan a complete tower and antenna replacement in the future. Because the temporary antenna is so small a tower load it will be mounted as a standby when we replace it and tower.

This antenna replacement along with the other changes that had already been made to the transmitter is allowing us to increase the transmitter power.

Benefits:

- .Substantial improvement in coverage compared to helical or zig-zag antennas
- .Elimination of de-icers
- .Ability to safely increase power (6 1/8 inch input is standard)
- .Greater reliability

Estimated Costs:

Depending upon configuration	$30,000 - 250,000
Antenna installation (tower crew)	20,000 - 50,000

Small, side-mounted temporary antenna

Other General Transmitter Updates

There are also many minor updates which greatly improve the operation and reliability of older transmitters. The following is some of the more common retrofits which we have done.

Older generation water flow sensors and flow meters have always caused problems. Better protection of our klystrons has been achieved by replacement with positive action paddle type sensors.

VSWR protection was upgraded by changing to more stable and high precision RF detectors and solid state amplifiers.

Station engineers may wish to consult with a knowledgeable electrical contractor and review the National Electrical Code (NEC) to ascertain if the present electrical configuration is adequate and meets the current local and state codes. Current installations may not conform to present day standards even though they did conform when they were originally installed.

The most common electrical problems are undersized wiring, and improper or incomplete equipment grounds. Frequently the cause of reoccurring lightning damage can be substantially reduced by upgrading the service entrance and associated grounding systems. A total system ground plan should be developed and implemented. In most cases only minor rework or upgrading will be necessary if the electrical system was installed and maintained correctly.

Others easy improvements can be made in general plumbing by changing from gate water valves to quarter-turn ball valves and changing to Hansen connectors. Cabinet blowers can also be replaced by units that are more commonly available, quieter and more efficient.

Transmitter Cost Comparison: New vs. Updating

The following shows average costs to upgrade and increase power of a 30 kW General Electric UHF transmitter to 60 kW. This assumes a complete retrofit to a 60 kW klystron in the visual amplifier. Note that all costs are estimates and were obtained by NCPT's past experience, and conversations with industry leaders.

High Voltage Contactor	$9,000
Heat exchanger w/pump	8,000
Diplexer	50,000
Exciter	30,000
Pulser	22,000
Beam supply	40,000
Klystron and circuit assembly	55,000
Other General Transmitter Updates	10,000
	========
	$224,000

Price range for a new 60 kW UHF transmitter $350,000 - 600,000

(Depending on overall quality and range of options)

Labor costs are not reflected in these estimates. These costs will be a function of the skills available within the station's staff versus what must be contracted externally.

Other Considerations

Operation at higher power (above 30 kW) may require replacement or refurbishment of the transmission line in order to reliably handle the power increase. The existing transmission line should be thoroughly evaluated. Experience has shown that worn center conductors, anchor insulators (bullets), and general contaminants in old transmission line are the prime cause of line failure when RF power is increased.

The antenna power handling capability is another area for concern.

If the transmitter upgrade is done in stages, you may be unable to increase power until the diplexer, antenna, or transmission line is replaced.

CONCLUSION

Total UHF transmitter upgrades are not for everyone. The extensive planning and specialized installation skills needed may be more than some stations can comfortably handle. Modifications can be time consuming. Some of the work needs to be done during non-broadcast hours if the broadcast schedule is not to be affected.

Each station must carefully choose which upgrades are appropriate to their needs and how they will be accomplished in terms of budget, manpower, and timetable.

In many ways a UHF transmitter is similar to an automobile. It not necessary to replace the car just because the tires and brakes are worn out. Upgrading older transmitters is an effective way to increase reliability, efficiency, performance, and longevity.

ACKNOWLEDGMENTS

The author wishes to acknowledge the support of the Engineering Division of North Carolina Public Television for their work to improve and refine these transmitter updates. Additional thanks is given to Kip Campbell and Wayne Estabrooks for their assistance in the preparation of this document.

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KLYSTRODE TECHNOLOGY UPDATE

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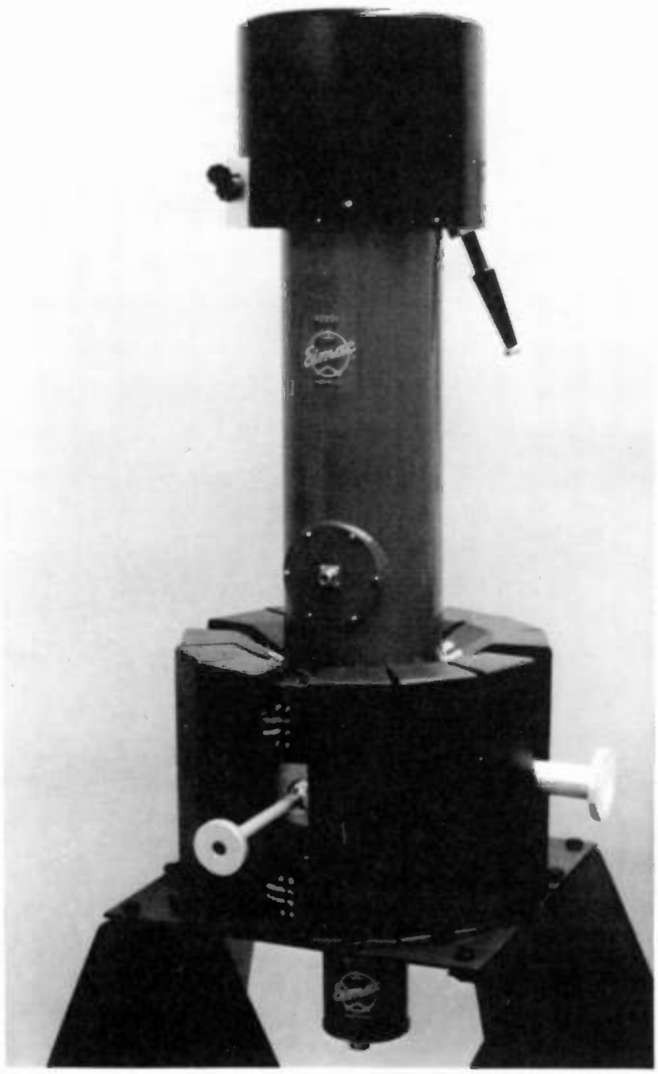
San Carlos, California

Introduction

Since the paper given at the NAB Conference three years ago by G. Badger[1] on the 60KW Klystrode Development, a great deal of effort has been expended in finalizing the product development and integrating the 60KW Klystrode into the first 120KW transmitter, as described by Comark at this years' conference. In addition, Klystrode technology and product development has been expanded into other power areas in UHF TV as well as a number of totally new potential applications. The fundamental advantages of the Klystrode in efficiency, size and weight plus Class B operation with amplitude modulation are making the device attractive for consideration in other UHF applications. These applications range from rf sources for Free Electron Lasers, Synchrotron Light Sources, Scientific Particle Accelerators to Space-Borne applications such as Neutral Particle Beam Accelerators. The Varian-sponsored development of the 60KW Klystrode for UHF television has made possible the potential application of Klystrode technology to these new requirements for rf sources in UHF. Likewise the acquisition of contract support from SDI programs has aided and complemented our continued development and productization of our UHF TV programs. The net result has been a greatly expanded effort in our programs to bring new Klystrode products to the market.

60KW Klystrode

The past two years' effort has largely been spent on characterizing the operation of the Klystrode in the intended 60KW service and productizing the hardware. The Comark[2] paper presented at last years' NAB summarized very well the test results obtained in TV service. Table 1 is an example of power consumption advantages and shows that a Figure of Merit of 1.29 can be obtained without pulsing or collector depression.



Hardware development has progressed from the early developmental model shown in Fig. 1 to the production model shown in Fig. 2. Double tuned output cavities are used to obtain the required bandwidth and the antenna is coupled into the second cavity - Fig. 3. This has turned out to be a very attractive way to couple power out of the Klystrode since no coupling loop or probe is required in the primary cavity which can be a problem at the high end of the UHF band where there is so little space between a coupling loop and the ceramic output window. On one occasion where our water load inadvertently opened, the Klystrode continued to operate with full beam voltage and no arcing in the output cavity. This is definitely not a recommended procedure, but it gave us additional confidence as to the Klystrode's reliability in this type of circuit.

The input circuit is fundamentally the same except a much neater high voltage blocker has been fabricated eliminating the long, awkward HV connection shown in Fig. 1. The magnet circuit has been simplified and focusing power of under 200 watts has been demonstrated.

A low band version of the amplifier shown in Fig. 2 has been developed and differs only in the height of the output cavities and the length of the input circuit. Measurements on the low band tube are similar to the high band version and indicate good bandwidth, efficiency and power gain. Fig. 4 is a photo of the low band 60KW Klystrode. It differs from the high band tube only in the height of the output ceramic window.

In summary, the 60KW program is nearing completion and most of the effort is directed toward integrating the amplifier into the 120KW transmitter which is being built by Comark for the Georgia Public Broadcasting System.

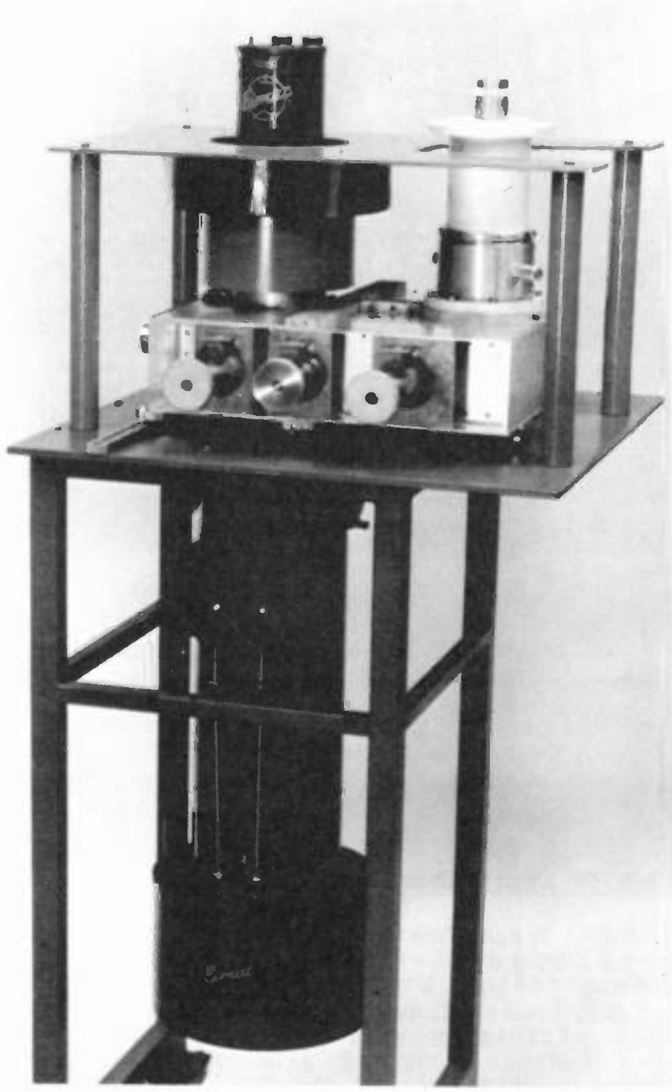


TABLE 1

	Peak Output Power	Avg. Beam Power 50% APL	Figure of Merit*	Advantage Ratio
Klystrode	60kW	46.5kW	1.29	1.87
Pulsed Klystron	60kW	86 kW	0.69	

* Peak RF Output divided by Average Beam Power

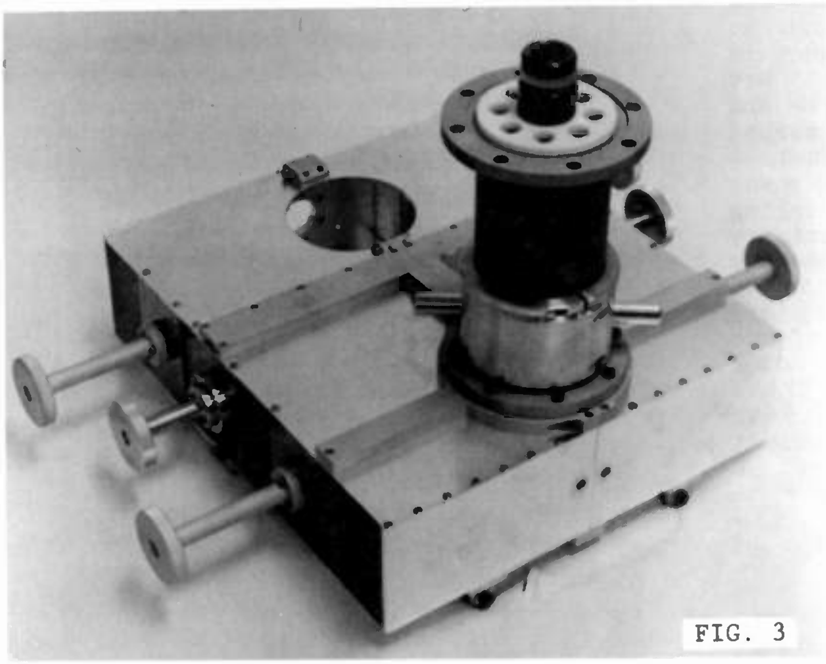


FIG. 3

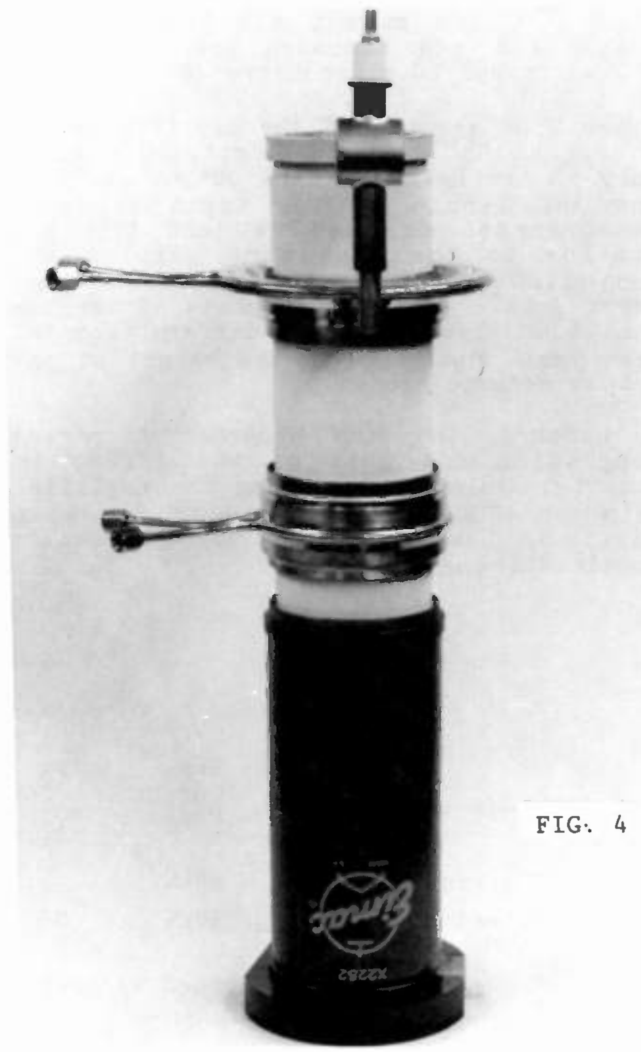


FIG. 4

10-15KW Klystrodes

A 10KW Tropo Scatter Klystrode has been developed for the 755-985 MHz communication band (Fig. 5). Light weight, compactness, broad bandwidth and high efficiency were the motivation for this contract support from the Rome Air Development Center of the U.S. Air Force.

For UHF television at lower power levels a 15KW air-cooled Klystrode was developed and reported on by M. Chase[3] of Varian/EIMAC at Montreux. A photo of this tube is shown in Fig. 6. From the scale shown the extremely compact structure is highlighted, but even more important the Klystrode collector only needs to dissipate an average power slightly greater than black level. The decrease in average collector dissipation compared to a Klystrode results in: a much smaller collector fin structure, a lower air flow, a much lower pressure drop across the fin structure. Therefore the Klystrode makes possible a more cost effective and quieter cooling system. 30, 40 and 60KW air-cooled Klystrodes are now being considered. Tables 2 and 3 give the performance data for the 15KW Klystrode.

Super Power Klystrodes

An increasing number of applications for high power (200KW to 1MW) are developing for rf sources in the UHF Spectrum. As the atomic particles which scientists wish to accelerate become heavier, the frequency of the rf accelerating cavity structures move down into the 200-800 MHz range. This frequency range is hard for conventional triodes and tetrodes to perform efficiently, and klystrons become almost unmanageable in size and weight. Fig. 7 is a comparison of a state of the air klystron for 1MW at 400 MHz and a Klystrode for the same power and frequency.

For space applications such as Neutral Particle Beam Accelerators, the nearly 5000 pounds of the klystron versus 250 pounds for the Klystrode has prompted quite a bit of attention. The net result has been contract from SDI monitored by Los Alamos National Labs to develop a Klystrode for 500KW peak power at 425 MHz. Fig. 8 is a cut-away view of this tube which is currently under development. Size and weight objectives are 44" long, 18" diameter, and 110 lbs.

FIG. 5

Table 2 X2254 UHF Performance Data

Frequency.........................772 MHz
Bandwidth...........................9 MHz
Beam Voltage......................18.2 KV
Focusing Power....................240 Watts

Condition	Beam Current (A)	Power Output (KW)	Conversion Efficiency (%)	Drive Power (W)	Gain (dB)
Peak of Sync	1.59	15.5	53	86	22.6
	1.21	9.8	45	56	22.4
Black Level	1.10	8.6	43	50	22.3
Ave.Picture Level	0.66	3.4	28	38	20.8

Table 3 X2254 UHF Television Data

Operational Mode	Conventional	Multiplexed	Multiplexed
Peak of Sync Power (KW)	15.5	8.0	4.0
Quiescent Current (A)	.020	.67	.67
Nonlinearity (%)	-61	-48	-19
Sync Compression (%)	-41	-51	-26

Differential Gain..................-40% *
Differential Phase.................-8 Deg *

* Expected Value

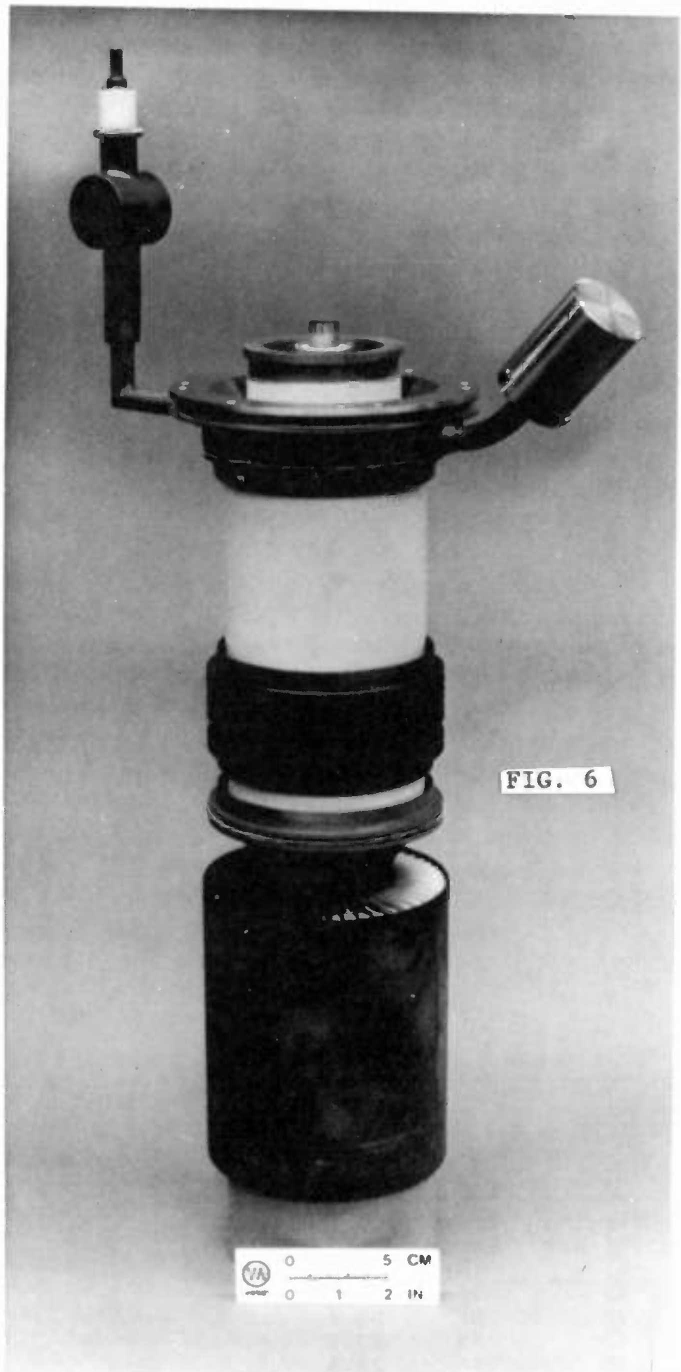
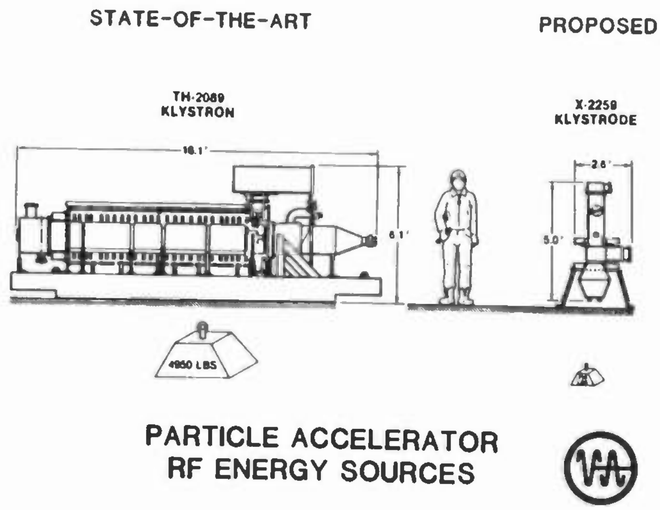


FIG. 6

FIG. 7



Conclusion

Klystrode technology is being expanded into a line of new products from 10KW to 1MW. The synergism between Varian-supported IR&D and government contract support for the higher power versions has enhanced our overall programs measurably. The 60KW TV Klystrode amplifier is about to begin television broadcast service, probably the last major hurdle before gaining wide acceptance in the broadcast industry. The fundamental operating principles upon which the Klystrode depends have been shown to be ideally suited for the applications that are evolving in UHF. Compared to existing devices which compete in this growing field, the Klystrode is a youngster; however, many of it's genes have come from its mature parents. This gives us confidence that further improvements in performance, reliability and cost will be made.

NAB Conference 1988

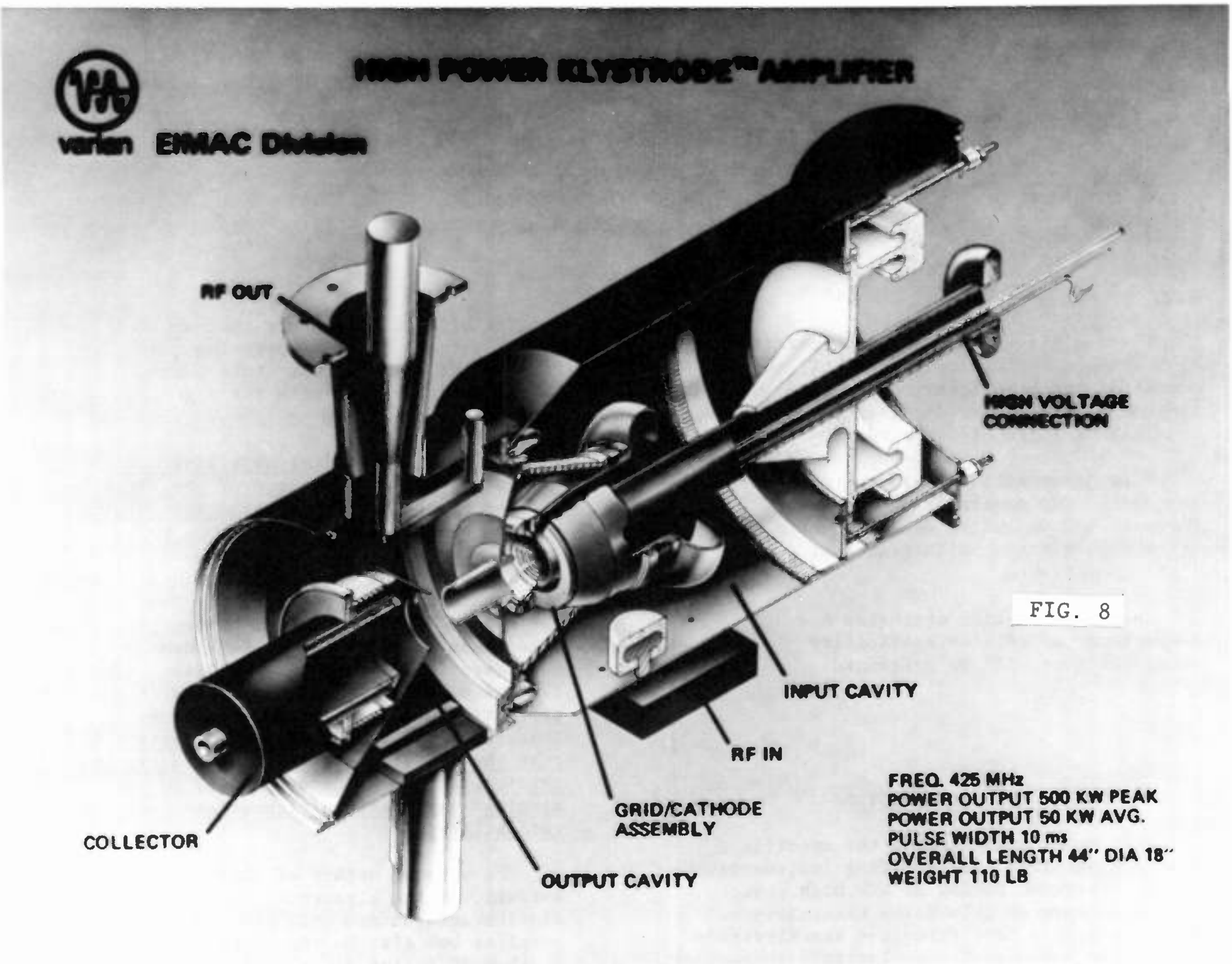


FIG. 8

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A 120 KW KLYSTRODE TRANSMITTER FOR FULL BROADCAST SERVICE

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The new Klystrode tube which permits ultra high average efficiency in UHF full service operation has been incorporated into an actual broadcast transmitter designed for its use. This is the first of its kind.

This paper will discuss the objectives of the design and construction of this transmitter, including details of the solid state driver stage which generates 600W of linear power.

Cost of operation estimates and comparisons as well as reliability considerations will be presented.

I. INTRODUCTION

This paper will discuss the specific design objectives and resulting implementation of the Comark SK Series of UHF high power klystrode powered television transmitters. Previous papers have discussed the klystrode tube[1,2] and the use of the klystrode in general terms in transmitter design[3]. This paper will set out the design objectives of the SK Series as well as discuss the equipment's systems and expected performance.

The equipment described herein, at the 120kW power level, is scheduled for installation in the world's first all klystrode transmitter site at Wrens, Georgia during the Summer of 1988.

II. DESIGN OBJECTIVES AND CONSIDERATIONS

The 60kW klystrode tube offers the UHF television transmitter designer one overriding advantage over klystrons. That advantage is efficiency. The average efficiency of the klystrode in television service has previously been demonstrated to be between two and three times better than the best, fully pulsed conventional klystron tube and at least four times better than many older systems in service today. The tube provides this advantage without the need for complex supporting systems or multiple power supplies.

The high efficiency is the result of the tube's natural Class B operating mode. In Class B the beam current (tube power requirements) varies with the magnitude of the input drive power.

One of the principal objectives of the SK design is to take advantage of the efficiency offered by the klystrode while maintaining klystron-like reliability.

The klystrode derives its unique behavior from the use of a control grid. This grid is not new to highly reliable, proven power tube types such as TWT's but it is a design consideration. It requires a bias system to set the tube's operating point and it requires protection from catastrophic failure. Therefore, the second primary objective is that the SK Series design must satisfy the grid's bias and protection needs in the simplest way possible without compromising reliability.

The varying nature of the klystrode beam current, unlike klystrons, places additional requirements on not only the tube's power supplies but also on the transmitter cabinet. Another SK series design objective is therefore to design a support cabinet that meets the EMI/RFI suppression needs of the klystrode while maintaining klystron capability.

The klystrode does not provide the same high gain as a klystron. Its gain is, however, considerably higher than a tetrode. Thus the klystrode requires more drive than a klystron, but at 60kW output levels it still can be driven from a practical solid state amplifier. It is the objective of the SK series design to provide a fully redundant solid state amplifier system capable of 600W peak output which can survive random failures of individual devices or power supplies without loss of meaningful output power.

The klystrode has also been demonstrated to be a reasonably linear device. It does, however, have unique non-linearity problems that must be compensated for in the system's exciter. Therefore, an SK Series design objective is to provide the highest standard of picture quality while keeping the level of exciter/modulator complexity to a minimum.

Finally, the single overriding objective of the Comark SK Series design is to not compromise the overall reliability of the on air television broadcast signal while providing the high power UHF broadcaster a significant savings in operating costs.

III. MEETING THE DESIGN OBJECTIVES

A. Efficiency without Complexity

Previous papers[4] have established a multiplying factor that can be applied to a klystron's fundamental "out of the box" efficiency to determine the maximum theoretical pulsed efficiency that the tube can be expected to achieve. This number is 1.69. Thus a high efficiency klystron, fully pulsed and tuned with full linearity compensation may be expected, under ideal conditions, to achieve a figure of merit of 1.69 times its out of the box performance. Given a tube that has a 50% basic efficiency, a pulsed "figure of merit" of 84.5% is the best that can be expected. These levels are never achieved in actual broadcast operation since the fully pulsed linearity correction is difficult to achieve on a stable basis. Practical values of 60% to 70% are however common.

"Figure of Merit" is defined as follows:

$$\text{Figure of Merit} = \frac{\text{RF Peak Power Output}}{\text{Average D.C. Input at 50\% APL}}$$

Since a klystron is a Class "A" device, "Figure of Merit" and Efficiency can be used interchangeably. This is true since the Average DC input power does not vary with picture content.

The system that is required to attempt to approach a high figure of merit requires complex high voltage pulsing on the mod anode or lower voltage pulsing of the ABC/BCD element or low frequency grid in the klystron. In any case, any of these systems require critical adjustments of both pulse timing, pulse rise time and overall video linearity correction. This complexity along with the high voltage nature of the problem can lead to a degradation of total system reliability.

In short, the klystron's high reliability is compromised by its complex pulsing support system. After all, the broadcaster is principally interested in the on air reliability of his signal and the radiation of a high quality product.

The klystrode, on the other hand, provides not only beam power variation during sync pulses, as in a pulsed klystron but it also varies its beam power over the entire video waveform. Thus, the klystrode provides full time beam modulation as a result of its inherent structure. This is achieved without special timing or shaping circuits, multiple power supplies or advanced linearity correction circuitry. The figure of merit for the klystrode has been consistently measured at 125% or higher.

In the SK Series design, the advantage of Class B operation is achieved without the complexity associated with today's pulsing systems or the proposed complex and yet to be invented arrangements for full time grid modulation or the multiple power supplies of the proposed MDC klystron systems.

The only inherent system complexity in the SK design associated with Class B operation are those that all the previously mentioned proposed system must also address; that is, video modulated beam current. There is only minor impact on the klystrode power supply requirements and there is no complex modulating system in the SK design.

Thus, by simply using the klystrode itself, the SK Series achieves its first design objective. The equipment takes advantage of Class B operation without introducing reliability degrading complexity.

B. Biasing and Protecting the Klystrode Grid

The klystrode's most controversial feature is its pyrolytic graphite RF grid. This element is what controls the beam current and is driven with the RF input signal. The circuit required to place the RF drive voltage on the grid resembles a conventional input cavity found in tetrode designs. This resemblance is only superficial however, and the details of RF input circuit design could be the subject for an entire paper. For the purposes of this paper, the reader need only accept that the RF drive is matched to the grid so that a reasonable impedance is maintained and the grid is isolated from ground for high voltage.

The klystrode's beam current operating point is set by the d.c. bias placed between grid and cathode. Since the cathode is operating at the negative beam potential of minus 30kV, the grid bias problem could be difficult. However, a very simple and reliable solution has been implemented in the SK design for grid bias.

In the transmitter's shielded H.V. compartment, a series string of zener diodes is used between the cathode and the beam supply negative to establish the klystrode's fixed bias operating point. This voltage is between 35 and 75 volts.

This simple solution provides both stable bias voltage while limiting the energy available to the grid during internal tube arcing events. Protection of the zener diodes, while present, does not guarantee absolute survival during catastrophic failures. Therefore, the zener string is fabricated as a replaceable, plug in module and is spared with the equipment. The probability of catastrophic failure is however small when the grid protection circuit is considered.

The protection of the klystrode grid begins in the tube design itself. The internal construction of the klystrode has placed the grid in a protected location away from direct arc paths.

The klystrode's physical design mounts the grid on a massive metal electrode which has proven to be the focus point of any tube arcing measured during the test program.

The SK design, however, does not assume that the grid is protected by the tube's internal layout. Instead, the design includes a very fast, high energy crowbar system. This system limits the energy that can be delivered to any transmitter component during an arc to that necessary to just punch a pinhole in aluminum foil.

Figure 1 is a block diagram of the SK crowbar system. The authors wish to thank the engineering staff of Varian/Eimac for the design of this circuit. The features include a status indicator showing circuit condition, total number of high energy discharges and an operator circuit test and ready control.

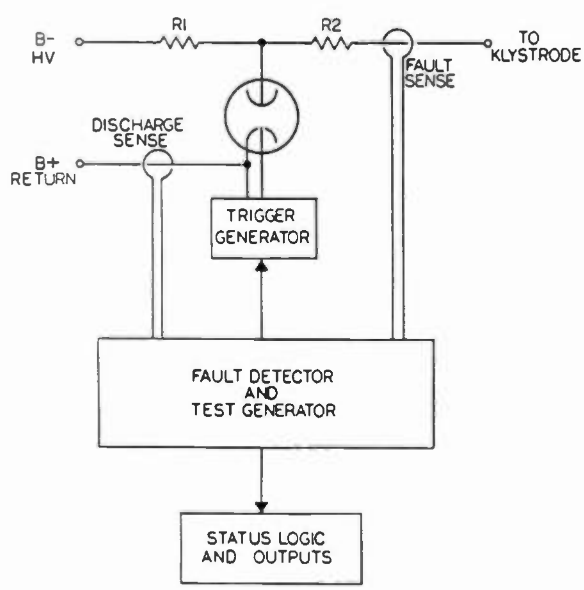


Figure 1 - SK Series Crowbar

Some of the more interesting performance specifications of the SK crowbar are as follows:

1. Trip current set point - 50 Amperes
2. Response Time - Less than 10 microseconds
3. Peak current - 3000 Amperes
4. Total conducted lifetime charge - 750 Coulombs
5. Max conducted charge - 5 Coulombs

Thus, the SK crowbar will fire whenever the beam current exceeds 50 Amps and it will provide protection within 10 microseconds. In actual tests, the energy delivered from an arc of the full beam supply barely produced a pin hole in a piece of aluminum foil.

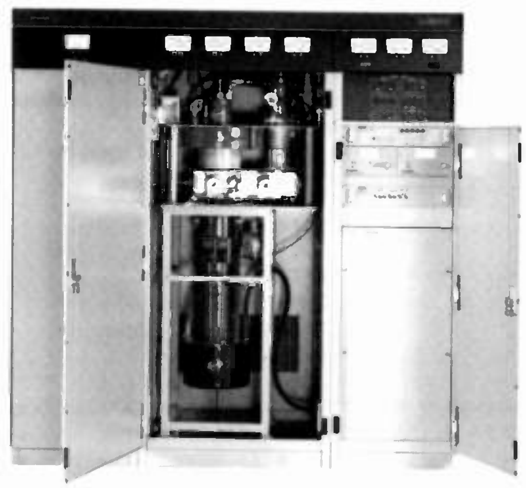


Figure 2 - 60kW Klystrode Cabinet including solid state driver (left) and control cabinet (right). Isolated H.V. compartment in lower right.

The SK Series design meets the initial objective of providing a simple grid bias system and a protection system that does not compromise reliability. In fact, the klystrode in the SK system may be better protected and live longer under field operating conditions than current day klystrons that don't have the benefit of the SK crowbar system!

C. Cabinet Design for EMI/RFI Capability with the Klystrode

The klystrode's beam current varies with the video waveform of the modulated drive signal. Thus the klystrode is in fact a very powerful video modulator. Considerable video power is represented by the beam current. At 30kV and 1.5 Amps of current change the peak video power is 45kW! The vast majority of the power is returned to the power supply. However, conventional klystron cabinet layouts, designed for steady state Class A environments, may not be suitable for klystrode operation.

Radiation of video frequencies from the klystrode's beam lead, positive return lead or high voltage components can cause difficulties with solid state control logic, metering circuits and in extreme circumstances the low level circuits of the IF modulator.

In fact, the radiation from the klystrode beam lead has been known to interfere with the proper operation of switching power supplies.

Therefore, the cabinet layout of the transmitter must contain some unique features if operational problems are to be avoided.

The complete high voltage section of the cabinet, containing all resistors, bypass capacitors, filament/bias networks and the crowbar must be closed and shielded.

In addition, the beam lead to the klystrode must be bypassed for video frequencies via a low impedance return path. Care must also be exercised to insure that all grounding connections within the cabinet are low impedance. In extreme cases the Beam lead to the power supply may have to be placed inside a bonded conducting tube connected to the cabinet and the power supply frame.

Fortunately, the Comark "S" Series of klystron transmitters contained, as an integral part of its cabinet design, an isolated H.V. compartment. Thus, in the SK design the basic "S" Series cabinets were easily adapted to meet the RFI/EMI requirements of the klystrode. Figure 3 shows the klystrode installed in the SK cabinet. Figure 4 shows the isolated H.V. compartment which is located to the lower right of the tube and is accessible from the front of the equipment. This compartment contains the bias supply, filament supply and SK crowbar system. The high energy triggered discharge tube is visible midway up on the left side of the cabinet.

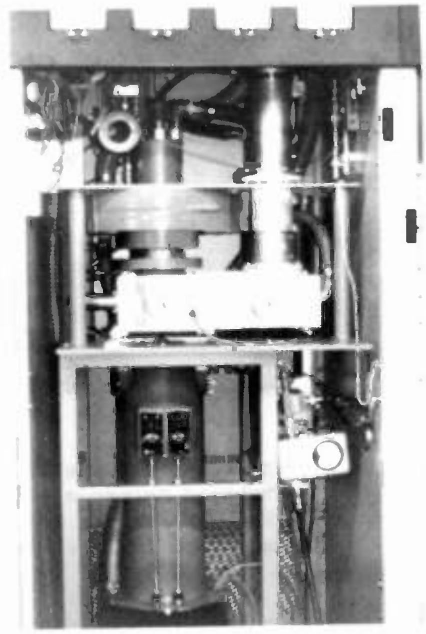


Figure 3 - Klystrode installed in SK Cabinet

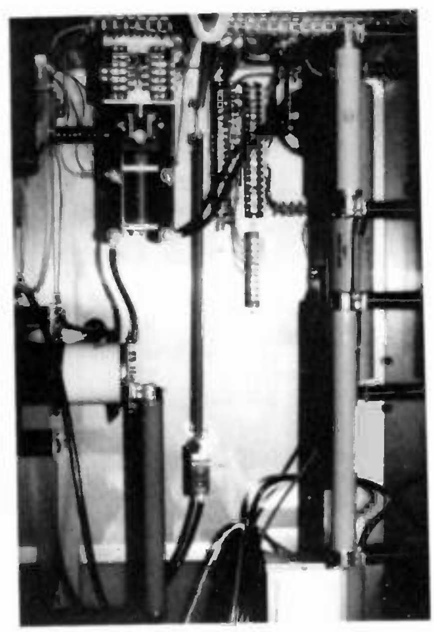


Figure 4 - Isolated H.V. Compartment

Thus the design objective of meeting the proper RF radiation needs of the klystrode can be met by adopting the standard Comark "S" Series klystron cabinet. Klystron compatability is maintained and the design objective is satisfied.

D. Solid State SK Driver

The 60kW klystrode amplifier requires between 300 and 600 Watts of peak RF drive. In the 120kW transmitter that means one 600 Watt amplifier per tube and one smaller amplifier for the aural klystrode.

While 600W is a reasonable power level for solid state, given today's technology, care must be taken to insure long-term reliable operation.

The SK driver utilizes multiple transistor amplifier modules combined to produce the 600 Watt peak output. Each individual module is, in itself, a combination of four devices hybrid combined.

Figure 5 is a photo of one half of a module showing a two transistor circuit. The circuit is mounted on a heat sink and cooled by forced air. Each full module provides 8dB minimum gain and 100 Watts peak output.

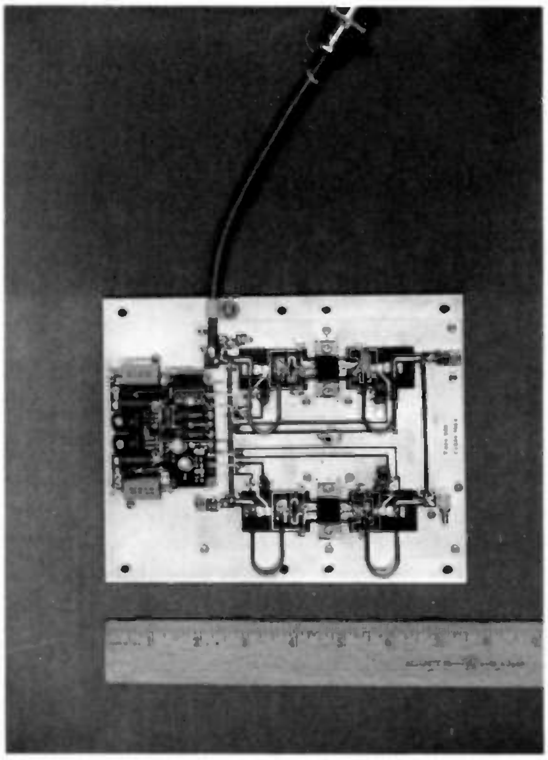


Figure 5 - 50 Watt Solid State Circuit

Figure 6 shows the general combination arrangement including the driver. The driver module is the same as the final amplifiers except that the power supply connections are split internally. This permits separate, independent power supplies for each pair of devices in the module.

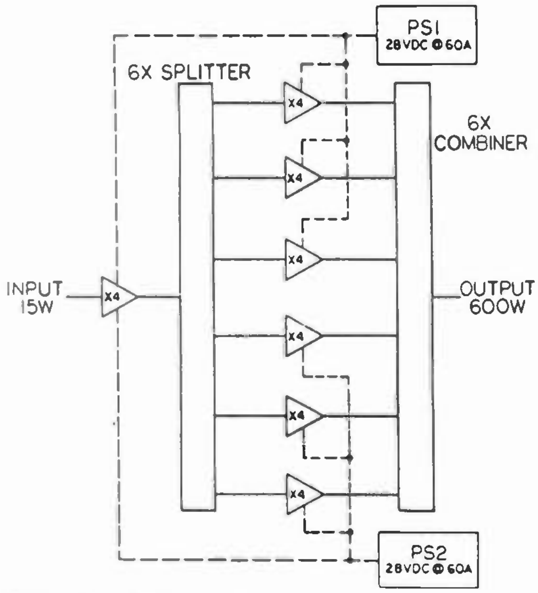


Figure 6 - Combination geometry of 600 Watt Solid State Amplifier

The split power supply philosophy is extended into the final amplifier array to permit each half of the array to operate on separate supplies.

Thus, the SK series solid state driver is both RF device redundant and power supply redundant. A failure of any single active component will not put the transmitter off the air.

The SK series design objective of producing a reliable 600 Watt driver which will maintain meaningful output power under random failure conditions is therefore achieved.

E. Linearity Correction of the Klystrode

The klystrode tube does not have the same hard saturation characteristic as the klystron. In the klystron the maximum output power is fixed by the pre-established beam current. As RF input drive increases, output power will be limited absolutely to the maximum allowed by the preset beam current.

In the klystrode the beam current increases with RF drive and thus output power is limited by voltage swing, impedances and cathode limitations, much like a tetrode. The result of this characteristic is that the klystrode high power transfer curve flattens more slowly than a klystron and continues to increase with increasing drive power.

At the small signal end of the transfer curve there appears an "S" shaped turn on characteristic typical of Class B operation. This can be partially overcome by setting the zero signal beam current to some small value such as 200 ma.

The correction of the klystrode's non-linearity is required at both white level and at black and sync. Further, this must be done at IF after the SAW filter to permit proper sideband cancellation.

Under multiplex operation the combined visual and aural carriers must be linearized to insure proper I.M. cancellation.

The linearity correction for the klystrode in the SK Series design addresses both transfer characteristic non-linearities as well as specific parameter non-linearities such as differential gain, differential phase and ICPM.

The "SK" IF modulator contains all of the non-linearity correction circuits and performs its task in a manner similar to a klystron modulator, including the special requirements of the klystrode.

Thus, the SK design objective of non-complex linearity correction is accomplished using techniques based on standard klystron modulators. Complexity is reduced and reliability is thus enhanced.

III. POWER CONSUMPTION

Comparison

An interesting exercise can be carried out that compares the projected cost of operation for various klystrode transmitters of the "SK" line against the best pulsed klystrons available today. The assumptions for this comparison are as follows:

1. 18 hours per day operation
2. 365 days per year
3. Mid channel efficiency of the pulsed klystron of 70%
4. Mid channel figure of merit for the klystrode 123%
5. 10db aural visual ratio
6. Normal losses for both RF systems and ancillary equipment

Table 1 is a tabulation of "SK" Series plant power consumptions compared to the power consumption of similarly rated pulsed klystron plants. These numbers are typical and are not intended to represent absolutely the best performance at any specific channel of either the klystrode or the klystron. They are for illustrative purposes only. They do, however, dramatically illustrate the power savings to the UHF station owner of the klystrode system.

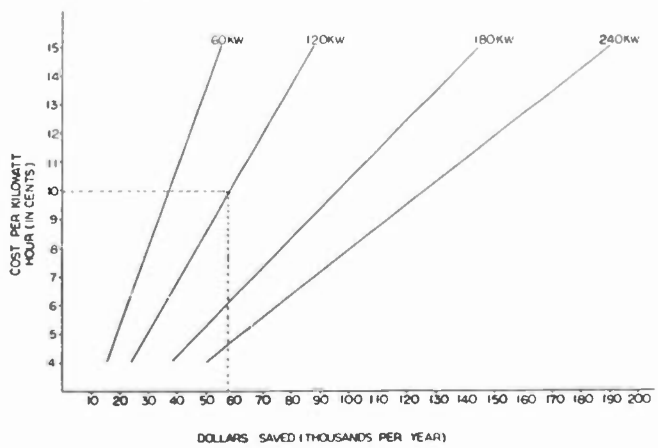
TABLE 1

PEAK OUTPUT	PLANT POWER CONSUMPTION (KW) KLYSTRON	KLYSTRODE
60kW	140kW	83kW
120kW	250kW	160kW
180kW	395kW	245kW
240kW	520kW	325kW
360kW	760kW	475kW

Courtesy of G. Morton - Comark

The data of Table 1 can be used to create a useful graph (Figure 7) showing the cost savings of a klystrode plant over a pulsed klystron plant for various power costs per kilowatt hour and RF power output.

Figure 7 can be used to show that, for example, a 120kW klystrode transmitter will cost approximately $55,000.00 less, per year, to operate than a pulsed klystron rig of similar power output when a.c. power costs are 10 cents per kilowatt hour. Other comparisons can also be drawn by changing power cost or RF power outputs and re-entering the graph of Figure 7.



Courtesy of J. DeStefano, Comark

Figure 7 - Operating Cost Savings Klystrode vs. Klystron vs. Power Cost

(Dotted line example is for a 120kW transmitter at $.10/kW hr. showing operating cost savings of $58,000 per year)

IV. RELIABILITY

The power savings of the klystrode are of course based on the assumption that the klystrode will have similar lifetimes to the klystron. While this may at first seem like a rash assumption, the limited data from present tests and established theory does not dispute the hypothesis.

Further, the lower complexity of the klystrode support systems will increase the mean time between failure of the overall transmitter.

Since every broadcaster is principally interested in the reliability of their air time signal, the klystrode's overall system is expected to perform at pulsed klystron reliability levels as a minimum. The "SK" crowbar system provides a level of protection for all high voltage components far above present day klystron transmitters.

Therefore, while only time will tell, the "SK" Series klystrode transmitting system is offered to the broadcast industry with the firm belief that the UHF station owner will receive high quality broadcast service at a significantly reduced long term cost of ownership.

V. TYPICAL 120KW SK KLYSTRODE SYSTEM

Figure 8 is a block diagram of the 120kW system scheduled to be installed at Wrens, Georgia in early Summer 1988. Features include a Magic Tee RF output system which permits operation at -3db from rated power in the event of a failure in one side of the parallel visual system. This is accomplished without the use of mechanical switch contacts. Emergency multiplex operation is also available through the use of just one mechanical switch. In addition, the failure of the aural klystrode can be compensated by operating one of the visual klystrodes as an aural and remaining on the air at 50% power.

The SK Series 120kW transmitter depicted in Figure 8 will be the first application of klystrode technology in broadcasting. The features and precautions in the design should insure long-term reliability and low cost of operation.

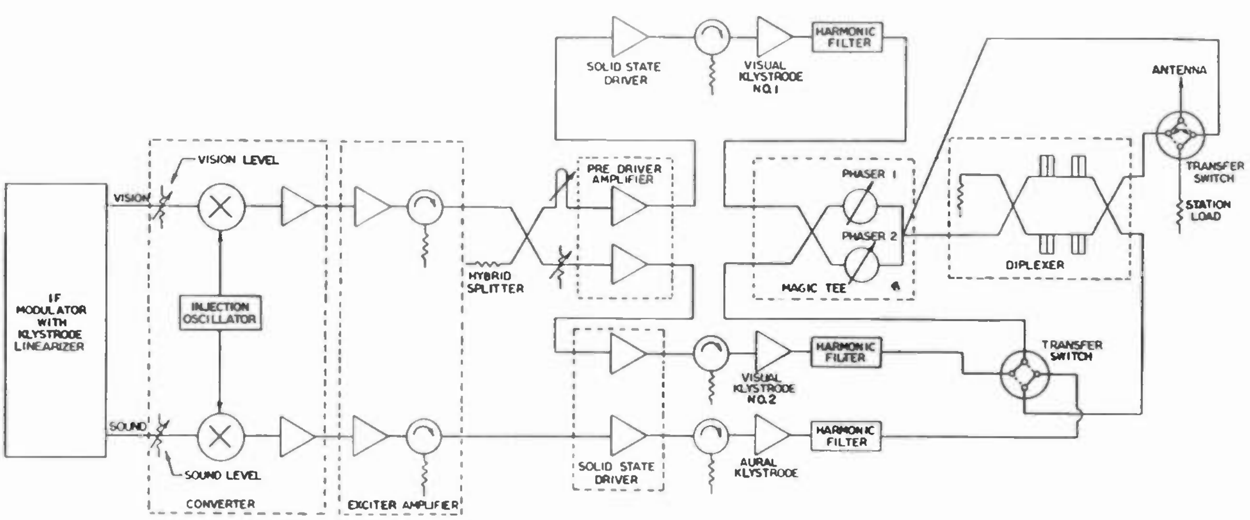


Figure 8 - 120kW SK Transmitter Block Diagram

VI. CONCLUSION

This paper has discussed specific design objectives that are necessary to successfully utilize the new klystrode tube in full service T.V. broadcast transmitters. Issues such as reduced complexity, RFI/EMI, solid state drive and tube protection have been addressed.

The cost savings offered by the klystrode as implemented in Comark's SK Series of equipment was also explored. Specifically, a chart was presented where the approximate cost savings for various output power levels and energy costs could be derived. The cost savings was based on a comparison with currently available pulsed klystron transmitters of equivalent power ratings.

As stated in the paper, the SK Series klystrode transmitter line at power levels from 60kW to 360kW is offered to the broadcaster with the firm belief that the system will prove to be both reliable and extremely cost effective.

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CIRCULARLY AND ELLIPTICALLY POLARIZED UHF TELEVISION TRANSMITTING ANTENNA DESIGN

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Over the past few years the utilization of circular or elliptical polarization has gained considerable acceptance in UHF television broadcasting. The purpose of this paper is to outline the trade-offs in various design considerations involved in the design of circularly and elliptically polarized antennas. Three different antenna configurations, with their particular advantages or disadvantages, are also compared.

I. ANTENNA TYPES

Over the years there have been three major antenna types developed for UHF television which provide circular or elliptical polarization. These are the normal mode helix antenna, various panel type antennas and the interlaced traveling wave array. The three types are described below.

A. The Normal Mode Helix

The normal mode helix consists of a supporting tube, made of structural material such as steel, and helical radiators supported around the tube with insulators. The entire antenna is broken up into sub arrays with each sub array fed from a feed system. A typical UHF television broadcast antenna might consist of anywhere from three to five sub arrays. This particular antenna is called the normal mode helix because radiation occurs normal to the axis of the helix, or perpendicular to the support tube. The antenna provides an omni-directional pattern. The polarization is circular with good axial ratio, ie the vertical and horizontal components are about equal and are 90^{o} out of phase as received by a viewer.

This particular type of antenna is reasonably simple to design and manufacture. The disadvantage of the helical radiator design is that it is reasonably fragile because the fairly thin helical conductors are exposed.

As the antenna only consists of three to five sub-arrays its elevation pattern is not quite as flexible as it might be if it consisted of more elements. In addition the azimuth pattern of the antenna is limited to omni-directional only. (Since the helix has circular symmetry mechanically it is limited to producing a circularly symmetrical azimuth pattern.)

B. Panel Antennas

This configuration is used primarily in Europe where multiplexing of several transmitters into one antenna is common.

The panel type antennas are easily modified to produce circular polarization by the addition of vertically polarized radiators to the already existing horizontally polarized elements.

The advantages of panel antennas are that they provide for a very broad bandwidth so it is possible to utilize one antenna for a number of channels by multiplexing the output of several television transitters into one transmission line and one antenna. A panel type construction also offers reasonable flexibility in producing various azimuth pattern configurations. By selecting the appropriate number of panels located around the tower and the proper phase and amplitude distribution to these panels, one may produce a variety of azimuth patterns ranging from omni-directional to various directional shapes such as one approximating a skull pattern or tri-lobe or peanut. Figure 1 and Figure 2 depict two such azimuth patterns.

The major disadvantage of panel antennas is that they require an extensive power distribution network consisting of a great number of transmission line segments with connectors which are vulnerable to failure. Typically a large number of panels are needed since in most cases a three or four sided tower needs to be covered with panels on all sides to produce an acceptable azimuth pattern. There may be up to 8-12 panels per tower face required to produce the needed elevation directivity, adding up to 36-48 panels per antenna. To achieve the proper aziuth pattern, the phase and amplitude distribution among the panels facing in the various directions on the tower faces needs to be closely maintained. When a complex azimuth pattern shape is required, this may prove to be rather difficult in the field. Changes that may occur at a later date, such as damage to a panel, icing, replacement of a damaged cable, could change the azimuth pattern of the antenna since conditions that existed to produce the original pattern may no longer be there after the changes have been made.

Mechanically the antenna itself is fairly light but produces a large windload because it requires that a rectangular or triangular tower section be completely covered with panels. The surface exposed to the wind by a panel antenna is many fold greater than the surface which is presented to the wind by a traveling wave or normal mode helical design.

C. Interlaced Traveling Wave Array

The third type of antenna which at this point in time is most commonly utilized is the interlaced traveling wave array. As the name implies the radiating elements producing vertical and horizontal polarization are interlaced along the array. See Figure 3. The traveling wave notation means that the energy is inserted into the antenna at its bottom end and is extracted by the radiating elements as it moves toward the top. By the time the energy reaches the upper portion of the antenna most of it has been extracted and what is left is radiated by the last two or three elements in the proper phase and amplitude.

A typical antenna consists of a cylindrical tube, ie a coax, which supports the radiating elements. The radiating elements are slots for horizontal polarization and dipoles which produce vertical polarization. The slots are cut in the outer surface of the co-axial line and the dipoles are located so that each slot has its dipole equivalent closely spaced to it.
See Figure 3.

The traveling wave type antenna offers rugged construction with a pressurized radome cover and a windload which is lower than the windload of a panel array. The radiating elements couple RF directly off the main input line so a power dividing network and distribution system is not used. This increases reliability. The design offers a great deal of flexibility in azimuth and elevation patterns including the possibility of producing different azimuth patterns for horizontal and vetical polarizations. See Figures 4-7.

This design is not adaptable for multi-channel operation, which is about the only disadvantage that it has.

II. SELECTION OF POLARIZATION

Circular polarization requires equal horizontal and vertical components. Provided there is sufficient transmitter power and antenna gain to achieve this combination, circular polarization should be selected. Often there is not sufficient transmitter power to produce circular polarization. In these cases a judicious choice must be made between the optimum ERP for horizontal polarization and vertical polarization. Often the criteria is that the maximum ERP of 5,000 kW must be produced for the horizontal polarization and whatever excess transmitter power is available will be radiated in the vertical polarization. This choice, although desireable for marketing purposes, may not be the optimum from the technical and actual performance point of view.

For example, reducing the horizontally polarized ERP by 1dB, or 20%, will reduce the calculated B coverage contours only by a distance of 1 mile. In practice this does not measureably affect the station coverage. However, diverting 20% of the available transmitter power to vertical polarization will produce a vertically polarized field intensity which is 1/2 of the horizontally polarized field intensity. This will measureably improve close-in coverage. The added vertical polarized component is predominantly useful for users who are not using an outdoor receiving antenna but rather rely on the rabbit ears or vertical whip antennas on the television sets.

III. DESIGN CONSIDERATIONS

A. Azimuth Pattern Shape for Horizontal and Vertical Polarizations

The first choice that confronts the designer is whether or not the azimuth pattern of the antenna for horizontal and vertical polarizations will be identical or somewhat different. Utilizing an azimuth pattern shape that is essentially identical for both polarizations will produce either circular or elliptical polarization throughout the coverage area in all azimuth directions. Although at first glance this option seems to be the optimum choice this is not always true. Consider the possibility where the population distribition is such that close-in coverage is required only in a specified azimuthal sector. This may be the case where the antenna location is not central to the population center. In such a case excellent close-in coverage may be required only toward the population center, whereas coverage to far outlying areas may be required all around the transmitter site. For a situation like this an antenna with an omni-directional azimuth coverage for horizontal polarization combined with a directional azimuthal coverage for vertical polarization might be optimum, see Figure 4.

Once the desireable azimuth pattern shape has been determined for both polarization, the actual development of the radiating element that will produce the required azimuthal coverage could begin. In order to produce optimum performance the radiating element must fulfill a number of requirements.

B. Coincident Phase Center of Vertically and Horizontally Polarized Elements

Obviously the element must produce the proper azimuth pattern shape for both polarizations. In addition to that the phase centers of the vertical and horizontal polarized radiating elements must be coincident or very closely spaced in the azimuth plane. Further, in order to maintain a good axial ration as a function of elevation angle, the phase centers of the radiating elements must also be either coincident or very closely spaced in the elevation plane as well.

A typical criteria would be that the azimuth phase centers of the radiating elements be coincident within 0.04 wavelengths and the elevation phase centers of the elements be coincident within 0.4 wavelengths. Using the proper design techniques and radiating element configurations it is possible to meet these requirements.

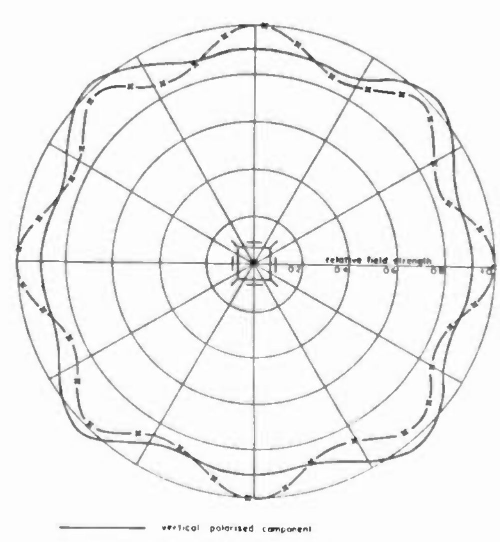
C. Proof of Performance Measurements

To ascertain the performance of the design configuration requires careful measurements. Azimuth pattern measurements of the combined horizontal and vertical polarized radiating element set is mandatory. The measurement technique which will best depict the overall performance of this element is the one that utilizes a rotating linear source, while the radiating element under test is rotated about a vertical axis. This measurement technique will immediately indicate any problems with phase center displacement, incorrect phase shift between the horizontal and vertical polarized elements, or incorrect pattern shape of either one or both elements. Figure 8 is the result of such a measurement. This pattern depicts vividly the overall azimuth pattern of the vertical and horizontal polarized components, the power split between them, and indicates that the 90° phase relationship has been maintained throughout the azimuthal coverage. As you see, the received power level has never dropped below the power level radiated by the vertical component, nor has it gone above the power level radiated by the horizontal component. This indicates a truly elliptical polarization with the proper vertical to horizontal polarized field ratio and the 90° phase shift between them.

SUMMARY

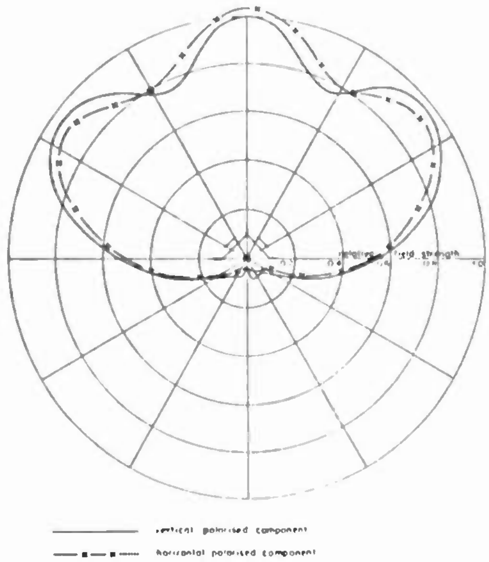
Circularly or elliptically polarized UHF television transmit antennas may be designed to provide a variety of azimuthal coverages. The azimuth patterns of the horizontal and vertical polarizations need not be identical. For optimum performance a 90° phase shift between the vertical and horizontal polarizations must be maintained. To fulfill this criteria the vertical and horizontally polarized elements and their interrelationship must be carefully evaluated and proven by pattern testing. The pattern testing is best done in an anechoic chamber free of extraneous reflections.

FIG. 1



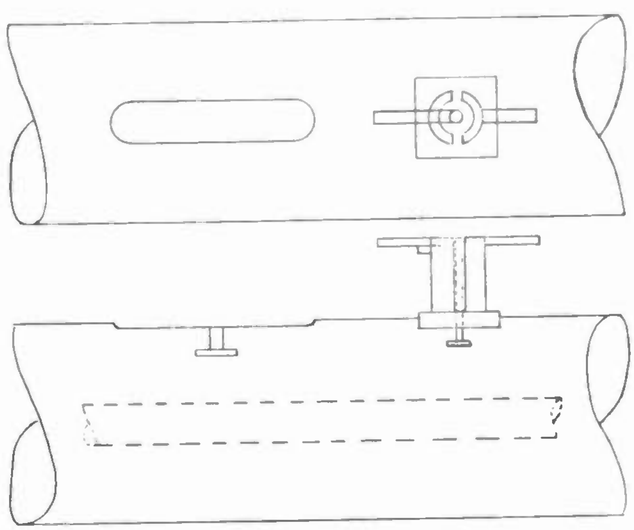
HORIZONTAL RADIATION PATTERN
H-POL OMNIDIRECTIONAL
V-POL OMNIDIRECTIONAL
FOUR SIDED PANEL ARRAY

FIG. 2



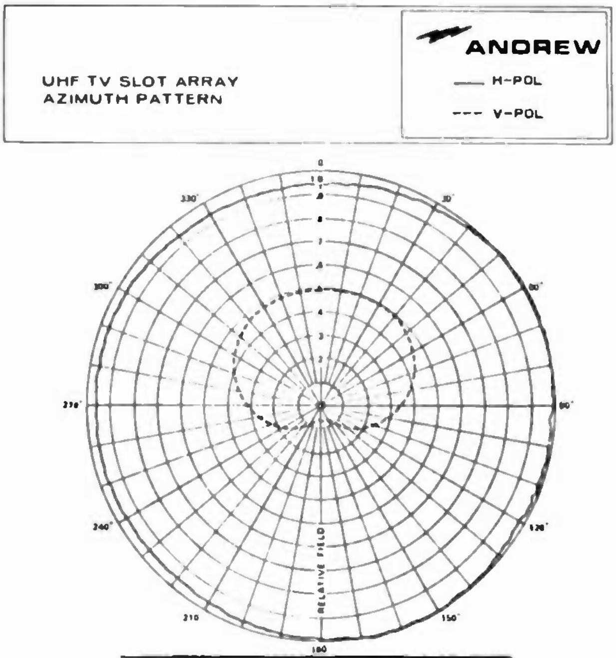
HORIZONTAL RADIATION PATTERN
H-POL BROAD CARDIOID
V-POL BROAD CARDIOID
TWO SIDED PANEL ARRAY

FIG. 3



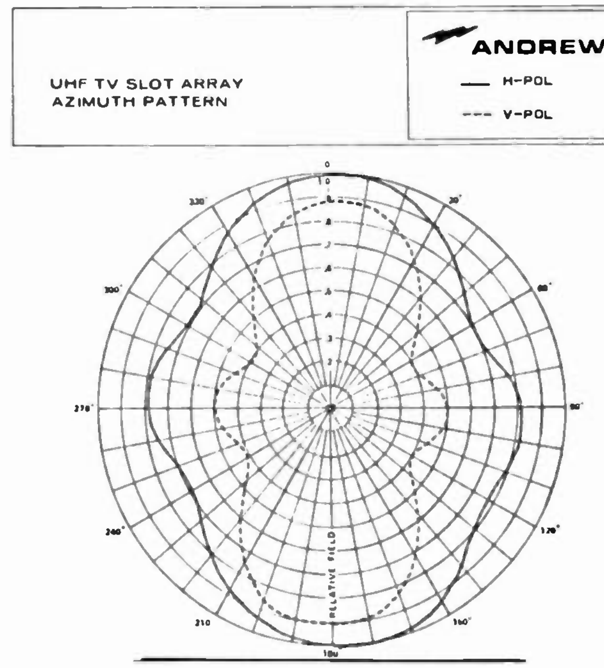
INTERLACED SLOT-DIPOLE PAIR
PRODUCES CIRCULAR OR ELLIPTICAL POLARIZATION.

FIG. 4



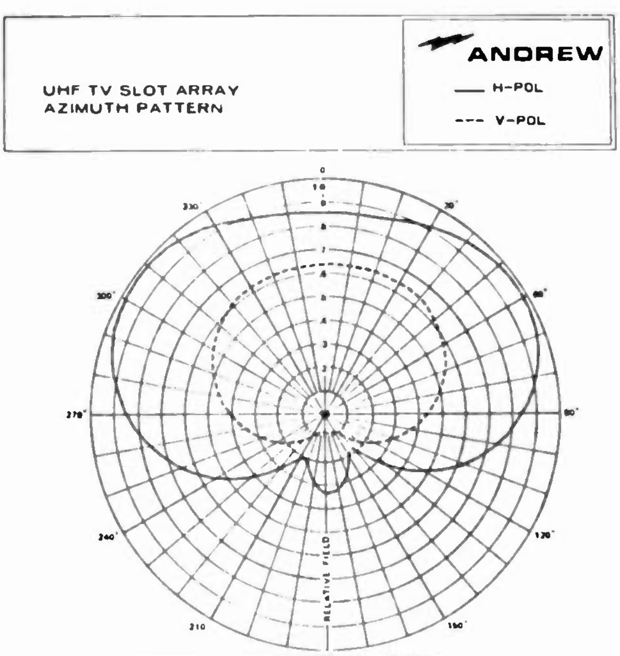
HORIZONTAL RADIATION PATTERN
H-POL OMNIDIRECTIONAL
V-POL CARDIOID

FIG. 5



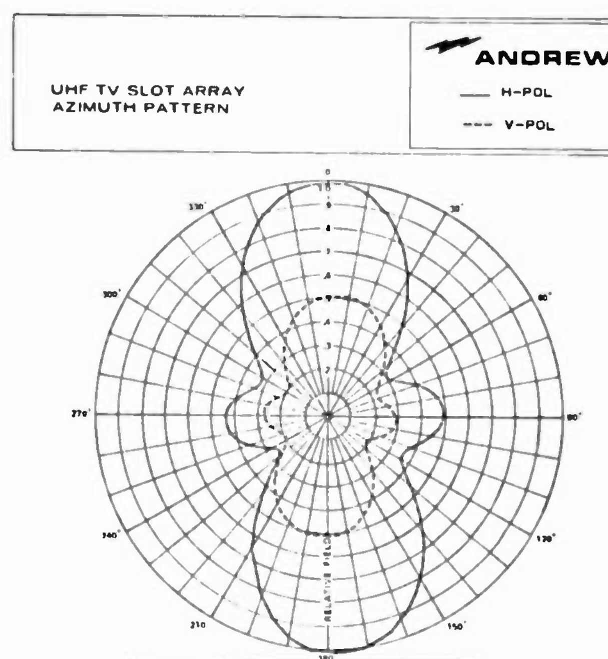
HORIZONTAL RADIATION PATTERN
H-POL BROAD PEANUT
V-POL NARROW PEANUT

FIG. 6



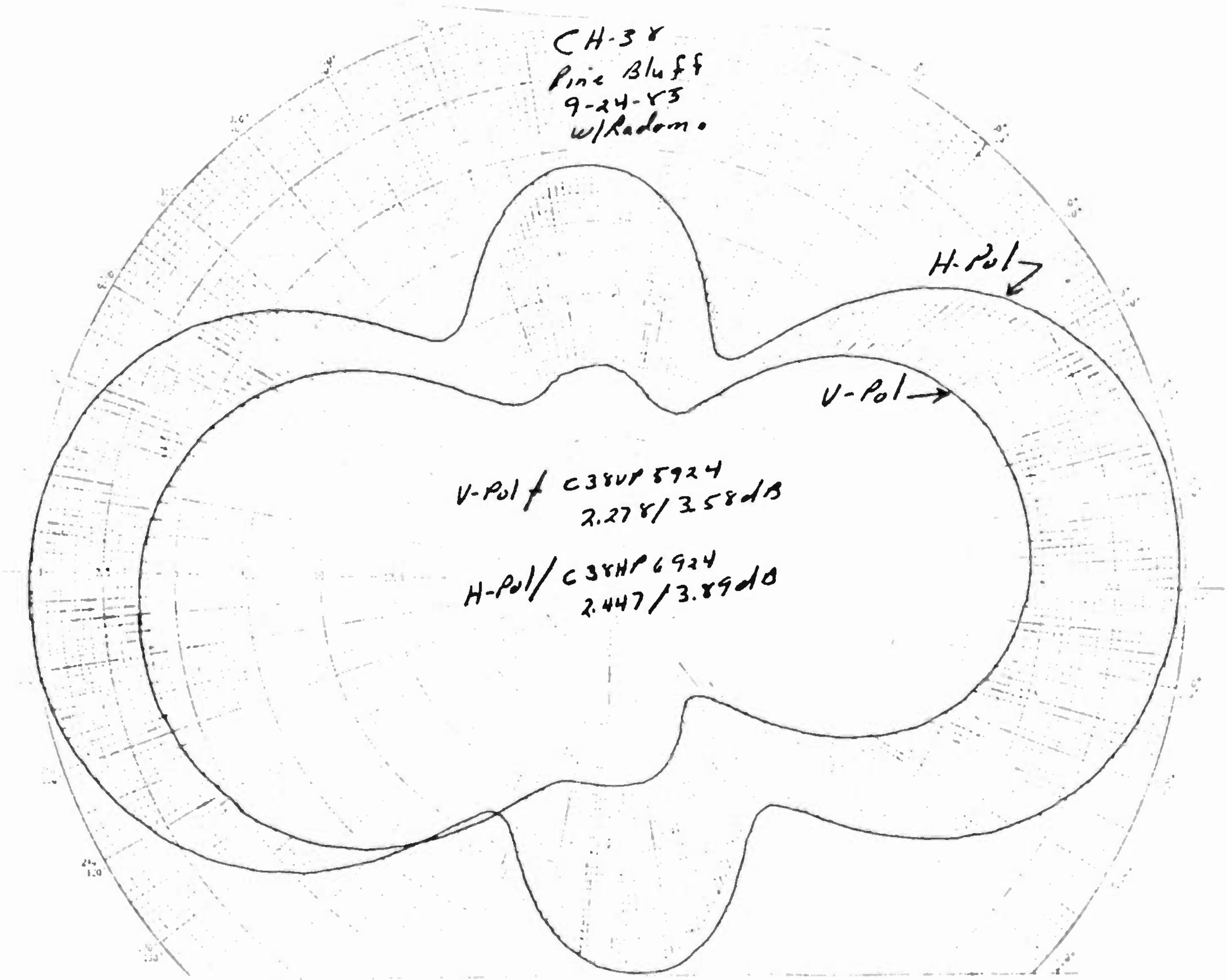
HORIZONTAL RADIATION PATTERN
H-POL BROAD SKULL
V-POL CARDIOID

FIG. 7



HORIZONTAL RADIATION PATTERN
H-POL NARROW PEANUT
V-POL ROTATED PEANUT

FIG. 8



HORIZONTAL RADIATION PATTERN
H-POL STANDARD PEANUT
V-POL MODIFIED PEANUT
MEASURED WITH A ROTATING LINEAR SOURCE

GD/djo
9000C

DEVELOPING ANTENNA PATTERNS TO MATCH DESIRED UHF TELEVISION COVERAGE

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INTRODUCTION:

There are many different type antennas available to the UHF Broadcaster; the new CP spiral antennas, the CP cavity, the CP ring panel, the traveling wave slot, the resonant slot, and the dipole corner reflector antenna. Each of these antennas are excellent and each one can outperform the other, depending on the location and application of the particular antenna. For example, developing an omni directional CP pattern antenna from a pylon slot type antenna, which by it's nature is a directional element, presents inefficiencies in the system. By the same standard, taking a CP spiral antenna which is naturally omni directional and trying to directionalize the pattern will create inefficiencies in that antenna. By utilizing the proper type of transmitting element in the beginning, we improve antenna efficiency, station coverage, and receptability at the same time.

There is little difference in the performance among properly engineered antennas. Element performance varies between 2%-5% in radiating characteristics. The proper application of the antenna element type with proper beam tilt, null fill, azimuth orientation, and antenna location represents 80%-90% of a systems performance. The remaining 10%-20% depends upon proper installation; did the riggers install the antenna properly. This paper will focus on the application (the 80%-90%), of "How to determine antenna parameters, to get the desired coverage". This is the most critical step in the selection of an antenna, and the most costly if not done properly.

In the selection of an antenna, both the technical and economic considerations vary, depending on the individual station. The practical application of antenna selection does not necessarily compromise the performance or desirable characteristics of an antenna system.

The capitol expense of a new antenna is small, compared to other equipment in the station. When amortized over 20 years or more, the antenna cost is very small. The most crucial part of any broadcast station is the antenna. If the signal doesn't cover the market properly, or radiate properly, the station can't be seen. The best studio equipment and the best programming cannot compensate for a signal that can't be seen. If you can't be seen - no one can watch.

By the practical design of coverage, we are able to increase the efficiency in market coverage, often reducing operating cost, while improving the market penetration. Practical antenna design and selection is based on (1) Antenna location in the community. (2) Location of the major population in the market (ADI). (3) Height, both HAAT, and Height Above Population. (4) Distance from the Antenna location to the nearest population center, and the furthest population center; at each radial. With this type information, we turn from providing just a pattern or antenna, to designing market coverage. Amazingly, a road map works well for preliminary layouts.

SELECTING AN AZIMUTH PATTERN:

The various antenna manufactures have numerous azimuth patterns to select. A customized pattern to fit the market place may also be necessary, many manufactures provide special designed patterns at no charge. When optimizing a pattern to a market place, we have to go beyond providing a relative field pattern, and plot the coverage in dB (on a road map see fig 1). The experienced broadcaster, isn't always aware that what appears to be a deep null in a relative field pattern is inconsequential when plotted in dB to terrain.

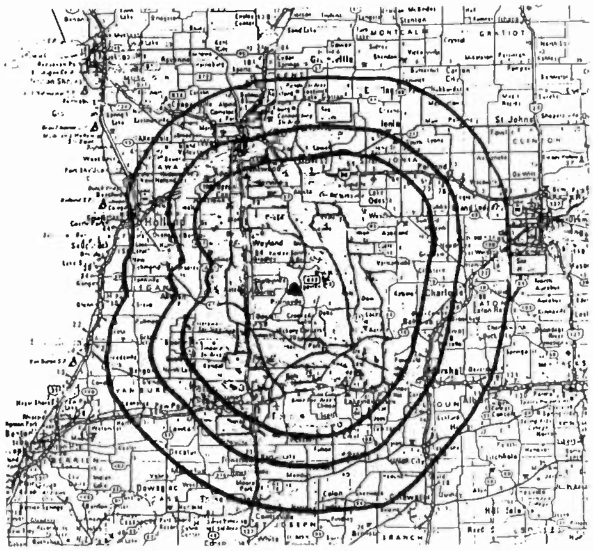


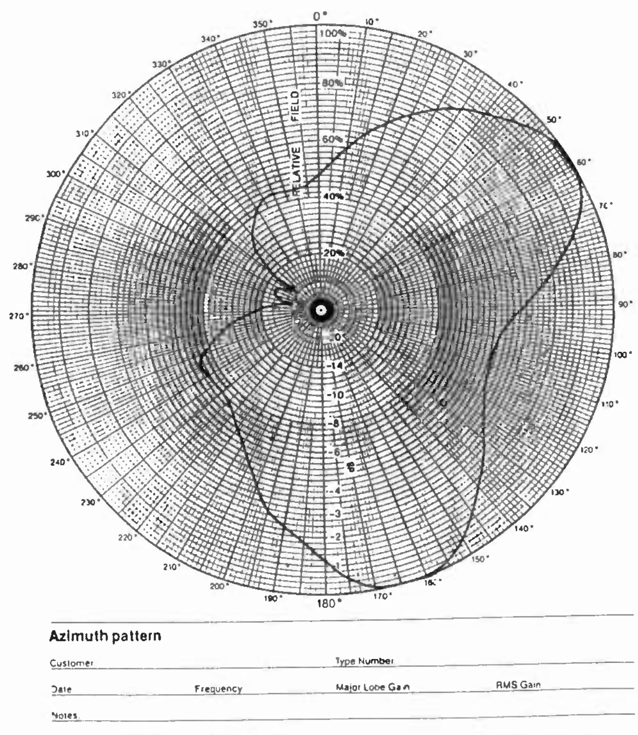
FIGURE 1

Using azimuth patterns with directivity of 2.0 or greater, a station can still retain a substantial ERP, (Eg X Ag = Total gain. Eg<32> X Ag<2.0> = Tg <64>. Power in X Tg = ERP, Pi<50kw> X Tg<64> = 3,400kw ERP). Gains of this type keep the transmitter size down and also reduce operating costs for the station. (Figures 2 and 3 show special patterns in relative field and their dB equivalent.)

The FCC 50-50 curves provide our City Grade (80 dBu), Grade "A" (74 dBu) and Grade "B" (64 dBu) contours, when plotted to terrain. The 2-10 mile HAAT, throw many broadcasters a real <u>CURVE,</u> in determining their actual coverage. In many instances, it's like putting one foot in a bucket of ice water and the other foot in a bucket of boiling water; on the average you'll be uncomfortable! The point to point elevations between the antenna center of radiation and the population you're trying to reach will provide a more realistic expectation for coverage, provided there is line of sight.

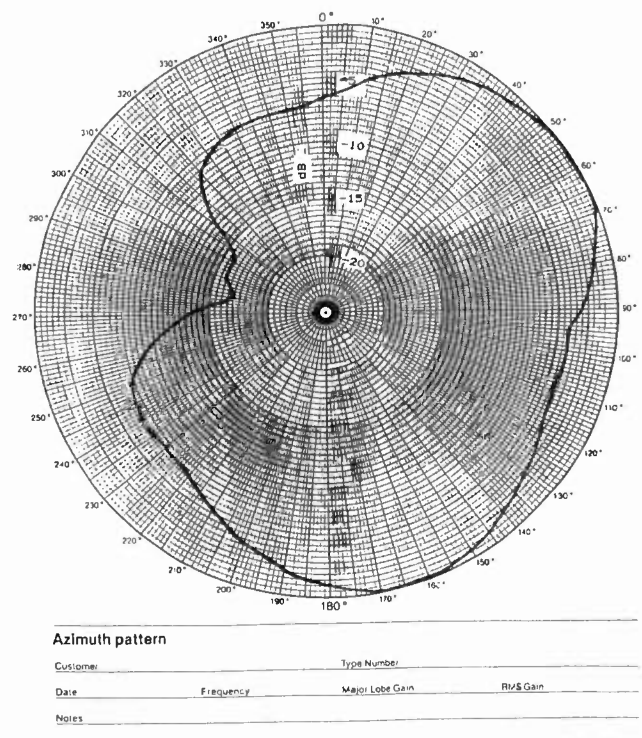
ELEVATION PATTERNS - DIMENSIONAL APPLICATION:

Our examples are the typical high elevation gain antennas, from 24 to 32 vertically stacked bays common in UHF service. These particular antennas have a narrow beam width from approximately 3 degrees to 1.5 degrees at -3dB (see fig 4). Because of these narrow beam widths, having sufficient null fill in the first and second null can produce a dramatic improvement in reception in homes located near the antenna site, and it is



JAMPRO ANTENNAS, INC.

FIGURE 2A



JAMPRO ANTENNAS, INC.

FIGURE 2B

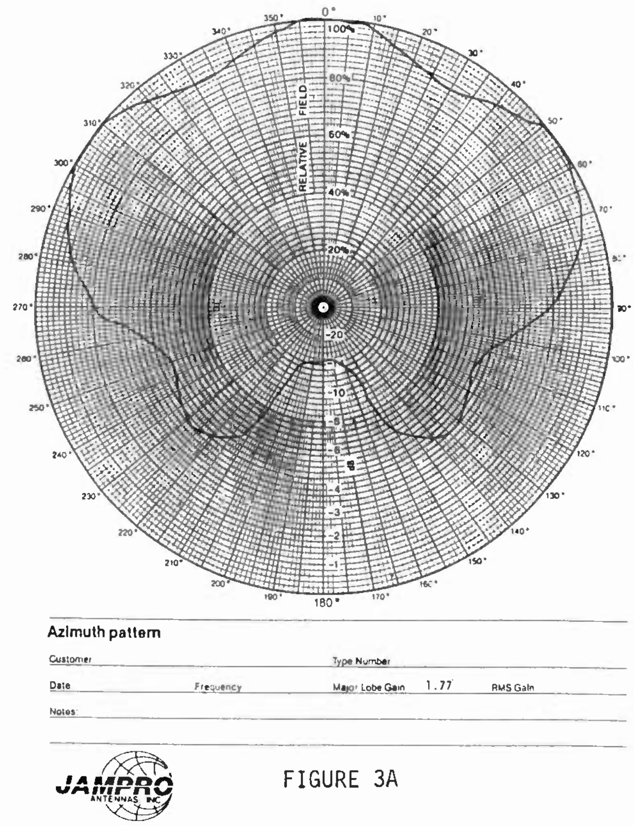


FIGURE 3A

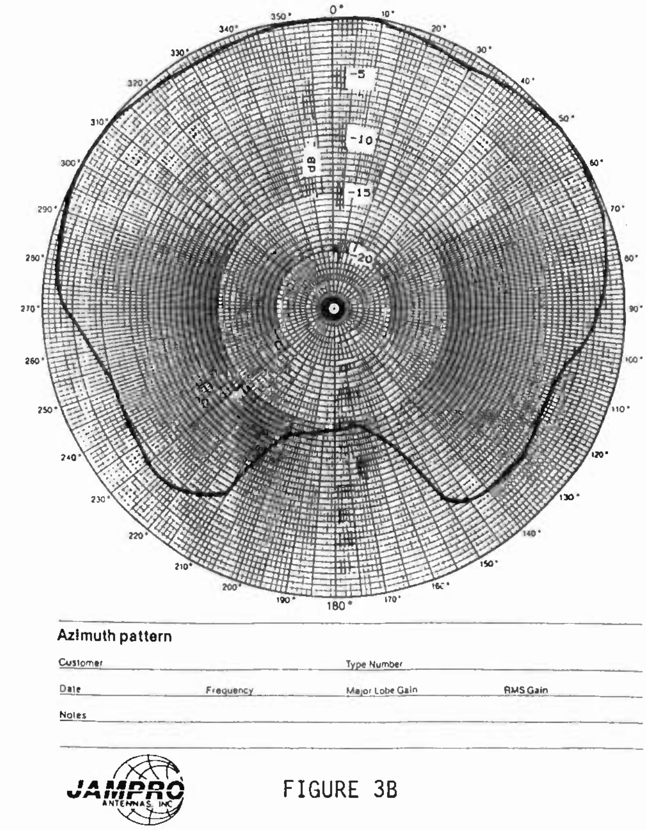


FIGURE 3B

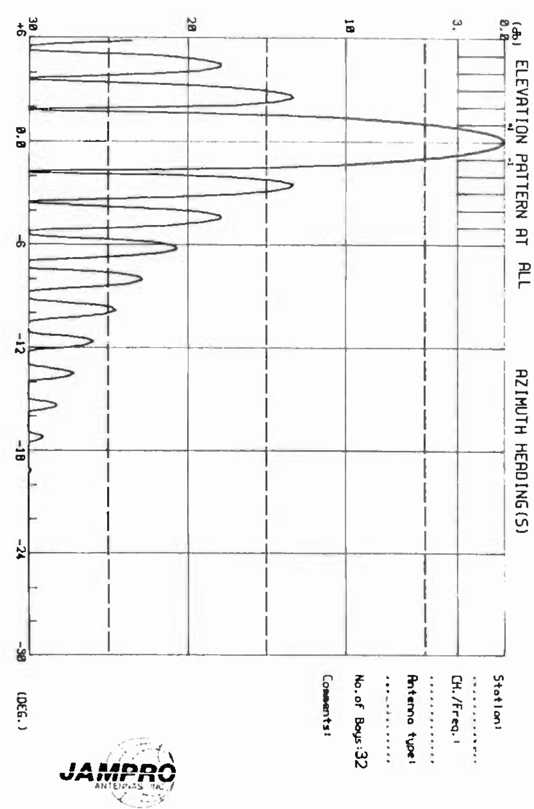


FIGURE 4A

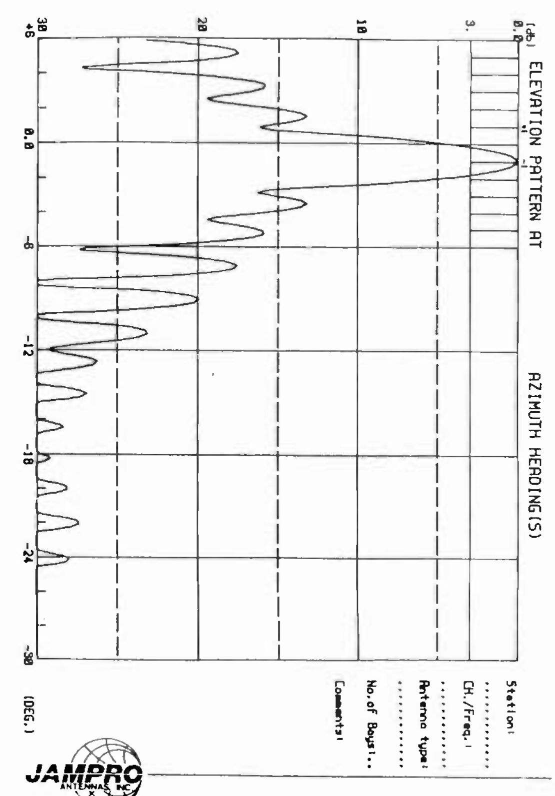


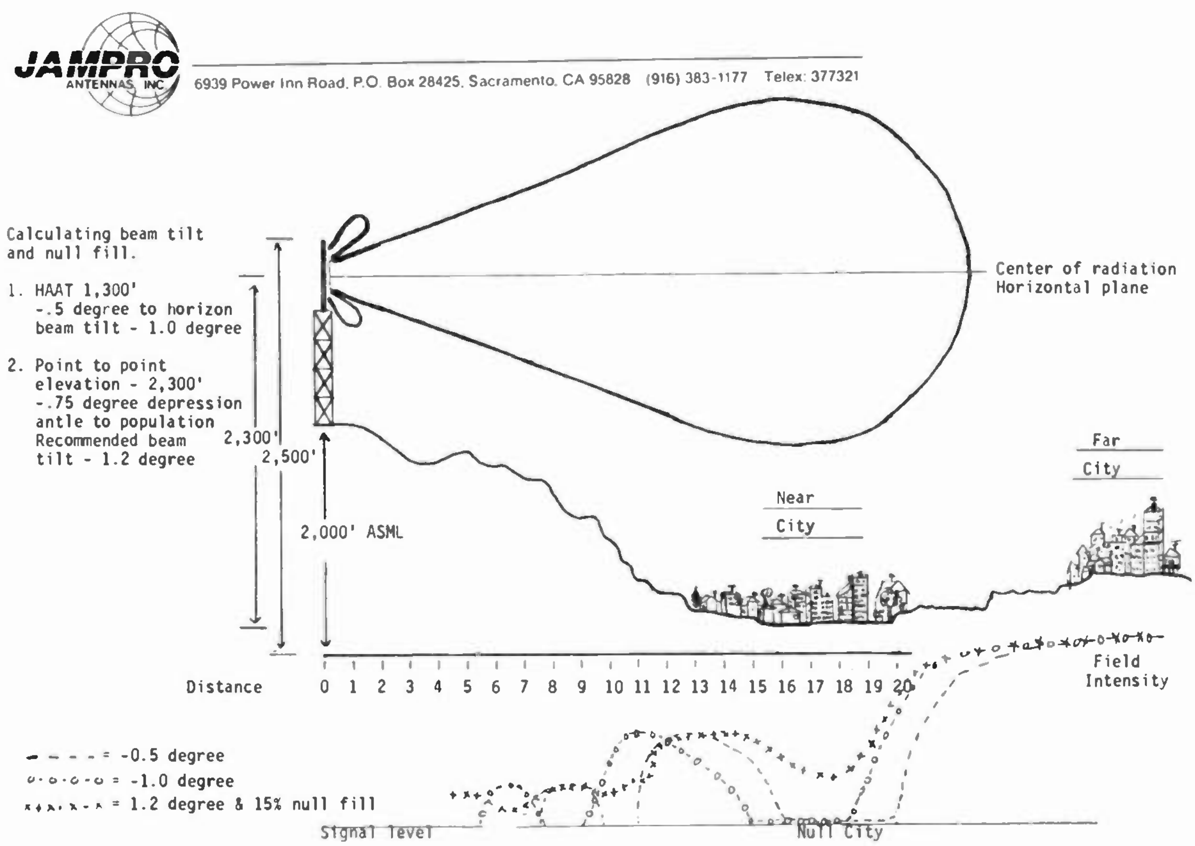
FIGURE 4B

particularly effective when the home is substantially below the elevation of the antenna. In this situation, you would want to calculate the depression angle from point to point to determine where the null falls (see fig 5), and then calculate the null fill necessary to maintain a solid signal level.

Proper beam tilt and null fill in an elevation pattern can either make or break a station. Historically the major lobe of the elevation pattern has been placed at the horizon. Therefore, approximately 50% of the station power has gone above the horizon, substantial amounts of power never coming near a receiving antenna. I have yet to find any one living above the horizon. Even placing the major lobe 1-2 degrees below the horizon may not be suitable. Using the HAAT to calculate the depression angle, can cause the signal to overshooting the principle community. By determining the point to point depression angle, adequate beam tilt can be applied to get the signal down to the population. Many of these problems do not occur in flat terrain, but become severe in mountainous areas where the antenna site has an HAAT that does not correlate to distance and the elevation above the communities being served.

An antenna located high on a ridge over a valley with population up and down the valley presents a need for differing amounts of beam tilt up and down the valley. In figure 6, community A and C requires -1.0 degree beam tilt for optimum signal, and at community B the signal will over shoot with -1.0 degrees of tilt. Community B needs -1.5 degrees. By applying -0.5 of mechanical beam tilt toward community B, and -1.0 degrees of electrical beam tilt we achieve the desired affect, of -1.5 degree tilt at community B and -1.0 at communities A and C. The beam is laid down in a uniform manor up and down the valley. The affects of this type tilt at 180 degrees from each point is just the opposite. For example, 180 degrees from community B, the beam tilt will be -0.5 degrees, and moving toward community A and C the tilt will increase to -1.0 degree. This method works well in the western mountains where there is little population behind the antenna, and often that population is higher in elevation than the valley floor. (See table 1 for depression angles. See table 2 for beam tilt on high gain antennas.)

FIGURE 5



HOME RECEPTION:

CAN CIRCULAR OR ELLIPTICAL POLARIZATION HELP?

When an acceptable antenna pattern (Elevation and Azimuth) has been selected, then determining if circular or elliptical polarization will benefit the station is the next task. This answer is as much or more a marketing decision by the station management, than a technical decision. There has been much discussion about the use of Circular polarization, and the reduced vertical vector, Elliptical polarization. The question is a practical one, "What does CP or EP buy the station?" The answer is situational; it depends on the location and what the station is trying to accomplish. Many people are looking to CP and EP for the wrong reason.

There is only one reason to use CP; To increase receptability of a signal. The difficulty in defining the benefit of CP and EP is not in the transmission but in the reception of the signal. This is an area which the station has little control. The viewer uses everything from the finest television receive antenna to wire coat hangers for antennas. An analysis of what happens between the transmitting antenna and the television set provides many answers as to whether a station will benefit from CP or EP, and which could be most beneficial.

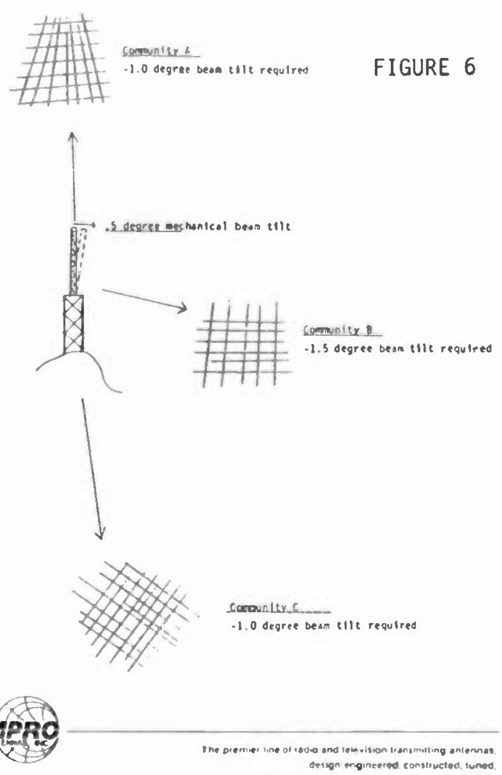


FIGURE 6

The Power received (Pr) at a set is the function of several factors, the Transmitted power (Pt), the free space Loss (Lfs), the mismatch loss between free space and the antenna and the antenna and the receive lead in (Lm), and the Polarization loss (Lp) due to improper antenna axis, alignment, and polarization skewing. The formula is Pr = Pt - Lfs - Lm - Lp. There are only two parameters which a station has control over, 1) the transmitted power,and 2) the polarization. The other parameters, Lfs is a function of distance established by physical laws, Lm is controlled by the viewer and the type of antenna and lead in used. Viewer education can help, but using the elements at our control can make a difference.

First the transmitted power, the station can operate up to the maximum ERP allowed by the FCC in the horizontal plain (H-pol) and using CP, can have the same power (ERP) in the vertical plain (V-pol). So we have a limit on power, but in effect it can be doubled using CP. At this time, we'll combine CP and EP reception theory. We'll examine the merits of each in just a moment.

The second parameter we control is polarization, but just in the transmitting antenna. The viewer controls the receiving antenna, and it is through the use of CP and EP we can regain control of the signal at the receiver. There are several types of receiving antennas, the outdoor horizontal (yagi), the indoor loop, rabbit ear (dipole), monopole, and the CP receiving antenna. At this time the CP receiving antenna is not in practical use.

The outdoor horizontal antenna has varying degrees of gain and directivity, but it's a horizontal antenna. The introduction of a CP signal into the plane of the horizontal antenna has little effect in improving reception where the signal is strong, but if the signal is strong you don't require the improvement.

In the fringe areas, where reception is marginal, the signal has traveled some distance, the probability is that the horizontal signal has been altered by terrain and ground clutter putting a tilt (or skew) in the polarization. This would affect the polarization loss on the receiving antenna, where power in the vertical from the CP would compensate, and the loss would not be seen.

Therefore, in marginal and fringe areas, the use of CP would negate the polarization loss in the outdoor horizontal antenna. Since, the tilting of the signal can be affected by the climatic conditions as well as terrain, the use of CP should stabilize fringe area reception. This can be very beneficial in keeping or bringing new counties into an ADI, and will economically benefit the station.

The indoor antennas can be "looped" into one category, a linear polarized antenna. Lay it flat it's Horizontally polarized, stand it up in the air vertically and it's Vertically polarized. This is true of the Rabbit ear, Monopole, and the Loop. The difficulty with these antennas is the not just the design, but the user doesn't know how to install and orient the antenna. With rabbit ears the dipoles are normally set in a "V" position. It's neither horizontal or vertical. With the loop antenna it is normally set vertically and then twisted into various shapes to get some type of reception. It's the classic situation that FM stations had with car radios until CP FM solved the problem.

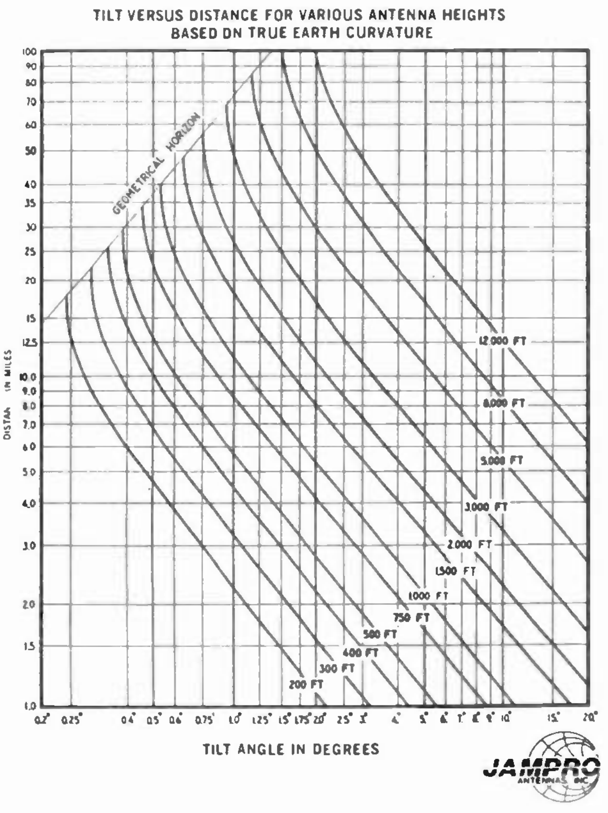


TABLE 1

Therefore, receivers using indoor antennas will benefit from the broadcast of a CP signal, and compensate for the polarization loss due to misorientation of the antenna. This is one of the most beneficial aspects of the CP signal, because of the increased number of secondary television sets, within a household not to mention portables, and hand held sets that are filling the market place. Many households have one set hooked to either the cable or the outside antenna, but there are often two or three secondary sets in the household that are viewed on a regular basis. This is the great untapped potential in the metropolitan markets of the country.

CP OR EP:

Selecting either CP or EP is an economic decision. It's not just a matter of operating a transmitter at twice the power to keep the same ERP, but it's a decision of population density, and ADI coverage.

CP is definitely the way to go with VHF. The low frequency and the long wavelengths of VHF lends itself to improved rabbit ear reception, and improved distance signal reception. Economically, with CP, a nominal increase in antenna gain and increase in transmitter size can accommodate the VHF station, and the power bill will still be reasonable.

To double the size of a UHF Transmitter is very costly. Not just the capitol cost but the operation cost of tubes and power. Therefore, careful consideration should be given to determining the necessity for improving fringe area reception by going full CP. For example if you have a large community on the edge of your Grade "B" signal full CP would be very beneficial. If the edge of the Grade "B" is rural or into another ADI, then CP may not be necessary. If on the other hand all the market was within the Grade "A", then using EP for indoor reception would be quite suitable.

SUMMARY:

The proper application of beam tilt and null fill to a selected azimuth pattern will provide excellent coverage. By examining the coverage plotted in dB to terrain, we get a general idea of the predicted coverage area. But when the coverage area is plotted in point to point elevation on selected radials, then we get a realistic approximation of the signal at that location. It's the beam tilt and null fill that insure the signal gets down to the population, and provide uniform coverage. No one lives above the horizon, and they never get diaries, so using the maximum allowable beam tilt is often the best approach.

Enhancing the reception is the second step after proper coverage design. Circular Polarization (CP) and Elliptical Polarization (EP) can improve receptability in the outlying areas where the signal is marginal, and in the metropolitan areas were secondary television sets use monopole, rabbit ears, and loop antennas. The CP and EP signal compensates for the poor performance of the indoor antennas and make it easier to receive a quality signal, a picture that is more stable and less susceptible to fading and interference. For the outdoor horizontal antenna it compensates for the skewed horizontal signal received at a distance from the transmitting antenna. CP and EP transmission will improve coverage and reception by masking the deficiencies in the receiving antenna.

All reception problems will not be solved by CP and EP. The debate on CP and EP improving reception by reducing ghosting, and reflections will continue, because each situation and location represents a different set of parameters, and we all have our opinions. Reception quality is as much or more a perceived value than a scientific fact. The final word is "IT MUST LOOK GOOD ON THE HOME TELEVISION SET".

Jampro Antennas has established several different types of antennas that meet the combinations of polarizations required by the broadcaster for optimum coverage.

	Model #:
Horizontal Only:	
1. Resonant Slot Antenna	JSM & JSH
2. Corner Reflector	JCR
Circular Polarized:	
1. Spiral Traveling Wave	JTC
2. Ring Panel	JRP
3. Cross Dipole Cavity	JSDP
4. Corner Reflector	JCR-CP
Elliptically Polarized:	
1. Cross Dipole Cavity	JSDP
2. Corner Reflector	JCR-EP
3. Resonant Slot Hybrid with Corner Reflector	JSM-JCR & JSH-JCR

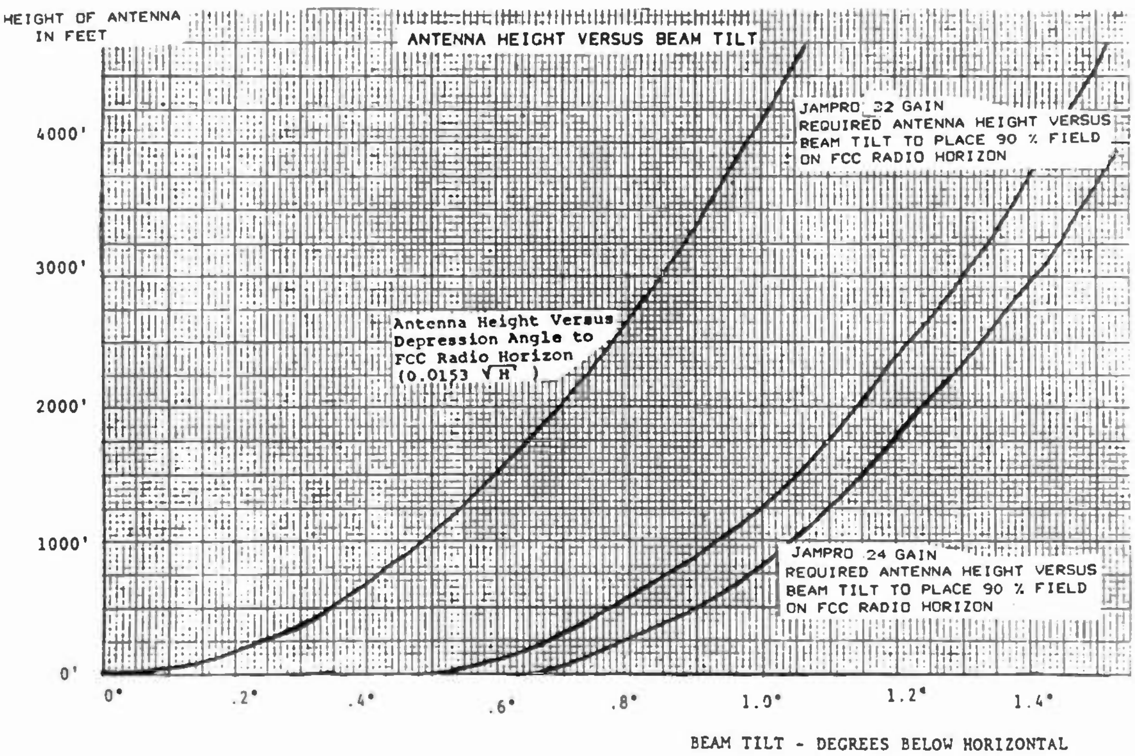

HEIGHT OF ANTENNA IN FEET
ANTENNA HEIGHT VERSUS BEAM TILT
4000'
3000'
2000'
1000'
0'
0°
.2°
.4°
.6°
.8°
1.0°
1.2°
1.4°
BEAM TILT - DEGREES BELOW HORIZONTAL
JAMPRO 32 GAIN
REQUIRED ANTENNA HEIGHT VERSUS
BEAM TILT TO PLACE 90 % FIELD
ON FCC RADIO HORIZON
Antenna Height Versus
Depression Angle to
FCC Radio Horizon
(0.0153 √H')
JAMPRO 24 GAIN
REQUIRED ANTENNA HEIGHT VERSUS
BEAM TILT TO PLACE 90 % FIELD
ON FCC RADIO HORIZON

TABLE 2

PLUMBICON* TUBE FOR HIGH-DEFINITION TELEVISION

Ad Franken
Philips
Eindhoven, The Netherlands

The dynamic resolution of an HDTV system is just as important as its static resolution. So camera tubes for such a system must meet requirements with respect to lag at least as severe as those for a conventional system. Moreover, the requirements for sensitivity are even more severe than they would be for a conventional system because of the broad bandwidth demanded of HDTV systems.

Plumbicon tubes, with their lead oxide layers, already have a proven reputation for low lag and high sensitivity. So it was an obvious step to develop a 30 mm Plumbicon tube specifically for HDTV applications.

The tube has a newly developed tetrode gun which generates high beam current and yet produces a smaller spot diameter than former guns. The tube also has electrostatic deflection and magnetic focusing.

INTRODUCTION

The current widespread interest in high-definition television (HDTV) is stimulating many developments in HDTV equipment in Japan, the USA and Europe. Many manufacturers, in fact, have actually started developing hardware to investigate the kind of imaging methods that will ultimately be preferred and to make practical assessments of the possibilities that HDTV can offer.

Many of these developments, of course, have centred around the new colour camera itself, and more especially around the imaging system to be employed, since it is this, probably more than anything else, that influences the resolution of a camera system.

There are two choices of imaging system: solid-state sensors and camera tubes. Although the performance of colour cameras employing solid-state sensors has been most impressive over the past few years, opinion as to their suitability for HDTV is currently divided, chiefly owing to the uncertainty of being able to produce a solid-state HDTV sensor at an acceptable price. For that reason, we decided some time ago to concentrate our efforts on developing an HDTV camera tube. This article briefly summarizes the results of this development programme.

* Registered trademark for television camera tubes

TARGET SPECIFICATION

The target specification we set for the tube (Fig.1) was based on consultation with several TV camera manufacturers and users.

One of the most important parameters for a high-resolution camera tube is modulation depth, which governs the tube's static resolution. Here, we set ourselves a target modulation depth of 40% for a frequency of 800 TV lines per picture height. This was an overall value that included the contribution made by the camera lens. Allowance for the degrading effect of the lens, therefore, meant that the tube itself should have a modulation depth of 50%.

Dynamic resolution or lag was, of course, another parameter we had to set, and with the demanding nature of so much present-day TV programme material, this could be considered at least as important as the tube's static resolution. Furthermore, experience indicates that HDTV systems are far more sensitive to lag than are conventional systems, so we set ourselves a rather stringent target here: a lag at least as good as that of the present 30 mm Plumbicon tube. The lag figures given in Fig.1 are for a low signal current of 20 nA, corresponding to high camera amplification - just the sort of situation in which lag becomes an important factor. The figures, therefore, are more appropriate to real situations and far more demanding than they would be if we had quoted them for a signal current of say 300 nA, a current at which it is comparatively easy to quote low lag figures. Moreover, it was decided to provide the tube with an internal light bias facility to allow improvement in lag by internal as well as external light bias.

Because of the much broader bandwidths needed for HDTV systems, the signal-to-noise ratio is around 10 to 15 dB lower than for conventional systems. This means that the requirements of high sensitivity and low output capacitance for an HDTV tube become even more critical than they are for a conventional tube. We set the sensitivity target about equal to that of the present 30 mm Plumbicon tube and although we didn't actually set a specific target for output capacitance, we made the proviso that it should be as low as possible.

Finally, we stipulated that the tube should be capable of delivering sufficient current for effective highlight handling and that its burn-in performance should be as good as possible.

Fig.1 Target specification for the HDTV tube

Modulation depth (800 TV lines/picture height)	
including lens	40%
excluding lens	50%
Decay lag (I_s/I_b = 20/300 A)	
after 3 fields	< 18.5%
Internal light bias facility (and possibility for external light bias)	
Sensitivity	
red	105 µA/lmF
green	145 µA/lmF
blue	36 µA/lmF
Low output capacitance (measured in the yoke)	
Highlight handling ability	

DESIGN PHILOSOPHY

Having set our target specification, we then had to settle on a design philosophy to meet it (Fig.2). First, to get the high modulation depth and high sensitivity required, we decided that a 30 mm tube would be the most appropriate.

We then had to choose the photoconductive layer. In view of the demanding requirements the HDTV tube would have to meet, the choice was naturally limited to lead oxide or selenium. We settled on the former because of its higher sensitivity, better lag and burn-in behaviour and better high-temperature performance. The new tube would therefore be a Plumbicon tube.

To get as small a spot as possible and to allow for a compact tube/yoke construction, we decided on magnetic focusing and electrostatic deflection. Electrostatic deflection has, moreover, the advantage that it allows for far easier registration correction. Its only drawback from the designer's point of view is that the electron-optical calculations are rather complicated and their implementation rather time consuming.

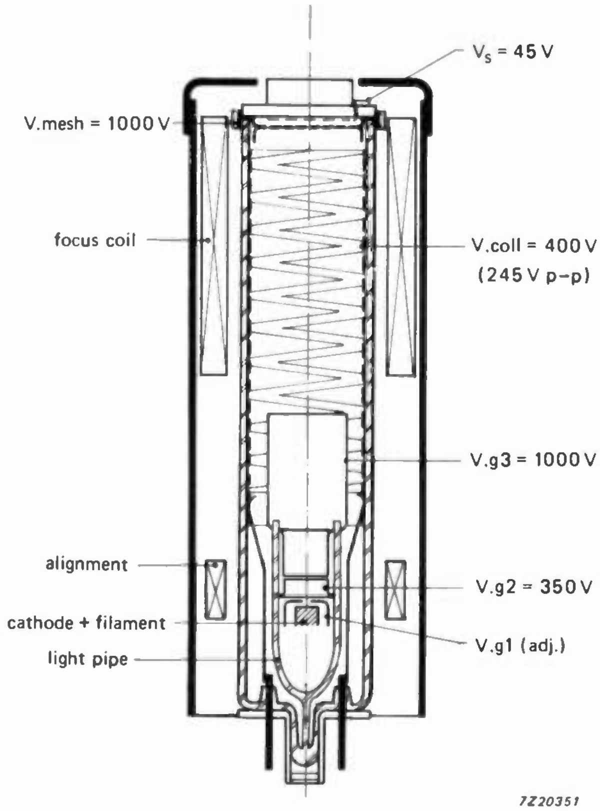
Finally, we decided to go for a completely new gun system: a tetrode gun rather than the diode or triode gun systems used in former Plumbicon tubes. This gun gives low beam temperatures (necessary for low beam-discharge lag), plus high beam current combined with a narrow beam angle, the latter giving good corner resolution.

Fig.2 Tube design parameters

Scan diagonal for high modulation depth, high sensitivity (30 mm tube, scan 10.5 x 18.7 mm)	: 21.4 mm
Photoconductive layer for high sensitivity, low lag, good burn-in	: lead oxide
Focusing/deflection for high modulation depth, short tube, small yoke diameter	: magnetic/ electrostatic
Electron gun for low lag, high available beam current, narrow beam angle (good corner resolution)	: new tetrode gun

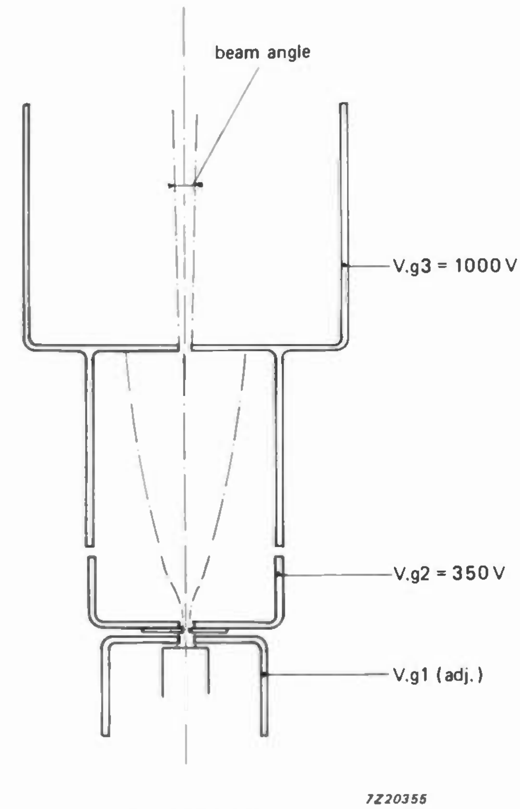
TUBE CONSTRUCTION

Figure 3 shows the tube (development number 89XQ) in its yoke. To minimize output capacitance, the area of the signal plate is only slightly larger than the scanned area. For the same reason, the mesh-to-layer distance was made as large as possible consistent with low beam bending. The tube operates with a mesh voltage of 1000 V and an average collector voltage of 400 V. The figure also shows the tetrode gun and deflectron pattern.



HDTV—PLUMBICON 89XQ

Fig.3

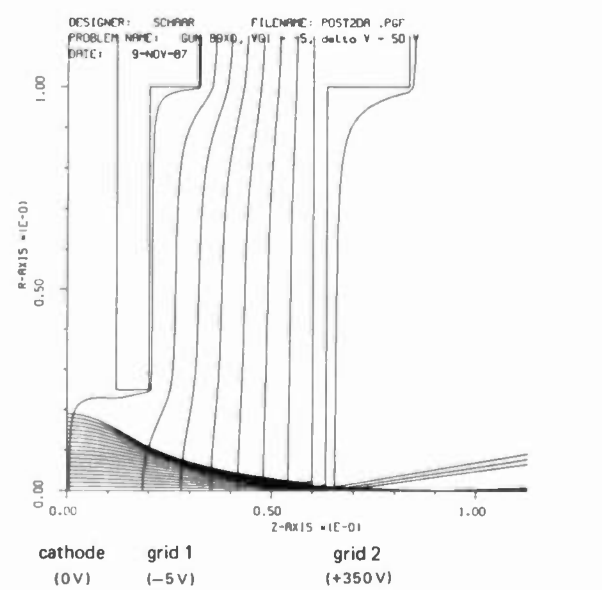


TETRODE GUN

Fig.4

Figure 4 shows the tetrode gun in more detail. The gun uses a 1.2 W impregnated cathode, and besides grids 1 and 2 found in conventional guns, there is an extra electrode: grid 3, maintained at 1000 V to restrict the beam angle and produce a very small spot on the screen. The new gun, moreover, allows the tube to be shorter than conventional tubes with no increase in beam landing error and distortion.

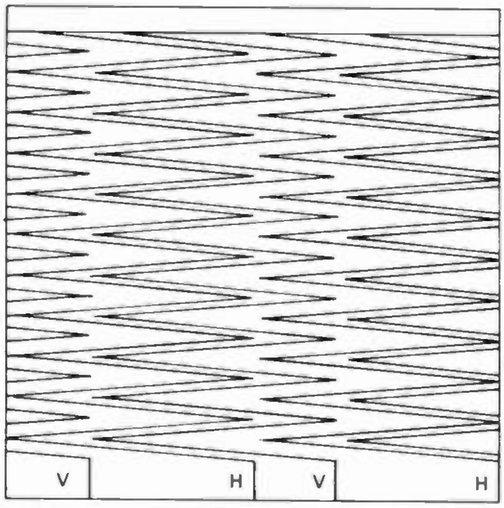
Figure 5 is a computer print-out showing the electron current distribution between grids 1 and 2. From this figure it's clear that the highest current density occurs in the neighbourhood of grid 2, and it's for this reason that the gun is capable of delivering a high beam current.



ELECTRON CURRENT DISTRIBUTION

Fig.5

The deflectron pattern itself can be seen in a somewhat simplified form in Fig.6. The tube is designed to operate with an aspect ratio of 16:9 and the width of the horizontal and vertical electrodes are in this proportion. The horizontal and vertical deflection voltages therefore are about equal at 245 $V_{p\text{-}p}$.



$V_{defl\,H} = V_{defl\,V} = 245$ V p–p (16 : 9 aspect ratio)

DEFLECTRON PATTERN

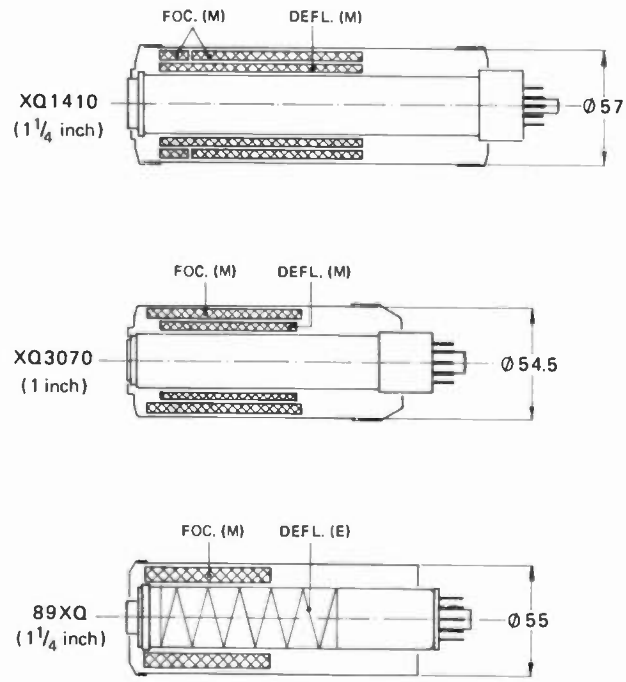
7Z20352

Fig.6

Figure 7 is a computer plot of the complete electrostatic field and an electron beam. The complexity of the electrostatic field in the neighbourhood of the deflection electrodes is quite evident from this figure.

PERFORMANCE AND POSSIBLE FUTURE IMPROVEMENT

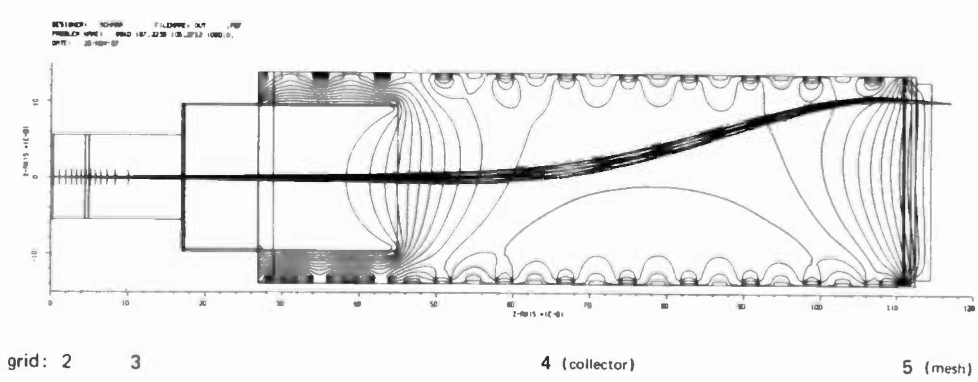
Figure 8 compares the size of the 89XQ tube with two conventional Plumbicon tubes: a 30 mm (1 1/4 inch) XQ1410 and a 25.4 mm (1 inch) XQ3070. It's clear from this figure that the new 30 mm tube is nearer in both length and diameter to the conventional 25.4 mm tube.



DIMENSIONS OF
1 INCH & 1 1/4 INCH
TUBE–YOKE COMBINATION

7Z20350

Fig.8



EQUIPOTENTIAL LINES WITH DEFLECTED ELECTRON BEAM

7Z20353

Fig.7

The significant improvement in modulation depth compared with the XQ1410 is shown in Fig.9, although there's still some way to go before the new tube reaches the target we set. (The modulation depth of the 89XQ was measured with a special High-Definition Amplitude Response test chart using slant burst patterns.)

Finally, from Fig.10 it's clear that with the exception of modulation depth (which is currently about 45% after correction for the lens), all targets have been reached. Further improvements can be expected in modulation depth, and with a redesigned tube configuration, we can expect to get significant reductions in output capacitance.

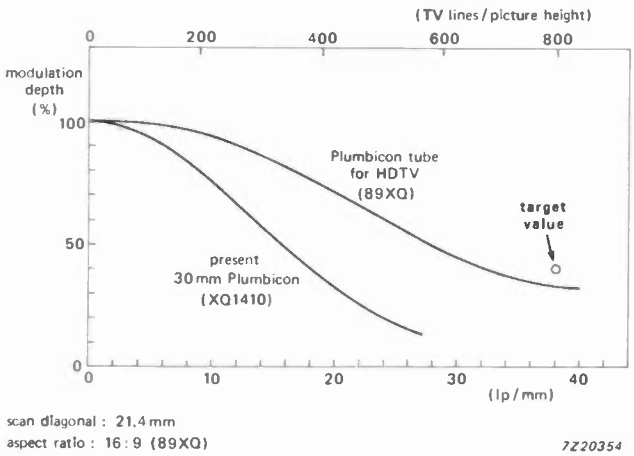


Fig.9

Fig.10 Development-sample data

	target	**test results**
Modulation depth (800 TV lines/picture height)		
including lens	40%	35%
excluding lens	50%	45%
Decay lag (3 fields)	< 18.5%	16%
Sensitivity (μA/lmF)		
red	105	120
green	145	145
blue	36	36
Output capacitance (pF)		
including yoke	low	4.7

CONCLUSION

To summarize, we can say that the new 30 mm 89XQ tube is well on the way to satisfying the requirements of coming HDTV equipment. In combining the qualities of high resolution, high sensitivity, low lag and excellent burn-in performance, the tube builds on the already well-established excellent reputation of the Plumbicon tube.

PROJECT MANAGEMENT IN BROADCASTING

Marvin C. Born
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Some major corporations in our industry are using management science techniques to enhance their operations. However, most medium and small market properties are not. The normal procedure to start any job in these smaller operations is to draw up a set of plans and go to work. These drawings and their preparation are thought to be planning, the complete planning. While the actual drawings of the placement and connections of the equipment are a critical part of the planning process, they are only a single part and not necessarily the most important part. What is not included in the drawings is who will supply the equipment, who will install it, when it will arrive and where it will be stored when it does arrive. The result of not including this information in your plan can be equipment that arrives too early costing both storage fees and interest charges. Labor may be improperly assigned, either too much or too little for a given task, or worse yet, the correct labor may be there, but no equipment or materials with which to work.

This paper looks at Project Management through the use of CPM (Critical Path Method), Pert Charts and Gantt Charts in the construction of major projects in the field of broadcast engineering.

CPM involves the use of a calendar which is broken down into a timetable with the starting and ending dates of each task. Assigning these tasks requires detailed knowledge of the complete job and detailed knowledge of the indivdual tasks, as controlling detail is the key to success in building any project. By assigning these dates one can order equipment at the proper time to control the arrival date and the installation date. The order date is actually entered as a task on the CPM chart. Proper planning results in a savings of both manpower and operating expenses. This paper explains the function of the Pert Chart, Gantt Chart, and resources calendar and how they were used to control the construction of a transmission facility project. This planning technique is known as Project Management and has been around for several years; however, its use has become widespread since being programmed on small desk top computers.

BACKGROUND

Thefirst question to ask is "What is a Project?" The MBA school definition of a project is a series of related tasks performed to reach a specific objective.The translation of that from business language to engineering language,tells us that the tasks must be related to be considered a project. An engineer who runs around his facility fixing one emergency problem after another is not working on a project, he is putting out fires, because the tasks he is working on are not related. Next the series of related tasks must have a beginning, middle and end. Building a transmitter building is a project, while operating the equipment in the building is not a project. If the tasks are developmental rather than operational, then the tasks become a project.

WHAT IS PROJECT MANAGEMENT

The field of project management is divided into four major areas, strategic planning, tactical planning, managing and evaluation. Each area addresses a specific need in controlling and managing the series of tasks which comprise a project.

Each of the areas should be given sufficient attention in detail to insure that the planner understands the tasks and how they function in each of the four areas. Strategic planning involves considering the the over-all impact of a project. Tactical planning involves looking at the details and arrangement of the various tasks, while management is the placing of both the strategic and tactical areas into operation. Evaluation is simply reviewing what has been completed with an eye toward the next time.

STRATEGIC PLANNING

The proper way to understand strategic planning is to stand back, look at the overall picture including both the present and future perspectives, and address these questions. "Is this a viable project?" "Does this meet the long and short-term needs of my company?"

"What would be the impact on the company if we did not do the project?" What is the benefits if we do the project? Strategic planning evaluates both the short and long-term needs of the company, translating those needs into terms of cost to the company, and asking, how will this project meet the present and future needs of the company.

TACTICAL PLANNING

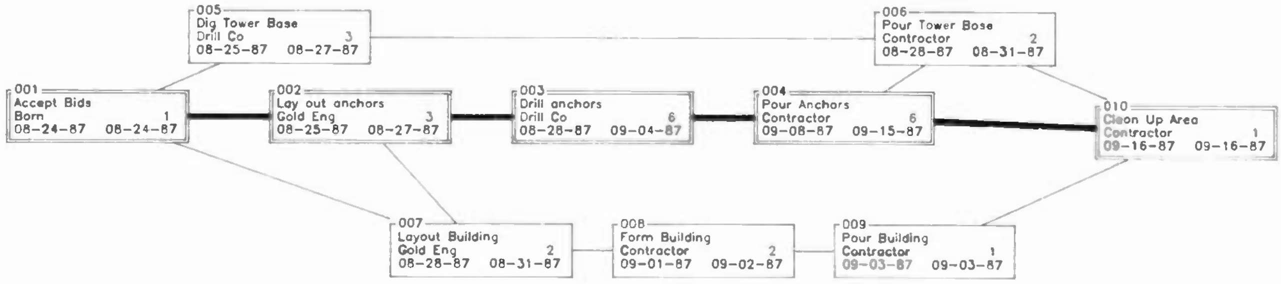
Tactical planning looks at the individual aspects of a project in detail such as what tasks are needed from start to finish, who will be the one to do the individual tasks and how long will these tasks take? How much will each task cost? What is the relationship of one task to another in time, equipment and people? Proper tactical planning requires detailed knowledge of each task and its relationship to every other task and the overall project. Computer project management is designed to automate the paper work of managing these relationships and is of great value in remembering details.

MANAGEMENT

To properly manage a project, we should implement the tactical planning by subdividing the various charts comprising the concept of Critical Path Method. By utilizing the information contained in the charts and from analysis of the relationships between the tasks, we can control the work with a minimum of lost time and money. Again a computer allows the charts to be revised quickly to try various ideas with minimum effort.

The Pert Chart The Pert Chart (figure 1) is a visual aid comprised of small boxes containing the name of the task, the resources required, the amount of time required to do the task, plus the starting and ending times of the task. After each task is assigned a box with the required information, the task boxes are linked in order of their occurrence. The concept of linking is the basis of project management and should be understood completely. The charts presented in Figures 1 to 4 are a reduced version of an actual program that was used to build the tower. Notice the double, darker line linking tasks 1 to 2 to 3....to 10. This is the critical path. Changes on this path will move the project completion date. The lighter line connecting tasks 5 to 6 to 10 for example is a non-critical path and minor changes may not change the critical path; however, changing the duration of a non-critical task can cause that task to move to the critical path, therefore sliding the ending time forward or backward. This movement is a useful tool in projecting "what if" cases such as bad weather, missing materials or labor problems. Major changes show up as spectacular movement among the tasks on the Pert chart. Some small modifications only change a single date and are not noticed as easily. Minor time movements are better seen in the Gantt chart.

Consider the sub-project of constructing the antenna tower bases. The first step was to accept bids. The resource allocated to this task was assigned to the chief engineer and the time required to open, evaluate and accept a contractor was one day. When this project was planned, three similar work activities were to be

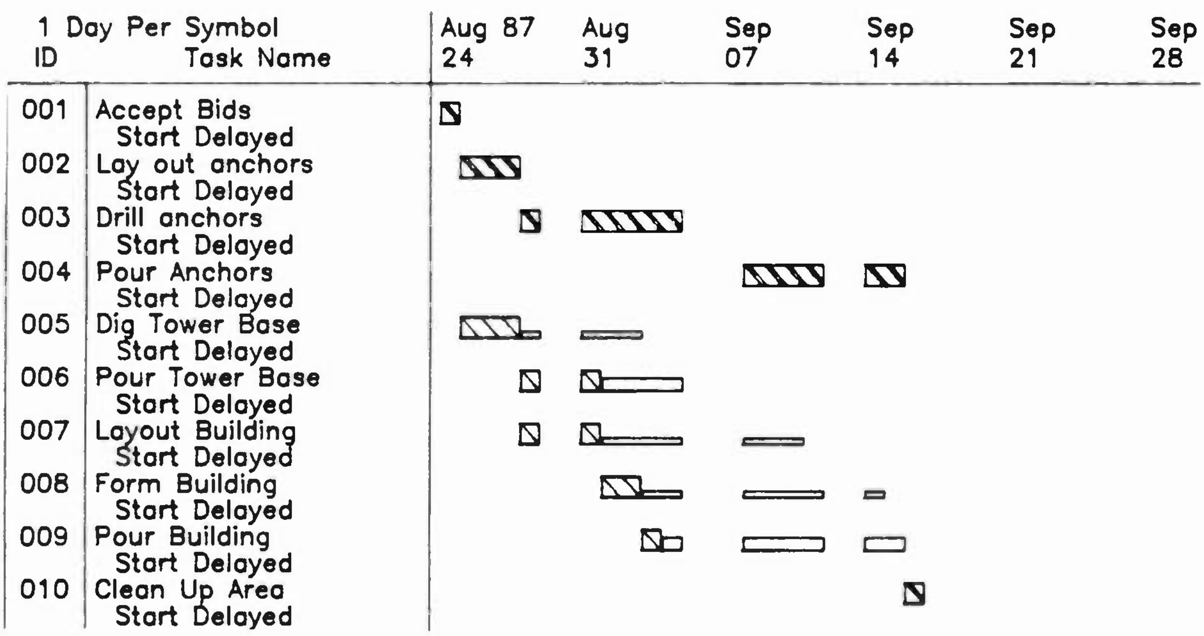


performed simultaneously. The anchors for the guy cables, the actual tower base and the building to house the transmitter were to be constructed simultaneously. The resources allocated were the surveying company (Gold Engineering Co), the contractor (Triangle Construction Co.), a drilling company (North Drilling) and the chief engineer (Born). The object of the planning session was to minimize the amount of time the various resources are on site and to assure that there are no time conflicts between the tasks assigned to a single resource.

The survey crew was to lay out the tower base and the guy anchors and later to lay out the foundation of the building. The drilling company was to dig the tower base with a back-hoe and to drill six anchor holes. The contractor was to pour the holes, the base and the building foundation, then clean the area in preparation for the tower erection. Using computer project management, each task was assigned a number, a resource, and the amount of time necessary to complete the task.

Each task was then linked to the next succeeding task. The computer then completed the Critical Path and non-critical paths for the project, thus completing the Pert Chart while checking for conflicts in resources and preparing a Gantt chart and a Resource Calendar.

Notice the "Form the building" (task 008) requires two days to complete as a non-critical task and (is represented by single slashed rectangles connected together) is followed by several float symbols. The project manager can quickly see that he can delay the forming of the building as much as a week without delaying the project. However, Task 009 which is pouring the building foundation starts one day later than task 008 and would be delayed by the same amount that task 008 is delayed. Task 009 has several days of free float available. Pouring the building foundation can be delayed up to seven days without delaying any part of the project. The Gantt charts allow a manager to see what he can do timewise to a specific task without disrupting the total project. Likewise, the manager can enter a change as a "what if" question to see what that change would do to the project completion time. Making a change then running the "update function" not only changes the Gantt chart, but updates the Pert Chart and the Project Calendar. Such changes can be made in either the Pert or Gantt chart data screens for a quick "what if" look. The planner can quickly switch screens to whichever is best able to display the answer to his question and then quickly return to the original.



The Gantt Chart The Gantt chart (figure 2) shows the scheduled time of each task on a time line basis. Each symbol in the chart represents a day or a number of days depending on the length. The symbol with the double slashed line represents days on the critical path and may not be changed without effecting the completion date. The smallest open rectangles are float or delay days which represent non-critical path tasks that can be moved or delayed.

The Project Calendar The project calendar (figure3) contains all holidays that apply to the project as well as weekends. Each day is assigned a number of hours to work, usually eight; however, the number of hours can be increased or decreased as needed on a daily basis and hours over eight can be assigned as overtime hours. Rain days can be inserted into the general project calendar and a new calendar can be generated with a new general schedule.

Calendar for: All Project Resources

1987	Sun 0	Mon 8	Tue 8	Wed 8	Thu 8	Fri 8	Sat 0
May	17 WKND	18	19	20	21	22 St road	23 WKND
May	24 WKND	25 Holiday	26 Road	27 Road Cont	28 Road	29 Road	30 WKND
May Jun	31 WKND	01 Fill Road	02 Fill Road	03 Fill Road	04 Lev Area	05 Lev Area	06 WKND
Jun	07 WKND	08 Drill #1	09 Dril #2	10 Drill #3	11 Drill#4	12 Drill #5	13 WKND
Jun	14 WKND	15 Drill #6	16 Dig Base	17 Dig Base	18 Dig Base	19	20 WKND

An individual resource calendar is assigned to each resource, and all information from the general project calendar is transferred to that calendar along with the exact work days from the pert chart. (Changes in the Pert or Gantt charts will automatically be reflected into these calendars.) An example of the value of calendars is the question, should I pay overtime to complete a task before the weekend or hold the crews over and work them straight time next week? Holding a crew over the weekend involves paying expenses for lodging and food, and it delays the project. However, paying time and a half or double time could be more expensive. The question of time vs money moves the discussion to yet another chart.

Project Details The Project Details Chart (figure 4) contains information with the billing rate for each resource, the overtime rate, and fixed expense costs for food and lodging.

With all the information for each resource entered in the calendars and project details chart, the answer to the question can be found quickly by generating a report using the overtime situation and generating a second "what if" situation using the "hold over" scenario and choosing the less expensive option. The calculations will be quicker and more accurate as the computer will catch that single task that must be completed on time when manual checking may overlook it.

Resource Details **Project: CPM-DEMO.PJ**
02-09-88 01:03 **Revision: 11**

Gulf Coast Broadcasting Tower Project

Rsrc Name: **Contractor**

Costs: **Begin**			Defaults		Totals	
Total Overscheduled:	0	Rate Mult: 1.50	Hours:	40	Var:	2400.00
Calendar Variance:	0	No. Units: 1	Fixed:	0.00	Fix:	0.00
			Rate:	25.00	Tot:	2400.00
			Priority: 50		Act:	0.00
			Allocation:	8x	Hrs:	96

	Sun	Mon	Tue	Wed	Thu	Fri	Sat
Workday	0	8	8	8	8	8	0

ID	Task	Pr	Hrs	Allc	Un	Ovr		Dur	Start	Finish	
006	Pour Tower Base	50	16	8x	1	0		2	08-28-87	08-31-87	
008	Form Building	50	16	8x	1	0		2	09-01-87	09-02-87	
009	Pour Building	50	8	8x	1	0		1	09-03-87	09-03-87	
004	Pour Anchors	50	48	8x	1	0		6	09-08-87	09-15-87	C
010	Clean Up Area	50	8	8x	1	0		1	09-16-87	09-16-87	C

EVALUATION

Evaluation is the final step in your project. Step back and look at what you have accomplished. Did you accomplish your goal? Did your resources produce and cost as expected? Were your time estimates close to the actual times. Pull out your original charts and compare with the final working charts and analyze your mistakes for future use. The evaluation stage prepares you for the next project. You gain insight into the value of services, people and equipment. Careful analysis of each person's performance reveals strong and weak areas that can be utilized, corrected or avoided as necessary. The time required for a piece of equipment to do a task should be compared to what a larger unit would cost to do the same task in a shorter time. The key to evaluation is to look for detail. Some useful information will jump out at you, other information is buried and you must dig for it.

CONCLUSION

Computer project planning is a tool that allows you to plan your work in detail. While this is useful, the real value is the ability to re-evaluate and re-plan as you go along and know what each adjustment will cost in time and money. You can evaluate multiple schedule options to arrive at the optimum plan for your project.

The average broadcast project will fit into a desk top computer with room to spare. Consider how handy your friendly computer will be with project management, a CAD program, word processor and spreadsheet installed . Your whole engineering department from design to scheduling to budgets can be handled on one machine. And, if you still have some time left you can read your mail box from the news department.

Acknowledgments:

Primavera Systems Inc.;Balla Cynwyd, PA

Computer Associates; San Jose, CA

Rebecca Born, MBA; Corpus Christi, TX

Equipment:

Hewlett Packard Vectra,

Compaq Plus

PROPOSED CAD DRAWING STANDARDS FOR RADIO AND TELEVISION ENGINEERING

Dr. Walter Black
Video Design Pro
Las Cruces, New Mexico

While equipment manufacturers have been quickly changing to more microprocessor controls and automation, the basic engineering functions of design and installation have largely remained manual. Many individuals have pioneered in using Computer Aided Design (CAD) to help mechanize the drafting function, and over 100 major groups are using VidCAD and AudCAD design engineering libraries, but demand for truly mechanized design engineering is only beginning. In order to avoid all of us going down costly dead-end roads, we must examine the issue of CAD drawing standards.

This paper is the result of an extensive training and integration program done for the USAF Broadcasting Service, under the Armed Forces Information Service. VDP has trained all major USAF engineering personnel, plus others from AFIS, AFRTS, Navy Broadcasting, the Pentagon, TASC Engineering and Voice of America. The workshops involved hands-on training in VidCAD and AudCAD, CableDOC (our automatic cable documentation program) and PathROUTER (which graphically represents signed paths, calculates H and S/C phase, and cable Bill-of-Materials). Initial training revolved around Generic CADD, but subsequent training was on AutoCAD Release 9. (Note: With the new pull-down menus of AutoCAD and the extensive redesign of the user interface by VDP, AutoCAD is now our system of choice in CAD engineering applications. It is far faster and even easier to use than other CAD systems for our present applications.)

From this work, we have developed engineering conventions and standards that can provide an initial platform for standards discussion.

WHAT IS A CAD STANDARD?

In an engineering drawing, typical CAD elements include:

- Symbols
- Placement rules
- Layer structure (similar to overlays)
- Colors
- I/O flow
- Text layout and size
- Plotting size/scale
- Legend and title blocks
- Cable numbering
- Cable run-list
- Bill-of-Materials
- Abbreviations

Of these, only I/O flow from left to right is widely accepted enough to be a standard, but even this does not address the complexity of the issues. For instance, is a 422 bus an input or output? How do you lay out re-entry systems where outputs loop back to inputs? Is camera triax or multicore an input or output?

There is fairly universal agreement on certain symbols, such as triangles for amplifiers or DA's, and > < symbols for jack fields, but the actual representation in size, line layout, etc., is not fixed.

Hardware/Software Limitations

Since CAD useage is largely constrained by the cost and quality of software and hardware, a few notes on this subject are appropriate.

Software: Most CAD packages offer up to 256 layers, usually only numbered. AutoCAD allows either numbers or text labels of layers of an unlimited number of layers. Text labels are most functional, and do not require a legend or key, but this does limit transfer of drawings to number-legend programs. Many programs also limit the number of components or blocks in a drawing, while others do not allow nested blocks or libraries over 500 items. Some also limit maximum size of components to 20K, which makes drawing a GVG switcher or ADM console very difficult. As a general rule, if the CAD program does not allow nested blocks, large blocks or large quantities of blocks, don't use it.

Hardware: EGA (640x350 pixels, 16 colors) and VGA (640x480, 256 colors) have become 2 defacto monitor and graphics card combo's for most PC computers. Consequently, most CAD buyers are encouraged to purchase these. We firmly believe that 800x600, 16 colors is the bare minimum for CAD applications, and we strongly recommend 1024x768. The power of boards, such as Verticom TwinFOCUS, that instantly zoom to between an over-all view and 1 or 2 interactive drawing windows, relieves the time an engineer waits for a drawing to regenerate from minutes to 1 second per zoom. Because drawing output is nearly doubled, the extra \$3-\$5000 cost is probably the best investment a company can make toward serious engineering automation.

Other Standards

In other fields, there are varying degrees of standards. In electronic and electrical engineering, a good and rigid list of standard components have evolved, and even rules for auto PCB layout have become well accepted. However, in architectural layout there are few totally accepted CAD standards, other than the traditional symbols for doors, windows, etc., and ANSI standard hatch and fill patterns. In terms of layer structure, Figure 1 lists a recommended standard for color selection and layer numbering based upon the CSI/AIA specification standards. In essence, the layers are numbered in groups of 10 and color number (e.g., Division 6--Wood and Plastics, user layers 60 to 69 and color 6). This is a simple strategy, and could be used in architectural jobs, but since all of our drawings would be forced into Division 10--Specialities and Division 11--Equipment, this standard is too restrictive for our needs.

PROPOSED STANDARDS

Symbols

Figure 2 shows samples of one of our components. We have 3 views for each piece of equipment-face, top and system block view. VDP component block names for each of these are 12 characters long, with the first 3 showing manufacturer abbreviation, 8 characters for model name, and the 12th character designating the view. For example, the names for the 3 views in Figure 2 are: The DOS nested view name can only be 8 characters, but AutoCAD allows longer internal names.

DIVISION	COLOR	LAYER #	CONTENT
1-General Requirements	1	10-19	Summary, standards, procedures, quality control
2-Sitework	2	20-29	Site preparation, paving, utilities, landscaping
3-Concrete	3	30-39	Formwork, reinforcement, accessories
4-Masonry	4	40-49	Mortar, unit masonry, stone
5-Metals	5	50-59	Structural metal framing, joists, decking, sheet metal, ornamental metal
6-Wood and Plastics	6	60-69	Rough carpentry, finish carpentry, plastic fabrications
7-Thermal and Moisture Protection	7	70-79	Waterproofing, insulation, fire-proofing, roof specialties
8-Doors and Windows	8	80-89	Doors and frames, entrances and storefronts, hardware, glazing
9-Finishes	9	90-99	Flooring, wall coverings
10-Specialties	10	100-109	Access flooring, fire protection, partitions, telephone, toilet and bath
11-Equipment	11	110-119	Library, theater, stage, audio-visual
12-Furnishings	12	120-129	Window treatment, furniture, multiple seating
13-Special Construction	13	130-139	Special purpose rooms, sound, vibration, recording instrumentation
14-Conveying Systems	14	140-149	Elevators, moving stairs and walks, lifts
15-Mechanical	15	150-159	Fire protection, plumbing, HVAC
16-Electrical	16	160-169	Service and distribution, lighting, communications

FIGURE 1. AIA LAYER AND COLOR SPECIFICATIONS

DOS Name of Nested Block	SONBVU82.DWG	Drawing conventions
System Block Name	SONBVU820--B	10" rectangle showing all I/O's 1" apart, signal name, connector, termination, on a 6" line, with .6" text (color by layer)
Rack Front View	SONBVU820--R	Real size, basic features shown, 1" text
Rack Top View	SONBVU820--E	This view will soon be replaced by a 3-D face view of the front

FIGURE 3. SYMBOL DRAWING STANDARDS

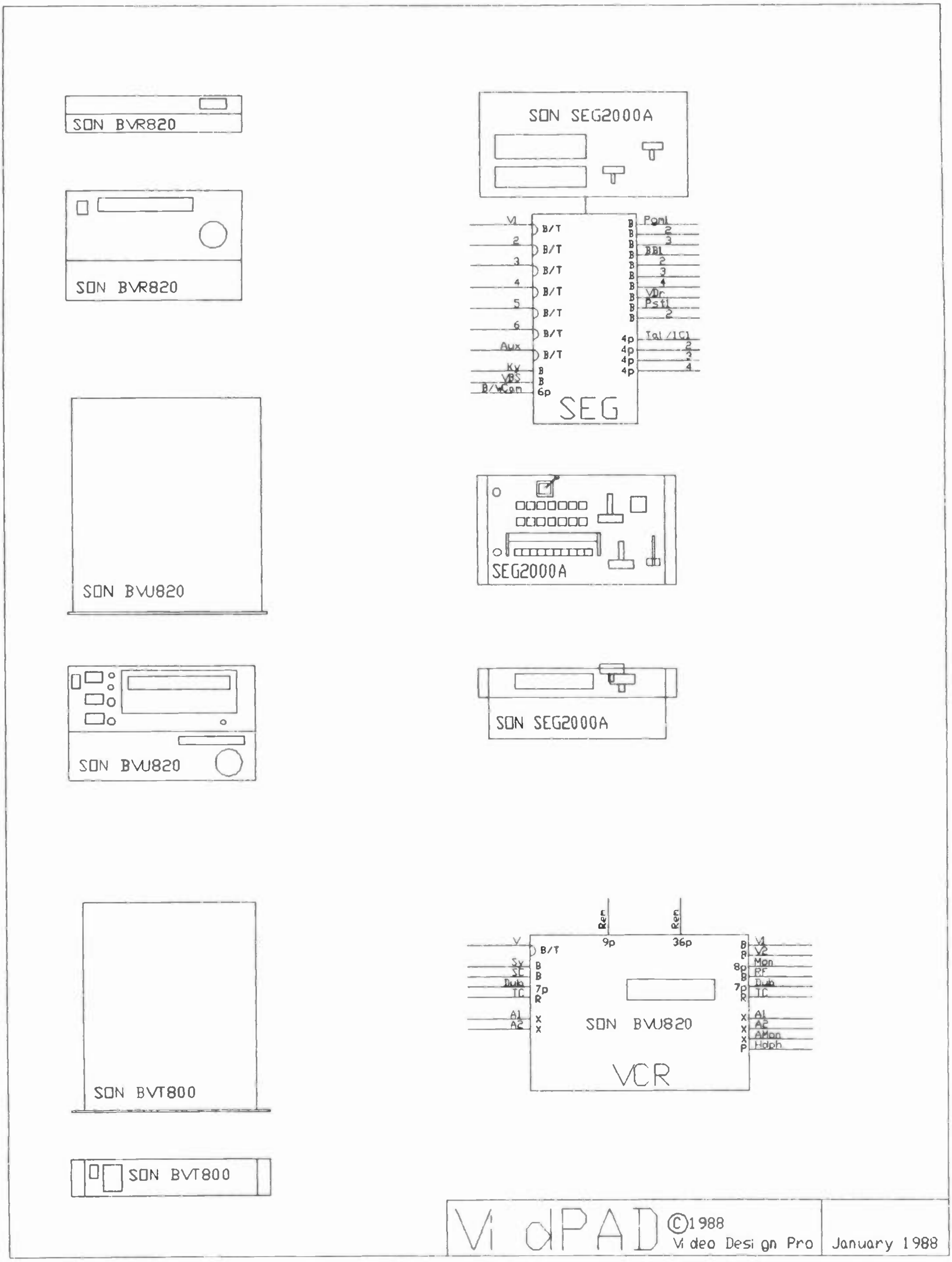


FIGURE 2. SAMPLE VidCAD SYMBOLS LIBRARY

Symbol Detail

We have chosen a stylized drawing for all views because:

1. It is more impressive to management and customers.

2. It reduces mistakes during installation for untrained help.

3. It takes no more time or effort for users to make aesthetically pleasing drawings.

As NBC stated in their evaluation, "There is enough detail to be interesting, but not too much to clutter the plotting." Of course, you can always just use boxes with names on them, like you did on the drafting table, but.... Notice that the system block view lists all the inputs on the left, outputs on the right, remotes on top; and gives the signal name, connector type and termination information in 1 drawing. However, some people prefer a 1-line drawing instead of an all-line. To separate video, sync and audio lines into different drawings would take twice the components and probably triple the DOS disk space--a tall order as our libraries are already in excess of 15K drawings and 30 Megs of disk space. Since AutoCAD allows easy customization, many of our customers delete detail, add detail or separate line functions.

Colors and Layers

Since color is an effective way to distinguish component and signal types, Figure 4 shows a proposed set of color values, coupled with corresponding layer numbering and layer naming schema. Since AutoCAD supports named layers, numbers would normally be used if a drawing must be transferred to another CAD package.

Some engineers believe plots should only be done in black, which has been traditional because of the limitation of plot copies. However, color is very functional for identifying layers of signal types. It is much easier to detect a green video line from a blue audio line, and since installers are more prone to look than read, any help from color will reap benefits.
Besides this, color plots are more pleasing aesthetically, especially to management-types who must pay the bills.

Text

Text is the most critical element in video and audio engineering. Block drawings are only functional if text is clearly readable, and rack elevations are more useful if the manufacturer, model name and system name (e.g., VTR1) is readable. Text must be readable on the monitor and on the plot. If you have a low-res monitor, it will be necessary to zoom to each saved piece of equipment to read the text, but a large component like an SEG may require several successive zooms to read the text. On a 1024 monitor, text on nearly a whole studio can be read at once.

Plotters are also limited by text. A pen plotter makes beautiful architectural plots, but system diagrams with extensive text are difficult to plot because plotters are slow, run letters together, and quickly flatten pens on text (1-3 plots). Many of our customers use the JDL printer/plotter because it is 3-5 times faster than a plotter, and it does a beautiful job on text. In fact, an E-size drawing (36x52) that will not be readable on a D-size plot (24x36) can be read in a C-size (18x24) on this printer/plotter.

Drawing speed is also affected by text style. A curved, proportional font such as Simplex or Complex is much slower on regeneration than an angular text such as TXT. Keep text short and simple for best results. Also consider placing text on separate layers so they can be turned off.

CONTENT	COLORS		LAYER #	LAYER NAMES
Floor Plans	1	Red	10-19	Floor-Plans
Video Block Drawings	2	Yellow	20-29	Diagram-Video
Audio Block Diagrams	3	Green	30-39	Diagram-Audio
Control	4	Cyan	40-49	Diagram-Control
Component	5	Blue	50-59	Diagram-Component
Sync	6	Magenta	60-69	Diagram-Sync
Intercom/Comm	7	White	70-79	Intercom
RF	8	Grey	80-89	Diagram-RF
Racks/Cabinets	9	Charcoal	90-99	Cabinets
Furniture (chairs, desks)	10	Orange	100-109	Furniture
TCT	11	Dk Green	110-119	AV-Equipment
Data	12	Lt Blue	120-129	Diagram-Data
Video Rack Components	14	Purple	140-149	Rack-Equipment

FIGURE 4. LAYER AND COLOR STANDARDS

The essential purpose of plotting is to use the drawing you have engineered, but most plots are for checking drawings in progress--proofs. It is therefore essential that plotters have better resolution than the monitor. This is not the case with an A (8.5x11) or B (11x17)-size plot, especially with a hi-res monitor. D-size has traditionally been the drawing standard for final plots, because a plotter cannot effectively get the information in a smaller format. Many of our customers find that users do not like D-size (24x36) plots, and often reduce them on a copier so the plot is easier to handle. Again, the JDL allows quick plots that are both smaller and easier to read.

Regarding scale in CAD, all drawing should be done in real size (1"=1"), and only reduced or enlarged during a plot. You should only then fit or scale a drawing to your paper size. On a plotter, 6 or 8:1 is the maximum reduction possible in order to read the text, while a printer/plotter functions at 15:1 on .6" text.

Legend and Title Blocks

With a numbered layer system, a detailed layer legend is essential but is not as necessary with AutoCAD named layers. For final plots, however, a description must be placed on the drawing, normally in a title block. While there is no standard for the title block, it should include job name/number, description, company, engineer, dates, revision data, and possibly a page number. You can easily make a component block for this information, then scale the block to fit the drawing to be plotted. The same scale can be a guideline for plot scale. In other words, if you must rescale this block 4x up, the plot should use a 4:1 scale.

Cable Numbering and Run Lists

Besides the physical rack elevations and system block drawings, the next most essential output of an engineering CAD system is cable documentation. We have developed a program called CableDOC to extract this information from the drawing. From this database, cable labels and run-lists can be easily printed without re-typing the data. Currently we are working on ways to simultaneously enter CableDOC data from within a CAD drawing and automatically update the CableDOC database. But regardless of how you get the data, a cable database should maintain the following fields:

Cable #	Cable type
Source	Room #
Destination	Rack #
Signal name	Panel #
Signal type	Date
Length	Remarks

Placement Standards

The central element in a system block drawing should occupy the geographic center. In a station this would normally be the routing switcher or patch panel, while in an editing system it might be the SEG. This is normally the largest piece of equipment. Most engineers try to place other components as horizontally as possible around the center. It may be necessary to use groups of similar components on different layers if you have a large number of similar pieces. For instance, in a system with 20 source playback VCR's, a group of 4 VCR modules on 5 successive layers that can be hidden is more functional than 20 VTR's stacked on the left on 1 layer.

the computer (usually 16 Megs), or larger if you allow it to swap to disk, our standard calls for a drawing large enough for most video and audio facilities, 150'x100'. The lower half of the screen is designated for floor plans, the top right quadrant for rack elevations, and the top left quadrant for system block drawings. This allows you to see the whole system at once, or zoom in to particular views. In a large drawing like this, color is very critical so you can easily see what you must zoom into. High Res is also important because AutoCAD allows dynamic zooms at up to 1/32 without pulling a slow regeneration. (It can take minutes on a large drawing and slow computer.) If you need to zoom in closer than 1/32 to read text, you force 2 or 3 regens.

AutoCAD also allows for zooming to a named view, but this also normally forces a regen. However, this may be helpful if your resolution is limited because it then allows you to use dynamic zoom to 1/32 of this named view without a regen.

It is possible to off-load sections of the drawing by creating a block and writing (saving) this block to the disk. The block can be edited in this smaller form because blocks are simply drawings. However, if you insert this block back into the overall drawing, you must explode (or use insert *) to change the block back into separate entities. If you do not follow this step, you will not be able to edit anything in that portion of the drawing. Exploding also restores the entities to their original layers.

Other tips on using blocks of sub-drawings include:

1. If you intend to restore it exactly as it was horizontally and vertically, use 0.0,0.0 as your block reference point.

2. If you need to get rid of any unused blocks (e.g., views not placed), then write block the entire drawing. This reduces the size of the drawing, and is especially useful when you have finished modifications and are archiving the final. Purge can also be used, but it is much slower on large drawings since it requires a confirmation before each deletion.

3. Subdrawings management is still not as easy to manage as using other layers (and freezing when not needed). Try increasing your memory first, except when sub-drawings are needed by other engineers during a team, multi-machine drawing project. However, this requires extreme diligence in sub-drawing management and record locking, well beyond normal AutoCAD operations.

CONCLUSION

In conclusion, our project with the Air Force has shown that CAD standards are possible, particularly in the areas of color choice, layers, placement and symbols. These standards need to be more widely tested and adapted so they can become accepted by the industry.

In the other areas, we have offered some guidelines plus recommendations on monitors and plotters, but your choice in these areas will depend upon the scope and complexity of work, frequency of use, plus budget.

ACKNOWLEDGEMENTS

The author would like to thank Capt. David Nuttall, Al Merewitz, Sgt. Rafael Alcantare and John Schoellman of the USAF Broadcasting Service for their assistance.

CONSIDERATIONS IN DESIGNING AND CONSTRUCTING AN AM/FM BROADCAST FACILITY

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Susquehanna Radio
Dallas, Texas

The decision to relocate a broadcast facility is an expensive endevor requiring much planning by both management and engineering. This paper is a summary of the intensive planning involved with the recently completed broadcast facility for KPLX and KLIF in Dallas.

There are many parameters that must be considered in finding the right location to construct a studio complex. The search for a new location should begin at leasst one year before the need to vacate the existing studio. This puts a station in a better bargaining position and it's less likely to make costly mistakes due to incomplete or hastily prepared construction blueprints. Station Management and Engineering need to formulate lists of what is needed and desired in a new location.

The Management list included:

- Highly visable location near downtown
- Easy accessability
- Free and close parking for employees and guests
- Security
- 24 hour accessability

The Engineering list include:

- Permanantly unobstructed STL Path (10 year)
- Good telephone and electrical service
- Emergency generator source
- Satellite reception capability
- No close RFI Sources
- No zoning restrictions against or leasor problems with antennas or towers
- Short cable runs to roof and towers
- Low building induced vibration and noise
- Solid floor construction-not pre stressed
- 24 Hour HVAC capability

When the right building is being negotiated make sure all of the agreements between the leasor and leasee are included in the lease. Buildings are being sold and financed on a regular basis and the new owners might not like antennas on their roof. Satellite reception is an area that should not be taken lightly in a rooftop environment. Have a computer study done for about $250. it will let you know that T.I. might be a problem. We had a full measurement done at the building of our choice before we proceeded with further negotiations.

The search for the right location in a large metro area can be time consuming but enjoyable. We actually selected several buildings by flying in the traffic helicopter. It's a great way to check out rooftop accommodations for STL Shots. Once you select several buildings that meet your needs its time to begin planning the studio layout. Most multi-tennant buildings have a space planner that can be helpful in the original layouts at no cost. Its a good idea to sit down with them at your present facility and show them how the radio business functions. Explain what you like and dislike about the current studio and what you would like to achieve in the new one. Don't be afraid to show a competing buildings' space plan to the designer if there are some concepts that you like. Keep in mind most interior designers have never seen a radio station before, much less designed one. The more input you give them the better your layout will look and function. The actual studio layout is an area that engineering and programming should be heavily involved with. We had worked on studio remodeling plans for two years before deciding to start over in a new building. We knew what we wanted to achieve and constructed traffic flow charts for the designers to study.

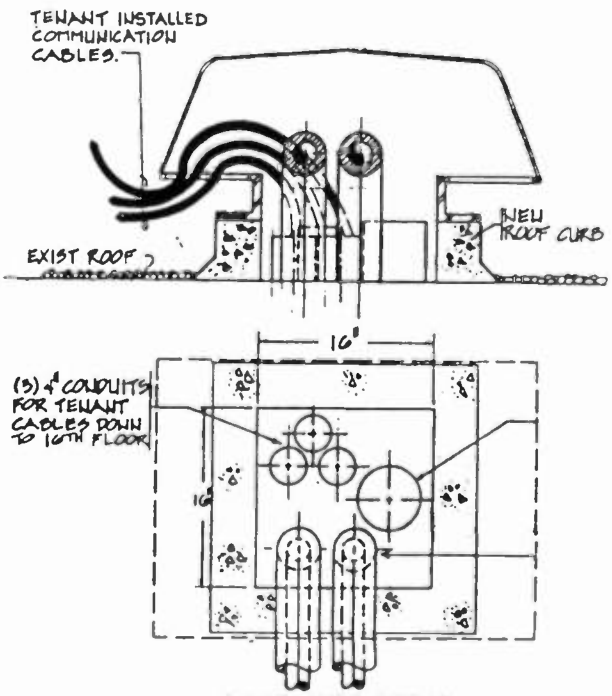
The studios at KPLX/KLIF began with a space designer plan which we modified back and forth for three months. We then enlisted the services of Joiner-Ross Inc. of Dallas to help in the acoustical design and final layout of the studio part of the project. Russell Burger and Richard Schrag worked with us over the next month to give us just what we were after: a very functional and space efficient studio complex with properly located core areas. The next step was to develop actual construction documents. Russell and Richard were instrumental in specifying low noise HVAC, electrical, grounding, acoustical performance and other parameters in the studio. We all worked together with a mechanical engineering firm, two interior design companies and building management to develop the documents in five weeks. They were then sent out to bid and the construction company was selected. Choosing the proper construction company is another area where caution is needed. The building manager was more familiar with the construction companies and their projects than we were. She even threw out the lowest bid knowing that the bidder would have to put cost overruns in as the project progressed to break even. We trusted her opinion and let her choose the construction company.

The special construction needed in studios will normally take longer than any general contractor estimates. There will be material delivery delays and subcontractor scheduling problems. The planning and construction meetings can eliminate most of these problems before they happen. There are some long lead time items that the building pre-ordered for the general contractor and stated so in the bid sheet. We ordered the sound isolation doors and frames 60 days before construction began so there would be no delay. Other items including engineering racks, U.P.S. systems, tower sections, antennas and line were on site when construction began. The Audio Consoles and cabinetry were ordered over 90 days in advance along with the bulk of the studio equipment. During construction the engineer in charge of the project needs to be on site every day to monitor progress and make sure the construction is being completed exactly as the specifications state. The most critical areas are sound walls and doors in the studio. To develop a good rapport with the workers we bought them all pizza lunch, gave them promotional t-shirts, and 200 special hats made for them. We also explained why it was so important for the drywall seams to be sealed with the expensive silicone sealer.

The rooftop antenna installation required a city zoning height variance due to location of Love Field runway 2 miles north of the 18 story building. The variance was granted for two 15 foot towers to hold the three STL Dishes, RPU Rotor, and radar dome. We also mounted a horizontal run of unistrut to secure the feed lines and act as a support for other Yagi and Omni Antennas mounted between the two towers in the middle of the building where they are less visable. The towers are custom five foot segements of 24" solid PI Rod. The short length allowed them to fit into the elevator. Each 25' tower came with two brackets that were welded to the top and lower horizontal beams of the buildings parapet next to the Penthouse. The bottom of each tower is 24" above the roof to prevent any problems with leaks. The towers have Cortana stati-kittys on them and are bonded to the building's lighting spike system.

The parapet around the building provides protection from fixed microwave terestrial interference and also protects from wind gusts. We installed three satellite dishes inside this area to provide us service from Satcom 1R, Westar 5, and a steerable video that doubles as a backup. The Equatorial dishes are mounted to the bottom of the parapet beam to keep from resting on or penetrating the expensive roof membrane.

The cable access the roof area and all studios is accomplished using 4" conduit. These paths are as direct as possible and have extra cabling in them for future needs. There are three runs to the weather cover which also has the HVAC coolant pipes and fresh air return for the studio's 24 hour HVAC system. ("Fig. 1") One conduit has 3 - 7/8" and a ½" spare STL line. The second conduit has 8 - ½" Heliax runs in it which we use for 450 receive and transmit antennas. The third conduit has all of the receive cables for master antennas, satellites, etc.. The conduit has pull boxes at both turns so adding cables in the future will be simple even though we have many spare cables presently. All roof conduits terminate in master engineering above the designated rack area it serves.



ROOF CURB DETAIL

("Fig. 1")

The studio conduits terminate under the computer floor next to the wall behind the racks. The control rooms and news room have two runs while the production rooms and news booths have one four inch conduit. There are direct runs from the AM control room to the Talk Stuido and the producer booth. There are speaker, air lights, master clock and remote control printer conduits as well. The conduits were all core drilled in the floor over an existing tennant so all work was performed on the weekend. The furniture hides all penetrations into the studio through the floor.

The studios all interconnect in master engineering via 16 pair Gepco Audio and 27 pair control cable. The Gepco multi pair cable in numbered every inch and fully jacketed, simplifying installation dramatically. Termination at both ends is provided by Seimens B66 Punch Blocks. There are 55 blocks on the wall behind the racks. The studio end of the cables are located directly underneath the rear of the consoles. The cabinetry was designed with removeable panels for easy access to the blocks and console connectors. There is over 5,000' of RG 58 and 59 between rooms as well as low loss 75 OHM video cable. We ran twice as many of these cables as were needed, anticipating future usage.

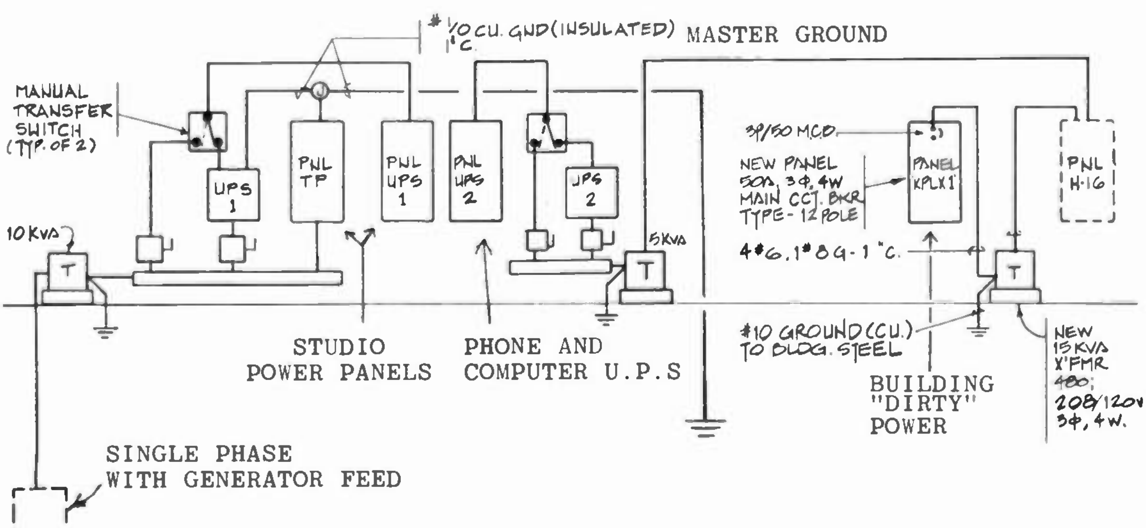
The various audio sources were isolated with distribution amps using +4 DB as house reference standard. The on air consoles are the only deviation and are fed directly to processing. The D.A.'s and audio routing switcher are located in the middle of the 7 racks in engineering. The punch blocks for these devices are also in the middle of the wall with FM control and production studios fanning to the left. AM and News to the right. This helps distribute the wiring evenly through the wiring duct.

The 7 engineering racks are 30" deep with 78" of rack rail front and rear. This allows plenty of room for deep equipment plus power and wire duct to keep the wiring neet. Each one has a primary function:

> Rack one houses radar, master clock, tellabs, and future
> Rack two is satellitte
> Rack three houses all 450 MHZ receivers and related Eq., etc.
> Rack four is audio distribution
> Rack five is modulation monitors and tuners
> Rack six contains AM and FM audio processing
> Rack seven houses all STL and Base station transmitters

The master ground buss for the studios is located behind the racks. There is a 1/0 insulated copper cable running to the basement of the building with its own ground rod. The studios and technical power are fed from this isolated ground with 2/0 welding cable in a star configuration from engineering. All A/C plugs in studio are isolated ground type and have seperate ground wires to the ground bar in each studio. This ground scheme was suggested by Joiner-Rose and has worked well.

The studio electrial power plan was also developed by them. All technical power is fed from one phase of the building feed or generator if main power is lost. There is a 10KW Topaz isolation transformer feeding the technical power distribution panel as well as the 5KW Best U.P.S. system. ("Fig. 2") The conduit or building ground is only connected to the cases of these



("Fig. 2")

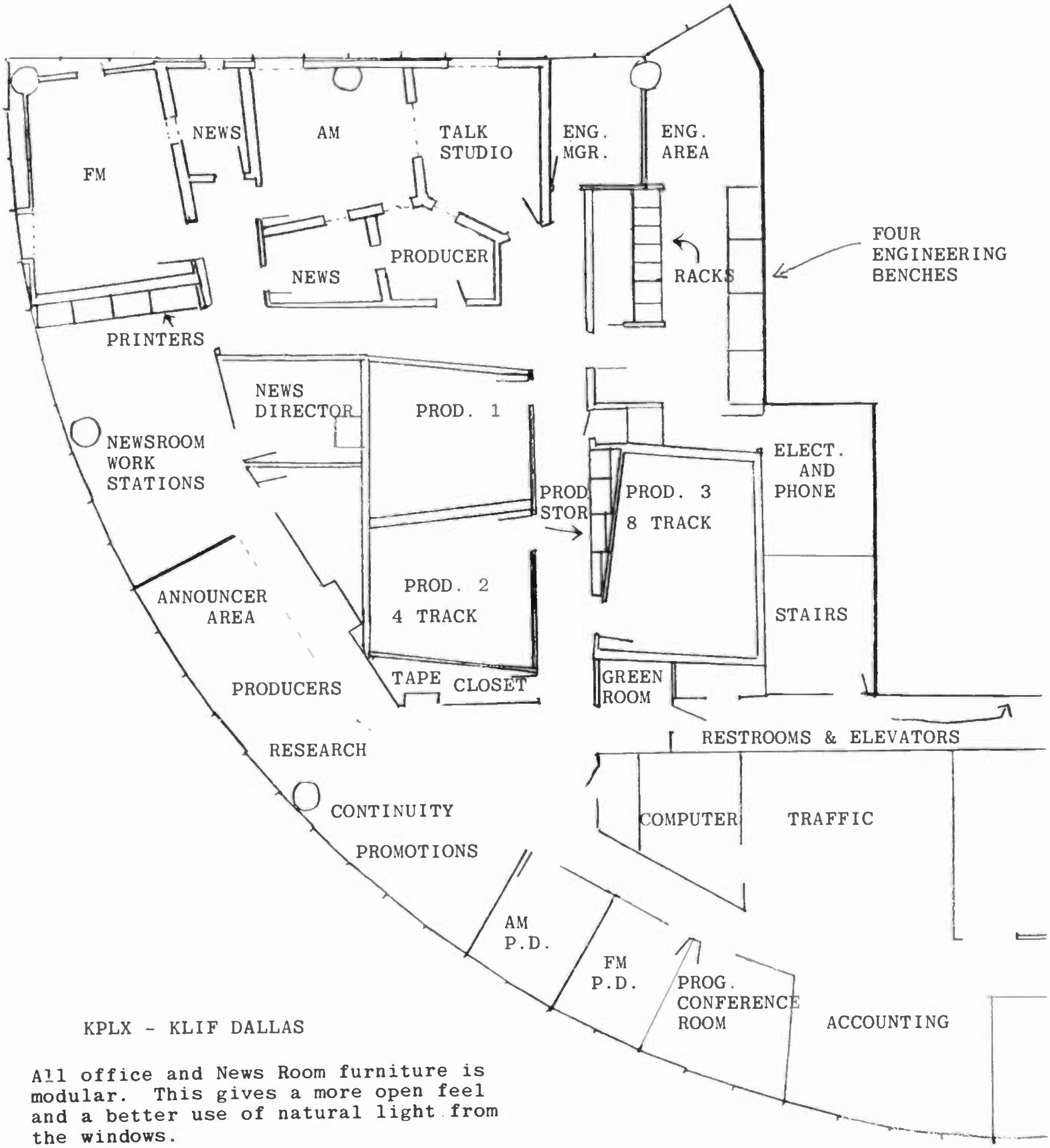
devices not the common or isolated ground, to prevent noise.

The studio is located just north of Downtown Dallas on floor 16 of an 18 story building. The lobby entrance is in the middle directly off the elevators. The conference room is in front of you with the Dallas skyline. Sales, Management and Support Staff are to the left. If you proceed down the hallway to the right you pass the Break Room, Accounting, Traffic Areas and Programming. It is at this point the traffic pattern splits. If you continue straight you'll pass Promotion, Continuity, Music Research, AM Talk Show Producers, and Announcer area then in to the news Room. The Goal was to keep traffic out of the Producer and News areas so the main path is in the Production room hallway. This hallway is almost 6' wide with no sound lock door at it's mouth. Noise has not been a problem due to the design of the rooms themselves. The Main TalkStudio is at the far left end of the hallway with Engineering at the right. The hallway turns to the left leading you past the Producer Booth, AM News Booth, AM Control then FM News Booth with FM Control Room on the corner. The studios were designed to give quick access to and from the News Room but keep the normal traffic out of the news area. ("Fig. 3")

The five studios are keystone in shape with sloped floating ceilings and RPG Diffusors on the rear walls. The AM Control Room has full vision to the Talk Studio, Producer Booth, and News Booth. The FM Control and news Booth have vision also. Much of the outside building glass was covered over with a second wall in the studio areas to eliminate noise. People can look outside the studio windows and watch jets take off without hearing them. The isolation from room to room is also excellent. The speakers are suspended on 2" pipe that is shot to the concrete ceiling pan. The placement of the speakers is almost a near field monitoring position. The 200 Watts per channel amplifiers and #8 Copper Speaker cable sound clean even when jocks really crank 'em up.

The seven consoles are all P.R.& E. Included are an ABX-26 in eight track production with BMX III-26 in AM Control and BMX III-18 in FM. The other production rooms are BMX II-14 with 10 mainfraimes in the News Booths. Both control rooms have a computer keyboard and video terminal next to the console. The On Air Software, Inc. computer program being used in the control rooms eliminates the paper clutter in them by storing liners, plug cards, and all other information in the computer. All computers are networked together with the Program Directors' offices, allowing the P.D.'s to update any information in the system from their offices. Video screens near the console have eliminated the need for copy stands in the control rooms.

We visited many radio stations in different cities before planning this facility. Concepts from them and others we developed made the time and money spent in planning and building woth it. We have constructed a beautiful, efficient and functional plant that serves us now and allows for growth in the future.



KPLX - KLIF DALLAS

All office and News Room furniture is modular. This gives a more open feel and a better use of natural light from the windows.

"Fig. 3"

AN INTRODUCTION TO WIRELESS MICROPHONE MULTIPLE SYSTEM FREQUENCY COMPATIBILITY

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The advantages of wireless microphone equipment have made its use popular and occasionally mandatory wherever sound reinforcement or recording is necessary. The freedom of movement and advantage of concealment it affords allows production options impossible with wired microphones.

Wireless microphones are radios and therefore are subject to the laws of radio frequency physics. These physical laws and the regulatory laws of the Federal Communications Commission put practical restrictions on the selection of operating frequencies. When two or more wireless microphone systems are operated simultaneously, a complex set of restrictions apply to the equipment design and selection of operating frequencies. The following is a description of those restrictions, what causes them and how they can be overcome.

DEFINITION

Multiple system frequency compatibility is defined as the condition of two or more wireless microphone systems operating simultaneously, with no degradation in performance due to the presence of the other systems. A system is defined as a transmitter and a companion receiver.

For example, suppose there are five wireless microphone systems operating simultaneously. If each system functions equally well with the other four systems turned on as well as with them turned off, the five systems are said to be compatible. If, however, the presence of any of the four systems degrades the performance of the fifth, they are said to be incompatible. Only one of the other systems may be responsible for the "interference". Alternatively, a combination of the other systems could be responsible. The following describes how a system or combination of systems can interfere with one another.

CAUSES OF FREQUENCY INCOMPATIBILITY

The restrictions on the selection of operating frequencies are imposed by the FCC and by the limitations of the transmitter and receiver circuits. The FCC reserves specific frequencies and bands of frequencies from which to choose. The eight specific frequencies reserved under FCC part 90.265(b) were welcomed, as they are available for wireless microphone use. Unfortunately, due to the channel spacing chosen, only combinations of three of these frequencies are compatible in spite of the eight available. Selecting frequencies in the TV channels allows for more compatible systems.

The restrictions imposed by RF physics and by the limitations of the equipment circuitry fall into six categories as follows:

1. Separation between operating frequencies.
2. Transmitter spurious signals.
3. Two signal intermodulation.
4. Three signal intermodulation.
5. Receiver local oscillator radiation.
6. Receiver image frequency sensitivity.

Each category is distinct and requires its own explanation. Note that all six restrictions considering all system frequencies must be satisfied simultaneously to achieve complete compatibility.

SEPARATION BETWEEN OPERATING FREQUENCIES

The separation between operating frequencies is simply how close together in frequency the systems are spaced. The limiting factor in the receiver design is the selectivity of the intermediate frequency (IF) filter and the dynamic range of the RF preamplifier and mixer circuits. The more selective the filter and the higher the dynamic range the closer the operating frequencies can be. The guideline is that all operating frequencies be separated from one another by 400 KHz or more.

This problem can be identified by turning on all receivers and then turning on only one transmitter at a time. If any receiver other than the companion receiver unsquelches, your operating frequencies may be too close together. Calculating differences in operating frequencies will prove or disprove this.

The best solution to this problem is to change system frequencies. To do this, calculate the difference between the system frequencies. If any of them are less than 400 KHz apart, the system frequencies must be changed. For example, suppose the following system frequencies:

F1 = 174.8 MHz
F2 = 175.4 MHz
F3 = 175.7 MHz

You find F2 and F3 less than 400 KHz apart. Either F2 or F3 must change so we choose to change F3. An acceptable frequency for F3 would be 178 MHz. Now all system frequencies are separated by more than 400 KHz.

Alternatively, you can turn on the transmitter companion to the offended receiver in an attempt to "capture" the receiver and thereby reject the interfering signal. Capture is a phenomenon in FM receivers whereby the stronger of two co-channel signals suppresses the weaker one. This alternative may or may not be successful.

TRANSMITTER SPURIOUS SIGNALS

In addition to the desired signal, all transmitters emit energy on other frequencies as well. Most wireless microphone transmitters use a quartz crystal as a frequency determining element and "multiply" it up to the operating frequency. A multiplier circuit is one in which the output frequency is a multiple (eg times two) of the input frequency. For example, consider a transmitter operating on 160 MHz. Starting with a 20 MHz crystal and multiplying by two yields 40 MHz. Multiplying by two again yields 80 MHz and once again yields the desired 160 MHz. This transmitter is a "times eight" (2 x 2 x 2 = 8) transmitter. However, it also radiates energy at frequencies other than 160 MHz. Signals are present at 80 MHz (x4), 140 MHz (x7), 180 MHz (x9), 200 MHz (x10) and so on. Granted, they are significantly weaker than the signal at the desired 160 MHz but they exist. A receiver operating on one of these undesired output frequencies will receive the transmitter spurious signal possibly causing audio degradation.

For example, a spurious signal might be transmitted at a level 70 dB less than the desired signal. Its transmitted power might then be -53 dBm. That is 57 dB above the threshold sensitivity of the receiver and will unsquelch it.

The solution is to select frequencies so these spurious signals simply do not fall on or near the other operating frequencies. Such undesired signals should be at least 250 KHz from any operating frequency. Multiples of one through 16 times the crystal frequency should be computed and compared to all the other operating frequencies.

This problem can be identified by turning on all receivers and turning on only one transmitter at a time. If a transmitter unsquelches a receiver other than the transmitter's companion receiver, you may have a transmitter spurious problem. Calculating undesired crystal harmonics will prove or disprove this.

The best solution to this problem is to change system frequencies. To do this, calculate the crystal harmonic which is interfering with a system. You can then choose to change the offending or the offended system. For example, suppose the following system frequencies:

F1 = 190.8 MHz
F2 = 214.8 MHz

The ninth harmonic of the crystal for system one occurs on 214.65 MHz which is within 150 KHz of F2. Calculations are done as follows:

1. 190.8 MHz/8 = 23.85 MHz (crystal frequency).
2. 23.85 MHz x 9 = 214.65 MHz (ninth crystal harmonic).
3. 214.8 MHz - 214.65 MHz = 150 KHz (less than 250 KHz apart).

An acceptable frequency for F2 would be 214 MHz because it is 650 KHz away from the ninth crystal harmonic of system one (214.65 MHz - 214 MHz = 650 KHz).

Alternatively, you can keep the transmitter on companion to the offended receiver in an attempt to mask the undesired signal. Success with this solution will vary.

TWO SIGNAL INTERMODULATION

First, lets begin with the definition of intermodulation (IM). When two signals are input to any nonlinear circuit, additional signals are created which are the sum and difference products of each of the fundamental input signals and their associated harmonics. The following output components will result:

- Fundamental: F1, F2
- Second Order: 2F1, 2F2, F1 +/- F2, F2-F1
- Third Order: 3F1, 3F2, 2F1 +/- F2, 2F2 +/- F1

- Fourth Order: 4F1, 4F2, 2F1 +/- 2F2, 2F2 +/- 2F1
- Fifth Order: 5F1, 5F2, 3F1 +/- 2F2, 3F2 +/- 2F1
- Higher Order Products

Order is defined as the sum of the numerical coefficients which multiply the F1 or F2 terms.

Note that the even order products usually occur very far removed in frequency from F1 and F2 and therefore are omitted here for simplicity.

If F1 and F2 are very close in frequency, the 2F1-F2 and 2F2-F1 terms also fall very close. If F1 and F2 are separated by 1 MHz, those products will also be separated from F1 and F2 by 1 MHz. For example, if F1 = 160 MHz and F2 = 161 MHz, close intermodulation signals will occur as follows:

- Third order 159 MHz (2F1-F2) and 162 MHz (2F2-F1)
- Fifth order 158 MHz (3F1-2F2) and 163 MHz (3F2-2F1)
- Higher Order Products

An RF spectrum analyzer display (in the frequency domain) is shown in Figure 1 as follows:

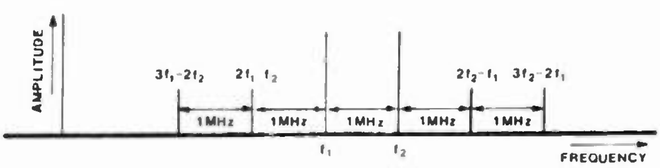

Fig. 1
Two Signal IM

If any of these products should fall on or near any system frequency, interference and incompatibility will result. The guideline is that these IM products should be at least 250 KHz away. Note that two signal IM will occur when two systems are operated simultaneously but no such interference will occur when operating only two systems. This is because the close in IM products (2F2-F1 and 2F1-F2) will be at least 400 KHz away from either system frequency. Two signal IM can cause interference when three or more systems are used.

Such IM can occur in the output stages of two closely held transmitters and will actually be retransmitted by the transmitters. The IM might also occur in the receiver RF circuitry due to close proximity of the transmitters to the receiver antenna. In any case, a product on or near a system frequency will unsquelch the offended receiver.

This problem can be identified in two ways. First, the interference occurs only when two transmitters are turned on. Turning either one of them off removes the interference. Second, the interference is more severe when the two transmitters are in close proximity to one another or, alternatively, are close to the receiver antenna. The interference may disappear completely when the transmitters (and receiver antenna) are separated from one another.

The best solution to this problem is to change system frequencies. To do this, IM product computations need to be performed and all IM products need to fall at least 250 KHz away from any system frequency. For example, suppose the following system frequencies:

F1 = 174.8 MHz
F2 = 175.4 MHz
F3 = 176.6 MHz

However, when F1 and F2 transmitters are both turned on and in close proximity to one another, the receiver on F3 not only unsquelches, but also receives the audio from F1 and F2. The following formulas need to be computed.

1. 2F1 - F2
2. 2F2 - F1
3. 3F1 - 2F2
4. 3F2 - 2F1

You determine that 3F2-2F1 = 176.6 MHz which is F3. You must select a new F3 while being sure the above four formulas don't indicate an IM product within 250 KHz of your new F3. In addition, you must take all of the combinations of two of the three system frequencies and be sure the above four formulas don't indicate an IM product within 250 KHz of the third system frequency. An acceptable frequency for F3 should be 178 MHz.

Alternatively, you can be careful to keep the transmitters separated from one another and from the receiving antenna by more than ten feet. Also, turning on the transmitter companion to the offended receiver may help to mask the problem.

THREE SIGNAL INTERMODULATION

Just as two signals input to a nonlinear circuit cause sum and difference products to be created, the same happens with more than two signals. Although not as severe, three signal IM has been shown to be a problem. The following output signals will result from three signal IM:

- Fundamental F1, F2, F3
- Third Order: 2F1 +/- F2 type and F1 +/- F2 +/- F3 type
- Higher Order

The even order products are usually far removed in frequency and are omitted here for simplicity. Higher order products, although of interest, are also omitted here for simplicity.

Consider systems operating on the following frequencies:

F1 = 159 MHz
F2 = 160 MHz
F3 = 161 MHz

Third order IM products will occur as follows:

Frequency	IM Product Formula
157 MHz	2F1-F3
158 MHz	F1 + F2-F3 and 2F1-F2
F1 = 159 MHz	2F2-F3
F2 = 160 MHz	F1 + F3-F2
F3 = 161 MHz	2F2-F1
162 MHz	F3 + F2-F1 and 2F3-F2
163 MHz	2F3-F1

Note that in the example given, third order IM products fall exactly on the system frequencies themselves. For example, F1 + F3 - F2 = 160 MHz which is F2 itself. Hence, equal spacing of system frequencies will result in two signal IM interference (eg 2F2-F1) as well as three signal IM interference.

An RF spectrum analyzer display (in the frequency domain) is shown in Figure 2 as follows:

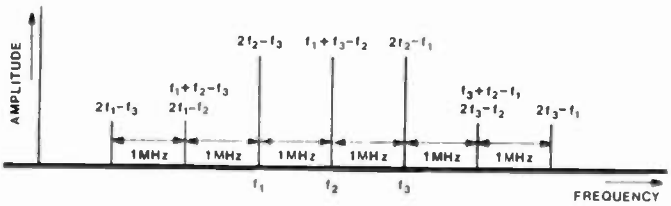

Fig. 2
Three Signal IM

If any of these products should fall on or near any system frequency, interference and incompatibility will result. The guideline is that these IM products should be at least 250 KHz away from any system frequency.

The characteristics of three signal IM are identical to those of two signal IM. Three signal IM can occur in the output circuits of transmitters or the input circuits of receivers. The problem can also be identified in two ways. Turning off any one of the three offending transmitters eliminates the interference. Also, the interference is more severe when the three transmitters are in close proximity to one another or to the receiver antenna.

As with two transmitter IM, the best solution to three transmitter IM is to change system frequencies. To do this, IM product computations need to be performed and all IM products need to fall at least 250 KHz away from any system frequency. For example, suppose the following system frequencies.

F1 = 181.4 MHz
F2 - 183.4 MHz
F3 = 184.8 MHz
F4 = 186.8 MHz

However, when transmitters on F1, F2, and F3 are turned on and in close proximity to one another, the receiver on F4 not only unsquelches but also receives the audio from transmitters F1, F2, and F3. The following formulas need to be computed.

1. 2F1 - F3
2. F1 + F2 - F3
3. 2F1 - F2
4. F3 + F2 - F1
5. 2F3 - F2
6. 2F3 - F1

Note that only third order product formulas are listed here. Fifth order formulas (eg 3F1 - F2 - F3 and 3F3 - 2F1) may also be significant but are omitted here for simplicity.

You determine that F3 + F2 - F1 = 186.8 MHz which is F4. You must select a different frequency for F4 while being sure the above six formulas don't indicate an IM product within 250 KHz of your new F4. In addition, you must take all of the combinations of three of the four system frequencies and be sure the above six formulas don't indicate an IM product within 250 KHz of the fourth system frequency. An acceptable frequency for F4 would be 204.8 MHz.

Alternatively, keeping the offending equipment physically separated from one another and the transmitter of the offended receiver turned on may help to minimize the problem.

RECEIVER LOCAL OSCILLATOR RADIATION

Lets begin with an explanation of a receiver's local oscillator by reviewing a basic superhetrodyne receiver block diagram. Refer to Figure 3.

The local oscillator (LO) is a circuit which generates a signal to be mixed with the signal received at the antenna in order to generate a signal at the intermediate frequency (IF). The local oscillator is actually a low powered transmitter type circuit whose output is wired to the mixer circuit. Although undesirable and weak, this LO signal is coupled to and radiated by the receiver antenna.

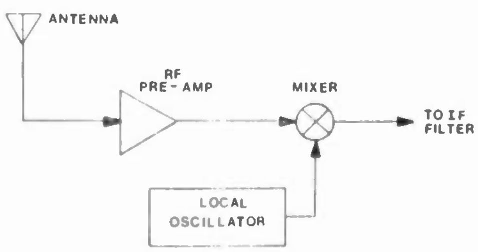

Fig. 3
Receiver Block Diagram

Consider a receiver operating on 160 MHz. Assuming an IF of 10.7 MHz and "low side" injection the LO will be on 149.3 MHz. This frequency will be radiated by the receiver antenna and will be detected by any receiver operating on or near 149.3 MHz.

If the LO signal falls on or near any system frequency, interference (and incompatibility) will result. The guideline is that no receiver LO should be closer than 250 KHz to any system frequency.

This problem can be identified by turning off the offending receiver causing the interference to disappear. Of course this effect is most obvious with all the transmitters turned off.

The best solution to this problem is to change system frequencies. To do this, calculate the LO frequency of each receiver and be sure each is at least 250 KHz away from any system frequency. For example, suppose the following system frequencies.

F1 = 203.3 MHz
F2 = 211.4 MHz
F3 = 214 MHz

However, when the receiver on F3 is turned on, the receiver on F1 unsquelches. You determine that the LO frequency of receiver F3 = 203.3 MHz (214 MHz - 10.7 MHz) which is F1. You must select a new F1 or F3 being sure it is at least 250 KHz away from the LO of the other receivers. An acceptable frequency for F1 would be 210.8 MHz.

Alternatively, physically separating the offending receiver and its antenna from the offended one will minimize the problem. Also, turning on the transmitter companion to the offended receiver should minimize the problem.

RECEIVER IMAGE FREQUENCY SENSITIVITY

Understanding this effect will be aided by a refresher on the explanation of a receiver image frequency. An image frequency is one which is separated from and on the opposite side of the receiver local oscillator (LO) frequency from its operating frequency by an amount equal to the intermediate frequency (IF). For example, consider a receiver operating on 160 MHz. Using a typical IF of 10.7 MHz and a common "low side" local oscillator (LO) at 149.3 MHz (160 MHz minus 10.7 MHz), the image frequency will be 138.6 MHz (149.3 MHz minus 10.7 MHz).

Typically a receiver is 70 dB less sensitive at its image frequency than at its operating frequency. This would yield a threshold sensitivity of -40 dBm at the image frequency. A transmitter output is +17 dBm or 57 dB stronger than the receivers threshold sensitivity at its image frequency. If a transmitter is operating at or near a receivers image frequency interference (and incompatibility) will result. The guideline is that all transmitters should be at least 250 KHz from any image frequency.

This problem can be identified by noting if the offending transmitter is 21.4 MHz lower than the operating frequency of the offended receiver. These numbers assume low side injection and an IF of 10.7 MHz although the principle is valid regardless of LO frequency.

The best solution to this problem is to change system frequencies. To do this the image frequency of each receiver must be calculated and be at least 250 KHz from any system frequency. For example, suppose the following system frequencies:

F1 = 186.6 MHz
F2 = 205.4 MHz
F3 = 208 MHz

However, when F1 transmitter is turned on, the receiver on F3 unsquelches. You find the image frequency of F3 = 186.6 MHz (208 MHz - 21.4 MHz) which is F1. You must select a different system frequency for F1 or F3 being sure all image frequencies are at least 250 KHz away from any system frequency. An acceptable frequency for F1 would be 204.8 MHz.

Alternatively, you can physically separate the offending transmitter from the offended receiver although the probable success of this solution is remote.

DON'T MISDIAGNOSE OTHER PROBLEMS AS COMPATIBILITY PROBLEMS

As mentioned earlier, wireless microphones are subject to the laws of RF physics and limitations of the equipment circuitry. Problems can occur such as outside RF interference or dropouts which need to be distinguished from possible compatibility problems in order to expediently diagnose and resolve a problem.

A diagnosis technique was indicated for each of the six possible causes for frequency incompatibility. These techniques combined with some arithmetic should lead you to the specific cause of the compatibility problem, if one exists.

SUMMARY

Recall that to achieve complete frequency compatibility all six potential causes of frequency incompatibility must be satisfied simultaneously with all system frequencies considered. More than 14,500 calculations are performed for only four wireless microphone systems. Obviously, a sophisticated computer program is required to analyze frequency combinations in a timely fashion. Furthermore, programs can be written to automatically search for a desired number of compatible frequencies given lower and upper frequency bounds.

Like other forms of interference, there are degrees of compatibility. In some cases the problem is mild and in others it may be severe. The stronger the interfering signal and the closer it is to a system frequency, the more severe the problem will be.

For example, if a system is on 160 MHz and an IM product is on 160.05 MHz (50 KHz away), the problem is likely to be severe because they are close in frequency. If, however, the IM product is on 160.24 MHz (240 KHz away) the problem will be mild or non-existent.

Mild compatibility problems may be masked by keeping all of the transmitters on. The signal strength of the transmitters will usually be sufficient to "capture" their companion receivers. If a transmitter is turned off, any signal stronger than 1 microvolt will be heard in its companion receiver. If, however, the transmitter is turned on, an interfering signal may have to be as strong as 1000 microvolts (60 dB stronger) to be heard in that receiver.

Note that the recommended solution in each of the six cases is to change system frequencies. The six categories causing incompatibility are a result of "laws of mother nature". Well designed, state of the art equipment can minimize the problems encountered, but, with a few exceptions, can never totally eliminate them. Since all quality equipment has been designed to minimize these problems, the final step to totally eliminating compatibility problems is a judicious choice of system frequencies.

Sophisticated computer programs capable of hundreds of thousands of calculations are required to select and analyze frequencies for a large number of systems. Your equipment manufacturer should offer this service to assist you in the formidable task of selecting frequencies to assure multiple system frequency compatibility.

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FURTHER CONSIDERATIONS ENG MICROWAVE ANTENNA POLARIZATIONS

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The television networks began experimenting with Electronic Newsgathering (ENG) 15 years ago and have performed studies that included microwave transmission trials, wherein the effects of clutter in the propagation paths were evaluated. As a result of these limited trials, ENG, as it is practiced today, became a fact shortly thereafter. Naturally, the prevailing competitive spirit that existed at the time precluded the possibility of devoting the years of study that radio frequency wave propagation deserves. As with most new technologies introduced to TV Broadcasting, ENG evolved into its present state as a result of many people diligently performing the mundane day-to-day tasks of television production. In short, the engineers introduced the concept, and the producers made it happen.

Although little or no organized experimentation occurred from which hard data could be collected, it was widely assumed that given certain theoretical considerations, employing circular polarization (CP) of the propagated electromagnetic wave was very important in the reduction or elimination of multipath effects. Unfortunately, there has been little to corroborate this proposition from observations made over the years. This has led to a more detailed investigation of the theoretical aspects of wave propagation. The results have revealed an incompleteness in the earlier theory regarding behavior of CP reflections.

This presentation will provide the ENG operator with the results of this technical investigation to ensure that better use may be made of microwave assets already on hand or those to be acquired and to also help with budgeting by making things simple. First, let us review how CP signals are generated and how the belief in CP antennas' capacity for rejection of reflected signals came about.

It is well known that a CP wave is generated by combining two linearly polarized electromagnetic field elements in space and time quadrature. The first requirement of space is provided simply by crossing two dipole elements, one vertical, and the other horizontal. The time or phase quadrature is obtained by delaying either the horizontal or vertical wave component by 90 degrees before they are combined. The result is a propagated wave whose polarization vector is continually rotating either clockwise or counterclockwise depending upon which of the linear components is delayed. In order to recover this launched energy with a minimum of loss, another antenna with space and time characteristics identical to that of launching antenna is required. In other words, both transmitting and receiving antennas must be capable of launching or receiving a circularly polarized wave of the same sense of rotation, whether clockwise or counterclockwise. Should one antenna be clockwise and the other counterclockwise (analagous to crossed linear polarization) an isolation of 20dB or so will occur between the two. This point is essential in order to understand how a CP antenna might distinquish between a directly received signal, and one that has been reflected. If, during propagation of a CP wave, one of the linear components is shifted in phase by an additional 180 degrees, the sense of polarization is reversed and what was originally clockwise becomes counterclockwise and vise versa. Such a phase reversal can sometimes occur when an incident wave is reflected. Since a CP wave is composed of both a horizontal and vertical component, let us consider how each component behaves when reflected by obstructions in the path of propagation and from there, determine how reflection of a CP wave might be affected.

Theory is that a wave, horizontally polarized, will normally retain most of its incident amplitude when it is reflected. Its phase, however, will always lag that of the incident wave by 180 degrees. Furthermore, unlike a vertical wave component, the degree of phase shift for a reflected horizontal component is independent of grazing angle at which the incident wave strikes a reflecting surface. However, when a vertically polarized wave is reflected, there may be little or no change in its character provided certain conditions are met. Therein is the core of a prevailing misunderstanding concerning the behavior of CP waves since the conditions are rarely met outside an antenna test range. It is understandable how this state of confusion regarding the behavior of circular polarization came about when much of the present literature does not emphasize or, in some cases, even mention those qualifying conditions under which CP may be found useful. Figure 1 shows behavior typical of waves propagated over land. I will now discuss how reflected vertically polarized waves are affected in an environment typical of that encountered in ENG. Under such conditions, there are few smooth surfaces except for an occasional body of water. Usually, propagation of microwave signals for ENG occur over land and are reflected by fairly rough surfaces and objects such as buildings, bridges, and ground cluttered with automobiles and other vehicles. As shown in Figures 2 and 3, most elevation angles of propagation resulting from a mobile microwave transmitter to a receiver will generally be 10 degrees or less. Therefore, most resulting reflections will be those due to small incident grazing angles. The same may be said, although to a lesser extent, for many grazing angles of reflections from scattered vertical obstacles such as tall buildings. Figure 1 shows how a reflected vertical polarized wave can vary in amplitude from that of the incident wave as a function of grazing angle. Moreover, the phase of the reflected vertical wave may lag that of the incident wave by any value between 0 and 180 degrees, depending upon the grazing angle with which the incident wave strikes the reflecting surface. Figure 1 may be used to graphically illustrate the relationship of amplitude and phase delay of reflected waves as a function of grazing angle for both reflected vertical and horizontal linear components of CP waves over average terrain. Notice that at a grazing angle of approximately 14 degrees, the magnitude of the reflected vertical component goes to a minimum at the same time that the phase angle goes through a sharp transition from 180 to 0 degrees. The grazing angle at which this occurs is known as the Pseudo-Brewster angle (corresponding to the Brewster[11] angle of optics). It is worthy to emphasize that no such change in amplitude occurs for horizontally polarized components, and that phase delays of these reflected waves are always 180 degrees from that of the incident wave for all grazing angles. It is also worth noting that only for angles greater than the Brewster angle is the phase lag near zero for a vertical wave component. In fact, for grazing angles of less than the Brewster angle, phase lag for both polarizations can change similarly and result in a reflected wave of the same sense as that of the incident wave and thus fail to provide protection against reflected signals.

Now that some of the ideas underlying this matter have been explored, we may turn our attention to how best to use this information. As an example of how an analysis may proceed, let us assume that an ENG receiving antenna will be located on a TV transmitter tower at a height of 1000 feet above ground. Assume also, best case propagation conditions of gently rolling terrain. For purposes of obtaining rough orders of magnitude, our computations will be approximate with some rounding of quantities. Again, referring to Figure 1, we find that the depicted Brewster angle is about 14 degrees. This means that any reflections due to ground clutter must be reflected at grazing angles of more than 14 degrees if there is to be any hope that they will be attenuated as a result of reversed sense polarization. At this height, this can only occur at a distance of 0.8 miles or less from the receiving anetnna. It appears then, that we can expect different performance from a CP antenna depending on whether the grazing angles of reflected energy impinging upon it are less than or more than the Brewster value. This angle is a function of both antenna height and length of radio path.

Let us now proceed with a comparison of performance of all three polarizations over a given microwave path. First, a look at close-in performance, i.e., 1000 to 4000 feet path lengths and large grazing angles of 14 to 40 degrees. Charts contained in Figure 1 will provide magnitudes and phases of reflections. Next, the case for horizontal polarization. In this case, both magnitude and phase shift remain constant at unity and 180 degrees respectively relative to the incident wave. It is also obvious that in this case, signal cancellation and "ghosting" is a fairly good probability while generally poor performance is expected.

For the case of vertical polarization, over the same close-in path length, the amplitude of the reflection is expected to vary from 0.15 to a maximum of 0.6 of the incident wave. Relative phase lag for these reflections will vary from about 130 degrees to zero. It can be seen from these results that any reflections will always be somewhat attenuated from that of the incident and that the probability for cancellation is not high because of these wide variations in amplitude and phase.

Combining results for the horizontal and vertical cases, the case for CP becomes evident. What results from such a combination is a wave with an elliptical polarization of opposite sense from that of the incident wave but containing a large horizontal component. Unfortunately, this horizontal component can be responsible for as much signal disruption as that due to a pure horizontally polarized wave. Thus, a comparison of results from using each of these three polarization modes for close-in receiving conditions shows that the use of vertical polarization is marginally better for avoiding any negative effects of multipath transmission than the other two, while the result of using CP is only slightly better than that of using horizontal polarization.

For cases in which the grazing angle is less than the Brewster angle, as is the case for most terrestial radio paths, a similar analysis follows. For the first case, that of horizontal polarization, it is evident that conditions for reflections are the same as for the close-in analysis. Reflections are at the same levels as for the incident while phase shift remains at 180 degrees for all grazing angles.

For the case of vertical polarization,

Figure 1 indicates that the levels of reflections vary from a value of 0.15 to 0.8 of that of the incident, depending upon the grazing angle. Phase lag is at or near 180 degrees. Combining horizontal and vertical as before, results in a CP wave that starts very elliptical at the Brewster angle and becomes more circular as the grazing angle is reduced. Unfortunately, this improved circularity is for a CP wave that is of the same sense, thus there is little, if any, difference between reflected and incident signals. Again, it may be concluded that the polarization that offers the most protection from signal interruption or distortion due to effects of multipath reception for mobile terrestial microwave communications is the vertical. Similar results have been reported by workers engaged in propagation studies commissioned by the United States Navy./2

While the evidence seems to indicate limited use for CP in terrestial microwave communications, it has been found very useful in the field of aviation. Wide application in this field is the result of the CP's ability to prevent antenna decoupling that might arise as a consequence of aircraft maneuvers. With CP antennas, an aircraft can perform rolls, banks, and other maneuvers without the probability of cross-polarized antennas. An example of such an application is one in which an aircraft is used as a relay between two ground stations. The CP ground antennas are oriented straight up while the aircraft CP antennas are projected downward. Both antennas are like polarized. Successful operation of this system depends on the following condition: the angle of incidence of the energy going between air and ground is at or close to 90 degrees. As we have learned, reflections emanating as a result of such large grazing angles will be circularly polarized and of opposite sense from the incident so that they do not interfere with reception of the desired signal. There are several users of such systems reporting positive results.

Conclusions

We have discussed the behavior of electromagnetic energy when reflected by obstacles in the path of propagation. We've shown that of the three polarizations, one (circular) is constructed from the remaining two (linear). In judging the quality of CP, a figure of merit; axial ratio is important. Its quantity is measured in decibels and is a measure of polarization ellipticity. This ellipticity is a ratio of the linear components that make up the elliptical or circular polarized wave. The greater this ratio, the more the behavior of the wave will resemble that of a linear polarized wave. It is well known that an axial ratio of greater than 2dB will nullify characteristics, desired or otherwise, attributable to CP. Figure 4 shows how we can readily see the circularity of a reflected CP wave is deteriorated as the grazing angle of reflection is reduced resulting in increasing ellipticity. As a result of this phenomena the reflection will behave much as one of the linear components, depending upon on which one is dominant.

As a result of this analysis, it can be concluded that vertical polarization, when used for mobile microwave communications, is the most effective in minimizing reflections and other multipath effects. Since a linear system costs a fraction of that of a CP system, this is indeed a fortunate conclusion. An improvement worth considering is that of a dual polarization system, whereby a facility for switching between two linear polarization at either or both ends of the link is incorporated for the purpose of obtaining polarization diversity reception. Such a solution can lead to a much improved system over that of existing CP systems and at a reduced cost. A simplified scheme may entail nothing more than rotating a linear antenna or antenna feed about its axis of propagation by 90 degrees.

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/1 The equations and their derivision from which these graphs were plotted may be found in Reference 1.

/2 Numerous charts and mesurements recorded on the effectiveness of the various polarizations may be found in Reference 1.

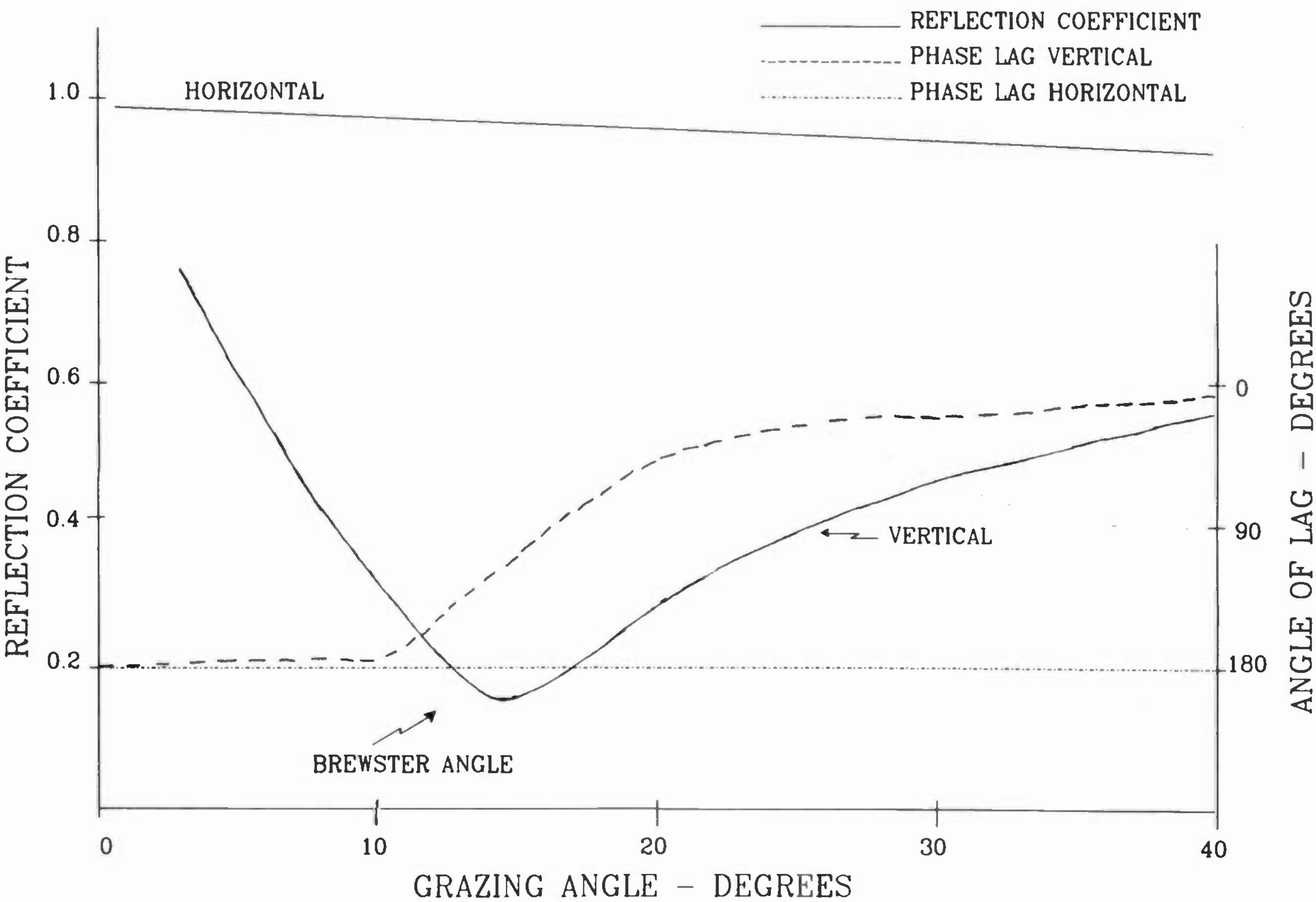
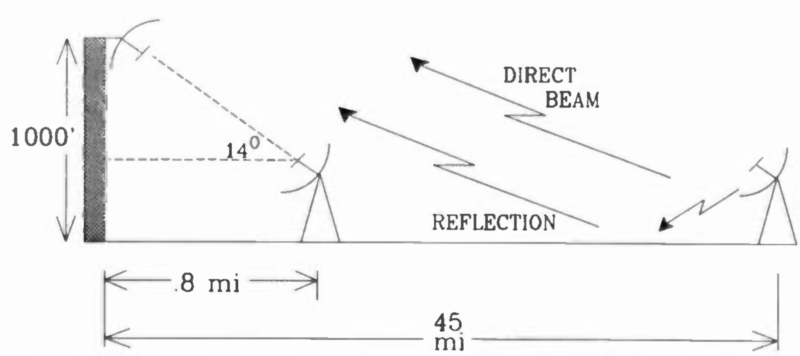


FIGURE 1 – REFLECTION CHARACTERISTICS, AVERAGE TERRAIN – S BAND



DISTANCE AT WHICH BREWSTER ANGLE IS EXCEEDED FOR 1000' TOWER
FIG. 2

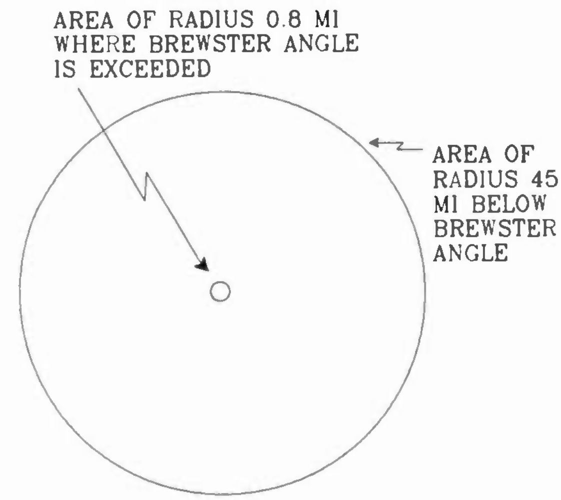


FIGURE 3

COMPARISON OF MICROWAVE COVERAGE AREAS ABOVE AND BELOW BREWSTER ANGLE FOR 1000' TOWER

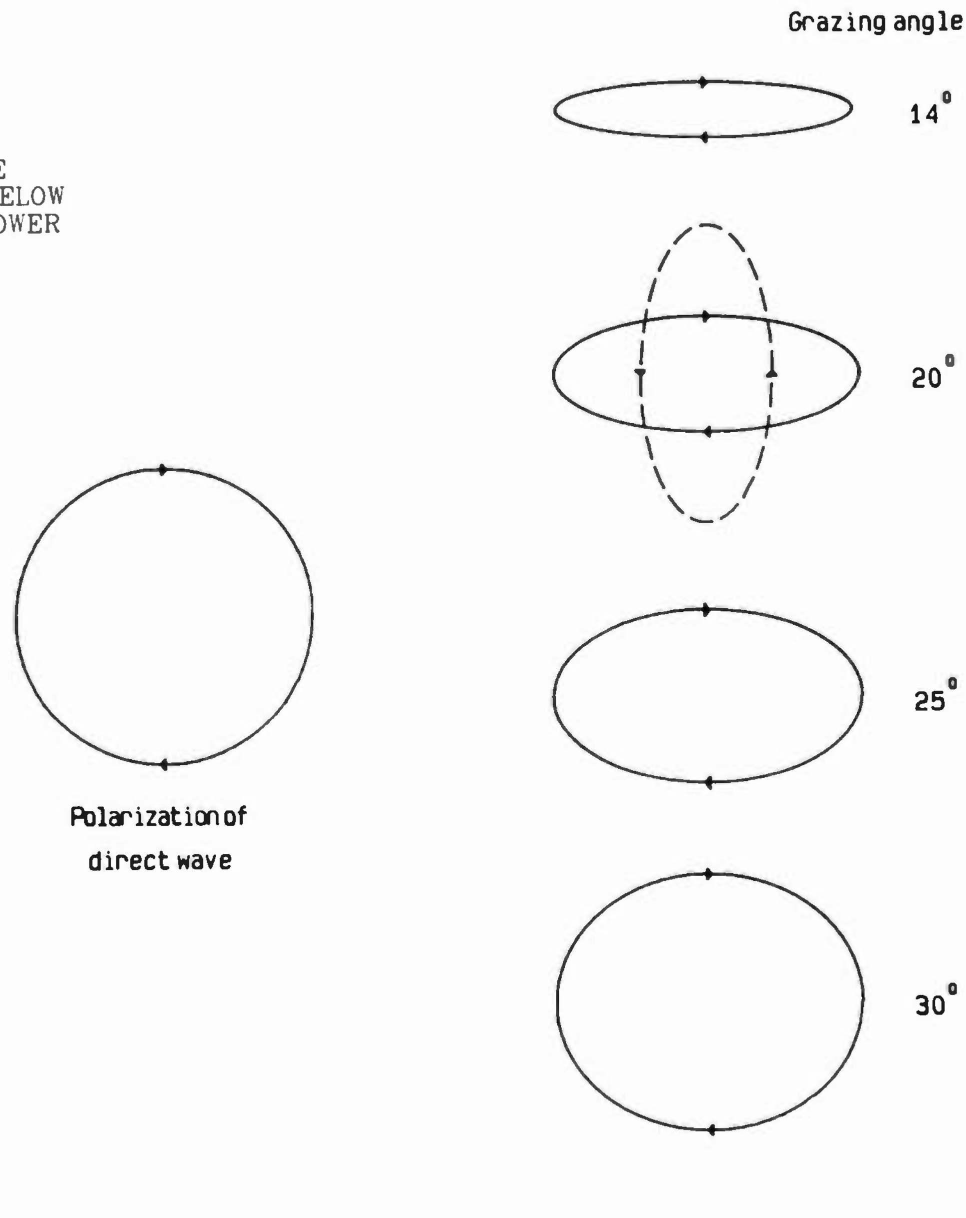


Figure 4 Shape of polarization ellipse of reflected wave over land.

NEW GENERATION RPU ENHANCES PERFORMANCE

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Abstract

A new RPU system is described which brings novel features and enhanced levels of performance to this class of equipment. In designing the RPU system, a "Wish List" of performance requirements and features was established and fulfilled. A technical discussion of TFT's patented RF generation scheme is presented, as well as other technical details of the RPU transmitter and receiver.

Introduction

Remote broadcasting continues to be an extremely popular part of a radio station's programming strategy. It gives radio stations and their personalities a higher profile in the community and provides novel promotional opportunities for advertisers. Engineers who must sweat the technical details of remote broadcasting welcome the opportunity to break the daily routine of station maintenance and administrative humdrum and look upon the technical challenge with relish.

One of the major technical considerations when doing a remote is conveying the program to the studio for broadcast. While phone lines are still in frequent use, equalized phone lines are quite expensive and often require considerable advance notice. Furthermore, dial lines are only adequate for voice and, even with the available signal processing, are not considered suitable for more demanding programming.

When given a choice, radio links are preferred for they have the potential to offer improved audio quality as well as convenience and economy. TFT now gives the engineer a choice among RPUs.

When TFT set out to produce an RPU system, we conducted several surveys of both broadcasters and equipment distributors. With broadcasters, we conducted personal interviews and uncovered individual needs. From distributors, we obtained sales statistics to spot purchasing trends among broadcasters as a whole. The information provided by both groups resulted in a "Wish List" of performance and convenience requirements.

RPU WISH LIST

Frequency Agility and Bandwidth Agility

Frequency agility was the most frequently requested feature for an RPU. However, since each of the different frequency bands allocated to RPUs has a different restriction on occupied bandwidth, an RPU system capable of frequency agility must must also have switchable deviation. An agile RPU transmitter avoids the inconvenience of crystal changing and operation is possible in more than one portion of the 450 MHz band without equipment dedicated to each band.

"Base" Station Transmitters

Remotes of even modest complexity often involve a van full of audio production equipment. A versatile, high quality RPU transmitter is needed at the base location of the remote operation to convey the signal to the studio (perhaps via a repeater site).

The ideal transmitter should generate an RF power output of 20 watts or more in order to cope with long or difficult paths. It should possess mixing capabilities for occasions when minimal equipment is needed and be simple to operate. Frequency selection should be made by front panel selection of 2 pre-selected frequencies; frequency programming should be uncomplicated. There should be easy selection of transmitter deviation in keeping with the different bands in which the transmitter can operate.

For use by non-technical operators, front panel layout should not be intimidating. Easy to understand labeling and warning indicators should invite intuitive operation. Flexible power requirements would permit ac or battery operation.

Repeater-capable Receivers

The receiver of an RPU system should be capable of operating both at the studio or repeater site. Special requirements are needed for repeater operation. One major consideration is that a repeater receiver must operate well under congested RF conditions since repeater locations are often at broadcast transmitter sites; nor must they be compromised by the presence of high power transmitters. Since repeater locations are most often unstaffed, a means of remotely controlling the repeater receiver is highly desirable.

Superb Audio Performance

Since the RPU is an integral part of the audio chain, wide audio frequency response with low noise and distortion is, of course, imperative. Performance even at lower deviations and bandwidths must also be superior. The transmitter must accommodate multiple audio inputs, both microphone and line level, with transformers in the microphone input for best rejection of noise.

System Intelligence

If an RPU transmitter is frequency agile, it is desirable to control the receiver's frequency and other operational characteristics from the transmitter. When repeaters are used, it is also necessary to ensure that positive control is exercised over the operation of the repeater transmitter's activation.

Since it is not unusual for a station to have more than one remote facility and since several stations may share a frequency, the common method of using a subaudible tone to identify the proper user is inadequate. A means--such as a digital code--of identifying the user as well as a specific repeater is desirable.

Implementation

The RPU products that TFT has developed incorporate all of the above features and technical requirements. Unique to RPU systems is the inclusion of frequency agility combined with switchable deviation in the transmitter and switchable bandwidth in the receiver. To improve audio signal-to-noise performance, noise reduction or other signal processing can be employed. The RPU transmitter includes connectors for looping external equipment through its audio section. The receiver includes DTMF (Dual-Tone Multiple-Frequency) circuitry to control the receiver from the transmitter. A modular jack on the transmitter accepts a standard touch-tone telephone input as the DTMF signal source.

The RPU Transmitter

The RPU transmitter offers frequency agility, set by the user with DIP switches. A patented RF scheme (described below) ensures superior audio quality and virtually no non-harmonically related spurious output. A front panel switch permits convenient switching between the two frequencies programmed by the user. The transmitter can be tuned to any of the group N_1, N_2, or S frequencies, and a switch selects between the applicable deviations of $\pm$ 10 kHz, $\pm$ 5 kHz, and $\pm$ 35 kHz, respectively.

There are two microphone inputs and one line level input, each with XLR connector and level control. The microphone inputs incorporate input transformers. A bar graph display indicates modulation levels. A headphone output with level control allows monitoring of the "mix" in situations where an external mixer is not used. A diagnostic meter with a multi-position selector switch monitors critical internal voltages as well as forward and reflected RF power. Each observable parameter is accompanied by its allowable range of values as an aid to troubleshooting. A separate LED indicator warns of excessive VSWR, and the unit can be operated from an ac source or external battery.

FREQUENCY MULTIPLYING STL TRANSMITTER

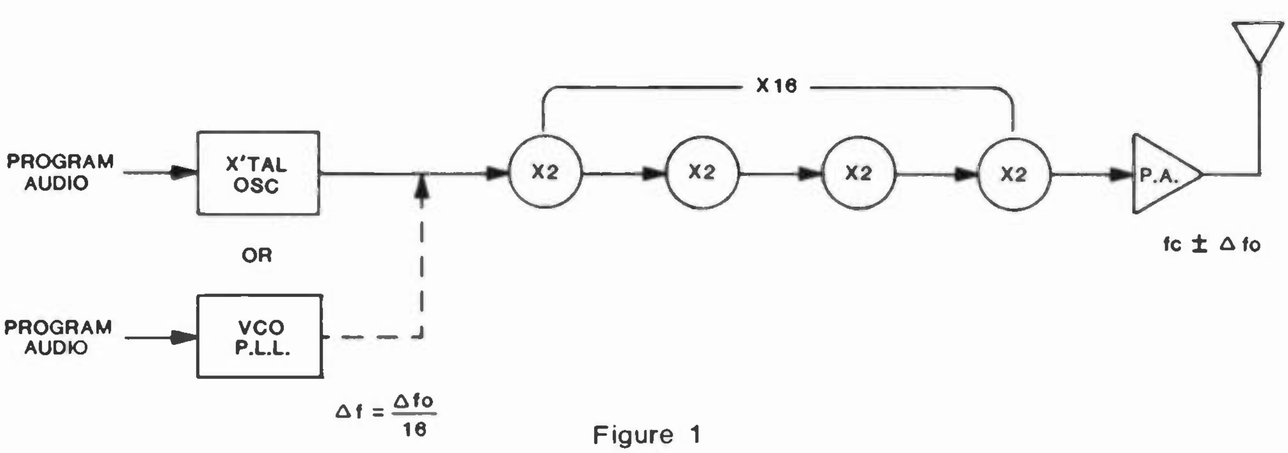

Figure 1

Older transmitter designs (Figure 1) used multiplication of a modulated crystal oscillator and/or low frequency VCO by as much as 32 times to arrive at the transmitter's output frequency. This process degrades the signal-to-noise ratio in the crystal oscillator or VCO by 20 Log N, where N is the multiplication factor. The result of a x16 multiplier is an SNR degradation by at least 24 dB.

Modulation of a crystal oscillator is itself an undesirable scheme for today's requirements for high quality audio. Because of the narrow region of linear frequency swing (pulling range) of the crystal oscillator, audio distortion is inherently higher. Furthermore, a transmitter using this modulation method cannot be frequency agile; a new crystal is required when changing frequency.

An improvement to this type of transmitter can be classified as the first generation of IF modulated transmitter, which uses a low frequency PLL VCO that is modulated by the audio (Figure 2). Rather than multiplying the resulting signal, the output is mixed with a UHF crystal-controlled frequency source operating near the output frequency. Bandpass filters are necessary after the mixer output to remove spurious components resulting from the mixing process. Since the ouput from a mixer is relatively low in level, considerable amplification is required. This adds to noise and distortion. Since bandpass filters can drift, spurious output may be difficult to control. These filters are costly as well.

TFT's transmitter design (Figure 3) improves on the older IF modulated transmitter by using three interrelated VCOs. It uses an IF modulated VCO rather than a modulated crystal and also eliminates the mixing step of the previous type of transmitter. TFT has been awarded U.S. Patent No. 4,710,970 for this design.

Using a phase-locked loop scheme, the transmitter's output frequency, generated by VCO3, is equal to the difference of frequencies generated by VCO1 and VCO2.

VCO1 provides frequency agility and serves as the FM modulator. It includes a divide-by-N phase-locked loop referenced to a crystal oscillator and is preceded by a high speed, two-modulus pre-scaler. This pre-scaler allows the frequency divider to be programmed in finer increments; therefore, the transmitter can be programmed in 2.5 kHz steps via DIP switches.

For ease of frequency synthesis by a divide-by-N loop and to achieve linear FM modulation, VCO1 operates at approximately 1/8 of the transmitter's output frequency. The PLL for VCO1 is designed with a low bandwidth (less than 1 Hz) so it can be directly modulated by the program audio. The modulator resembles the type used in high quality FM broadcast transmitters, yielding superior audio performance with distortion less than 0.1% and SNR better than 85 dB in the VCO.

FIRST GENERATION I.F. MODULATED TRANSMITTER

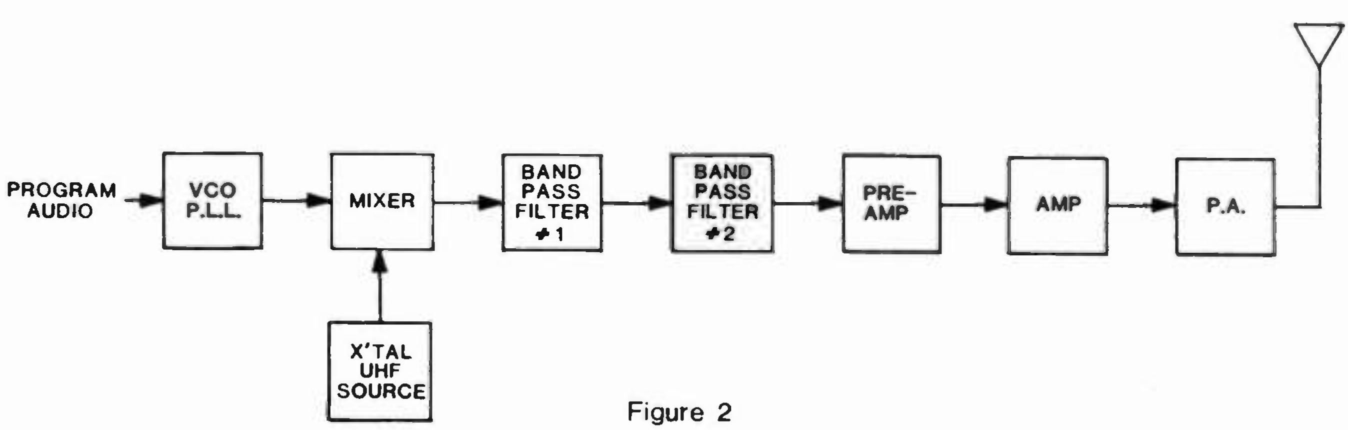

Figure 2

VCO2 provides a stable and low noise UHF source. It incorporates a wide bandwidth PLL with an oven-controlled crystal reference oscillator. These design elements ensure spectrum purity and provide high frequency stability with phase noise (side bands) exceeding 100dB below the carrier (dBc).

VCO3 generates the carrier frequency directly. With a high output level of +10dBm, less amplification is needed in the output stage. VCO3 is also a wide bandwidth PLL, yielding high spectrum purity and low phase noise (exceeding 100 dBc).

In order to translate the IF modulation (VCO1) to the output, a sample of the VCO3 output is down-converted by heterodyning it with VCO2's output. The down-converted signal, in the frequency range of 53.5-54.5 MHz, is compared in a phase detector to the output of VCO1. The output of the phase detector is used to correct the frequency of the output VCO. Since this control signal consists of the audio modulation applied to VCO1, it varies the output frequency in direct proportion to the audio modulation. With no frequency multiplier in the entire transmitter, the characteristics of the RF carrier are therefore exactly that of the modulated VCO1. The absence of any multipliers and frequency mixing for generation of the transmitter frequency ensures spectrum purity and high RF level at the source. Because no spurious harmonics are generated, costly tuning components and filters are not needed, thus resulting in a cost-effective design.

The RPU Receiver

The RPU receiver is also frequency agile with DIP switch programming and front panel selection between the two pre-programmed channels. IF bandwidth is also switchable to accommodate the transmitter's deviation or to compensate for spectrum crowding.

Provided on the front panel are a headphone output with level control, a diagnostic meter with multi-position selector switch, and a squelch adjustment. In addition to the audio output, the rear panel includes a repeater transmitter enable connection that is controlled by a DTMF security code.

The TFT RPU receiver (Figure 4) is a frequency synthesized, triple-conversion super-heterodyne receiver. It covers the 450 MHz and 455 MHz RPU bands and, within each band, is DIP switch programmable ± 500 kHz from the center frequency in 2.5 kHz steps.

The first IF section's local oscillator (L.O.) is 54 MHz above the received frequency. It consists of VCO1, whose design is very similar to that of VCO2 in the RPU transmitter. It also incorporates a wide bandwidth PLL for low phase noise, which is a superior technique compared with multiplication of a crystal oscillator. The phase comparator of the PLL is referenced to one of two ovenized crystal oscillators: one for the 450 MHz band and one for the 455 MHz band.

NEW GENERATION OF MODULATED TRANSMITTER (U.S. PATENT NO. 4,710,970)

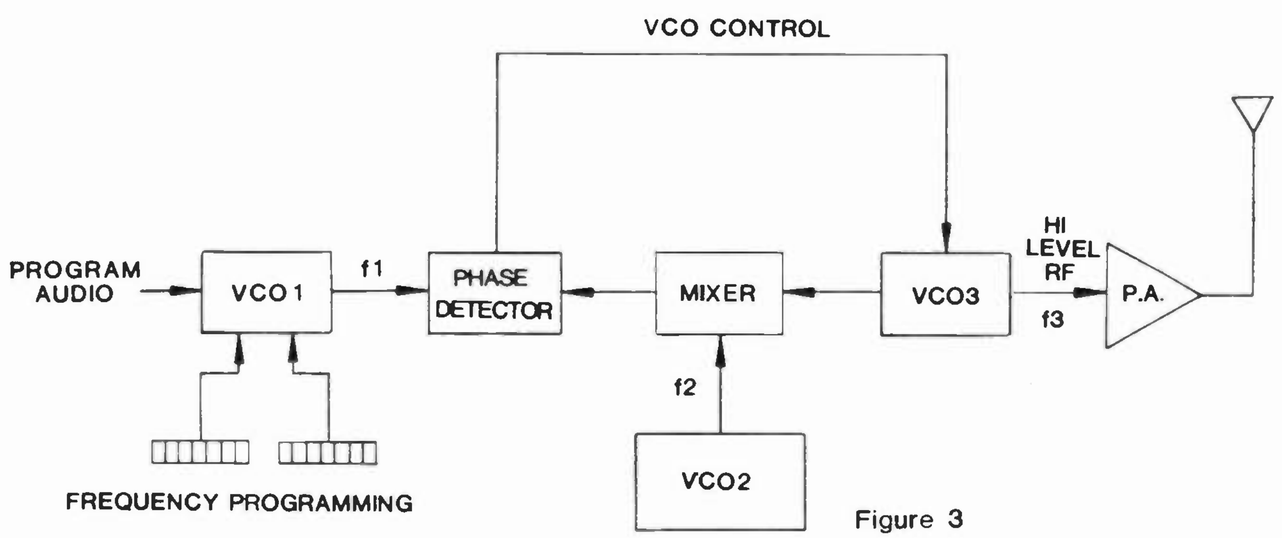

Figure 3

The first IF amplifier at 54 MHz incorporates a Surface Acoustic Wave (SAW) filter for excellent selectivity and image rejection. Since the image frequency is located 108 MHz away, it is greatly attenuated by the helical input filter. The second IF section utilizes an L.O. whose frequency source, VCO2, is very similar in design to VCO1 in the RPU transmitter and provides the frequency synthesis capability in 2.5 kHz increments. It is followed by a 10.7MHz filter.

The third IF at 10.7 MHz produces a 455 kHz output, which is passed through a user-switchable narrow or wideband filter. The Pulse Counting Discriminator provides audio demodulation. Its linearity is superior to that of a quadrature detector.

A DTMF decoder accepts a user-assigned 4-digit security code sent from the transmitter, and permits the selection of desired receiver channel, IF bandwidth, and activation of a repeater transmitter. External remote control functions can also be performed. For example, the remote operator can engage cart machines or automation equipment at the studio when a remote session is concluded.

These external remote control functions require external interface equipment supplied by TFT. The interface unit is capable of controlling eight functions and connects to the receiver via a twisted pair; up to nine units can be accommodated.

While DTMF tones are normally part of the received signal, a separate DTMF input is also provided as an alternate signal path. Consequently, control is not lost in the event that the received signal is not available.

RPU RECEIVER BLOCK DIAGRAM

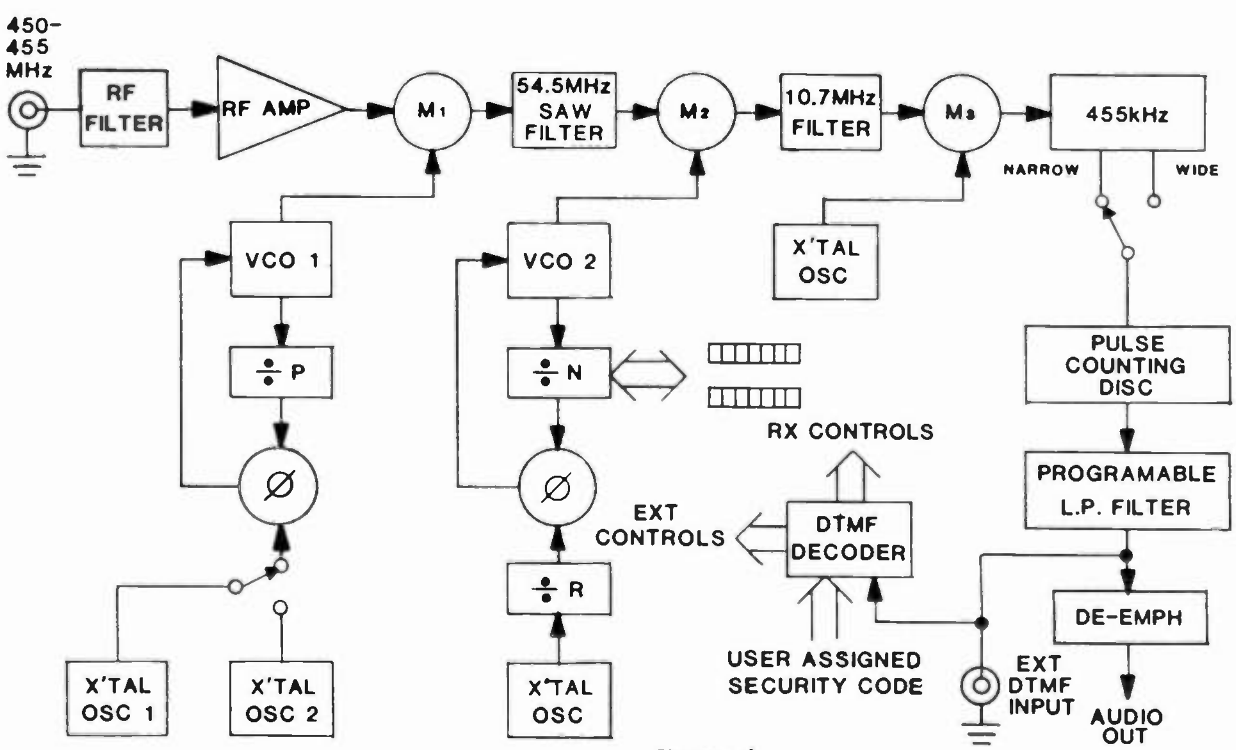

Figure 4

RF RADIATION REGULATION COMPLIANT AMMETER SYSTEM

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The regulations for human exposure to radio frequency (RF) radiation require special measures to control general public and occupational exposure.[1] The occupational exposure requirements present a unique problem to the AM broadcast engineer who is required to make periodic base current measurements.[2] Proximity to radiating components while performing the current measurements may expose the engineer to RF radiation for a time interval in excess of the guidelines.[3]

Since Delta Electronics, Inc. introduced the Model TCA RF Ammeter series in 1975, over 6200 ammeter systems have been installed in AM broadcast stations on a world-wide basis. In support of TCA users, Delta has been investigating a number of approaches to enable the broadcast engineer to accurately measure the base current without exposing the engineer to time-averaged field strengths exceeding the permissible levels, and thereby meet license renewal requirements with respect to ANSI exposure compliance.[4]

This paper will review the problems of occupational exposure to RF radiation occurring during measurement of base current. The paper will then explore the various approaches considered by Delta during the development of a solution to the metering problem.

INTRODUCTION

Every AM radio station in the United States must address the question of occupational exposure to nonionizing RF radiation during tower base current readings. For station license renewal, broadcasters at facilities licensed under Part 73 of the Federal Communications Commission (FCC) rules, which include commercial AM broadcast stations, are required to determine public and occupational exposure to RF radiation.[5] This, of course, includes exposure during tower base current readings.

Since health effects of exposure to nonionizing RF radiation are unclear[6], minimizing such exposure is a reasonable safety precaution and is, therefore, good engineering practice. Research on the biological effects of nonionizing RF radiation is continuing[7] and may lead to exposure standards that are tighter than the existing American National Standards Institute (ANSI) guidelines. Furthermore, the establishment of a standard for exposure implies a health risk for which the broadcaster may be held legally accountable. This risk is equivalent to any other business risk. For these reasons, the broadcast engineer should consider taking steps to minimize RF exposure, where feasible, to well within the ANSI guidelines.

Fortunately, methods have been developed to avoid excessive occupational exposure to RF radiation during tower base current readings. The remainder of this paper will discuss several of these methods, including those developed by Delta Electronics, Inc. First, however, the true magnitude of this problem and the state of ammeter technology must be described in order to clearly discuss the methods for solving the RF exposure problem.

THE PROBLEM DEFINED

The first step in addressing the potential RF exposure problem is to determine whether an exposure problem exists. Thus, the question to be answered is: will the engineer taking base current readings be exposed to electric and/or magnetic fields from all sources which, when averaged over any six minute period[8], result in exposure beyond ANSI guidelines? If, in the case of an AM only site, the electric field is less than 632 volts per meter and the magnetic field is less than 1.58 amperes per meter, the answer to this question is no and no RF exposure problem exists. That is, the radiation energy absorbed by the engineer would not exceed the ANSI guidelines. However, a prudent engineer might elect to take steps to minimize RF exposure for the reasons given above.

For multiple RF radiation sources, such as an FM station colocated with an AM station, the RF exposure is determined by adding the fractions of the ANSI exposure limit from each source.[9] This method of calculation is necessary because the human body absorbs RF energy more readily at FM frequencies than at AM frequencies.[10] Fortunately, the radiation source for FM is at the top of the tower and should, therefore, contribute little to the total RF exposure when measuring base currents. The engineer must not, however, overlook multiple sources when evaluating RF exposure during base current readings.

Unfortunately, the determination of the electric and magnetic field strengths in the vicinity of the tuning elements at the base of a tower is not a simple matter. Use of the tables, charts and antenna field equations contained in the FCC technical bulletin is not sufficient to guarantee exposure below ANSI limits because tuning elements produce locally intense fields[11] which cannot be taken into account in the tables, charts or equations. The best that the use of tables, charts or equations can do is to indicate a probability of exceeding the exposure limits. (Note that when using the tables and charts you must conservatively assume that all of the station's power goes to each antenna.)[12] Thus, the only acceptable method for determining the field strengths of interest is field measurements made with suitable instruments.[13]

Assuming that such measurements have been made and that the fields exceed the ANSI limits, base current readings may still be taken as usual if exposure times are limited so that the six minute time average exposure is less than exposure at the ANSI limits for six minutes.[14] As an example, assume that the measured fields are 2.45 times the ANSI limit: 1548 volts per meter or 3.87 amperes per meter. Remembering that power and, thus, energy absorption increases as the square of the voltage or current, the energy absorption rate is six times the rate at the ANSI limit. Therefore, to keep the same six minute averaged RF exposure, the engineer may make readings for up to one minute of exposure without exceeding the guidelines provided that for the previous five minutes and for the succeeding five minutes the engineer is not exposed to RF radiation. This time restriction is necessary to guarantee an acceptable exposure averaged over any six minute period. In such cases, the answer to the above question is, again, no. No RF exposure problem exists. Care should be exercised that anyone potentially exposed, especially new employees, be made aware of these exposure time limitations. This would ideally be implemented as part of the station's overall safety program.

In the above example, the one minute exposure limit would be plenty of time to make a base current reading. This appears an adequate solution but consider a four tower array. The time required to make a complete set of base current readings would be nineteen minutes. On bitter cold days, however, the person taking readings might decide to accept the unknown health hazard, excessive RF exposure, over the known health hazard. Clearly, the better policy is to find some ammeter technology that will allow base current reading away from fields that exceed ANSI limits.

AMMETER TECHNOLOGY

Two types of RF ammeters are used at the base of AM broadcast towers: thermocouple ammeters and the Delta TCA (Toroidal Coupled Ammeter) series ammeters. Thermocouple ammeters use a thermocouple to measure a temperature change in a dissipative element inside the ammeter. Since the dissipation and, therefore, the temperature increases as the square of the RF current, thermocouple ammeters are nonlinear. Their readings under modulation differ from their readings with no modulation. Thus, for the engineer to determine carrier power into the antenna, he must wait for a period of low modulation. But with today's heavy audio processing and emphasis on no "dead air", such periods are rare and directly result in longer exposure to RF radiation during base current readings.

Thermocouple ammeters are installed either permanently or, for lower power stations, temporarily via a meter jack. The proximity of the thermocouple ammeter to the tuning elements may expose the engineer to locally intense fields. For lower power stations, the use of a temporarily installed thermocouple ammeter may not cause RF exposure problems due to the low currents involved. However, measurements must be made to determine the field strengths.

The Delta Model TCA series RF ammeters consist of a toroidal current transformer feeding a patented, linear detector circuit. Since the current reading responds linearly with current, the readings are free of modulation effects except for small fluctuations due to extremely low modulation frequencies and changes in carrier regulation (carrier shift). This freedom from modulation effects means that the engineer may quickly make base current readings and avoid long exposure to RF radiation.

The original design for the Model TCA series of RF ammeters did not include provisions for remote metering. However, this design is no longer available and all TCA models have remote metering and are designated by the suffix EX. A variation of this model, designated by the suffix EXR, has remote controllable relays that allow the meter detector to be switched out of the circuit for lightning protection and allow scale changes for dual scale meters. A further variation removes the meter from the detector enclosure and is designated by the suffix XM or XMR. Although originally intended for phasor applications, these meters may find use in RF exposure avoidance applications.

Delta Electronics will be introducing a newly designed low current ammeter called the TCA Jr. The meter and associated circuitry are mounted on a meter plug suitable for temporary installation in a meter jack. Due to the low currents involved, the use of this meter is unlikely to cause RF exposure problems. However, as previously noted, measurements must be made to determine RF field strengths.

APPLICATION OF EXISTING TECHNOLOGY

Assuming that the engineer has an RF exposure problem or wishes to minimize RF exposure, his first step would be to try to adapt his existing system so as to avoid excessive RF exposure. For thermocouple ammeters, the answer would probably fall into the category of exotic solutions. For instance, the engineer might use a telescope or a television camera to view the ammeter from a distance.

With Delta Model TCA series ammeters, several opportunities exist to reduce RF exposure. Since the meter enclosure of a TCA system is separated from the toroidal current transformer by a coaxial cable, the meter enclosure may be moved to a location of minimum exposure. When planning a new installation, keep in mind that a TCA series ammeter may be ordered with a coaxial cable up to twenty feet long. For optimal accuracy, TCA series ammeters are calibrated with the coaxial cable supplied so cable substitution is not recommended.

Ideal coaxial cable is free from the influences of external fields but real coaxial cable is not. Although this is rarely a problem, the coaxial cable should be routed to avoid induced currents in the cable that could cause erroneous readings. A simple method to check for such errors is to power down; disconnect the coaxial cable from the toroidal current transformer, terminate both the coaxial cable and the current transformer in 50 ohms; reestablish the ground connection between the coaxial cable shield and the current transformer and power up (see CAUTION note). If an up scale reading occurs, an error signal is present and the coaxial cable should be routed to reduce the error or be encased in conduit.

CAUTION

Never disconnect a Model TCT-4 or TCT-5 current transformer under power as internal arcing will occur. Be sure that the termination resistors can dissipate sufficient power.

Utilizing the remote metering provisions of the TCA series ammeters is another method to reduce long term exposure to RF radiation. The engineer need only enter the high field area at the base of a tower often enough to ensure accurate remote readings.[15] This is, of course, much less often than is necessary for base current readings. Use of these provisions with remote control would further reduce RF exposure. In either case, the EXR model is preferred because its remotely controllable relays isolate the detector circuit from lightning damage.

FURTHER DEVELOPMENTS

What does an engineer do, however, if the RF fields are so high that his base readings can never be taken without violating the ANSI guidelines? Legal releases are deemed unacceptable[16], so a technical solution must be found. Again, exotic solutions could solve this problem, but Delta believes that cleaner, simpler solutions exist. To this end, Delta engineers are investigating several possible answers.

The following are important considerations for any solution to this problem:

1. The solution must yield accurate, repeatable readings that do not require calibration against the base meter.

2. The solution must be acceptable to the FCC.

3. The solution must be compatible with the thousands of existing TCA series RF ammeters.

4. The solution must be simple to operate.

5. The solution must be low cost, contributing only a small fraction of the ammeter cost.

6. The solution must be maintainable.

The first two considerations were discussed with the FCC and they advised that as long as Delta can guarantee the accuracy of meter readings, such meters would be considered base meters and not remote meters. Therefore, if consideration 1 is met, then it appears that consideration 2 is also met.

The first idea was to use a Model TCA XMR ammeter with the DC meter cable extended to outside the ANSI field limit. Coaxial cable would be used between the detector enclosure and the meter movement to limit RF pickup. The meter movement would have to be protected from weathering and RF fields in a metal enclosure. A control cable and power supply would also be supplied to operate the relays inside the detector enclosure.

The disadvantages of this scheme are the need for a protective enclosure, a power supply at the base requiring AC power, and a maintenance problem for the coaxial cable to the meter movement. This coaxial cable must be removable since it is part of the meter system necessary for calibration. Thus, this cable could not be permanently buried.

The second idea was to simply extend the coaxial cable until the meter enclosure is outside the area of high field using double shielded coaxial cable or, perhaps, triaxial cable. However, review of the literature on coaxial cable susceptibility to external RF fields reveals that even the best triaxial cables or super-mu coaxial cables are surprisingly subject to RF fields. The resulting error currents in an RF ammeter are, of course, not acceptable.

Burial of the coaxial cable as a means of avoiding RF fields is not entirely effective because the shield of the coaxial cable will, to some extent, become part of the antenna's ground system carrying RF currents. Burial also presents a maintenance problem. The coaxial cable is part of the ammeter system and must be returned with the ammeter for repair and calibration. The cable could be buried in plastic pipe to facilitate removal, but the plastic pipe and the labor to install it become a "hidden" expense.

Use of triaxial cable requires triaxial connectors on one or both ends of the cable. These connectors require metalwork modifications to the aluminum castings of the meter enclosure and/or the current transformer enclosure. Since the best connection configuration for the outer shield has not been determined, the extent of these modifications is not known.

Preliminary field trials show promising results. The extent of the error currents in carefully installed coaxial cables of less than sixty feet, as evaluated using the method described above, reveal that the error currents are at acceptable low levels. However, since the error signal is indistinguishable from the desired signal, further field tests are necessary before this solution to RF exposure can be adopted.

The third idea was to convert the external output of a Model TCA EXR ammeter to an audio tone for easy transmission to an area of low field intensity. The audio tone frequency would be proportional to the ammeter reading and be factory calibrated to, say, 10 kHz full scale. The chief advantage of this scheme would be transmission of this tone via media that are insensitive to RF fields and do not require expensive, buried cables. One option is a short length of the low price, plastic version of multi-mode fiber optic cable buried just below the surface. Another option is a short range VHF radio link using a device similar to a police radar gun.

Low cost, precision voltage-to-frequency and precision frequency-to-voltage circuits were built and successfully tested in the Delta development laboratory. However, with the addition of power supplies and housings at each end and conversion circuits for RF or fiber optics, costs escalated to significantly high levels. Therefore, this solution does not satisfy consideration 5 above.

The fourth idea is incorporation of a DC reference source within a Model TCA EXR meter enclosure so that a remote meter can be calibrated remotely. The remote meter is then renamed a secondary meter. Physically, this involves the addition of a small printed circuit assembly inside the TCA EXR meter enclosure. The resulting meter is called a Model TCA EXRS. The same metering and control cable that would be used with an ordinary TCA EXR is used with the TCA EXRS. Thus, if TCA EXR meters are installed with remote metering, the solution to the problem is particularly easy.

For secondary meter readings, positive 24 volts on the control lines activate a relay that connects the RF sample from the current transformer to the patented, detector circuit and the secondary meter reading is taken just as with a Model TCA EXR. Meter linearity errors are taken into account by adding the remote correction curve supplied with the TCA EXRS to the correction curve supplied with the secondary meter and adding this total correction to the secondary meter reading. To calibrate the secondary meter, a negative 24 volts is applied to the same control lines activating the reference source. The reference voltage is applied to the output of the detector circuit yielding a reading at both the base meter and the secondary meter. The meter multiplier potentiometer for the secondary meter is then adjusted to read at the calibrate scale mark. The secondary meter is ready for accurate base current readings beyond the high field area.

The metering and control cable need not run all the way back to the transmitter building. A fence post placed just outside the ANSI field limit holding a standard weatherproof electrical outlet box is an option for use with a battery operated, multiscale, portable secondary meter named the Delta Model TCA PSM. After base current readings are taken, the portable secondary meter is installed in a charging stand at the transmitter building to recharge the batteries.

This scheme has several important advantages. The metering and control cable may be permanently buried with no calibration maintenance problem. Any changes in this cable characteristics due to absorbed moisture are calibrated out. RF pickup in the metering cable is not a problem since the DC signal is easily distinguished from such interference. The method of using DC signaling is field proven in TCA EXR remote metering applications. With the station off the air, the calibration circuit can be checked by observing that the base meter reads at the calibration reading, that is, the major scale division at 70 or 75 percent of full scale depending upon the scale.

This solution appears to meet all the acceptance considerations listed in 1 through 6 above. The only real disadvantage is that Model TCA EX and TCA EXR ammeters would have to be returned to the factory for conversion and test as Model TCA EXRS ammeters.

As of the writing of this paper, none of these schemes have been field tested sufficiently for release as a Delta product. However, we intend to complete our testing program in time to announce at the 1988 NAB convention which options successfully completed field testing and are, therefore, available to the broadcast community.

SUMMARY

The contribution to RF exposure of locally intense fields produced by tuning elements at the base of an AM broadcast tower can only be evaluated by measurement of those fields to assess conformance to ANSI RF exposure limits. If the fields are above these limits, base current measurements may still be taken without exceeding the ANSI exposure guidelines by limiting exposure times or by moving the measurement location outside of the high field area. Several methods of avoiding excessive RF exposure already exist using existing metering techniques and methods for dealing with extreme cases are under development at Delta Electronics and will soon become available.

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[12]Ibid p. 15

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ASSESSING PERSONNEL EXPOSURE TO MAGNETIC FIELDS ASSOCIATED WITH AM RADIO BROADCAST TOWER MATCHING NETWORKS

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ABSTRACT

Federal Communications Commission (FCC) rules require that broadcast stations insure that station employees or contract personnel not be exposed in excess of the radiofrequency (RF) radiation protection guide developed by the American National Standards Institute (ANSI). An often neglected aspect of this issue is that of occupational exposure of individuals working in close proximity to AM radio broadcast tower matching networks. AM tower impedance matching circuits, typically installed near the base of AM towers, can lead to relatively intense magnetic fields associated with the inductors used and relatively large RF currents flowing in the circuits. These magnetic fields may commonly exceed the limits recommended by ANSI. This paper addresses the magnitude and spatial extent of these magnetic fields for commonly used inductors and provides practical guidance on how to avoid exceeding the ANSI limits by controlling the exposure time. A calculational technique is presented, along with confirmatory measurement data, which has been applied to a large number of typical AM matching circuit inductors, yielding clearance distances from the inductors as a function of inductor current that will result in magnetic field strengths less than 1.58 A/m, this being the ANSI limit in the AM standard broadcast band.

INTRODUCTION

The recent advent of new environmental processing rules by the FCC which require that broadcast stations, both applicants for new licenses and license renewals, insure that station employees, contract personnel or the general public not be exposed to RF fields in excess of the ANSI RF protection guide is beginning to have an effect on many broadcasters. For the most part, this impact has been on stations showing that they comply with the ANSI limits relative to exposure of the surrounding area about the station transmitter site; i.e., that the station is not responsible for excessive exposure of the general public which lives in the vicinity. However, the FCC rules state that measures should be taken to keep worker exposures within acceptable levels as well and it is within this area that broadcast stations, as a whole, have the most difficulty in coping with the new rules. This is because, in many cases, it is frankly difficult to change operational methods that have been ingrained after many years and in others, it is technically difficult to avoid high levels of electromagnetic fields. The case of AM standard broadcast radio operations is a prime example of at least two of these technical problems; how to avoid excessive RF exposure during (1) live tower climbing for tower painting, rigging, etc., and (2) performing base current readings and other tasks at the base of the tower where personnel may come into extremely close contact with tower impedance matching circuitry. This paper addresses the latter of these issues.

THE ANSI STANDARD AND AM RADIO FIELDS

The ANSI standard is based on the concept of limiting the rate at which energy is absorbed from the field by the tissues of the body. The specific absorption rate, or SAR, is related to the strength of the incident fields and it is the fields which are usually specified as the limits of the ANSI standard. The fields are derived from a consideration of limiting the SAR to 0.4 watts per kilogram (W/kg) as averaged over the entire mass of the body and 8 W/kg in any one gram of tissue. Due to the fact that the body tends to act like an antenna, i.e., it possesses a frequency response, RF field strengths are relatively less restricted in the AM band as compared to limits in the VHF portion of the spectrum where the body can exhibit relatively strong resonances.

For RF fields in the 0.3-3.0 MHz range, the ANSI standard recommends maximum field strengths of 632 volts per meter (V/m) and 1.58 amperes per meter (A/m) for electric and magnetic fields respectively. For most situations involving AM radio, field strengths which might approach the ANSI limits will typically be very close to the tower and thus in the so-called near-field. For AM tower installations, near-field exposure almost always consists of low-impedance fields. This means that the field impedance which is equal to the electric (E) field strength divided by the magnetic (H) field strength is less than 377 ohms, the value for free space plane waves. Thus, as far as tower radiation is concerned, it is the H field which has the greater probability of exceeding the ANSI limit when compared to the E field.

RF exposures found around the tower tuning circuits are similar in that it is more common to find excessive H fields than excessive E fields. It is the magnetic field associated with the inductors (coils) used in these tuning or matching circuits which is the subject of this paper.

THE PROBLEM

Because matching circuit inductors, when carrying the normal currents found in AM radio broadcast systems, can produce intense magnetic fields close to them, and because workers often must occupy areas very close to these inductors, it is of special interest to evaluate the fields of inductors. The objective of the project reported here was to develop an approach for determining the magnetic field strengths near such inductors and to find the distances away from the inductors at which the magnetic field would be equal to the ANSI standard. With this information it was believed that guidance on limiting RF exposures could be developed that would be realistic and at the same time provide for the required protection.

THE SOLUTION

Following an analysis of helix coils by Gailey (1987), the inductors used in AM tower matching circuits were represented by a spiraling helical coil represented in Figure 1. Through application of the Biot-Savart law, the magnetic field components at any point about the coil can be derived and represented by:

$$B_x = \frac{\mu_0 I}{4\pi} \int_0^{2\pi n L} \frac{\left[a\left(z - \frac{\theta}{2\pi n}\right)\cos\theta - \frac{1}{2\pi n}\left(y - a\sin\theta\right) \right] d\theta}{D}$$

$$B_y = \frac{\mu_0 I}{4\pi} \int_0^{2\pi n L} \frac{\left[a\left(z - \frac{\theta}{2\pi n}\right)\sin\theta - \frac{1}{2\pi n}\left(x - a\cos\theta\right) \right] d\theta}{D}$$

$$B_z = \frac{\mu_0 I}{4\pi} \int_0^{2\pi n L} \frac{\left[a^2 - ay\sin\theta - ax\cos\theta \right] d\theta}{D}$$

where a = coil radius
L = length of coil
n = number of turns/unit length
u_0 = permeability of free space

and

$$D = \left[a^2 + x^2 + y^2 - 2ax\cos\theta - 2ay\sin\theta + z^2 - \frac{\theta z}{\pi n} + \frac{\theta^2}{4\pi^2 n^2} \right]^{3/2}$$

X,y and z are the coordinates of the desired calculation point; z is the value along the coil axis measured from one end.

These equations represent elliptical integrals and can be solved by numerical integration on a computer. It is found, that for any given distance from the surface of a coil, the maximum value of magnetic field is located on the axis of the coil. For the axial component of magnetic field, the above expression for B_z can be simplified to a closed form:

$$B_z = \frac{\mu_0 I n}{2} \left[\frac{z}{(a^2 + z^2)^{1/2}} - \frac{(z - L)}{[a^2 + (z - L)^2]^{1/2}} \right] \qquad (1)$$

The analysis reported here made use of the more complicated expressions to determine a surface about a particular coil geometry at which, for a given amount of current, the magnetic field strength would be 1.58 A/m. The integral expressions were numerically integrated on a VAX 11/780 computer using an algorithm which iteratively found the distance from the coil at which the magnetic field strength was 1.58 A/m. A coil geometry of 23 cm diameter , 40 cm length and 9 turns was used for this calculation to illustrate the general shape of the ANSI magnetic field surface. Figure 2 illustrates what this surface looks like for the chosen coil geometry. This specific geometry was used to simulate some of the larger inductors found in AM tower matching systems and, at the same time,was also used in the fabrication of a model coil which was then used in laboratory measurements to verify that the computation scheme worked correctly.

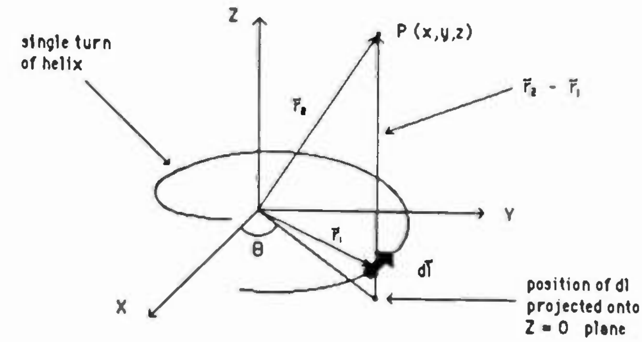

Figure 1. Illustration of a single turn inductor. Taken from Gailey (1987).

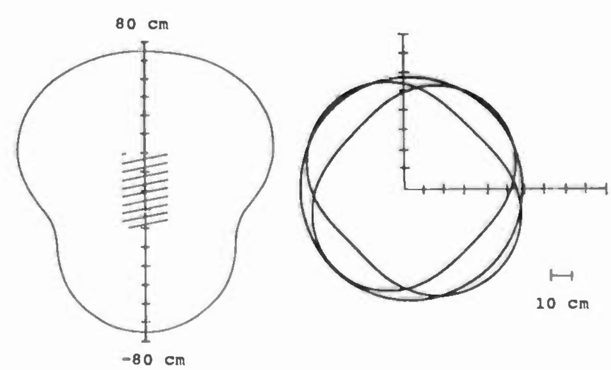

Figure 2. A: Side view of cross-section of 1.58 A/m contour for experimental coil used in tests. The ends of the coil are the areas which require the greatest clearance to avoid exceeding the ANSI magnetic field limit. B: Axial view of 1.58 A/m contour cross-section showing the oblateness caused by the pitch of the coil. Were it not for the pitch, the surface would be a perfect ellipsoid.

Figure 2-A, a side view of the coil showing the ANSI contour, shows that the areas near the ends of the coils are the areas of greatest magnetic field strength; i.e., the distance on the axis of the coil to the point where the magnetic field strength is 1.58 A/m is the greatest compared to other points on the surface of the coil. Due to the pitch of the coil windings, an asymmetry appears toward both ends with an oblateness to the generally circular cross-sectional area of the solid volume. This is apparent in

Figure 2-B which provides cross-sectional views of the 1.58 A/m contours as viewed looking down the coil axis. The circular contour occurs at the center of the coil's length. The other two contours are those toward the ends of the coil where the oblateness due to the coil's pitch is most apparent.

Laboratory measurements were made of the magnetic field strength at various points on the axis of the model to assess the validity of the calculational approach. Figure 3 shows both the computed axial component of magnetic field strength and the measured value. The measured values are seen to be in good agreement with the theoretical values. The deviation between measured and computed values was found to be in the range of about 0.7 to 1.4 dB depending on the position along the axis of the coil. Magnetic fields were measured with a Holaday Industries, Inc. Model HI-3006 isotropic broadband magnetic field strength meter which had been calibrated in a transverse electromagnetic (TEM) cell. The coil was driven from a linear amplifier at 1 MHz with 1 amp RMS.

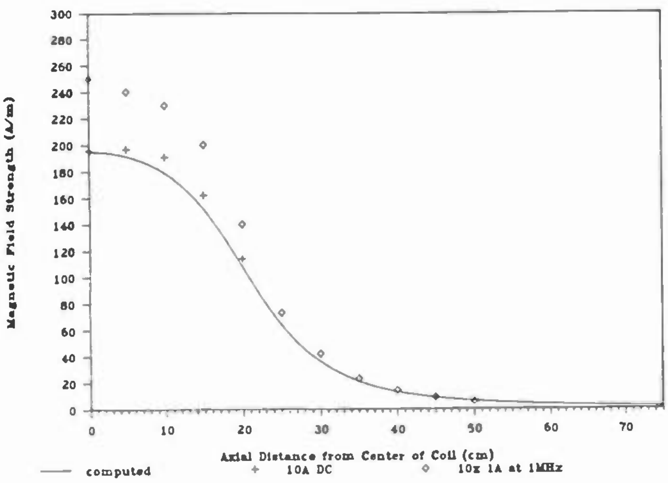

Figure 3. Computed and calculated values of the magnetic field strength along the axis of the experimental coil.

With verification that the calculational approach produced correct results, it was decided that computed values of magnetic field strength would be obtained for a variety of real-world inductors typically found in AM station antenna networks. Through the cooperation of KinTronic Laboratories (P.O. Box 845, Bristol, TN 37621-0845), detailed data were obtained on the physical specifications of 179 different inductors ranging in diameters from 4 inches to 24 inches, lengths of from 2.5 to 52 inches, number of turns from approximately 3 to 65 and current carrying capacities from 10 to 150 amperes. These inductors, designed for use in AM radio antenna networks, were taken to be representative of the kinds of inductors likely to be found in most AM radio antenna installations. The detailed physical specifications were used for subsequent computations.

RESULTS

Using the closed form of the expression for axial magnetic field strength, equation (1), calculations were made for each inductor to obtain the distance from the end of each inductor to the point on the inductor axis at which the magnetic field strength was equal to 1.58 A/m as a function of inductor current, up to the rated limit of each inductor.

Figures 4 and 5 illustrate the nature of this variation for two different inductors selected from the entire group. These two inductors are representative of the extremes in sizes from the group but show the tendency of how the axial distance to the ANSI magnetic field limits varies with current; this same general relationship was found for all of the inductors, the specific values being simply dependent on the inductor dimensions and current.

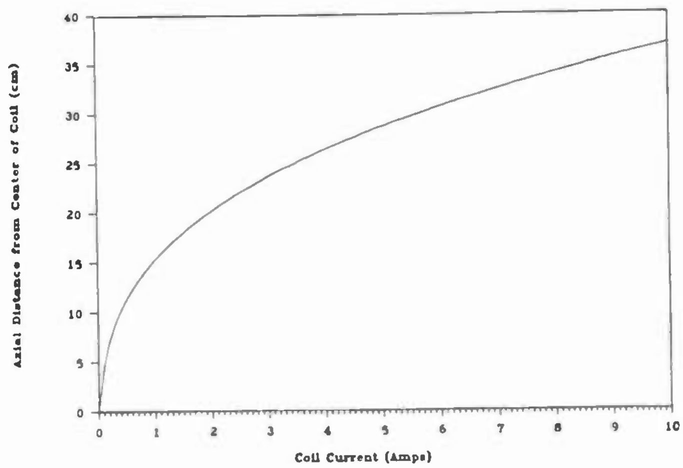

Figure 4. Distance from either end of KinTronic inductor Model L5-10 at which the magnetic field strength is equal to the ANSI guide as a function of inductor current.

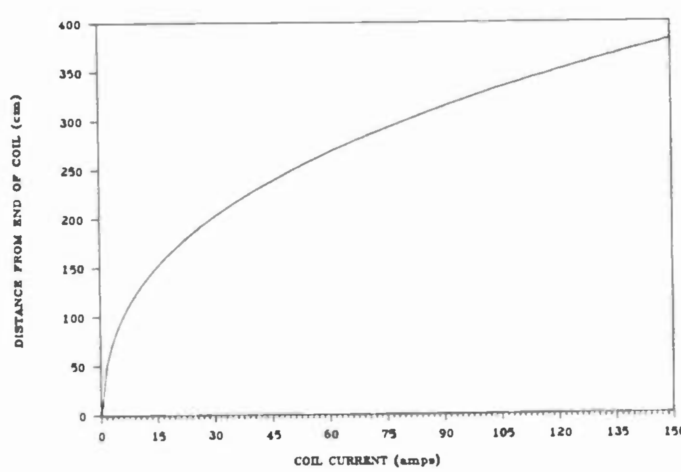

Figure 5. Distance from either end of KinTronic inductor Model L98-150 at which the magnetic field strength is equal to the ANSI guide as a function of inductor current.

The large number of inductors evaluated prevents graphical presentation of the results for all inductors but the overall findings are summarized in Table 1. Table 1 shows, for each of the categories of current ranges for inductors, the associated distance range within which a magnetic field strength of 1.58 A/m will occur on the axis of the inductor when the rated current is flowing through the inductor. Each of the current values shown represents between 9 and 19 different models of inductors for which calculations were performed. The table represents calculations for a total of 179 inductor models from KinTronic Laboratories.

Table 1. Summary of Magnetic Field Data for Inductors

Rated Current (A)	ANSI Magnetic Field Distance (cm)
10	41.3 - 79.8
15	42.8 - 93.6
20	47.5 - 106
30	70.6 - 156
40	84.2 - 168
50	98.4 - 170
60	136 - 230
80	177 - 314
100	194 - 342
150	194 - 381

The data given in Table 1 are meant to be used as a guide only; actual field strengths are a function of the specific coil diameter, number of turns, spacing of the turns and current. The primary conclusion of the analysis is that relatively high magnetic field strengths can be produced near the inductors commonly used in AM radio tower networks and care must be exercised when working near such inductors to avoid exceeding the ANSI recommended level of 1.58 A/m.

ANSI TIME AVERAGING PROVISION

While the magnetic fields close to AM tower network inductors can exceed the ANSI recommended level of 1.58 A/m, this does not mean than exposure to even higher field levels is not permitted. The ANSI standard actually specifies that the value of 1.58 A/m magnetic field strength is the value not to be exceeded for exposure as averaged over 6 minutes (Tell, 1986). For example, a higher field would be allowed for periods less than 6 minutes, provided that there is sufficient time in the remaining part of the 6 minute period at low or zero levels so that the average is within the limit of 1.58 A/m.

The way the ANSI standard is written, this time averaging provision is specified as follows:

$$H^2(A^2/m^2)t(\text{min}) \leq 15\ A^2\text{-min}/m^2$$

If a worker had to occupy an area near a high current carrying inductor, for example in the 'dog house' at an AM tower to perform tower base current readings, where the magnetic field strength was measured to be 6 A/m, then the above relation would restrict the time of exposure to 0.42 minutes or 25 seconds. Following this 25 second exposure to 6 A/m, the worker's exposure would have to be reduced to zero for the next 5 minutes and 35 seconds to meet the averaging rule.

The time averaging provision can be used to advantage in AM radio operations to avoid exceding the standard for RF exposure used by the FCC. Stations should provide for some means of measuring the magnetic field strength near AM tower matching networks and the time that personnel must occupy high magnetic field strength areas. Submission of information describing the above procedure and the resulting averaged exposure levels ($\leq$ 1.58 A/m) to FCC should be sufficient to document that the station has met compliance with the FCC application of the ANSI standard.

As an alternative to the described measurment process, stations could elect to limit exposure to acceptable levels through other means. An example would be to provide for any necessary antenna current measurements via remote means such that personnel do not have to be in the immediate vicinity of the current carrying inductors near the tower base. Such an arrangement, however, would still require an initial measurement to verify that the composite fields caused by all inductors at the remote measurement point are less than 1.58 A/m.

The analysis presented in this paper has addressed the fields caused by single inductors but in reality, often several inductors are used in the matching networks. Because of the large variety of network layouts, with an infinite number of variations in how the inductors are configured, it is not practical to compute the resulting fields at all possible positions. The results given here show that maximum field strength occurs off of the end of the inductor, on its axis; common sense in avoiding such positions should be used to minimize exposures.

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TAMING LIGHTNING AROUND BROADCAST TOWERS

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Abstract

In a time when lightning activity is becoming more of a problem to broadcasters, the protection industry is creating more and more confusion. The lack of standards addressed to the broadcasters, and the constantly changing situation has made it difficult for the station engineers to make a proper choice for protection. Claims seem similar, while costs vary by many thousands of dollars.

This paper deals with one aspect of that problem, the direct strokes to towers. It defines the problem and the resulting design requirements imposed on a protective system. It evaluates the products on the broadcast market against these requirements, and identifies their weaknesses. Finally, it identifies the Dissipation Array™ System concept (US Patent No. 4180698) as the only option that can assure a 100% lightning free environment for the broadcasters, transmitter site, and presents 17 years of history to vindicate that premise.

INTRODUCTION

Since the inception of the broadcast industry, lightning has proven to be a serious challenge and a costly adversary. In recent years that challenge and the related cost factors have become a significantly greater problem. Increased exposure through use of sophisticated programming equipment; and, micro-miniaturization has proliferated these problems exponentially. Simply stated, these new circuits cannot function satisfactorily in the presence of the lightning related environments that were considered not to be a hazard a few years ago. Suppression devices only deal with part of the problem - not the direct strike.

During the past 15 years, there has been a great deal of research and development in the field of lightning protection. Several companies have been formed for the express purpose of providing various forms of lightning protection devices. Most are devoted to power conditioning. Some have worked well, others are of little use particularly to broadcasters.

Broadcasters usually have a greater exposure factor than any other industry. This is because they are normally in remote locations; and, the transmitter towers are natural lightning rods. The presence of those towers are the greatest contributing factor to lightning damage. They are frequently struck, and the resulting primary and secondary effects can cause minimal to devastating damage. The actual impact depends on the character of the stroke and the age or susceptibility of the related broadcast facility.

In 1971, a new form of lightning stroke protection was introduced to the broadcast industry; a system called the Dissipation Array™ System (DAS). Through proper use of the DAS™, it has been proven that lightning can be prevented from striking a broadcaster's tower. As in all R & D efforts, the DAS system encountered some failures in the early years of its use. In spite of that, it has achieved a 99.7% reliability. Recently, the market has been invaded by incompetent attempts to duplicate the DAS success. Their failures have created doubts within the broadcast industry, as to the DAS effectiveness. This paper deals with the differences and discusses the critical design factors.

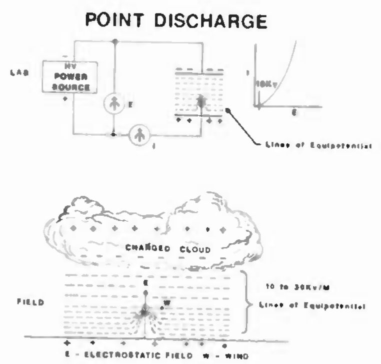
THE DISSIPATION ARRAY CONCEPT

The DAS concept is based on a phenomena known as "Point Discharge", "Corona Discharge" or St. Elmo's Fire. Each of these expressions describe an electrostatic phenomena involving the use of nearby air molecules to carry away an electrostatic charge from a highly charged body.

Point Discharge, as a phenomena, has been known and understood for years. Atmospheric physics teaches us that up to 90% of the storm energy is dissipated through this phenomena. Trees, grass, fences and edge tend to form "natural points". However, these natural points are not very efficient. They are usually too close; or not enough per unit area. As an example, a pine forest has millions of points (pine needles) however they are too close and are struck during severe storms.

To illustrate the point discharge phenomena, consider Figure 1, where a laboratory experiment is illustrated, then transposed to the field.

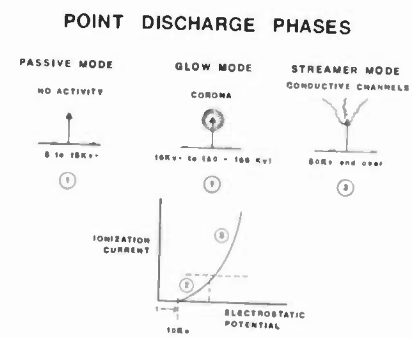
Figure 1, Point Discharge



A point in an electrostatic field will ionize the surrounding air, if elevated to a potential of at least 10Kv. Notice how the lines of equipotential form around the point. The concentration on the point indicates a high difference in potential between it and the surrounding air, thus facilitating the transfer of an electron-ionization. In the field, the same phenomena exist except the tower may be the "point".

As illustrated by Figure 2, points are not always ionizing the air. There are actually three modes of operation as illustrated.

Figure 2, Three Modes of Operation, Sharp Points in an Electrostatic Field.



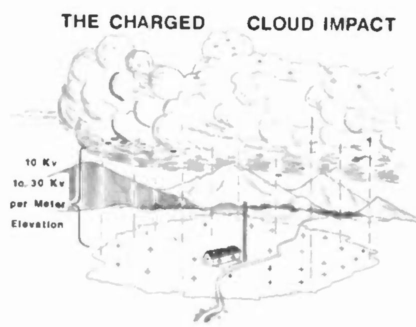
Point discharge or ionization takes place during the glow mode only. The streamer mode produces an avalanche of ions, forming a conductive channel. These conductive channels then become natural targets for lightning. They must be avoided to prevent direct lightning strikes. From this we see that a dissipative system must rapidly pass into the glow mode, but never beyond that mode, even under the worst of case situations (strong electrostatic fields.).

This leads to:

Design Rule One: The Dissipation Array System must remain in glow mode throughout the storm.

Next, a review of the storm character is in order. From Figure 3, we see that a storm cell induces an electrical charge on the surface of the earth, under that cell.

Figure 3, The Charged Cloud Impact

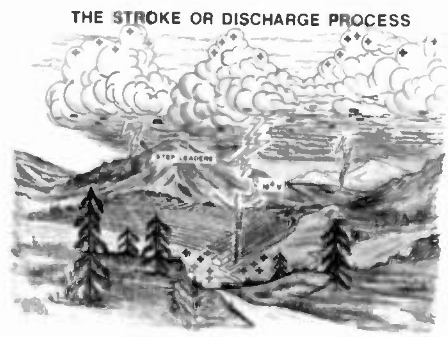


As the storm progresses, the electrostatic field follows that cell as it moves with the storm, inducing a charge on everything within its sphere of influence. The result may be thought of as an electrical shadow.

From Figure 4 we see that when the storm reaches maturity and the cell potential reaches about 10^8 volts, conductive channels move from that cell toward earth. It moves in steps, called: "Step Leaders". This process is similar in influence to unreeling a wire from the cloud toward earth.

The first downward moving leader that meets an upward moving streamer closes the circuit and lightning results. The stroke terminates only after the charge between the two bodies has been effectively neutralized. If there were no charged bodies with a significant difference in potential, there would be no strike. The stroke neutralizes the charge.

Figure 4, The Stroke or Discharge Process.



Design Rule Two: The Dissipation Array System must dissipate the site charge slowly, maintaining the potential below that conducive to lightning. Continuous current flows of up to 10,000 microamperes or more may be required.

From the pine trees we found that an abundance of points alone is not the complete solution. Other parameters are obviously influencing the rate of ionization produced. Work performed by many Atmospheric Physicists and summarized by Dr. Alan Chalmers in his book: "Atmospheric Electricity", provides some insight on the remainder of the ionization design problem. His data demonstrate that:

1. Single points provide only a few hundred

microamperes before the streamer/avalanche mode.

2. Many points in close proximity produce less current for a given situation, than the single point.

3. The greater the point separation and the higher the point above the conductor, the greater the ionization current.

4. Location and orientation of the points, significantly influence the ionization process.

LEC research has proven that when points are side-by-side (little to no separation, as in a wire brush), the assembly functions almost as if it were a solid conductor. Further, a thin wire of the same length will often produce more corona current.

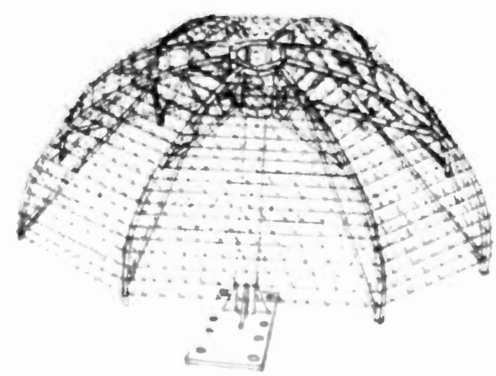
Strike Prevention in Practice

The foregoing has presented the salient parameters for the design of a lightning strike prevention system, based on point discharge. It is now incumbent on the designer to integrate these into a system configuration that fits each situation, without compromising any of these parameters. To that end, LEC has developed the Dissipation Array System™ (DAS)™, with many basic configurations all of which satisfy these requirements. DAS configurations have been developed that are applicable to any facility configuration.

Towers from 100 feet to 2000 feet have been protected using a derivative of one, or a combination of three basic ionizer configurations. Though they may vary in size and method of deployment. These are:

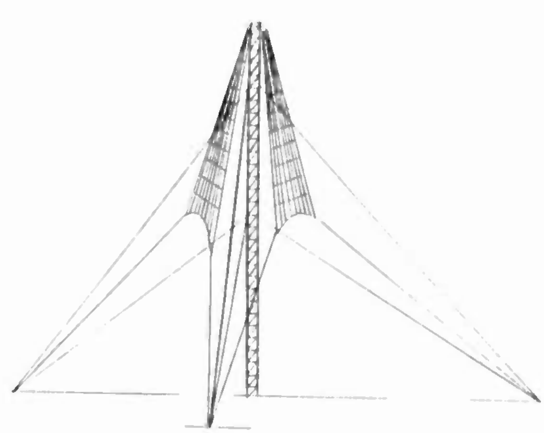
1. The Hemisphere Array is illustrated by Figure 5. It varies in size from 3 to 6 meters in diameter, and is shaped to conform with the "Lines of Equipotential" as they form around the DAS and the tower as a unit. Point height and separation is established to provide maximum ionization or dissipation efficiency per unit area of dissipator surface. This configuration has been proven effective for heights of up to 100 meters.

Figure 5, The Hemisphere Array



2. The Trapezoid Array is illustrated by Figure 6.

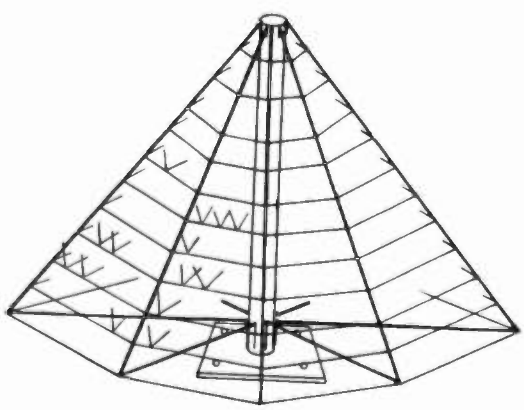
Figure 6, The Trapezoid Array



Each panel varies in area from about 300 square meters to over 500. Panel length must vary with tower height. Three panels are required, one in each of the planes formed by any two of the adjacent upper guy wires.

3. The Conic Array is illustrated by Figure 7. It is not often used on towers, but for unusual situations, it may provide the only viable solution.

Figure 7, The Conic Array

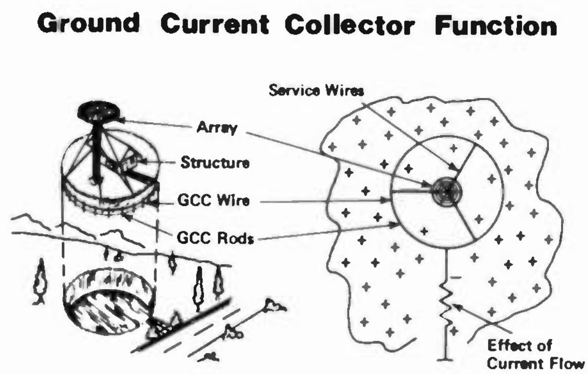


The Conic Array uses a center pole to support the apex. The face of the ionizer is in the shape of a truncated cone, providing a dissipative area in excess of 100 square meters, again selected to suit the specific application.

The Ground Current Collector Function

The Ground Current Collector (GCC) functions as the "charge collector". In contrast to a grounding system, the GCC must collect the charge as it is induced on the site, and provide a low impedance path to the ionizer. Figure 8 illustrates a typical GCC deployment for a broadcaster.

Figure 8, The Ground Current Collector Function



Without this function, a lightning strike prevention system is incomplete. To dissipate a charge, it must first be collected; then it must be carried swiftly to the ionizer. For example, a pump cannot pump water unless the water is first collected and piped to the pump. The DAS™ may be considered an electrostatic pump.

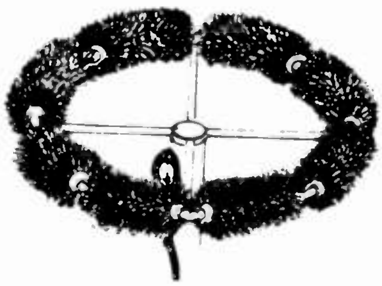
Comparing the Market Options

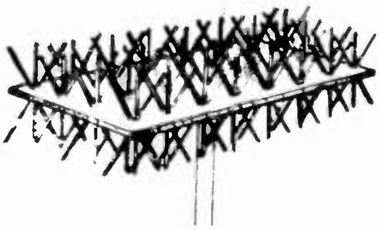
To date, there are at least four different devices, marketed toward the broadcaster, that are claimed to prevent lightning strikes; that is, in addition to the LEC Dissipation Array System. A comparison of the differences, and to the required design parameters, reveals some important factors. These differences are in the ionizer design, the grounding and the warranty. First, the DAS is a system, not a component or gadget.

1. Ionizer Design. Their ionizers are too small, the

points are too close and the shapes do not take the electrostatic field formation into account. Figure 9 illustrates their design concepts. Compare them to the Hemisphere of Figure 5.

Figure 9, The Hybrids





2. The Ground Current Collector function is not considered in any of the four devices. They depend on customer grounding, with no statement as to what is required of that ground system.

3. Warranties. Their warranty is carefully worded to protect the seller. The DAS is guaranteed functionally. These other devices may promise the money back; however, the customer must first pay for installation and removal, plus the freight both ways. That adds up to more than the cost of the gadget.

The DAS History

Since 1971, the Dissipation Array System has been used to protect the broadcast industry against lightning strikes. To date, over 670 systems have been installed, a large percentage of which are for broadcasters. The DAS has demonstrated a 99.7 percent reliability in preventing lightning strikes. A few examples have been selected to illustrate this point. All of these have been working for at least 5 years:

Table 1, Survey Samples

	Outages*		Years
Station	Before	After	Service
CKLW	12	None	16
KIVP	25+	None	9
WFPG	4+	None	10
WZAR	2+	None	8
KFSN	4	None	6
WKZL-TV	15	None	5
KINI-FM	(Many)	None	9
KLAS-TV	8	None	11
WBBH-TV	20	None	12

*Due to Lightning Activity

Summary Situation

In summary, there are two facts that need to be emphasized:

1. Lightning strikes can be prevented from terminating in any area of concern by slow ionic discharge.

2. The DAS is the only complete system; and, the only system that can be made 100% effective. It is the only system with 17 years of history and with the broadcast industry and a functional guaranty to back the claims.

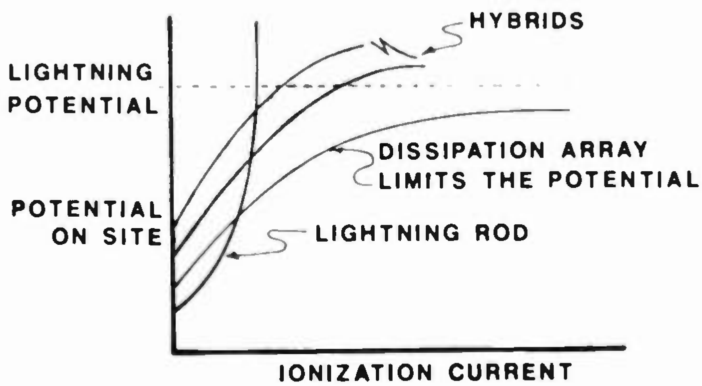
These other devices on the market do reduce the number of strikes to the towers they are to "protect". However, they often fail as a lightning rod, thus they may be rightfully termed a "Hybrid".

Hybrids deal to varying degrees with the weak strokes and fail when approached by the stronger stroke.

Figure 10 illustrates the reason for this situation.

A lightning strike comes about as the result of large potential differences between the incoming stroke channel, (the leader) and the tower. These high potential differences are the result of accumulated charge on the site, concentrating on the tower. The ionizer must constantly dissipate enough charge to keep the transmitter facility below strike potential, if it is to prevent a strike. Only the DAS has been proven to accomplish this consistently.

Figure 10, Performance Comparison



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ORGANIZING LOCAL PCB CLEANUP

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In 1986 broadcasters were alerted that many older transmitters (pre-1979) contained PCBs. The manager or engineer at such a facility was suddenly faced with a problem completely alien to his field of expertise. The purpose of this paper is to serve as a guide for the safe, economical and legal disposal of PCBs.

Some have questioned the necessity for the strict rules which the EPA has set down for the handling and disposal of PCBs in light of the fact PCBs have been used for years and many of us have been exposed to them with no apparent ill effects. PCB fluid had superior insulating and heat transmission characteristics, the ideal insulating oil. Other properties of PCBs made it the ideal material for other uses. The toxic rating of PCBs was less than that for aspirin, so it was thought that PCBs were safe chemicals.

It wasn't until several fires involving PCBs in the mid-1980s that investigators uncovered problems. They found that PCBs, when exposed to high temperatures, produced by-products that were among the most toxic known to man. Our ideal fluid had now become somewhat of a monster. The EPA subsequently published rules prescribing handling and disposal of PCBs which provide for very large fines for violators.

Improper disposal of PCB material can subject one not only to a large fine but also to cleanup costs that can be staggering. With the consequences of improper disposal in mind, let's investigate the proper procedure.

It goes without saying that the first step in any cleanup operation is to determine if you have PCBs present in your transmitter. As simple as this appears, it may prove to be one of the more difficult parts of the task. If your transmitter was built prior to 1979, the possibility it contains PCB bearing components is relatively high. Calling the manufacturer isn't always conclusive; different suppliers were probably used during the model life of your equipment and some of the suppliers may no longer be in business.

The most thorough procedure would be to make a list of all oil filled transformers and capacitors in your transmitter. For each type of component find the model number, serial number, date code, and any other information which may aid in identifying the part. The manufacturer of the component can tell you if it contains PCBs if you give them this information. It is sometimes possible to get a sample of the oil analyzed by a chemical laboratory. Don't forget to include any spare components in your search. After you have ascertained that you do in fact have PCBs at your transmitter site, the parts should be listed by size and type.

The next step deals with disposal. Disposal charges can be broken down into two parts: transportation and elimination. Transportation charges can be shared with other stations in your area. The charges for elimination will naturally depend on the amount of material to be eliminated and what has to be done to eliminate it. The disposal charges for large transformers include flushing, burning the oil and the flushing fluid, then burying the case in a hazardous waste landfill.

If you haven't already done so, now would be a good time to contact other engineers in your area to see if they have PCB material. Assuming there are others in your area with PCBs, a committee should be set up to handle the disposal project. This can be done through your local SBE chapter. The purpose of the committee is to compile a list to include the types of PCB waste, its location and the owner's name. Armed with this information it is time to call in a REPUTABLE hazardous waste disposal firm.

To the best of my knowledge there are only four companies with national capabilities for disposing of PCBs; these are listed in Figure 1. There have been reports of small firms contracting to dispose of PCB waste and simply storing the material and absconding with the money. You need to be aware that the generator of the waste is ultimately responsible for proper disposal. YOU NEVER LOSE TITLE TO THE WASTE.

When selecting a company to dispose of the waste material, it is imperative that they be licensed by local, state and national authorities. They should also agree to arrange for any permits you may need to move the material. Be sure the contract you make with them contains a "hold harmless" clause. This clause will protect you to the limits of the contractor's assets should problems arise from the disposal.

General Electric
(800) 626-2001 Ext. 25

Westinghouse Electric
(412) 937-7140

Ensco
(501) 223-4160

Chemical Waste Management, Inc.
(312) 218-1500

FIGURE 1. MAJOR DISPOSAL COMPANIES

Some disposal companies make regular pickups (called milk runs) at locations scattered across the country. The most economical transportation charges result from getting included on one of these milk runs. The truck making the pickup will have to visit each transmitter site. Don't try to move all the waste to a central location because it is illegal to transport PCBs without a license.

Savings in disposal can also be realized by providing the labor necessary to load the truck. Heavier items can be loaded with a forklift provided by a local rent-all and driven by one of your group.

Prior to the actual pickup, the disposal firm you select will send forms which you are required to fill out with all the pertinent information on the waste. When the truck makes the pickup, the driver will give you a receipt for the material. After the material has reached the disposal location, a form verifying receipt will be sent to you. You will also receive notification of the final disposition of the material. These items should be made a part of the station's records.

Stations in smaller markets can also benefit by getting themselves included on a milk run even though they are the only one at their location. Charges escalate rapidly for special pickups.

Electrical power companies are faced with the problem of PCB disposal, too. Most of them are very helpful and will assist you in solving your disposal problem. Don't hesitate to ask them for help.

It would be impossible to cover every eventuality that may arise in disposing of PCB waste. Prior to undertaking PCB disposal, understand the EPA requirements and NEVER discard a small PCB capacitor in the regular trash.

ACKNOWLEDGMENTS

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ENGINEERING AND OPERATIONAL CONSIDERATIONS FOR MOBILE SATELLITE COMMUNICATIONS

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Abstract: Mobile Ku-Band satellite communications are used for a variety of applications including news and production. During the development of this relatively new application of the technology, users have gained a great deal of knowledge about the practical and engineering aspects of operating mobile satellite systems. This paper discusses various aspects of the satellite technology as applied to mobile use. Topics that are discussed include:

- How the application affects the choice of equipment
- How the system parameters affect the choice of equipment
- How weather affects operations
- What are some of the costs involved including licensing, operations, and maintenance

APPLICATION OF THE SYSTEM

The common element in all satellite systems is, of course, the satellite. How this element is configured and operated is the single most important contributor to system operation. Consider the basic link equation:

C/N (RCVR) = C/N(UP) * C/I * C/N(DN)

Where C/N (RCVR) = Received carrier to noise in dB
C/N (UP) = Uplink carrier to thermal noise in dB
C/I = Carrier to interference noise in dB
C/N (DN) = Downlink carrier to thermal noise in dB

Note that the * symbol denotes power addition.

Satellite operators (GTE, RCA, SBS, etc.) have different operating modes for the transponders that are leased for occasional use. The new brands of satellites all have very sensitive inputs, such that is it possible to saturate the satellite with flux densities ranging from -94 dBw/m^2 to -88 dBw/m^2 depending on your location. Unfortunately, these sensitivities also make the satellite receptive to interference as well as to desired signals.

To combat this, the operators place attenuators in the front end of the satellites so that an uplink signal must also overcome this level, thus raising the carrier power relative to the thermal noise. The attenuators range from 0 to 12dB usually in 3dB steps. These attenuators are adjustable from the ground control facility but typically are set at 6dB. The control facility will normally lower the attenuator for short news stories based on the assumption that the quality of news feeds is not as critical as program feeds, (i.e. more interference is acceptable). The more important the quality of the feed, the higher the attenuator setting. Also read "quality" to include duration of the feed.

The higher the attenuator setting, the larger the uplink carrier power must be to saturate the transponder. However, this increased power requirement can be very advantageous to the user because it increases the uplink C/N portion of the equation. For production work, RS-250-B performance is mandatory. Achieving the required 56dB signal-to-noise ratio depends both on the transmitting power and the receiving antenna size, and since the latter is variable (different customers use different sizes), the safe play is to insure that the uplink is not the limiting factor in performance.

For news work, the receiving antenna is generally known, unless the stations are networking. RS-250-B performance is not generally required except at the network level. Hence TV stations are more likely to compromise on performance by using smaller antennas and lowering the satellite attenuator setting.

SYSTEM CONSIDERATIONS

From the link equation it should be clear that the best C/N (RCVR) is achieved when the satellite transponder is saturated. Logically, the antenna gain and transmitter power are chosen to be consistent with the operating mode of the system. Since the uplink saturation flux density (S.F.D.)is a product of antenna gain and transmit power at the feed, there are several options available as shown below:

Table 1
Uplink Performance Capabilities

Antenna	HPA Configuration	S.F.D.
A. 3.7M	300 watt hub mounted	$-85 dBw/m^2$
B. 3.7M	500 watt hub mounted	$-83 dBw/m^2$
C. 3.7M	500 watt rack mounted	$-86 dBw/m^2$
D. 2.4M	300 watt hub mounted	$-88 dBw/m^2$
E. 2.4M	500 watt hub mounted	$-86 dBw/m^2$
F. 2.4M	500 watt rack mounted	$-89 dBw/m^2$

There are, of course, advantages and disadvantages to all of the configurations above. Let's take a moment to examine several concerns that have been expressed by potential users:

Large Antennas

Because of the vehicle width restrictions, antennas larger than 2.4 meters must be folded. The disadvantage to this is that it requires an additional set-up on the part of the operator (it requires about 45 seconds). In actual practice, the only disadvantage of the larger antennas is that they require a heavy truck. Heavier trucks are not necessarily a disadvantage however. They are stronger, more powerful, and can carry larger payloads (shelters, equipment, etc.).

Small Antennas

The main advantage of a small antenna is that it can be mounted on a smaller truck. This may be important for news operations where speed and maneuverability are important. The major disadvantage to small antennas is their inability to receive the coordination channel (or IFB) and to monitor the downlink signal during adverse weather conditions. There have been numerous reports of these situations, although they likely comprise only a small percentage of the daily feeds. The other disadvantage is their poor quality in maintaining the downlink. The small antennas are practically unusable when monitoring the downlinks of dual transponder feeds in many parts of the USA.

High Power Amplifiers

HPA's come in several varieties as shown above. If the unit is mounted near the antenna feed flange, it is referred to as a "hub mounted" unit. These configurations have the advantage of delivering maximum available power to the feed since there is little waveguide loss with which tc contend. This translates into lower cost. However, hub mounted HPA's are exposed more to the elements. Care has to be taken to insure operation over a wider range of environmental conditions than rack mounted HPA's. Another disadvantage of hub mounted HPA's is that when the TWT or supporting circuits fail, they are more difficult to repair because they are in an outdoor environment. If the unit is mounted in the rack, then the user must either buy a larger unit (eg. 500 watt) or must resort to phase combining two identical 300 watt HPA's to overcome the waveguide loss (typically 2.0 to 2.5dB). Either of these two alternatives translates into higher cost and lower reliability.

Let's return now to the link equation and examine what uplink S.F.D. is required for various operating conditions. The values of uplink saturation power and downlink E.I.R.P. published by the satellite owners are end of life values, and these are the ones that you must design your system to accommodate. When satellites are new (as they are now), the actual values are 1.5 to 2 dB better than shown. The potential user should not be fooled by claims that smaller systems have demonstrated the capability of saturation from the edge of the satellite footprint. While these claims are probably true today, they may not be true in a few years when the satellite has aged.

Satellite flux densities of $-94 dBw/m^2$ to $-88 dBw/m^2$ are values that must be received by the satellite to achieve the maximum EIRP on the downlink with the attenuator settings set to zero. Which value to use depends on which satellite and from where the user is transmitting. Most engineers prefer the conservative approach of using the $-88 dbw/m^2$ value since the transportable may be required to operate from the edge of the satellite footprint (why pay that much money for something that will be limited in the area from which it can operate). Nor does the user always know which satellite he is going to use (the preferred one might be busy at the desired time).

EFFECTS OF THE WEATHER

The satellite flux density must be received regardless of the atmospheric conditions on earth. As you know, Ku-Band suffers from scattering and absorption of the rf signals by moisture (rain, ice, snow). Dalsat was the company selected by NBC to conduct the initial rainfall attenuation studies that have since become the industry standard.

According to National Weather Bureau data, rain does not occur above approximately 14,000 feet. Therefore, the vulnerable path length is that portion of the rain path which is below 14,000 feet. As shown in Figure 1, vulnerable path length varies as a function elevation angle to the satellite. The path lengths represented here would occur at Portland, Main and Miami, Florida.

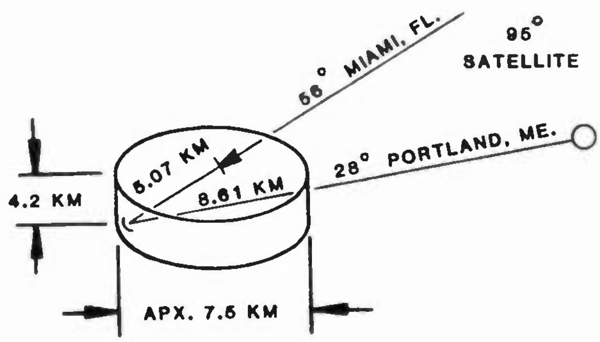

Figure 1

Historically, engineers have used a standard of 1dB/Km/inch/hour to predict attenuation of Ku-Band signals due to rain. Data taken during the survey and supporting observations taken since that time suggest that attenuation is more closely described at 0.6 dB/Km/inch/hour. Table 2 below shows a comparison of various assumptions:

Table 2
Rainfall Attenuation (dB/inch/hour)

Elevation Angle (deg)	Coefficient (db/Km/in/hr) 1.0	0.8	0.6	0.4
10	24.6	19.7	14.8	9.8
20	12.5	10.0	7.5	5.0
30	8.5	6.8	5.1	3.4
40	6.6	5.3	4.0	2.6
50	5.6	4.5	3.4	2.2
60	4.9	3.9	2.9	2.0

Based on our observations, the following table is representative of the attenuation levels that an operator can expect under various conditions:

Rain		Snow	
Rate (in/hr)	Fade (dB)	Rate (in/hr)	Fade (dB)
0.5	1	2	1
1.0	4	4	4
2.0	6	6	6

For news feeds, the higher precipitation rates are not a serious concern because typically they last for short periods of time. One need only wait a few minutes, or simply accept a degraded picture. For program feeds, however, these conditions must be considered carefully.

A frequently asked question is, "how often can I expect such occurrences?" Unfortunately, this is a statistical problem and so there is no clear, concise answer. For program feeds, one must simply assume that at some point in the operational life, all of these conditions will be experienced.

With all of these considerations, let's look at an example. Let's say the truck is located in the "hot" area of the satellite footprint. The following table shows the S.F.D. margin available when trying to saturate various satellites whose input attenuators are set at 6dB, and when experiencing 1.0 in/hr rain <u>or</u> 4 in/hr snow.

System	Uplink Margin Satcom	G-Star	Spacenet
3.7 M, 300W hub	0dB	+1dB	-5dB
3.7 M, 500W hub	+2dB	+3dB	-3dB
3.7 M, 500W rack	-1dB	0dB	-6dB
2.4 M, 300W hub	-3dB	-2dB	-8dB
2.4 M, 500W hub	-1dB	0dB	-6dB
2.4 M, 500W rack	-4dB	-3dB	-9dB

For contrast, the reader can easily assume variations in the input conditions and can formulate how any one configuration would respond. For instance, a news operator could assume that the input attenuator could be adjusted to 0dB. With a 2.4 meter, 300W hub mount, he could easily saturate Satcom or G-Star, even in moderately severe weather conditions experiencing 4dB fades. And so on.

In the above calculations, we have assumed truck operation in the "hot" spots of the country which covers approximately 2/3 of the contiguous U.S. Most satellite antenna footprints roll off drastically near the coast-lines. Clearly, an operator must consider working from a location which is degraded by as much as 6dB. Remember that the whole purpose of the satellite system is to enable coverage of events in distant locations. News rarely seems to break in a convenient spot close to home, or does the local basketball team always play at home.

Let's now consider some operational aspects of a mobile satellite system.

FINDING THE SATELLITE

One of the most commonly expressed concerns of engineers considering the use of mobile satellite communications is how reliably can a mobile operator find the correct satellite? Our experience has shown that if an operator is equipped with a spectrum analyzer and a true indication of elevation angle from the antenna, a trained user can always find the satellite within two minutes of pointing the antenna in the general direction of the satellite.

Since each satellite has its own spectral "signature" it is quite easy to find the correct one with a little training. The procedure simply calls for raising the antenna to the proper elevation angle and then panning in azimuth until the satellite is found.

There are, or course, fairly low cost auto-positioning systems that use a Loran receiver to determine the location of the mobile system in E-W coordinates and a small computer to point the antenna accordingly. Other systems are based on a flux gate compass which is much less expensive, but is accurate only to 3 degrees. The operator must then complete the acquisition process.

DEPLOYMENT

The time required to deploy the system is important only in news coverage applications. The single longest time required is to warm up the high power amplifier (7 minutes). When a vehicle arrives on the site, the first step is to acquire power (usually an on-board generator). Once this is stabilized, the HPA's are turned on to begin their warm-up cycle.

From the time a user arrives on site to the time he can acquire the communications channel on the satellite is typically 9 - 11 minutes, even with the larger 3.7 meter TRI-FOLD antennas. All mobile systems should be designed for deployment within this time frame by a single operator, allowing the second person to set up cameras etc..

OPERATING COSTS

Dalsat recently conducted a survey of over two dozen mobile satellite operators, covering the operating costs of both large and small antenna systems. The following paragraphs summarize the findings.

FCC Licensing

The FCC requires the issuance of a permanent license. The initial filing is usually handled by a Washington law firm and will result in legal and filing fees of $400 to $1200. Licenses are processed in 6 to 8 weeks, depending on the backlog at the FCC.

Department of Transportation Licensing

For vehicles whose gross vehicle weight rating is 10,000 lbs and under, a standard operator's license is all that is required. Interstate travel is not restricted by weigh stations nor is the vehicle required to carry more than ownership and proof of insurance certificates. For all vehicles in excess of 10,000 lbs, the following 1988 D.O.T. rules apply:

- o Drivers must pass a written test that may be administered by the employer.
- o Drivers are required to pass annual physical examinations and to maintain records so stating.
- o Driver must file an application with his employer to be a driver of a large vehicle (in addition to his other duties).
- o Driver must hold a valid Class 4 Driver's License.
- o A log book must be maintained in the vehicle and made available upon request.
- o All drivers must pass a certified training course on the operation of large vehicles.

For further details on these items the reader should contact the local Department of Transportation.

Mobile satellite television systems are used either for contract work such as sporting events, or for news reporting. Many television stations use them for both, and actively market the use of their system. Payoffs for such applications are typically 12 - 15 months.

With such a wide range of uses, the annual mileage put on a mobile system ranges from 10,000 to as high as 65,000 over the systems that were surveyed. Average uses per month and other data are shown below:

Personnel Cost/year	$48,000
Maintenance/year	$ 7,500
Maintenance/mile	35¢
Licensing Costs/year	$ 800
Average Uses/month	19 days
Average Mileage	8.2 mpg
Average Fuel Cost	$1.12/gal

Personnel costs were construed to include the salary and overhead costs of one or more operators and/or maintenance personnel as their time applied to the operation of the system. It may or may not reflect additional hiring of staff. Maintenance costs were for all electronic and mechanical maintenance and did not include equipment upgrades or additions.

SUMMARY

The purpose of this paper has been to acquaint a person considering the purchase of a mobile satellite system with some of the more important engineering and operational aspects of ownership. Clearly all aspects of such a decision cannot be covered in such a limited forum as this paper, but the topics shown here are important first considerations.

Assuming that the market and competitive factors have led the reader to consider the use of a mobile Ku-Band satellite system, then the important first engineering considerations should be:

- o How is the system to be used?
  - News
  - Production
  - Both
- o How important is the quality of the audio and video?
- o How important is the timing of the feed?
- o Over what geographic area of the country will the system operate?

These questions will lead to the basic decisions concerning the size of the vehicle, antenna, and the type of RF electronics. From these, the engineer can then specify all of the remaining details and features of the system.

PRACTICAL DESIGN CONSIDERATIONS FOR DESIGNING KU-BAND VIDEO DOWNLINKS FOR BROADCAST TV STATIONS

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INTRODUCTION

Ku-band satellite services are becoming increasingly popular among broadcasters. Most of this popularity is due to the lack of terrestrial interference that makes site location for C-band video services difficult. As a result a Ku-band downlink can be located anywhere, frequently resulting in significant cost savings in installation. To program distributors it means that any location that needs to receive a program can do so without fear of having particular frequencies or parts of the sky blocked by interference.

You have probably heard the expression "It's Ku-band, you can use a smaller antenna"? If you've heard something along those lines, have you stopped to think if that's really true? If you already have a Ku-band system or are contemplating the installation of a broadcast grade TVRO it will pay to know the basic design information and some of the common "gotchas" that make Ku-band systems different from their C-band counterparts. The purpose of this paper is to help the reader design a TVRO, and avoid the common mistakes associated with the installation of Ku-band facilities; mistakes that someone else has already paid the price for.

THE BASICS

Carrier to Noise Ratio

The downlink carrier to noise ratio (C/N) is the single most important design goal to meet when designing a high quality downlink. The carrier to noise ratio is the ratio in db of the RF carrier power to the noise energy contained in a given RF bandwidth. It is given by:

$$C/N = EIRP - Pathloss + G/T + 228.6 - 10Log\ BW$$

Where EIRP is the satellite's downlink power in the direction of the earth station in dbW (This value is normally obtained from contour maps), Pathloss is the free space pathloss from the satellite to the Earth station (Approximately 205db in the Ku-band), 228.6 is Boltzman's constant, and BW is the receiver's 3db RF bandwidth in Hertz. Carrier to noise must always be specified with a bandwidth, since a bandwidth component is contained in the C/N derivation.

The measurement of a video signal's C/N may be approximated by using a spectrum analyzer looking at the same feed the video receiver is seeing. The video C/N using this method is given by:

$$C/N_{video} =$$
$$\text{Analyzer C/N} + 10Log\ (\text{Analyzer BW / Video receiver IF BW})$$

Where the analyzer C/N is measured from the noise floor to the tip of the carrier in db, and the two bandwidths are measured in the same units. For example an analyzer shows a video signal that is 36db from the noise floor to the tip of the carrier, using a resolution bandwidth of 300kHz, and the video receivers IF bandwidth is 36MHz or 36,000kHz. This yields a video C/N of:

$$36db + 10Log\ (\ 300/36{,}000\ )\ \text{or}$$

$$36db - 10Log\ (.0083)\ \text{or}$$

$$36db - 20.8db = 15.2db\ \text{Video C/N}$$
in a 36MHz bandwidth receiver.

Notice in these formulas that the C/N can be increased by reducing the receiver bandwidth.

Video signal to noise

The end signal to noise (S/N) ratio of FM video systems is determined by :

$$C/N + FMI$$

Where C/N is the carrier to noise ratio and FMI is the FM Improvement ratio. This formula applies to any signal that is above threshold. For purposes of this discussion threshold is the carrier to noise level where the FM detector is locked to the incoming carrier all of the time. Video signals below threshold will show sparklies and video will degrade rapidly as the signal drops. Most receivers will begin to show threshold effects in average video at a C/N of approximately 10db.

Threshold

The carrier to noise threshold number above is not to be confused with the threshold rating that is presented in the receiver manufacturer's data sheet. The data sheet threshold is defined as the point where the signal to noise vs carrier to noise relationship departs from a db for db relationship by 1db. This occurs after some threshold degradation has already taken place, and is a number that represents unusable pictures. This definition is used because it can be be accurately measured, whereas a sparklie threshold is slightly more subjective. A receiver with a data sheet threshold of 8db will show sparklies in average video at a 10db C/N, and sparklies in NTSC color bars at a 13-14db C/N.

In a full transponder system, reduction of the C/N from 10db to 5db will cause a 50+db S/N picture to degrade not by 5db, but to a screen full of noise that is almost unrecognizable as video. So in the design of any satellite system is is crucial to operate above threshold. Since almost any picture above threshold is useable in a pinch, the margin above threshold can generally be viewed as fade margin.

System C/N

Downlink C/N must be differentiated from system C/N. It is system C/N from which the receiver must reconstruct a picture. System C/N is the result of uplink C/N, Downlink C/N, and all the various Carrier to Interference (C/I) elements that may be significant in a given system. Interference elements include: the antenna's rejection of adjacent satellites, the antenna's crosspolarization isolation, the receivers IF filter rejection of the adjacent channels, and intermodulation interference. System C/N is given by:

$$C/N_{X1} - 10\text{Log}\,(1/X_1 + 1/X_2 + 1/X_3 + 1/X_4) / 1/X_1))$$

where X_1 is the lowest C/N or C/I component, and X_1,X_2, and so forth are the power ratios represented by the various C/Ns. For example a C/N of 12db is a power ratio of 16 to 1. For only two C/N elements the system C/N is given by:

$$C/N = 10\text{Log}\,(X_1 \times X_2) / (X_1 + X_2)$$

Generally, to assure adequate end to end performance for a wide variety of services, the downlink C/N should be at least 4db higher than the desired end to end C/N.

FM Improvement ratio

The FM improvement ratio (FMI) is the improvement over the carrier to noise ratio brought about by the use of FM. FMI is a function of deviation, baseband bandwidth, and pre-emphasis characteristics. For example the narrow bandwidth half transponder NTSC video that Conus uses in the SNG system has an FMI of about 35db, whereas full transponder video with its much wider deviation will have an FMI of over 43db. This means that a threshold picture in half transponder systems will present a video signal to noise ratio of about 45db, and full transponder threshold pictures can be as high as 52-53db S/N.

Figure of merit G/T

The Earth station figure of merit, or G/T represents the system's overall noise performance, and takes into account both noise and antenna gain. The G/T is given by :

$$G_{ant} - 10\log(N)$$

Where G is antenna gain in db and N is the system noise temperature in degrees Kelvin (LNA temperature + antenna temperature). A db change in G/T will result in a db change in the downlink C/N and a resultant db change in the video signal to noise ratio (S/N).

On the ground Ku systems Typically suffer from poorer noise performance than C-band systems. The best economy C-band LNAs (Low Noise Amplifiers) today are 60 degrees Kelvin or so, while Ku-band units come in at 160 degrees or so for similar cost. Ku-band noise performance is then typically 10Log 60/160, or 4.26db worse than C-band.

Antennas

Antenna gain in db for any circular parabolic antenna is given by:

$$G = 10\text{Log}\,(109.7\,F^2\,D^2\,n)$$

where F is frequency in GHz, D is diameter in meters, and n is efficiency. Some useful relationships can easily be remembered. Doubling the area of an antenna will increase it's gain by 3db, and doubling it's diameter will increase the gain by 6db.

A common problem with new, large Ku-band installations, is installing the antenna on a sidelobe. This sounds like one of those situations that can't happen to you, but most large full transponder Ku systems have enough margin to produce pictures under clear sky conditions with a 50+db S/N on a sidelobe. You will be absolutely confident that you've gone through the main lobe every which way, and the fine pictures will convince you that all is right, until it rains at which point you claim Ku-band is worthless since it goes away so easily.

Aiming any large satellite antenna must be done carefully the first time in order to avoid this problem. A sample link calculation should be done to estimate the receive level to be expected, and if you don't come within a few db of the estimate, suspect misaiming. It's not easy for the Ku-band novice to convince the experienced factory installer that the antenna's not aimed correctly, but this happens all the time, and it will happen to you.

Receive antenna gain is the cheapest system performance you can buy. A proper receive antenna will let you shop around for transponder time instead of buying the highest power and most expensive satellite time. You will pay the price for a good antenna only once, but you will pay for high power transponder time every minute you use it. A proper receive antenna will overcome some of the shortcomings of poor uplink performance. If you are involved, or are going to be involved in satellite news gathering a larger receive antenna will save otherwise lost shots when technical problems arise, or you must operate from edge of coverage locations. Whatever you do don't spend $350-450k on a satellite truck and through all the performance away because you think your saving $5-10k by buying a smaller downlink. $5-10k is not very many lost shots, and further, you'll wind up putting in the right size downlink eventually anyway, having wasted the entire expenditure on the small downlink.

THE SMALL ANTENNA MYTH

It is the energy density, expressed as power per unit of bandwidth (W/Hz or more commonly dbW/Hz) reaching the detector that determines the end carrier to noise ratio of any transmission system. For example low power SCPC (Single Channel Per Carrier) satellite systems use energy densities similar to high power systems like video. These SCPC systems employ a much smaller bandwidth than video, thus achieving similar energy densities to video with much smaller absolute power levels. The energy density for a given C/N performance level is a constant for FM systems regardless of the type of transmission.

Physics dictates that Watt for Watt the theoretical performance of Ku-band satellite systems will be dead even with C-band systems for antennas of any given size. The increased antenna gain at the higher Ku-band frequency will be exactly offset by higher Ku-band pathloss. Below is a comparison of a 5 Meter antenna, and pathloss in both bands.

	Gain of 5 Meter Antenna	Pathloss from satellite
C-band	44.1db	195.4db
Ku-band	53.7db	205.0db

When practical implementation considerations such as satellite size and design, ground segment noise temperatures, and rain fade margins are considered, Ku-band will require antennas that are larger than C-band for systems of equal RF power. It is these practical considerations that lead satellite designers to build Ku-band satellites that fly a lot more power than similar C-band satellites. Present Ku-band satellites typically employ transponder powers in the 20 Watt range with a strong trend toward higher power levels, while C-band satellites employ transponders in the 8 Watt range. This is done even though higher power will mean fewer Ku-band transponders per satellite.

Ku-band does present an advantage to the satellite designer for the spaceborne transmit antenna. Present day commercial communications satellites are constrained by the launch vehicle to practical antenna diameters of about eight feet. Since an 8ft Ku-band antenna will be much more directive than an 8ft C-band antenna, better beam shaping is possible with practical antennas on a Ku-band satellite. All present day satellites use more than one feedhorn to illuminate the reflector. By varying the power and phase relationships to the various feedhorns, the beam shape can be tailored to fit the satellite's service area. In the case of a well optimized domestic Ku-band antenna, the satellite will project a radio frequency image of the United States onto the United States much like a slide projector projecting the image of a map. Since antennas of equal beamwidth have equal gain, an ideal CONUS beam antenna (CONtiguous United States) will have the same gain regardless of what frequency band is employed. The practical antenna implementation in space will have 2-3db advantage in the Ku-band within the service area. This improvement in downlink power results from reducing power radiated into adjacent territories, and can be seen in the comparatively steep Ku-band contours as you move away from the U.S. border.

Ku-band also presents an advantage to the uplink designer, since a 2 degree compliant antenna can be smaller than at C-band. The fact that a smaller antenna is licensable does not, however, mean that performance will be acceptable.

If a smaller antenna is useable in the Ku-band it is because the Ku-band satellites transmit more power, and/or you are trading away rain fade margin for antenna size. Lets compare what we have so far using a 20 Watt Ku-band satellite with C-band as a reference:

Ku power advantage (TWT size + space ant effic.)	+6db
Ku ground segment noise penalty	-4db
Minimum additional Ku rain fade margin	-3db
Total Ku vs C performance	-1db

From above we can see, that depending on the rain margin allocated to Ku-band over C-band, Ku-band actually has a slight disadvantage over C-band for antennas of equal size. The rain margin assigned in link analysis is usually determined by the downlink location's climate, and the desired link availability. In the above case 3db was used as an example, but may be much higher depending on the link's availability requirement.

Satellite distribution systems are normally designed to provide a certain S/N ratio a certain percentage of the time, such as 52db with 99.99% availability. Since program distribution services want the same level of performance regardless of the method employed, higher fade margins must be assigned to Ku-band systems. The ABC and NBC network distribution systems both have similar program quality requirements, but use different bands. The ABC C-band system employs 6.3Meter antennas as it's baseline antenna. The NBC Ku-band system employs 6.1Meter antennas as it's baseline. Thus two distribution systems operating in different bands with the same objectives wind up with virtually identical antenna sizes.

You cannot use a smaller antenna in the Ku-band just because it's Ku-band.

Ku-band distribution systems like the NBC example above do enjoy one very distinct advantage. 99.99% of the time they will be better than C-band, since the large rain fade margin will, on clear days, be available to produce higher than normal video quality.

SATELLITES ARE NOT ALL THE SAME

In the C-band the satellite power levels and transponder bandwidths have been constrained severely do to required compatibility with terrestrial C-band sharing in both the uplink and downlink bands with co-equal users. This has resulted in a high degree of uniformity in both power and channel plans that lead to uniformity in RF bandwidth, deviation, and ultimately receive video levels.

Transponders

In the Ku-band there are no co-equal users, so a wide range of satellite characteristic have developed. Transponder bandwidths range from 43MHz to 72MHz. The number of transponders per satellite ranges from 6 to 16. Transponder power levels range from 16Watts to 45Watts with higher power satellites under construction. EIRP values in the Ku-band can vary by over 10db depending on the satellite, and half or full transponder operation. In designing a general purpose Ku-band downlink for broadcast use this range must be considered in order to insure adequate performance on all satellites likely to be accessed.

Half transponder EIRP levels will be at least 5db below the full transponder EIRP contour for any satellite using TWTs (Travelling Wave Tubes). The nonlinear nature of TWTs near saturation requires that any evaluation of half transponder performance must be made when both halves of a transponder are full. A video signal in one half of a transponder with out the other half full will get nearly all of the TWT's output power, thus creating a false impression of half transponder signal quality.

BANDWIDTH AND VIDEO LEVELS

Considerable confusion exists over video level and bandwidth. For example a receiver that produces the normal one volt of video on a full transponder (36MHz) Ku-band transmission will only produce half a volt of video on a half transponder signal, thus giving the appearance that the transmitting end is double terminated or receiving less than normal video input. Remember that FM stands for frequency modulation, and that when less bandwidth is used, and hence lower deviation, less amplitude will be produced at the output of the FM detector.

A useful mental visualization may help. Visualize a video waveform monitor display, now rotate the image 90 degrees clockwise and picture it on the face of a spectrum analyzer. By doing this you will start to see the relationship between RF bandwidth and video level. In half transponder systems the decrease in deviation and resultant low video level must be compensated for in the receiver's output amplifier by increasing the gain thus restoring the one volt output. In a similar manner, a system normally configured for half transponder video will produce two volts of output video when looking at full transponder transmissions.

Even so called "full transponder" signals in the Ku-band vary considerably in apparent video level. This depends on the individual system operator, and how much bandwidth they choose to use. At present no operator actually uses a full Ku-band transponder for one video. The transponders are so wide that at present is not practical to do so for NTSC video.

Video services attempting to reach very small antennas will employ narrow channels to increase C/N in order to stay above threshold in small receive terminals. These services will use only a small part of the transponder bandwidth, but all of the power. On the other hand, high quality services will instead use high deviation and larger antennas to make up C/N lost in wider bandwidths, but will benefit from an improved FMI due to the high deviation.

IF filter bandwidth

The IF filter bandwidth is one of the most frequently goofed up aspects of receive system design. Selection of a filter that is too wide will reduce the C/N and may increase C/I values by letting in adjacent interfering signals. Selection of a filter that is too narrow will slice off the edges of the transmitted signal resulting in edge tear, sparklies in areas of high color saturation, and video buzz in the audio channels.

For high quality single video per transponder transmissions Carson's rule can be safely followed. The filter bandwidth can be found by:

$$BW = 2(D + M_H)$$

where D is the deviation in MHz, and M_H is the frequency in MHz of the highest modulating frequency, normally 4.2MHz for NTSC. While audio subcarriers are higher in frequency their low amplitude leaves them out of all but very fine grain analysis. For half transponder services with close spaced signals above or below, the value calculated above should be reduced by 15%.

For example the Conus system uses 7.5MHz deviation with NTSC video, so:

$$2\ (7.5 + 4.2) = 23.4\text{MHz}$$

which must be reduced by 15% or:

$$23.4 \times .85 = 19.89\text{MHz}$$

The actual Conus specification for it's half transponder video is 20MHz at the 3db points.

When selecting a receiver be sure that the receiver manufacturer's filter specifications are accurate. Many filters we have encountered at Conus are substantially wider than the published information would suggest. Also check with the service provider to find out what they recommend for receive bandwidth, you'll save yourself a lot of trouble.

Low pass filter

Be sure that the receiver you choose has a video low pass filter suitable for removing the audio subcarriers that you will encounter. Many receivers do a poor job of rejecting 5.41MHz, and as a result give the appearance that there is more than normal amounts of high frequency noise present. Since sharp filters will cause some degradation of fine grain video characteristics, a receiver that must be used for more than one type of service could be filtered externally to allow for easy filter selection.

RECEIVE SYSTEM INTERMODULATION PRODUCTS

Most Ku-band receive systems and many new C-band systems employ low noise block down converters (LNBs) that mount to the feed horn allowing the use of relatively inexpensive cable between the antenna location and the receiver.

First be aware that all block conversion systems are not the same. Common LNB IF bands include: 270-770MHz, 900-1400MHz, 950-1450MHZ, 3.566-4.066GHz, and 3.7-4.2GHz. Be sure that the LNB frequency range matches that of the receiver, and that the frequency band is not inverted by using local oscillator injection on the wrong side of the band for the receiver in use. Frequency inversion, which is more common in C-band systems, will result in inverted tuning of the band as well as a polarity reversal in the video.

The use of LNB techniques result in the entire 500MHz satellite band being passed through many stages of amplification before the channel selection filtering is accomplished. Maintaining the required linearity in the block system limits the dynamic range of this approach. In order to prevent the generation of intermodulation products ahead of the channel filters, care must be taken to see that the receiver is presented with the input level that it expects. This problem is aggravated in the Ku-band by the 10 to 1 power differences that can be experienced between different services.

Many receivers are built with considerable excess gain in order to accommodate long cable runs. If such a receiver is presented with a short cable from the antenna and few other losses, the front end of the receiver will overload producing intermodulation products. When receiving heavily loaded satellites the number of products and the composite modulation of all the input signals will produce intermodulation products that are nearly identical to noise. This will increase the apparent noise floor, giving the false indication that the receiver is not getting enough signal. Intermodulation in FM systems does not always produce visible crosstalk. Overload and intermodulation are more common than lack of signal.

Symptoms

Indications of intermodulation or overload problems include: Video crosstalk; noisy pictures; normal operation when only one or two transponders are active, but degraded performance when the satellite is full; full scale signal strength indications in the absence of video; and other than a clean screen of noise when video is absent.

Setting input level

Some receivers employ an AGC to avoid intermodulation problems while others provide a built in adjustable IF attenuator. Receivers that do neither must have their input level set with external attenuators. If the receiver does not seem to perform as expected, try adding attenuation to it's IF input until the picture just starts to get worse due to lack of signal then take out 10db of attenuation. Be sure this is done on the weakest signal that the system will be asked to receive.

CABLE AND CONNECTORS

The use of inexpensive cables and connectors with block conversion systems creates the impression that the receive system is as simple to engineer as a VHF antenna system. The reality is that you are dealing with a microwave distribution system that is quite sensitive to loss and impedance mismatches. The "F" connector like the SMA connector derives it's characteristic impedance from the cable that it is attached to by maintaining the cable's center conductor and dielectric through the connector. "F" connectors when properly attached are useable up to several gigaHertz. The same cannot be said for BNC connectors and the host of cheap adapters that one is tempted to use in building an IF distribution system.

Impedance

Most LNB systems are 75 Ohm systems, and all of the components used in IF distribution must also be 75 Ohm. If BNC or N connectors are used they must be true 75 Ohm versions. The vast majority of BNC connectors used in broadcast stations are actually 50 Ohm connectors even though they are attached to 75 Ohm cables. These impedance errors are not important at video frequencies, but can be devastating at satellite IF frequencies.

The effects of poor IF distribution design include: ripple in IF frequency response, deep notches in the IF passband, and one or more video channels that work poorly while others work normally.

CONCLUSION

hopefully this presentation will help you design a broadcast grade Ku-band TVRO that does the job desired without breaking the bank. Remember that the most economical system is the one that is designed and built correctly the first time.

FIELD TESTING AN EARTH STATION FOR TWO DEGREE COMPLIANCE

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This paper will discuss the FCC rules and regulations regarding 2 degree compliance for earth station antennas, and provide a method in which to measure and record an antenna radiation pattern. A discussion of minimizing possible adjacent satellite interference during radiation pattern testing will also be covered.

INTRODUCTION

Since the introduction by the Federal Communications Commission of rules governing transmit sidelobe performance of earth station antennas, a special emphasis has been placed on the field testing of fixed and portable transmit antennas. Manufacturers test range data of representative earth station antennas (ESAs), while affording information on how well the particular product could perform, still does not take into account the installation and wear and tear of operational ESAs; especially in the case of transportable systems. With continuous deployment/stowing of the reflectors and feed systems, the repeatability of transmit radiation characteristics is in question. To ensure correct operation of the transportable ESAs, the FCC will introduce a test requirement to be performed by the owners of transportables. This plan will require owners to transmit test their transportable antenna(s) once per annum, and require the measurement of at least one transmit co-polar radiation pattern over +/-7 degrees of the geostationary arc (azimuth cut).

FCC RULE 25.209

Due to the large number of communication satellites in the geostationary orbit and tighter spacing between satellites, adjacent satellite interference from transmitting earth stations has become a major concern.

To protect adjacent satellites from undesirable uplink signals, the FCC created Rule 25.209. Rule 25.209 discusses the necessity of ESAs meeting stringent requirements for antenna sidelobe power levels. Specifically, the rule states that in the plane of the geostationary satellite orbit, the gain of a transmitting antenna must lie below the envelope as defined by:

$$\begin{array}{rl} 29 - 25\mathrm{LOG}(\emptyset)\ \mathrm{dBi} & 1 \leq \emptyset \leq 7 \\ +8\ \mathrm{dBi} & 7 \leq \emptyset \leq 9.2 \\ 32 - 25\mathrm{LOG}(\emptyset)\ \mathrm{dBi} & 9.2 \leq \emptyset \leq 48 \\ -10\ \mathrm{dBi} & 48 \leq \emptyset \leq 180 \end{array}$$

Where $\emptyset$ is the angle in degrees off boresight, and dBi refers to dB relative to an isotropic radiator. In field testing of earth stations, transmit radiation patterns are to be measured over $\pm$ 7 degrees, thus 29-25 LOG ($\emptyset$) becomes the governing requirement.

APPLICATION OF RULE 25.209

In plotting the required envelope on an ESA radiation pattern, the equation can be described as:

$$G - [29 - 25\mathrm{LOG}(\emptyset)],\ 1 \leq \emptyset \leq 7^\circ$$

Where G is the gain of the antenna. This form of the equation enables the peak of the radiation pattern to be established as a point of reference from which data points of the envelope can be calculated. An example is given in Figure 1. A radiation pattern of a 2.3 metre antenna with a gain of 49.1 dBi is shown. At $\emptyset$ = 1 degree, 29 - 25LOG($\emptyset$) = 29 dBi. Thus the point on the plot at degree is 49.1 - 29 or 20.1 dB down from the peak of the main beam. At $\emptyset$ = 2 degrees off boresight, 29 - 25LOG($\emptyset$) = 21.5 dBi. Thus at 2 degrees, the point on the plot is 27.6 dB down from the peak of the antenna. The remaining data points are calculated and plotted in the same manner.

The purpose of Rule 25.209 is to restrict the EIRP of the antenna in the direction of an adjacent satellite especially at 2 degrees from boresight where the nearest satellite may be located. If the EIRP of the earth station with the antenna shown is 75 dBW, then at the main beam 75 dBW is radiated. At 2 degrees from boresight, the EIRP of the earth station must be less than 75 - 27.6 = 47.4 dBW.

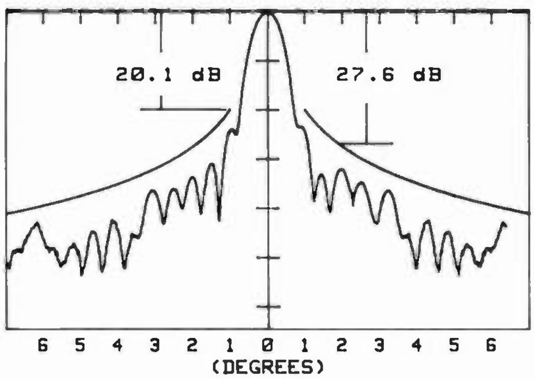

FIGURE 1. 2.3M ESA PATTERN

FIELD MEASUREMENTS OF EARTH STATION ANTENNAS

There are numerous ways in which to record an earth station radiation pattern. The method described here was developed by Andrew Corporation and Contel/ASC as an alternative to conventional methods for antenna testing. This method reduces the requirement for the high EIRP levels needed to provide the required C/N ratio of 50 dB for recording radiation patterns, by using a high stability synthesized source at the antenna-under-test (AUT), and a complimentary high stability spectrum analyzer at the cooperating earth station (CES). A C/N of 50 dB is required for the CES spectrum analyzer to track the CW signal through the entire AUT movement during test.

CONVENTIONAL METHOD

The conventional method of recording antenna patterns is to transmit a high level continuous wave (CW) carrier from the AUT using a relatively unstable source, such as a video exciter.

High EIRP levels are required when patterns are recorded using frequency sources that have insufficient stability. Instability in a test signal requires higher spectrum analyzer resolution bandwidths to maintain a stable reference. A minimum bandwidth obtainable may be no narrower than 30 kHz.

Conventional Method Disadvantages

Antenna testing using high EIRP levels has a number of disadvantages. First, a high EIRP requires the use of a high power microwave amplifier. If the AUT is a new facility, a large amplifier may not be available at the time of testing. Second, in today's two degree spacing environment, a potential exists for causing interference to traffic on adjacent satellites when performing azimuth transmit pattern measurements as the AUT is moved over $\pm$ 7 degrees. Third, a full satellite transponder may be required since a high EIRP may cause intermodulation distortion and possible service interruption to other carriers on the transponder.

ALTERNATIVE METHOD

The use of high stability source and high stability spectrum analyzer permit a decrease in resolution bandwidth at the cooperating earth station's spectrum analyzer. Bandwidths as low as 10 Hz have been used successfully. This considerable decrease in bandwidth from 30 kHz yields an improvement in C/N of approximately 30 dB, which permits a decrease in EIRP. With this substantial decrease in EIRP from the AUT, this pattern measurement technique can be employed on satellite transponders which are carrying SCPC type (Single Channel Per Carrier) traffic. However, due to the requirement of performing transmit patterns over a linear region of a transponder, extreme caution must be exercised since even a low EIRP CW signal could force the transponder into saturation, depending on transponder loading. Thus, coordination must always be made with the relevant satellite operations control center before accessing the transponder.

Figure 2 shows a typical equipment set-up for radiation pattern testing. At the (AUT), a synthesized CW source is generated by a Hewlett-Packard HP8672A frequency generator. This stable source ensures that no signal drift occurs at the cooperating earth station. This source has a frequency stability on the order of 1 x 10 -9/day. The output is amplified by the HPA and routed to the AUT. Typically, a 20 watt TWT is used.

The CES is required to receive and record the transmit data from the AUT. The CES receives the synthesized CW signal using an LNA, whose output is routed to a Hewlett-Packard HP8566 spectrum analyzer. The HP8566 is used since it is very stable and permits narrow resolution bandwidths.

As stated, this is an approximation but experience has shown that even for elevation angles greater than 40 degrees, accuracy has been within tenths of a degree. Patterns recorded this way are always checked against range pattern data.

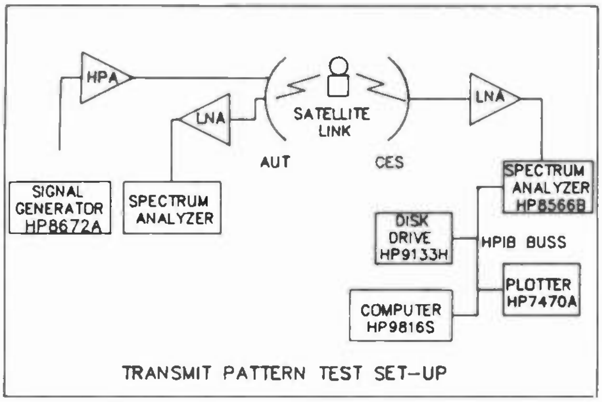

FIGURE 2. TYPICAL TEST SET-UP

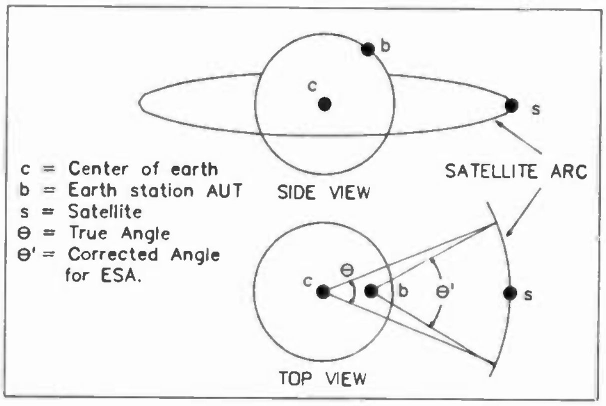

FIGURE 3. CORRECTION FACTOR

PRETEST CONSIDERATIONS

Though low EIRP transmissions are used in this method of pattern testing, consideration for possible interference to an adjacent satellite when moving the AUT in the azimuth plane must be considered. A method for determining the maximum EIRP allowable without causing interference to adjacent satellites is discussed later. This method is titled Maximum Test EIRP Calculations.

For azimuth patterns, a correction factor must be considered for sweep angles when testing non-polar mount antennas. When satellites are spaced by two degrees, this angle is relative to the center of the earth. With an earth station located at the earth's surface, this angle will appear greater as the latitude of the earth station decreases. The actual angle is approximately equal to:

$$2 \text{ degrees}/\cos(\emptyset)$$

Where Ø is the elevation angle of the satellite from the AUT. For example, if an AUT has an elevation angle of 35 degrees when pointing at the satellite, an azimuth pattern of +/- 7/COS(35) or +/- 8.5 degrees will be required for a pattern of +/- 7 degrees. See Figure 3.

TEST PROCEDURE

Coordination of the CW uplink from the AUT is made with the satellite operations control center prior to testing. The control center will check antenna peak and cross polarization discrimination. Once the uplink has been established, the testing can begin. For each step in the procedure, a figure detailing the spectrum analyzer display is indicated. The step-by-step procedure follows:

(1) The CES obtains the CW signal on the spectrum analyzer and reduces the resolution bandwidth to a minimum that will permit the analyzer to maintain the receive signal within that bandwidth. See Figure 4a.

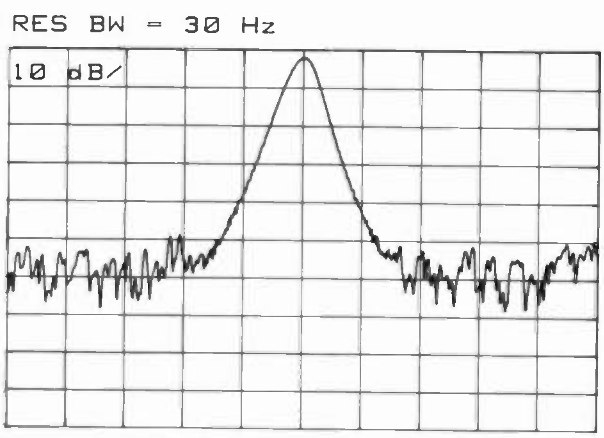

FIGURE 4a. STABLE CW SIGNAL

(2) The spectrum analyzer is then set to zero Hz span at the peak of the CW signal. Zero spanning the analyzer will change the analyzer from the frequency domain to the time domain. This domain permits the signal amplitude to be traced on the spectrum analyzer as the AUT is moved. See Figure 4b.

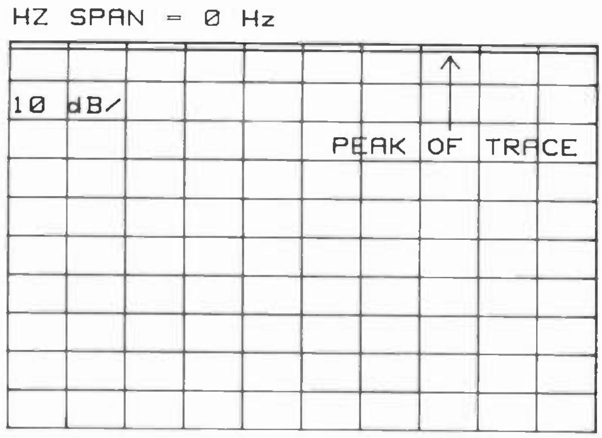

FIGURE 4b. 0 Hz SPAN

(3) The dynamic range should be checked to ensure that at least 50 dB of C/N is available by recording the peak level of the signal at the CES, instructing the AUT to terminate transmission, and recording the system noise floor level. The difference between peak signal level and system noise floor level is C/N. See Figure 4c.

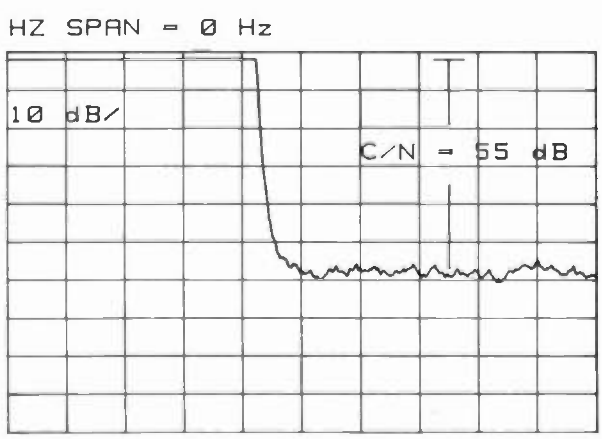

FIGURE 4c. DYNAMIC RANGE

(4) The linearity should be checked to ensure that the signal CW uplink signal has not saturated the satellite transponder. The AUT reduces the CW in known increments by reducing the output level of the HP 8672A generator. The reduction in signal level should be equivalent when measured at the CES. Once verified, return AUT signal level to original level. If reduction measured at CES is not equivalent, the AUT signal must be reduced until an equivalent range is reached.

(5) While maintaining phone contact with the CES, the AUT synchronizes a start time with the CES and begins moving the AUT to the east the equivalent of 7 degrees of the satellite arc (after correction). In most cases, a resolver assembly with a digital angular readout is used to provide degrees of movement. Upon reaching 7 degrees, the AUT synchronizes a stop time with the CES. See Figure 4d.

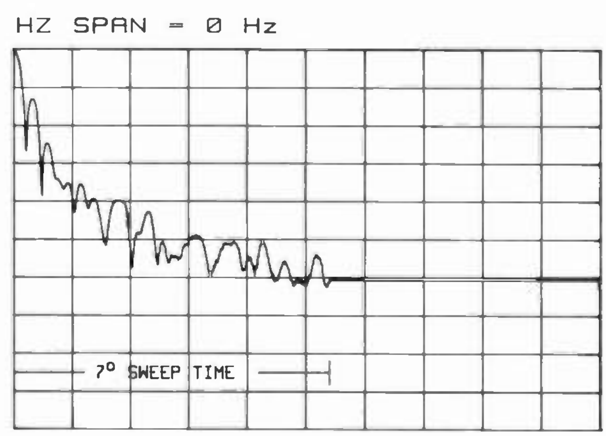

FIGURE 4d. MOVE TO EAST

(6) The time required to move the equivalent of 7 degrees of the satellite arc is doubled and entered into the analyzer sweep time at the CES.

7) The AUT is then moved to the west 14 degrees (after correction), with the start of the move synchronized between the two sites. The CES 'single sweeps' the trace on the analyzer to store it on the analyzer screen. See figure 4e.

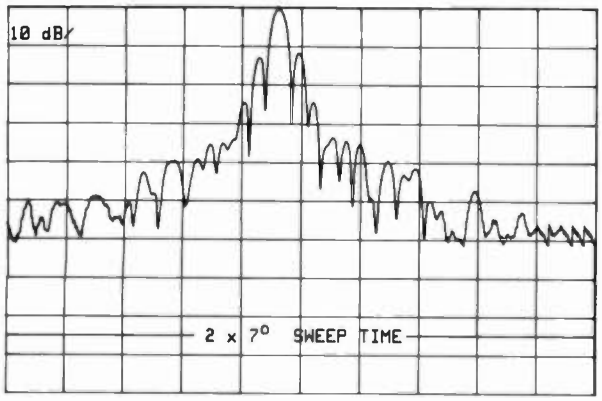

FIGURE 4e. FULL SWEEP

(8) The pattern trace stored on the analyzer screen is loaded to a Hewlett-Packard series 200 computer, 29 - 25LOG(Ø) curves are added, and the data is plotted on a Hewlett-Packard 7470A plotter. See Figure 4f.

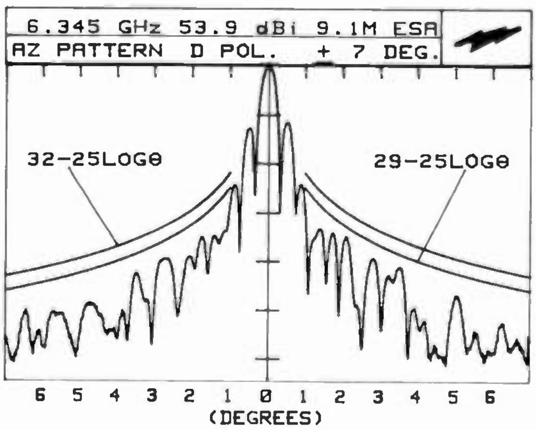

FIGURE 4f. PATTERN PLOT

(9) Elevation patterns are usually recorded with azimuth patterns. For elevation patterns, no correction factor is necessary for 7 degree sweeps when using an elevation over azimuth antenna mount.

SEMI-AUTOMATED ESA PATTERN MEASUREMENT

The technique and software used to measure antenna patterns has been under development for four years. During the measurement process, a Hewlett-Packard series 200 computer is used to retrieve the trace via IEEE 488 bus from the spectrum analyzer. The trace is then plotted to the CRT or to a Hewlett-Packard 7470A plotter for analysis. In the process of plotting the measured data, the following parameters are checked:

- amplitude stability
- actual antenna movement
- spec. gain at measured frequency
- FCC curves are plotted based on spec. gain
- labels and analyzer settings are plotted

This data can be stored on disk for future reference.

SOFTWARE ANALYSIS

To ensure that the angular scaling is correct, the software program finds the peak of the trace and centers it on the plot. The first half of the plot determines the angular coverage, with the second half of the plot scaled accordingly. This method of plotting eliminates the need for the peak of the trace to be at the exact center of the grid on the spectrum analyzer.

The peak of the signal represents the point at which the 29 - 25LOG(Ø) curve is referenced. From user inputs of gain and angular coverage, the software centers the trace on the plot, calculates the FCC envelope, and plots it on the trace, incrementing 0.1 degrees as it plots. A plot of a 2.3 metre antenna recorded by this method is shown in Figure 5.

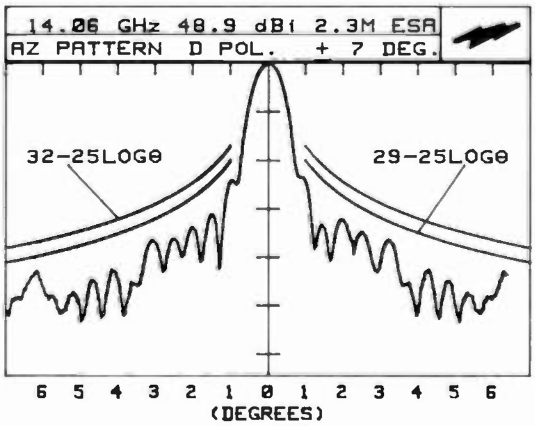

FIGURE 5. 2.3M PLOT

MAXIMUM TEST EIRP CALCULATION

A method by which to calculate the maximum EIRP allowable by an AUT to ensure that negligible adjacent satellite interference occurs during an antenna test is known as Maximum Test EIRP Calculation. Factors in determining the Maximum Test EIRP allowable include Site EIRP and Cut Isolation.

Site EIRP

When an earth station is licensed by the FCC, a corresponding maximum EIRP is permitted at that station. See Figure 6. This means that if full EIRP is radiated by the main beam of the earth station antenna, a certain power level is also being radiated off axis due to antenna sidelobe levels. By definition of the 29 - 25LOG (Ø) rule, the power in the sidelobes cannot exceed this curve. If for example the maximum EIRP for a given station is 84 dBW and the AUT gain is 50 dBi, then from Figure 6, the maximum permissible EIRP at two degrees off axis is given by:

Max. EIRP-(AUT gain-(29-25LOG(2)) dBW

84-(50-(29-25LOG(2)) dBW

= 55.5 dBW

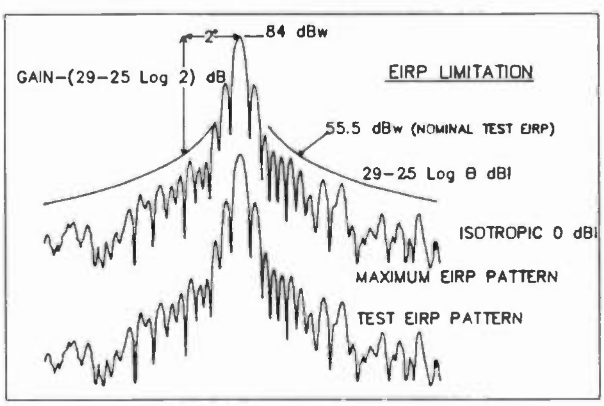

FIGURE 6. EIRP LIMITATION

Cut Isolation

The majority of transmit earth stations currently in use utilize mount A geometry of the Elevation over Azimuth design. See Figure 7. With this mount geometry, azimuth movement alone at the earth station does not accurately follow the satellite geostationary arc. See Figure 8.

In Figure 8, the earth station longitude matches that of the satellite at 99 degrees. If the azimuth is moved by two degrees of arc (to the azimuth angle for the satellite at 101 degrees), the boresight of the antenna will miss the satellite by an elevation factor (Arc Offset Angle) in degrees, which for a given antenna translates directly to a dB power delta termed here as Cut Isolation. The figure also shows that if the azimuth cut is made through a satellite other than at the earth station longitude (i.e. 75 degrees), the Arc Offset Angle and therefore Cut Isolation increases. Figure 9 is a chart showing the relationship between earth station latitude, earth station longitude, and Arc Offset Angle.

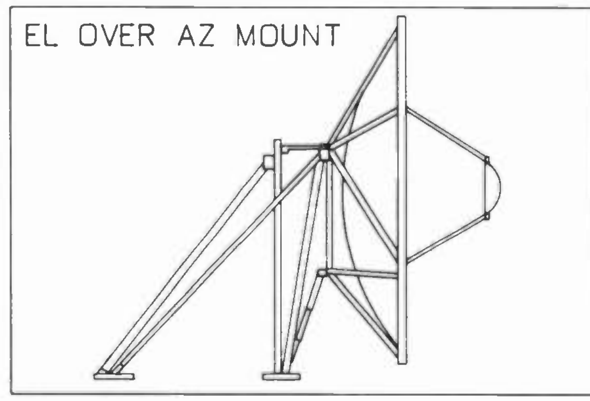

FIGURE 7. EL OVER AZ MOUNT

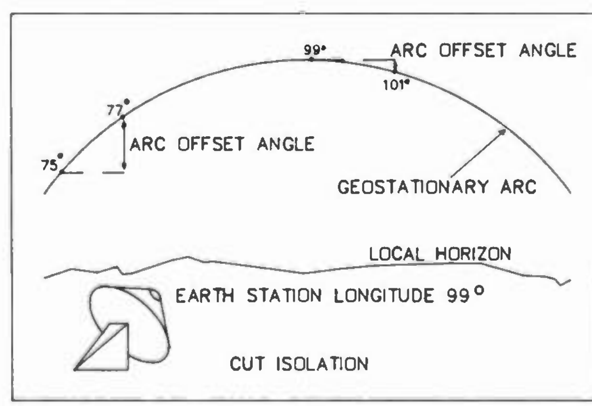

FIGURE 8. CUT ISOLATION

Cut Isolation can be determined by following this procedure:

(1) Given the ESA longitude and latitude, and the primary satellites longitude, determine a value for arc offset angle from Figure 9.

(2) Peak the AUT on the test satellite.

(3) Transmit a stable CW test signal at a nominal EIRP level. (nominal EIRP level = Power into AUT + Gain of AUT)

(4) While monitoring the downlink signal level at the CES, move the AUT off the satellite in elevation by the arc offset angle. The resultant change in the downlink power level when referenced to the main beam peak power level is the Cut Isolation. Typical values are 2-5 dB.

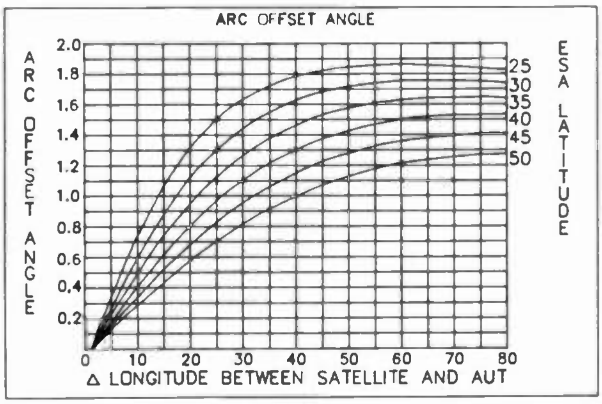

FIGURE 9. ARC OFFSET ANGLE

Maximum EIRP Allowable

Thus, the maximum allowable EIRP for the AUT with an adjacent satellite two degrees away is:

Max.EIRP-(AUT Gain-(29-25LOG(2))+Cut Isolation

CONCLUSIONS

A method of pattern testing earth station antennas has been presented. A step-by-step procedure in actual antenna sidelobe testing was covered, concluding with a description of a pattern analysis program developed by Andrew Corporation and Contel/ASC in 1984.

Since the onset of antenna pattern measurements, three significant innovations in the industry accepted method of verifying antenna performance have been introduced.

The first innovation, developed by Andrew Corporation and Contel/ASC in 1984, utilizes high stability uplink and downlink equipment to increase the dynamic range of the measurement system by 25 to 30 dB. This allows the EIRP radiated by the AUT to be lowered to minimize the possibility of adjacent satellite interference during testing.

The second innovation, also developed by Andrew Corporation and Contel/ASC, utilizes state of the art computers from Hewlett-Packard to record pattern data, analyze the data, and store the data on disk.

The third innovation introduced a procedure for calculating the maximum AUT EIRP and downlink dynamic range based on actual satellite spacing.

ACKNOWLEDGEMENTS

The concept of using a stable, unmodulated carrier was developed at Comsat Labs under the sponsorship of Intelsat. This concept was first proven during the In Orbit Testing of the Intelsat IV A satellites (F2 and F3).

All of the test equipment and portions of the software were purchased from Hewlett-Packard Co.

Further information on references made to the FCC can be obtained from the Phase I and Phase II FCC final reports generated from inputs of the Two Degree Satellite Spacing Committee.

The authors would also like to thank the following engineers from Andrew Corporation and Contel/ASC for their contributions in developing test procedures and software required to perform this testing method:

Fred Frey	Contel ASC
Kevin Linehan	Andrew Corp.
Behrooz Fakahari	Contel ASC
Mike Lee	Contel ASC

REFERENCES

Morgan, M., Linehan, K., Frey, F. "An On-Site Technique for Earth Station Antenna Radiation Pattern Measurements In A Two Degree Spacing Environment", Antenna Measurement Techniques Association, 1987 Proceedings.

CUSTOM DESIGNING A SATELLITE NEWS VEHICLE TO MEET STATION REQUIREMENTS

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In this paper we will outline the areas of consideration when planning your station's entry into the world of Satellite News Gathering. We will touch on all the issues that we encountered when talking and negotiating with the various vehicle and equipment manufacturers.

TRUCK CHASSIS AND BODY CONSIDERATIONS

When you purchase a satellite news vehicle, you are committing your station to a sizeable chunk of capital expenditure. Plan carefully! You will have to live a long time with any mistakes made here.

Size... Plan so that even with a minimum of production and editing facilities you are not so limited in space that working in the facility for extended periods is uncomfortable. Allow room for at least two persons, the truck operator and an editor. Ideally, there should also be room for talent and a producer. A work table for the producer will prove to be extremely useful. Adequate storage should be provided for a lighting kit, porta-reels, power cable, audio accessories, and anything else you may need for live feeds over a prolonged period. Remember, if you're not far from home when you use this truck, you probably didn't need it in the first place.

Fuel... Most medium to heavy duty truck vehicles use diesel fuel. Diesel engines are generally more durable and fuel effecient. Finding diesel fuel is usually not a problem, but it's not a bad idea to check your local area of operation for availability. Auxiliary fuel tanks should be considered to provide a working range of several hundred miles. Arriving on location without enough fuel to run the generator for several hours can be a bit frustrating.

Carrying capacity... Check the vehicle specifications against the delivered weight of the truck to be sure the payload capacity has headroom for the extra equipment and supplies you are likely to be hauling.

Manufacturer, domestic or foreign... There are good truck manufacturers both here and abroad. Usually there are no problems obtaining local service for domestic vehicles. If your interests are leaning toward a foreign manufacturer, be sure you can obtain nearby maintenance and repairs, especially during the warranty period. Also, are spare parts readily available?

Ease of maintenance... Can you get to the back of equipment racks easily? Once you're there, can you still move? Are the Travelling Wave Tube Amplifiers accessible? Where are the air conditioners mounted. If they're on the roof, they may be difficult if not impossible to service while the truck is in use. Are there separate air conditioners for the equipment and personnel? Is adequate ducting provided?

Safety... The truck should be properly interlocked to prevent starting the engine with the jacks down, the antenna deployed or the mast extended. Lights should be provided to illuminate the antenna and the mast as they are raised. If a mast is used, it should have a beacon mounted at the top. A means should be provided for raising the mast from outside the vehicle to encourage the operator to look up first. Reflective pylons should be carried and set up as soon as the truck is parked.

RF SYSTEM

In the RF system the antenna is probably the most critical item. The single most important point here is to insure FCC compliance. Specifically, will the antenna meet and sustain the present two-degree spacing requirements as defined in Part 25 of the FCC rules? Ask the antenna manufacturer to explain the range test procedures and to provide the corresponding data that is obtained. Comparison of this data may reveal some inherent deficiencies in the antenna design. Next, check the mounting of the antenna. Does it appear to be capable of preventing movement in high winds? (You say you are not going to cover hurricanes with your truck?) Speed of deployment? Don't put too much emphasis on this aspect. You will find that you have plenty of time time to put the dish up while TWTA's warming up, editors putting finishing touches on their stories, etc.

Some communications systems may be transmitting on a polarity different from the video. The Skyswitch communications package, used by CBS, CNN and other networks, requires a four port antenna for this reason. Other comms packages, such as the system supplied by Conus, do not have this requirement.

Look carefully at the specifications of you communications system before specifying your antenna.

Look carefully at the location of the TWTA relative to the feed point of the antenna. The best location for minimum waveguide loss would have the TWTA hub-mounted directly on the antenna. Our feeling is that we would prefer to have this unit located inside the truck, protected from the elements where it would be more easily serviced. The next best location then, would be directly under the antenna and inside the truck. Rack mounting works, but usually involves substantial waveguide loss and a waste of valuable rack space. TWTA's can be configured as a single unit with no redundancy, dual hot-switched with automatic or manual switching between the two HPA's, or dual phase-combined with no interruption of service caused by a failure of one of the PA's other than the temporary reduction of power. We often require less than 100 watts of RF driving our 2.6 meter antenna to saturate the transponder. A 300 watt amplifier provides adequate reserve for overcoming abnormal atmospheric attenuation. A smaller antenna or increased waveguide losses would naturally require more power.

A final RF consideration has to do with terrestrial microwave. A 40 foot pneumatic mast with 2 GHz antenna gives us the ability to receive from an ENG van, or to track a helicopter for extended coverage in areas not accessible to the satellite truck. Last winter WTKR-TV used the truck to cover skiing activities in the western part of Virginia. To cover the action at the top of the slopes, our truck operator set up a portable 2 GHz microwave there and provided our viewers with excellent coverage from two vantage points.

AUDIO/VIDEO SYSTEM

Let us not forget the purpose of this whole mission...that is to get pictures and sound back to the studio from a distant location. Keeping in mind that you may be using your truck for anything from a single reporter stand-up report to a dual-anchor remote newscast with edited packages produced on location, you should plan for audio mixing facilities for up to 4 microphones, plus tape playback, and the ability to switch between 2 to 4 cameras and video tape. Monitoring should include a good color monitor, a waveform/vectorscope combination, a spectrum monitor, a satellite receiver, and audio monitors and level meters. You will want cuts-only editing capability with separate monitoring for independent operation.

A source identifier with selectable full-field or vertical interval identification will be a tremendous help in locating your signal at the receiving end.

COMMUNICATIONS

You would think that since we are in the business of communications it would be a snap to handle any requirements here. Due to the very fact that we have a satellite truck that can go almost anyplace to bring pictures home to our viewers, it gets sent many times to locations where there are no reliable means of communications available from the truck. As mentioned earlier, the type of satellite communications package you choose will depend almost entirely on the news service in which you plan to participate.

Most comms packages provide for telephone communications through a satellite. This is great, but certainly not convenient for normal use as it requires the truck to be parked, levelled, antenna deployed, and the desired satellite acquired. However, many areas are covered by cellular phone systems and every truck should have cellular capability. Having an additional cellular phone antenna mounted on the truck will make it easy to use an additional portable "briefcase" type phone.

At least one land-line phone should also be installed for shoots, other than spot news, that can be pre-planned and require your truck to be on location for extended periods. The connections for the land line phones should be available on the outside access panel.

Although you will probably be operating your truck beyond the coverage of your remote-pickup 2-way system, you should have a mobile unit installed. A photographer and his news vehicle will almost always be assigned to accompany the satellite truck. With both vehicles having 2-ways on board communications can be maintained. Portable, handi-talkie type radios should also be available for giving cues and instructions to photographers and talent located near the truck.

DOWNLINK AND STUDIO FACILITIES

No matter what you do with the truck, it will all be for naught if you don't have adequate receive facilities. Practically all satellite news gathering operations take place on the KU band so we will limit our comments to equipment operating in that band.

Many satellite news gathering networks are using half transponders. This is an effecient way of utilizing expensive transponder time by allowing two uplink facilities to access a single transponder at the same time. This simply means that one uplink transmits on the lower half of a transponder while another transmits on the upper half of the same transponder. This method of transmission necessarily limits the transmitter deviation and subsequently reduces the signal to noise ratio. Let me be quick to point out that the resulting signal to noise ratio is quite satisfactory for a news gathering operation and is certainly justified when considering the economics of doubling up on transponder useage.

Considering the possibility of receiving half-transponder material, you must do your part when it comes to specifying the diameter of the receiving antenna in order to receive a good signal from the satellite. The better the received signal, the better the carrier to noise and subsequently, the better the signal to noise ratio of the video. At least one satellite news gathering service recommends a minimum 5.5 meter dish. At WTKR-TV in Norfolk, VA, we are using a 4.57 meter dish and are experiencing quite satisfactory results. We have found, however, that these satisfactory results are only obtained by very careful focusing and centering of the feed assembly at installation time.

If you are an NBC affiliate as is KTVY, you probably already have the necessary receive capability with the secondary antenna and the 2E receiver. Keep in mind, however, that NBC operates only in the full transponder mode. Additional equipment would be required to operate half transponder.

Some planning must also be given to cueing and providing IFB to the talent in the field. At WTKR-TV and KTVY, we have installed two private lines that do not go through our PABX switching equipment and which terminate in automatic answering devices. We have wired outputs from our audio consoles to these devices. You must be able to provide a "mix-minus" signal on these outputs. Mix-minus is simply defined as a MIX of audio sources sufficient to cue the talent in the field, usually the entire on-air mix, MINUS the audio from the satellite feed. This is done because of the transmission delays involved in satellite work. If we were to feed the talent's audio back to them over the IFB circuit it would arrive in their headphone approximately 1/2 second after speaking and would be very disconcerting to the talent. Provisions should also be made for interrupting the mix-minus feed with an intercom or talk-back feed for giving instructions to the talent in the field. Intercom facilities between a field producer, camera operators, and the studio should also be provided.

SUMMARY

Satellite News Gathering is an exciting and rapidly growing facet of our ever-changing business. We at WTKR-TV in Norfolk, and KTVY in Oklahoma City, have gone through a real learning experience this past year examining equipment at shows and conventions, checking specifications, finally ordering and then implementing our systems. We sincerely hope that relaying our experiences to you in this manner will make it easier when you put your system together.

FURTHER IMPROVEMENTS IN SATELLITE NEWS GATHERING

Makoto Kaijima
Toshiba Corporation
Kawasaki, Japan

General

Last year, at the NAB convention, Toshiba exhibited a portable earth station which it developed for use with a Japanese high powered satellite. The system employed a 1.2m antenna reflector and a 120W High Power Amplifier (HPA).
The subject of this papers is Toshiba's recently developed portable Ku-band Satellite News-gathering system (2) called the Satellite scoOpe System "Mt-3", which can be used to transmit video, audio and ancillary communications worldwide.

The Mt-3 employs a 1.8 meter antenna, a 300-watt HPA, a 150K Low Noise Block-converter (LNB), and transmit and receive capability for one video and two audio channels. In addition, the Mt-3 is configured with frequency agile Single Channel Per Carrier (SCPC) and Multiple Channel Per Carrier (MCPC) transmit and receive capability for edge of band communications. The system has been packaged so that each individual flight case is compact and sufficiently light-weight to meet International Airline Transportation Authority (IATA) regulations for "carry on" baggage. This type of portable Ku-band uplink system has been introduced in the past by Marconi(1), Midwest(2), Hubcom, etc. The Mt-3 is similar to previous system in it's architecture, but features ease of operation and deployment and less overall weight.

The Mt-3 was developed jointly by Toshiba Corporation and CBS Engineering and Development, based on a CBS specification of a Mini-RADET II system, Similar designs have been discussed in JIWP (The Joint Interim Working Party 10-11-CMTT/1 Comittee) for international Satellite Newsgathering use.

CBS and Toshiba have scheduled a joint test and evaluation of the system during March, 1988, utilizing a U.S. domestic satellite.

Fig. 1 Mt-3 ready for transportation

System Configuration

The components of the system are shown in Table 1.
The system ready for transportation is shown in Fig. 1 and the system setup for operation in Fig. 2.

Table 1 describes the functional unit the packages in size and individual flight case weight. As shown in the Table, the dimensions of the individual packages are such that the sum of the width, height and depth does not exceed 80 inches, and the weight of each package in shipping form is not more than 100 pounds.

The HPA section is shipped divided into the RF unit and PS unit to protect the TWT and to maintain weight below 100 lbs. The system is provided as 10 packages with an overall shipping weight of 884 pounds.

The Mt-3 system is shown in Fig. 3. As shown, the exciter incorporate a video modulator, audio modulator and an up-converter to produce an RF output between 14.0 and 14.5 GHz. The interface to the HPA is at a RF level of -10dBm. The 300W output of the HPA is routed via a flexible waveguide to the antenna.

Received signals in the 11.7 to 12.2 GHz band (or 10.95 to 11.7 GHz for Intelsat) are routed to the high stability LNB which provides an output signal of 950 to 1450 MHz (or 950 to 1700 MHz for Intelsat) to the receiving equipment which also housed within the exciter unit.

In this respect, the exciter has built-in MCPC and SCPC demodulators, which are used for communications between the base station operation center and the remote Mt-3 system.

The IF signal is split in the exciter and also sent to the video receiver and to the spectrum analyzer. A local oscillator signal is also generated in the exciter and applied to the high stability LNB.

Table 1 Composition of Mt-3

Function	Package Identification	W,H,D in inched	Weight in pounds
1.8mϕ OFF-SET Parabolic Antenna	Reflector 1	38x11.15x30.85	71
	Reflector 2	36.5x15.5x28	86
	FEED	26 x 16 x 32	55
	Mount A	30 x 12 x 35	99
	Mount B	23 x 28 x 29	96
	Tripod	22 x 12 x 46	92
300W HPA	TWT (RF)	26 x 19 x 34	98
	HPA SHELF	26x19.5x 32	99
Exciter	Exciter	26x19.5x 32	90
Monitor	Monitor	26 x 16 x 32	98

Antenna

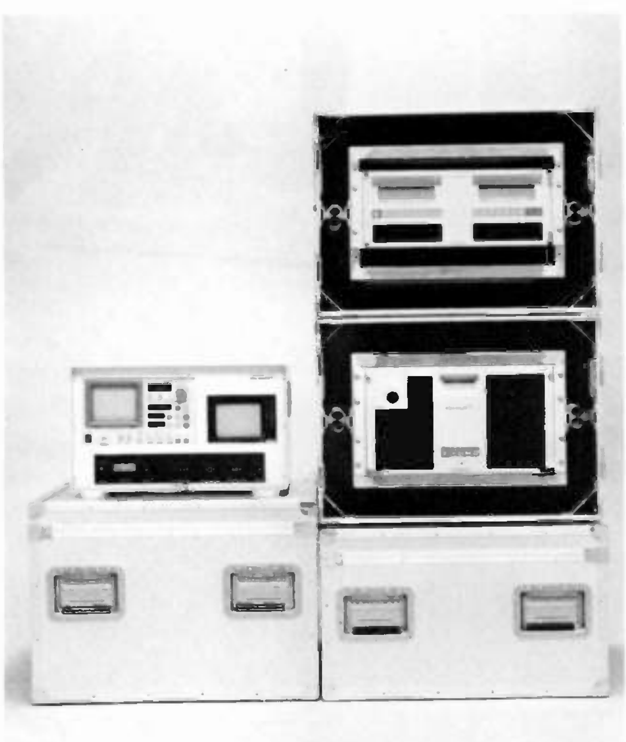
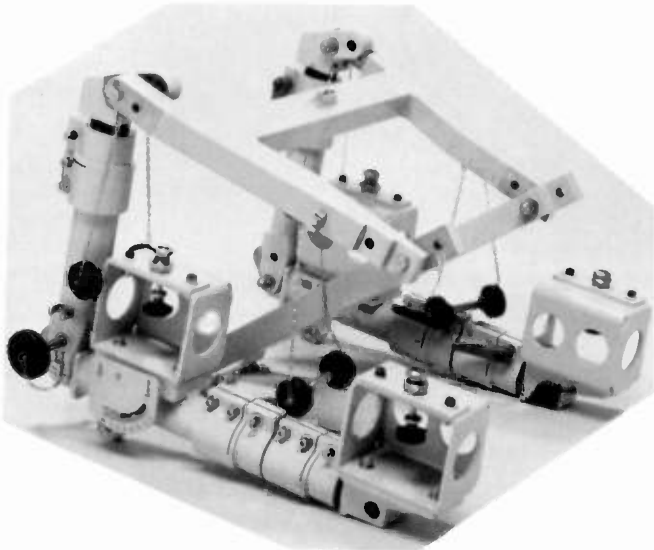


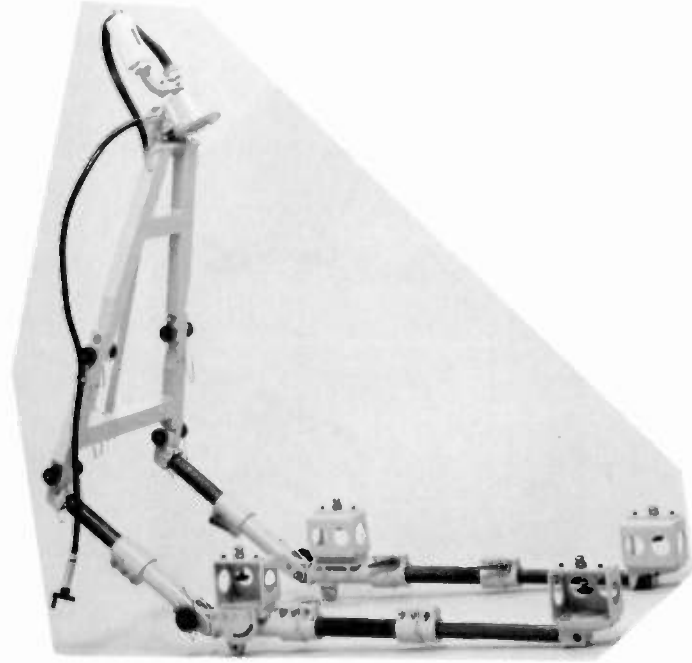
Fig. 2 Mt-3 in operation

Left:

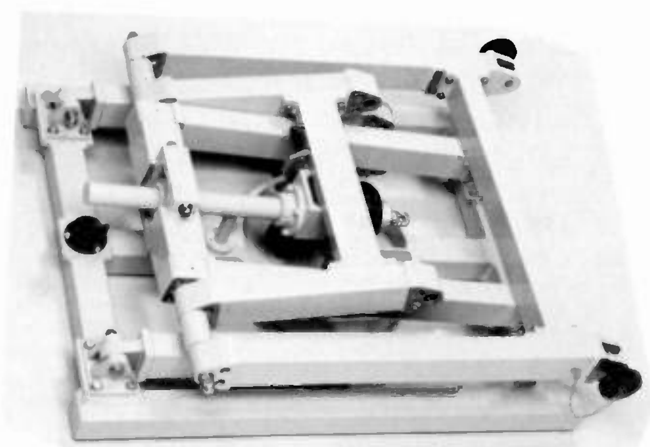
Electronics

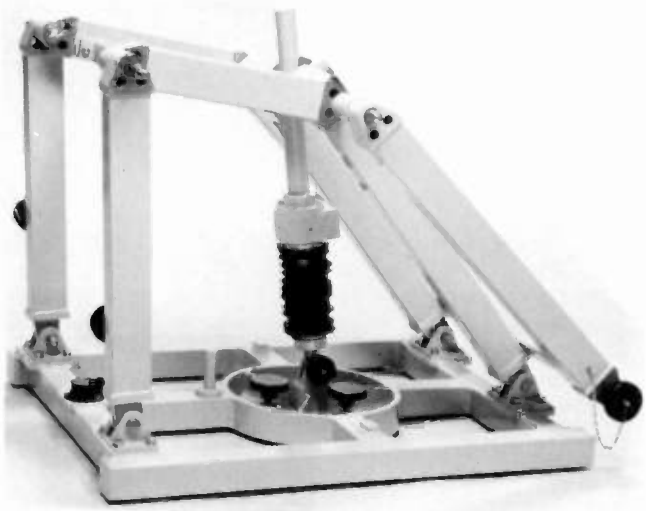
FEED MOUNT



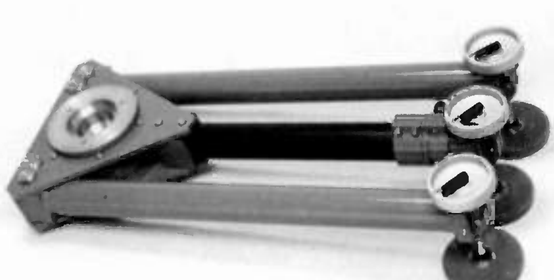


MOUNT



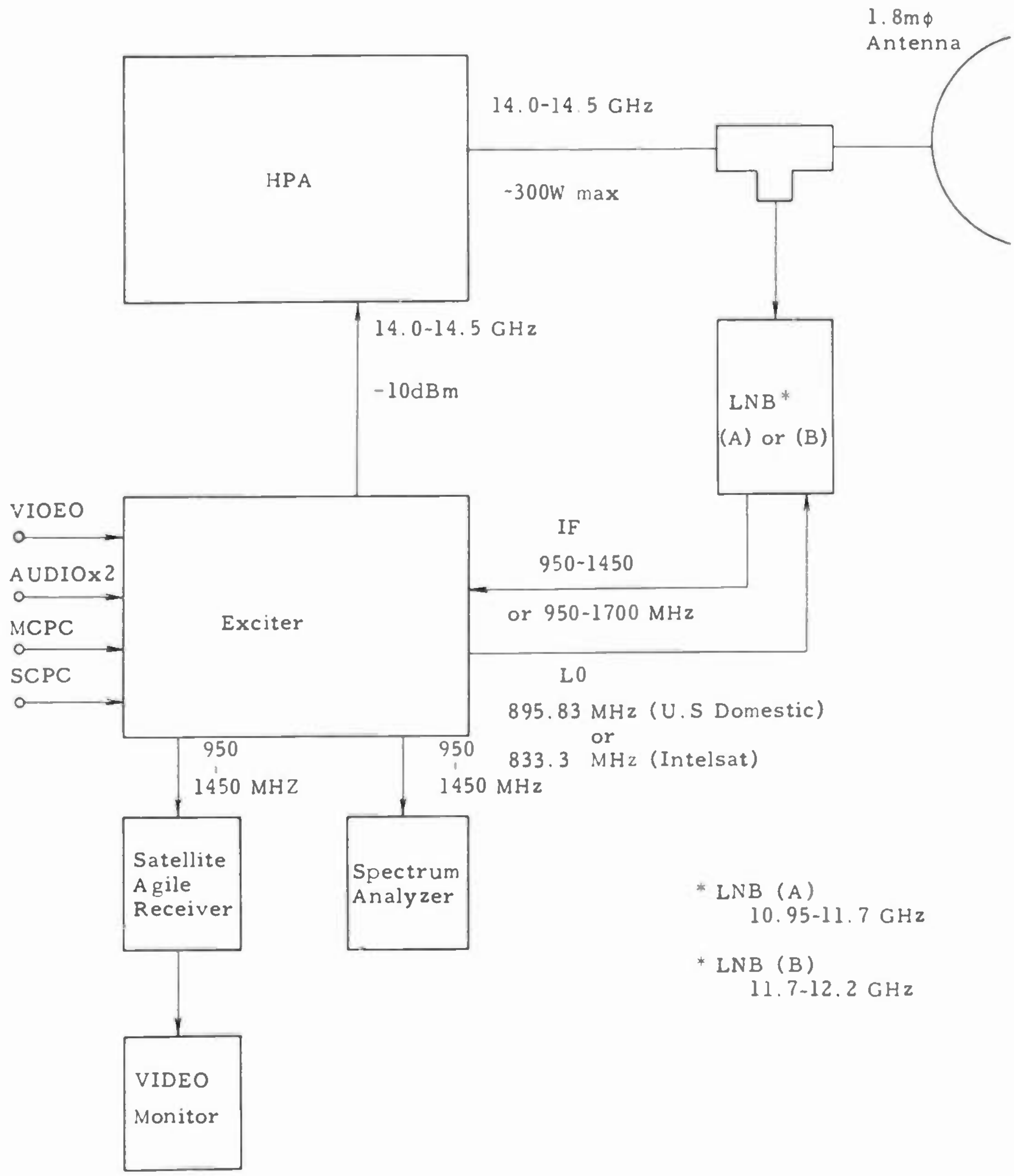


TRIPOD



Assemble Antenna

Fig. 3 System diagram of Mt-3



Features of Mt-3

1. Top Feed type offset parabolic Antenna

A top feed system allows the following:
(1) Wind resistance load reduction.
(2) Azimuth rotary mechanism is simplification, allowing 360 degree rotation.
(3) Plane of polarization can be easily adjustable from the back of the antenna.

2. New mechanism for Elevation adjustment

A movable mechanism is employed, which allows elevation angle adjustment of 5 to 80 degrees using a single jack.

3. Six segment Antenna Reflector

(1) The reflector is of nominally 1.8m in diameter and is divided into six segments. This design reduces dimensions and number of packages for easy transportation.
(2) The reflector is constricted using FRP (Fiber-reinforced plastic) which helps reduces the weight of the equipment. Segment reflector is shown in Fig. 4.

4. Synthesized Exciter

(1) The unique synthesizer design covers the full range of 14.0 to 14.5 GHz in steps of 125 kHz by the use of a micro Central Processing Unit (CPU).
(2) The MCPC and SCPC channel (push to talk operated) frequencies can be set independently to a maximum separation of 27 MHz below the center of the transponder in 250 kHz steps shown in Fig. 5.
(3) The combination of multiple synthesizers reduce the carrier phase noise of the MCPC & SCPC channels.

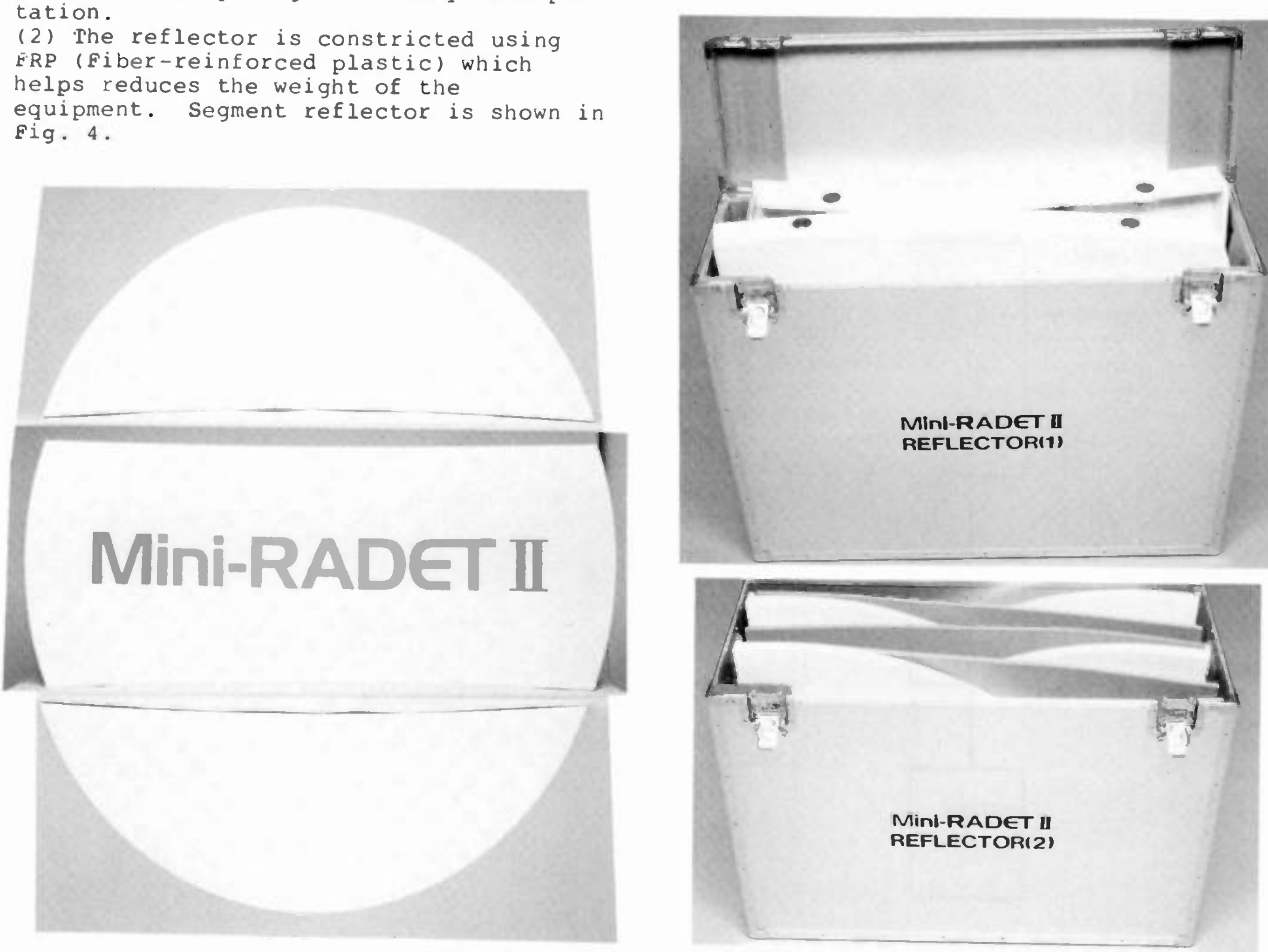


Fig. 4 Segment Reflector of Mt-3

5. All purpose Exciter

(1) Specifications

TV, MCPC & PTT
    Independent level setting
RF 30MHz/17.5MHz
    Selectable
Emphasis 525/625
    Selectable
Emphasis 50μ/75μ/J17/flat
    Selectable
Audio Subcarrier (2)
    5.8 ∿ 7.8 MHz
    100kHz step available
4 Mode operation
    P.T.T/P.T.T & MCPC/MCPC/VIDEO
    selectable

(2) The MCPC baseband signal is switched to the Video subcarrier automatically when the Video transmission mode is selected. The above functions are set via key switches on the front panel of the unit.

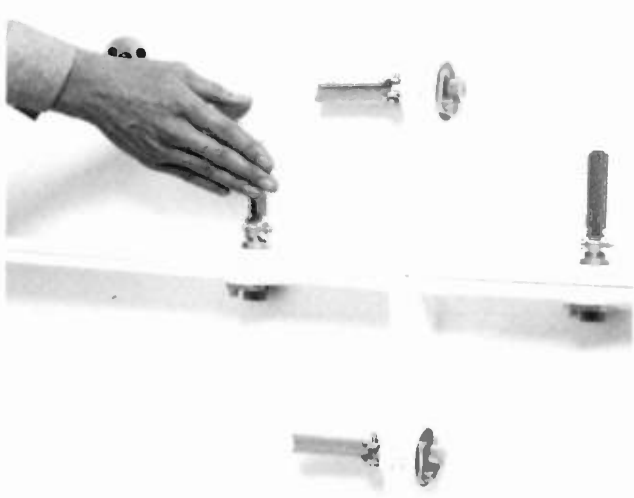
6. CPU controlled HPA

Transmitting power monitor accuraccy is assured over a wide range of temperature. Alarm detection and protective functions are provided with appropriate indications.

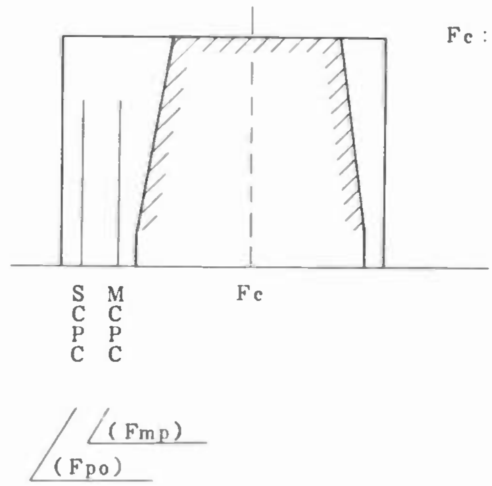
7. Quick clamps and Releases for Quick Setup

These parts have been engineered in thought to shorten the setup time and assure proper operation each setup.

(a) Reflector



Quick clamps



Fc : Center freq. of video carrier

$Fc = 14.0 \sim 14.5\,GHz$ (125kHz step)

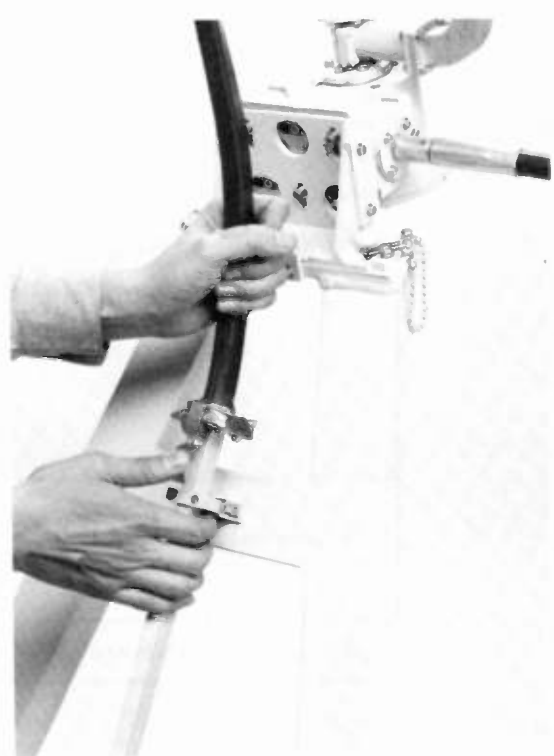
$Fpo = Fc + 0.25 * M$ (MHz)

$Fmp = Fc + 0.25 * M$ (MHz)

for $-108 < M, N < 108$

Fig. 5 Frequency agility of each carrier

(b) Waveguide

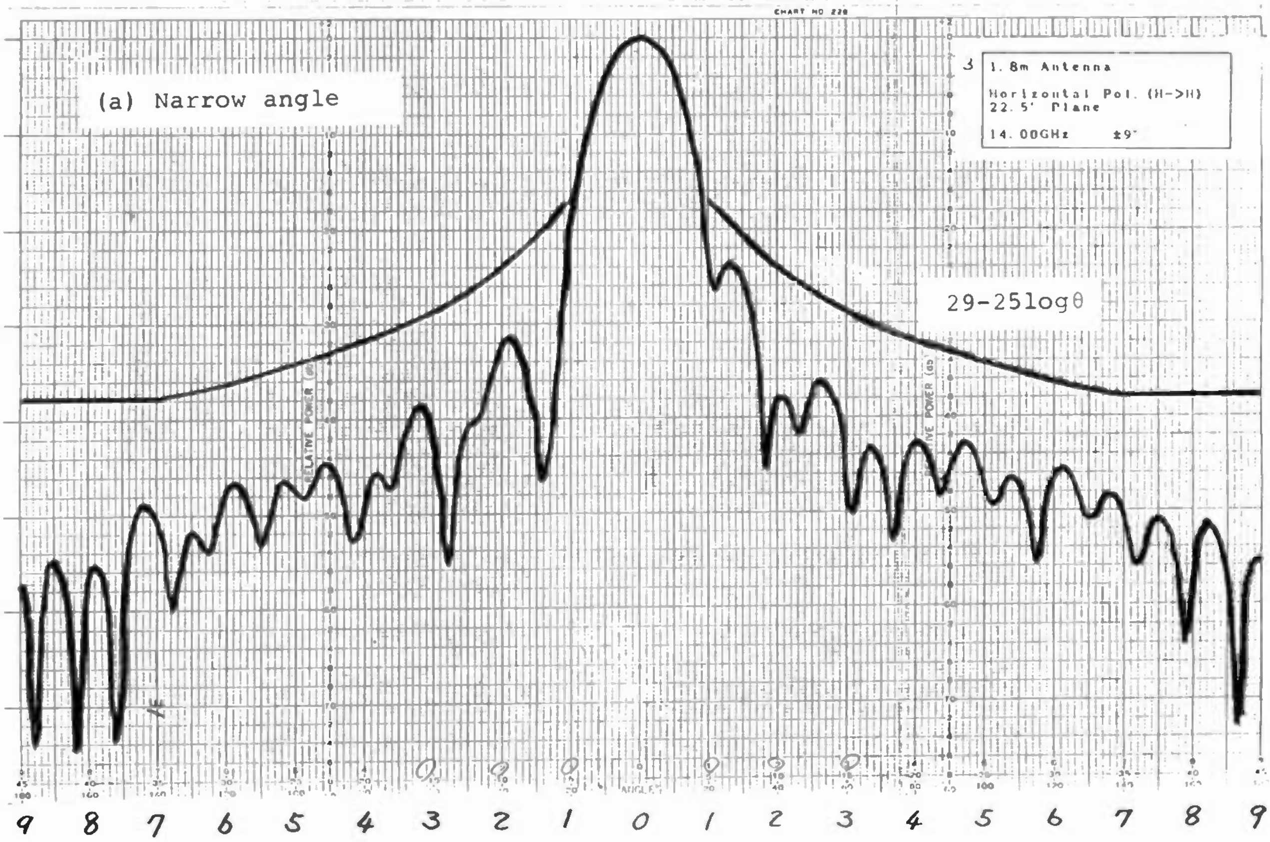


System Specifications of Mt-3

Main specifications of the Mt-3 are shown in Table 2.
Antenna performance meets the Federal Communication Commision (FCC) requirement as specified in Part 25-209. Part 25-209 requirements are met for various planes such as 22.5° and 45° as may be met in action deployment for various satellite. An example of antenna pattern measurement is shown in Fig. 6. Total system Power Consumption is 2.08kVA at 300w RF power output.

Table 2 Main characteristics of Mt-3

Eirp		14.25 GHz	69 dBW max.
G/T		11.15 GHz	21.3 dB/k typ.
Frequency		TX RX agility	14.0 - 14.5 GHz 10.95 - 12.2 GHz 125 kHz step
TX	VIDEO	VIDEO AUDIO MCPC (Subcarrier)	1 2 1
	MCPC		1
	P.T.T	(SCPC)	1
RX	VIDEO		1 (monitor)
	AUDIO		2 (monitor)
	MCPC		1
	P.T.T		1
Mod.	VIDEO		15 - 25 MHz p-p
	Dispersal		0.5 - 2 MHz p-p
	Audio Subcarrier		0.5 - 2 MHz p-p
	Audio		50 - 150 kHz p-p
	MCPC		50 - 150 kHz p-p
	P.T.T		75 kHz p-p
Operational Wind Velocity			withoutguy wire 30miles/hour max withguy wire 60miles/hour max
Operational Temperature			5 - 105°F



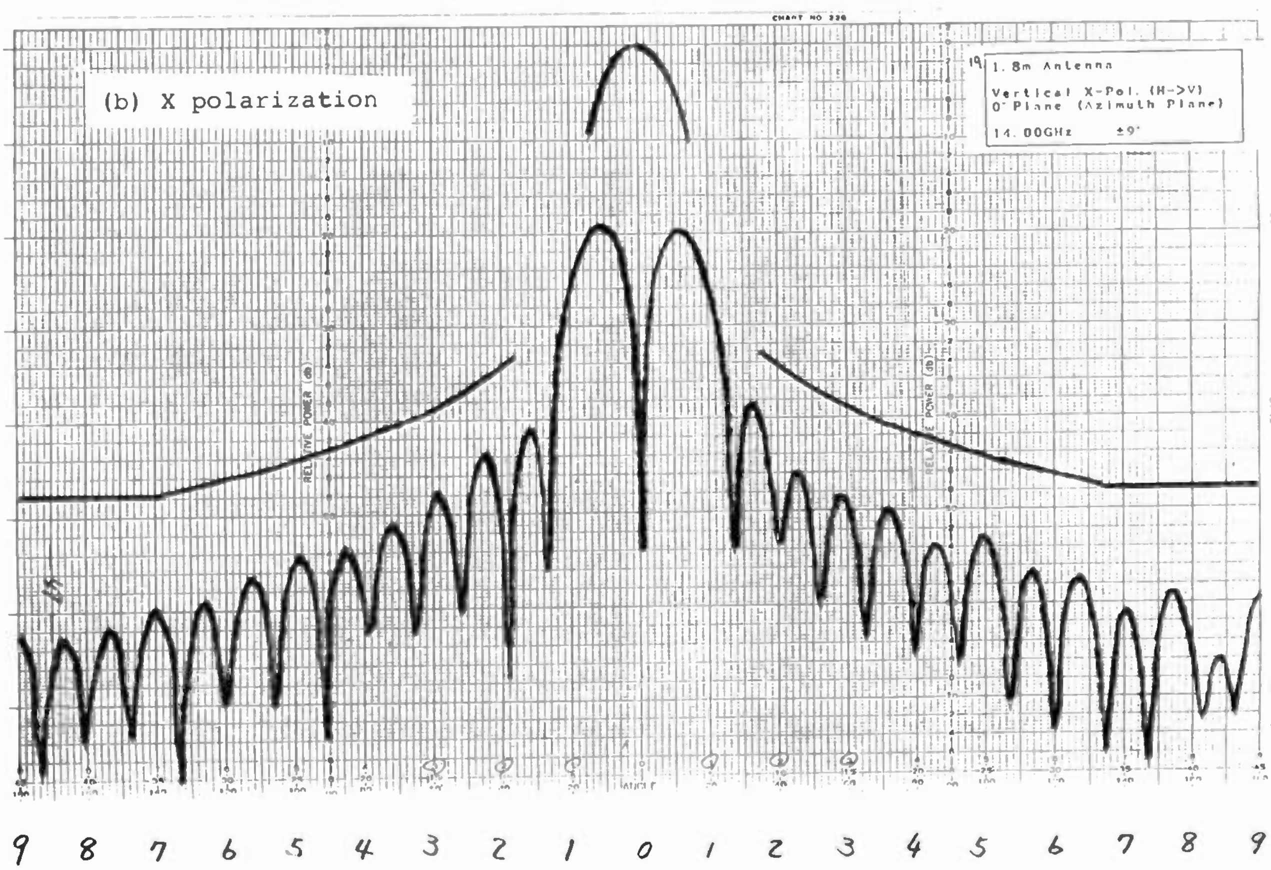


Fig. 6 1.8m Antenna Pattern

Summary

The Mt-3 system will prove to be an economical, reliable easily operated and easily deployed unit for satellite newsgathering. This unit incorporates new engineering features based upon industry experience with prior prototypes and is specifically designed for SNG. The unit is not a composite of avairable "off-the-shelf" units as are most SNG equipment.

Acknowredgement

Toshiba Corporation has developed this product jointly with News Organization in the United States.
Toshiba offers it's thanks to CBS Engineering & Development for their valuable advice and developmental assistance.

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(1) J.Ramasastry, G.Knights et al. "RADET - a new tool for news-gathering"; EBU REVIEW NO.222-APRIL, 1987.

(2) S.Hilaris, "Field Experience with Rapid Deployment Earth Terminals (RADETs) for Satellite newsgathering"; CBS Review, March, 1987.

A NOVEL METHOD OF MEASURING THE DEVIATION OF VIDEO FM MODULATORS

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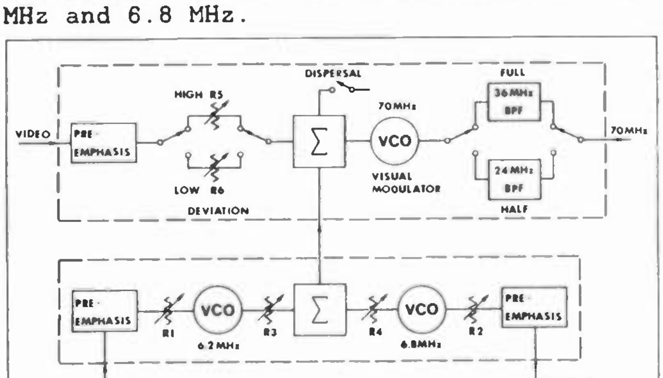
ABSTRACT: The purpose of this communication is to present a simple yet accurate method of verifying, and if necessary adjusting, the frequency deviation of the visual carrier and of the aural subcarriers without having to resort to the indirect method of Carrier Bessel Null. The method proposed is a direct method in that it measures the actual deviation of the FM carrier when this carrier is modulated with video test signals. This assures optimum use of the available transponder bandwidth for single or dual video operation.

INTRODUCTION: In establishing an SNG video transmission circuit it is important to optimize all of the transmission parameters for best overall performance.
These parameters include:
On the uplink side; antenna gain, HPA power, video deviation, aural subcarrier level and subcarrier deviation.
On the downlink side; Receive antenna gain and LNA noise figure for best G/T and, optimum receiver IF bandwidth for best C/N.
In the transponder; attenuator setting for best compromise between available uplink power and resulting transponder EIRP.

FM MODULATORS: A simplified block diagram of the visual and aural FM modulators is shown in Figure 1. There are two frequency agile aural subcarrier modulators in the exciter. These can operate on any frequency between approximately 5 MHz and 7.5 MHz. The nominal subcarrier frequencies are 6.2 MHz and 6.8 MHz.



BLOCK DIAGRAM VISUAL/AURAL FM MODULATOR

FIGURE 1

The audio signal is applied to a 75 usec pre-emphasis network and the audio levels can be adjusted by controls R1 and R2 ahead of the Voltage Controlled Oscillator (VCO) FM modulators.
The level of the two modulated subcarriers is adjustable by R3 and R4, ahead of the summing network. The summed aural subcarriers are applied to a second summing network, the output of which is applied to the visual FM Modulator. In essence, the audio signal modulates the subcarrier which in turn modulates the main carrier. The other input to the summing network receives the video signal which has been modified by its pre-emphasis network.
The exciter has two video deviation adjustment controls, one for "FULL" transponder operation and one for "HALF" transponder operation. In the "FULL" transponder mode, the modulated signal is routed through the 36MHz IF bandpass filter. For "HALF" transponder operation, the 24 MHz IF bandpass filter is used.

VIDEO PRE-EMPHASIS NETWORK: In FM systems the noise voltage spectrum is triangular, increasing linearly with frequency.
The color-subcarrier with its sidebands resides at the upper part of the band and is therefore subjected to high levels of noise. After demodulation, the high frequency channel noise becomes low frequency color noise. There is therefore an imbalance between the signal-to-noise ratios of the luminance and chrominance signals.

Fortunately, the amplitude of the higher spectral frequency components of a video signal are of decreasing amplitude. This means that high video frequencies can be pre-emphasized before transmission. A de-emphasis network, having complementary characteristics, attenuates the higher video frequency components in the receiver thereby restoring flat frequency response and at the same time, reducing the amplitude of the triangular noise to a considerable extent.

For transmission of NTSC signals over analog FM microwave links or satellites, the CCIR has defined a pre-emphasis network and its associated response curve as shown in Figure 2.

The 761.6 kHz frequency is referred to as the "crossover" frequency of the emphasis network. At this frequency the insertion loss of the pre-and de-emphasis networks are assumed zero. Above this frequency, the de-emphasis network removes noise, while below it, the noise is increased. In FM systems there is a net improvement because with triangular noise, most of it is above the crossover frequeny.

For NTSC,the low frequencies are attenuated by 10 dB and the chrominance signal at 3.58 MHz, is transmitted with +3.16 dB emphasis. For NTSC signals, intermodulation between chrominance and luminance results in differential gain and differential phase. These effects are greatly reduced by lowering the amplitude of the luminance component in the pre-emphasis network.

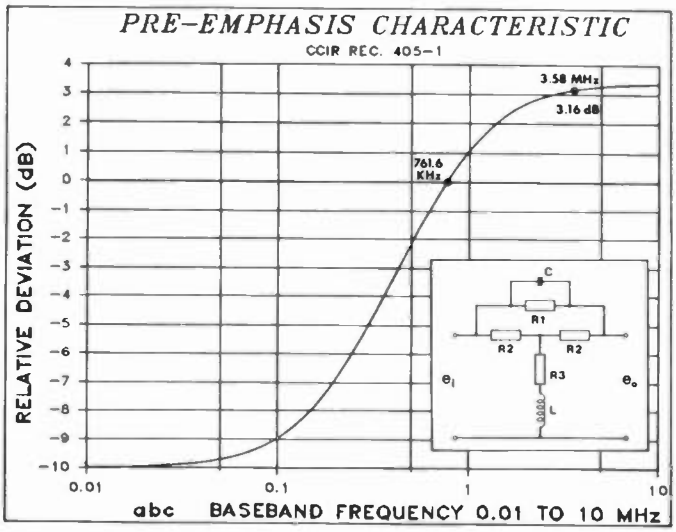


FIGURE 2

The introduction of emphasis appears to be a useful tool in the design of FM systems. Unfortunately, it also creates problems. For large and steep luminance transitions, the instantaneous rate of change of the luminance level is made even larger by the pre-emphasis network. This is clearly visible in Figure 3. The peak-to-peak luminance transition has increased from 1 Volt to 1.2 Volts.

The bandwidth of an FM signal is roughly proportional to both the amplitude and the frequency of modulation, therefore, these large black-to-white and white-to-black transitions may cause overdeviation in either direction.

The chroma signal too can cause large frequency deviations whenever a strongly colored object is present in the scene, since the pre-emphasis network boosts the chroma signal.

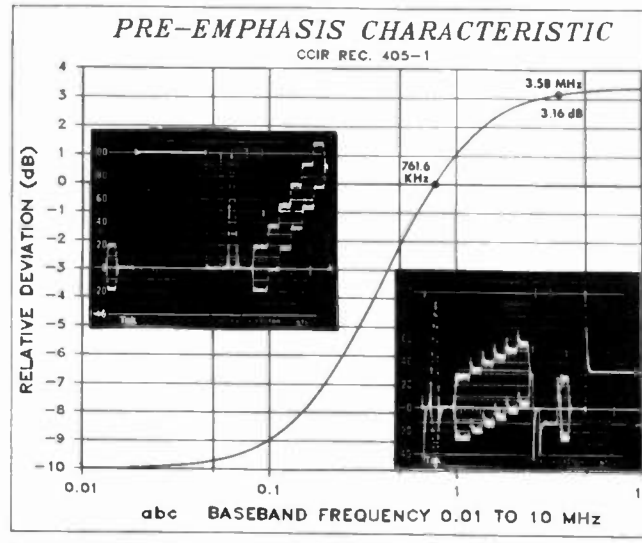


FIGURE 3

EFFECT OF PRE-EMPHASIS ON TEST SIGNALS:
Figure 3 shows the composite test signal; the BAR with 1T rise time, 2T PULSE, 12.5T PULSE and 5 STEP STAIRCASE signals before and after going through the pre-emphasis network.

The flat part of the 100 IRE BAR has been reduced by 10 dB to 31.62 IRE. The leading edge of the BAR becomes a sharp spike of approximately 100 IRE. The trailing edge too becomes a sharp spike of approximately 100 IRE starting at 31.62 IRE and reaching a level of approximately -70 IRE. The burst and subcarrier levels which measured 40 IRE Peak-to-Peak, are now increased by 3.16 dB and measure 57.56 IRE or almost 44% higher than before. The luminance component of each of the five steps has been reduced by 10 dB.

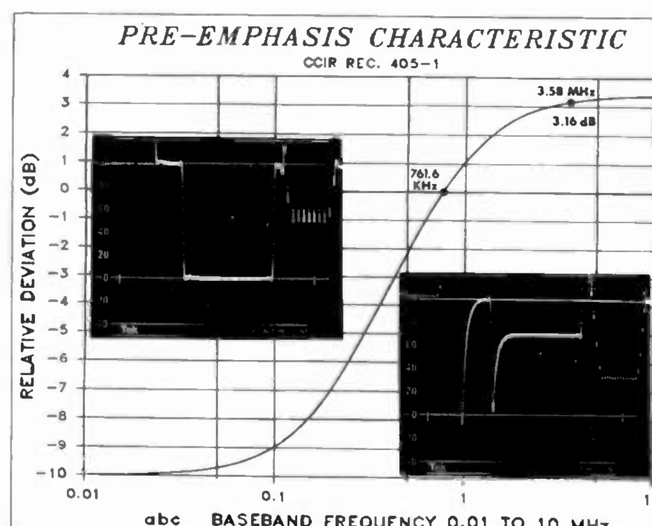


FIGURE 4

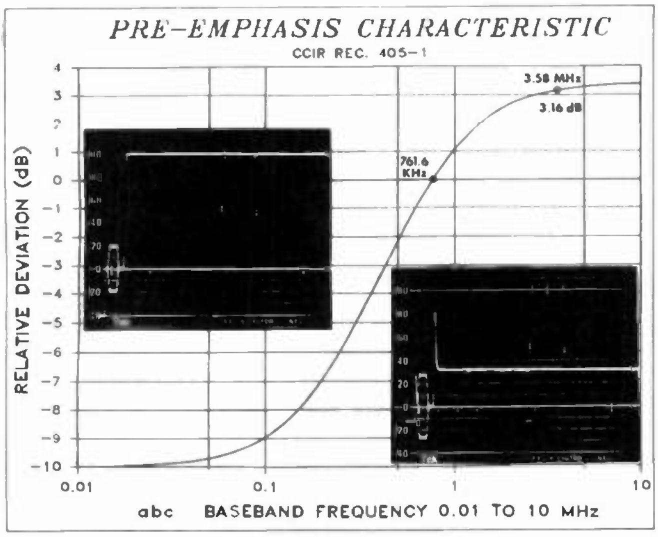


FIGURE 5

Figure 4 is a detail of the horizontal sync pulse expanded from 40 IRE to 100 IRE. Again, the leading edge of the sync pulse reaches full amplitude whereas the flat top is reduced by 10 dB therefore the -40 IRE becomes -12.65 IRE.

Figure 5 shows the WINDOW signal. For this signal too, the white level is reduced by 10 dB, from 100 IRE to 31.62 IRE but the leading edge spike now reaches 80 IRE, not 100 IRE as was the case for the BAR signal with 1T rise time. The rise time of the WINDOW signal is approximately 250 nsec and this is inadequate to build-up to 100 IRE when going through the pre-emphasis network as the time constant, which controls the shape of the waveforms, is approximately 180 nsec.

Figure 6 shows the 5 STEP STAIRCASE SIGNAL after going through the pre-emphasis network.

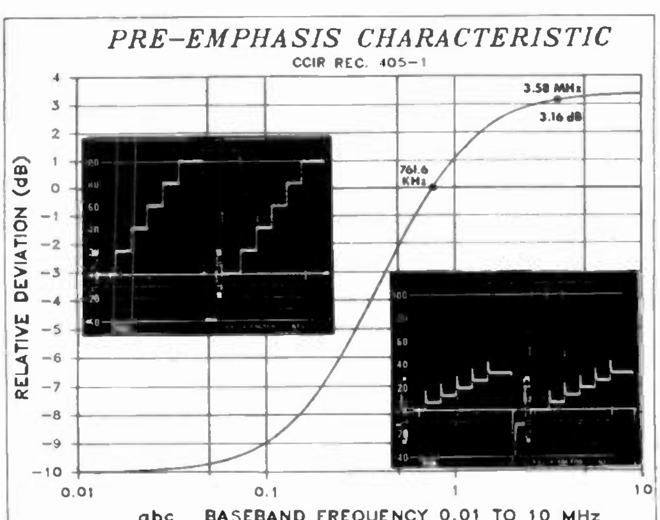


FIGURE 6

PRECISE METHOD OF SETTING DEVIATION:
Knowing that at the crossover frequency of 761.6 kHz, the gain is unity, this allows for a very precise setting of the deviation using the carrier Bessel null method. An example will illustrate this method.

Figure 20 shows the spectrum of an FM modulated signal for a modulation index m=1. The carrier which was unity before modulation, is now reduced by 2.32 dB. The first order sidebands J1 are 7.13 dB below the unmodulated carrier, the second order sidebands J2 are 18.79 dB below the unmodulated carrier etc. By increasing the modulation index, the carrier can be reduced to zero and this occurs for m = 2.4048. See Figure 19 and Appendix I.

> EXAMPLE: What is the level of the test tone for first carrier Bessel null so that the modulator will deviate the carrier 10 MHz for a 1 Volt peak-to-peak video signal?

For Fm = 761.6 kHz and m = 2.4048 for first carrier null, the deviation is:

$\Delta F = m x F_m = 2.4048 x 0.7616 = 1.8315$ (MHz)

Since the carrier frequency deviation is 10 MHz for a 1 Volt p-p video signal, the required p-p voltage for 1.8315 MHz deviation is:

1.8315/10 = 0.18315 V or 183.15 mV p-p, or 64.75 mV RMS.

Table 1 gives the level of the 761.6 kHz test tone for carrier frequency deviations between 7 MHz and 15 MHz in 200 kHz steps. The required test tone levels are listed in: mV peak-to-peak, mV RMS and dBm. Where 0 dBm equals 1 mW into 75 Ohms.

> EXAMPLE: The level of the 761.6 KHz test tone for 13.8 MHz deviation is 46.92 mV RMS or, 132.72 mV p-p or -15.32 dBm.

The carrier Bessel null method is only accurate if the level of the test tone can be set precisely to the desired value and if the video input impedance presented by the modulator is exactly 75 Ohms, and if the spectrum analyzer used to determine carrier null has adequate resolution. If the level of the test tone is in error by 1% than the deviation will also be in error by 1% and, if the carrier null observed on the spectrum analyzer is only 25 dB, an error of +/-4% will result. One way to determine the level of the test tone is to loop the output of the test tone generator through the vertical input of a calibrated waveform monitor (WFM) and to terminate the test tone source in the input impedance of the visual modulator. For a given test tone level for unity vertical gain and (X5) vertical gain of the WFM the corresponding

TEST TONE LEVEL REQUIRED FOR GIVEN DEVIATION

TABLE 1

DEV MHz	P-P mV	RMS mV	dBm *	DEV MHz	P-P mV	RMS mV	dBm *
7	261.64	92.50	-9.43	11	166.50	58.87	-13.35
7.2	254.37	89.93	-9.67	11.2	163.53	57.82	-13.51
7.4	247.50	87.50	-9.91	11.4	160.66	56.80	-13.66
7.6	240.99	85.20	-10.14	11.6	157.89	55.82	-13.81
7.8	234.81	83.02	-10.37	11.8	155.21	54.88	-13.96
8	228.94	80.94	-10.59	12	152.62	53.96	-14.11
8.2	223.35	78.97	-10.80	12.2	150.12	53.08	-14.25
8.4	218.04	77.09	-11.01	12.4	147.70	52.22	-14.39
8.6	212.96	75.29	-11.22	12.6	145.36	51.39	-14.53
8.8	208.12	73.58	-11.42	12.8	143.09	50.59	-14.67
9	203.50	71.95	-11.61	13	140.88	49.81	-14.80
9.2	199.08	70.38	-11.80	13.2	138.75	49.06	-14.94
9.4	194.84	68.89	-11.99	13.4	136.68	48.32	-15.07
9.6	190.78	67.45	-12.17	13.6	134.67	47.61	-15.20
9.8	186.89	66.07	-12.35	13.8	132.72	46.92	-15.32
10	183.15	64.75	-12.53	14	130.82	46.25	-15.45
10.2	179.56	63.48	-12.70	14.2	128.98	45.60	-15.57
10.4	176.11	62.26	-12.87	14.4	127.19	44.97	-15.69
10.6	172.78	61.09	-13.03	14.6	125.44	44.35	-15.81
10.8	169.58	59.96	-13.19	14.8	123.75	43.75	-15.93
				15	122.10	43.17	-16.05

* dBm = 1 mW into 75 Ohms

761.6 kHz TEST TONE LEVEL FROM WAVEFORM MONITOR

TABLE 2

IRE (x5)	ΔF MHz	IRE (x5)	ΔF MHz	IRE (x5)	ΔF MHz	IRE (x1)	ΔF MHz	IRE (x1)	ΔF MHz
90	14.24	110	11.65	130	9.86	30	8.55	50	5.13
92	13.94	112	11.45	132	9.71	32	8.01	52	4.93
94	13.64	114	11.25	134	9.57	34	7.54	54	4.75
96	13.35	116	11.05	136	9.43	36	7.12	56	4.58
98	13.08	118	10.86	138	9.29	38	6.75	58	4.42
100	12.82	120	10.68	140	9.16	40	6.41	60	4.27
102	12.57	122	10.51	142	9.03	42	6.10	62	4.14
104	12.33	124	10.34	144	8.90	44	5.83	64	4.01
106	12.09	126	10.17	146	8.78	46	5.57	66	3.88
108	11.87	128	10.02	148	8.66	48	5.34	68	3.77
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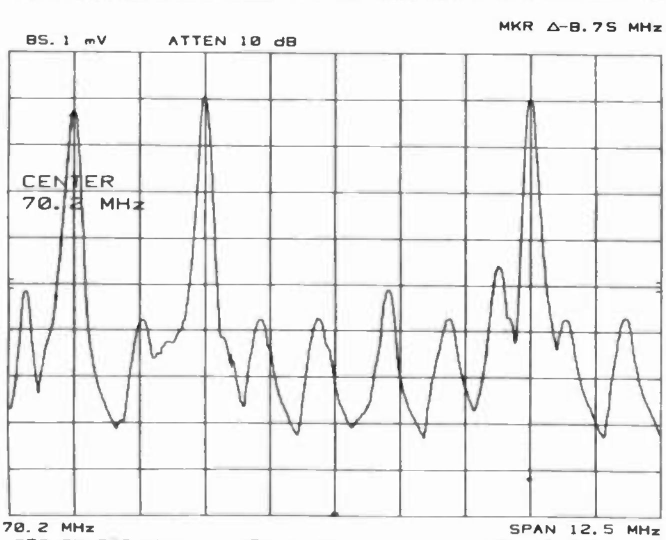


FIGURE 7

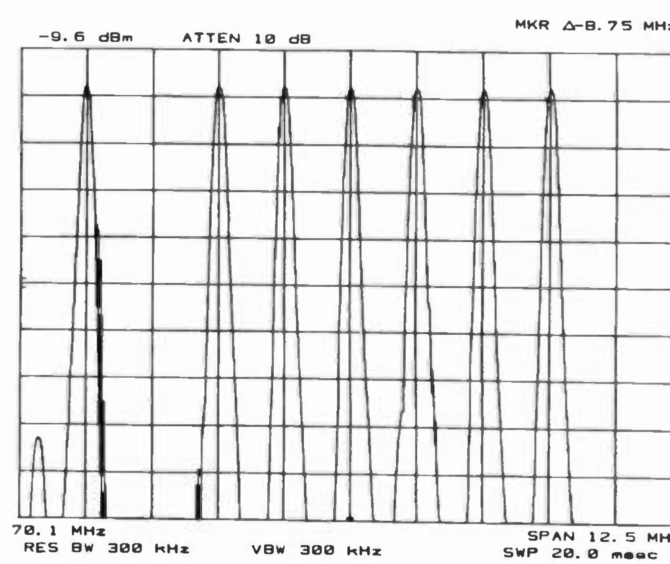


FIGURE 8

deviation for first carrier null is given in Table 2.

USING VIDEO TEST SIGNALS FOR CARRIER DEVIATION SETTING: With a video test signal such as the WINDOW signal, the spectrum of the FM modulated signal is as shown in Figure 7. The spectral energy corresponding to peak of sync is clearly visible, so too are the black level and the white level. The carrier has ample time to dwell for a short time at each of these steady state levels. By definition, the p-p carrier frequency displacement as measured between peak of sync and white level is 2 x ΔF. This display is the result of deviating the carrier with a signal which is 10 dB below the 1 Volt level, therefore the deviation is also 10 dB or 3.1623 times lower than what would result if a 1 Volt p-p signal was applied at a frequency of 761.6 kHz. Therefore, the measured p-p carrier frequency displacement must be divided by two and this result multiplied by 3.1623 to obtain the deviation ΔF.

EXAMPLE: The frequency span of the spectrum analyzer used in Figure 7 is 12.5 MHz, hence each major vertical devision is 1.25 MHz.The p-p carrier displacement equals 7 x 1.25 MHz or 8.75 MHz.
8.75 MHz divided by two is 4.375 MHz and this value, multiplied by 3.1623 (10 dB) equals the actual deviation.
In this case: 13.835 MHz.

Figure 8 shows the frequency spectra of the 5 STEP STAIRCASE SIGNAL. The frequency span of the spectrum analyzer was adjusted so that each step coincides with the vertical graticule of the spectrum analyzer. This display allows the verification of the low frequency linearity of the FM Modulator.

Figure 9 shows a graph of carrier frequency displacement versus frequency deviation for deviations between 8 MHz and 14 MHz.

EXAMPLE: The p-p carrier frequency displacement between peak of sync and white level for 13.8 MHz deviation is equal to 8.73 MHz. For 9.1 MHz deviation it is equal to 5.76 MHz. This can also be read from Table 3.

The spectrum analyzer used to generate the spectra of Figures 8 and 9, was an HP Model 8566 with continuously variable frequency span. This allows adjustment of the span so that the vertical graticules coincide with peak of sync and black level and also with each of the 5 steps of the stairstep signal.

ACTUAL DEVIATION VS. SPECTRUM ANALYZER (SP/AN) DEVIATION. TABLE 3

DEV MHz	SP/AN DEV MHz	DEV MHz	SP/AN DEV MHz	DEV MHz	SP/AN DEV MHz	DEV MHz	SP/AN DEV MHz	DEV MHz	SP/AN DEV MHz
4	2.53	6	3.79	8	5.06	10	6.32	12	7.59
4.1	2.59	6.1	3.86	8.1	5.12	10.1	6.39	12.1	7.65
4.2	2.66	6.2	3.92	8.2	5.19	10.2	6.45	12.2	7.72
4.3	2.72	6.3	3.98	8.3	5.25	10.3	6.51	12.3	7.78
4.4	2.78	6.4	4.05	8.4	5.31	10.4	6.58	12.4	7.84
4.5	2.85	6.5	4.11	8.5	5.38	10.5	6.64	12.5	7.91
4.6	2.91	6.6	4.17	8.6	5.44	10.6	6.70	12.6	7.97
4.7	2.97	6.7	4.24	8.7	5.50	10.7	6.77	12.7	8.03
4.8	3.04	6.8	4.30	8.8	5.57	10.8	6.83	12.8	8.10
4.9	3.10	6.9	4.36	8.9	5.63	10.9	6.89	12.9	8.16
5	3.16	7	4.43	9	5.69	11	6.96	13	8.22
5.1	3.23	7.1	4.49	9.1	5.76	11.1	7.02	13.1	8.29
5.2	3.29	7.2	4.55	9.2	5.82	11.2	7.08	13.2	8.35
5.3	3.35	7.3	4.62	9.3	5.88	11.3	7.15	13.3	8.41
5.4	3.42	7.4	4.68	9.4	5.95	11.4	7.21	13.4	8.47
5.5	3.48	7.5	4.74	9.5	6.01	11.5	7.27	13.5	8.54
5.6	3.54	7.6	4.81	9.6	6.07	11.6	7.34	13.6	8.60
5.7	3.60	7.7	4.87	9.7	6.13	11.7	7.40	13.7	8.66
5.8	3.67	7.8	4.93	9.8	6.20	11.8	7.46	13.8	8.73
5.9	3.73	7.9	5.00	9.9	6.26	11.9	7.53	13.9	8.79

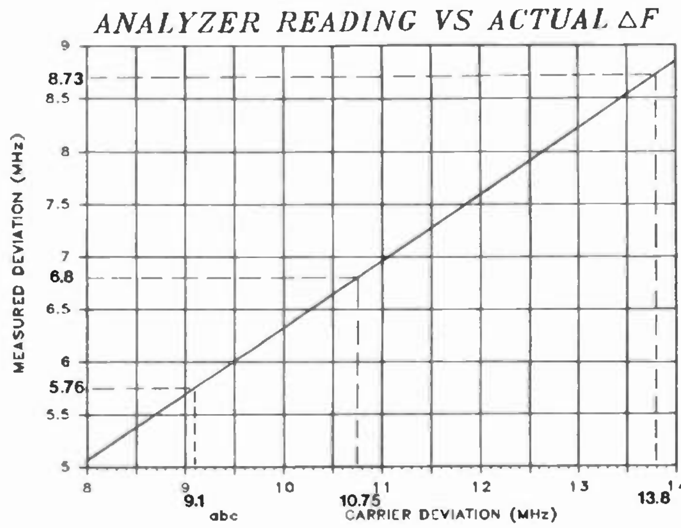


FIGURE 9

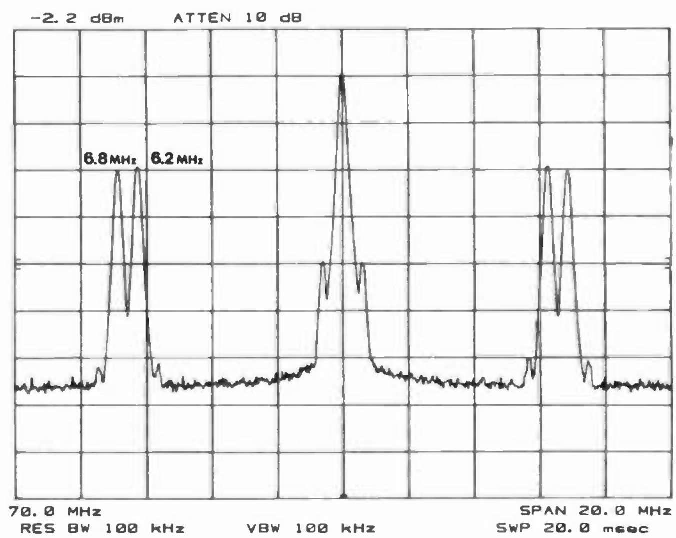


FIGURE 11

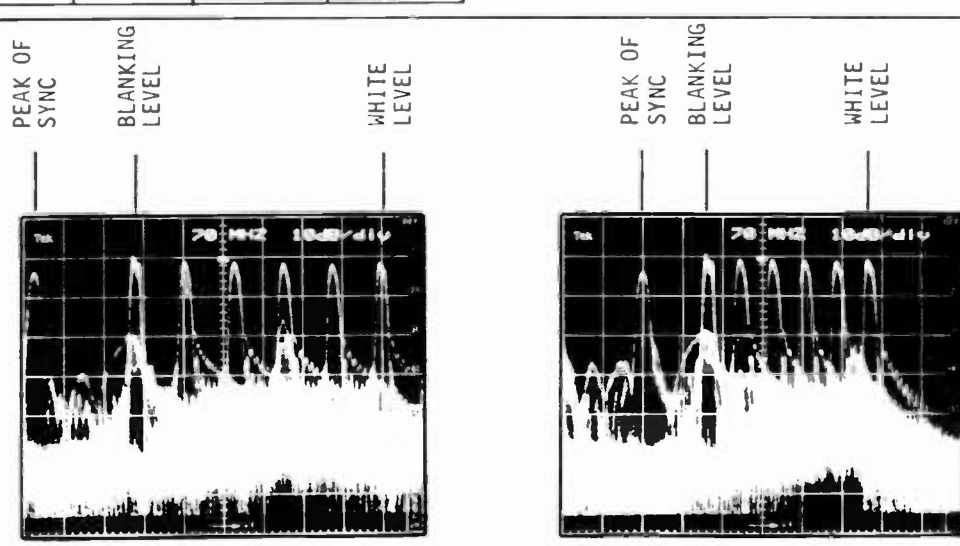


FIGURE 10a

13.8 MHz DEVIATION

FIGURE 10b

9.1 MHz DEVIATION

These same measurements can be made with the inexpensive Tektronix Spectrum Monitor Model 1705. This unit has a fixed frequency span of 1 MHz per division. This is not as convenient but as Figures 10a and 10b show, it is nevertheless possible to adjust the deviation for 13.8 MHz and 9.1 MHz without great difficulty.

AURAL MODULATOR SETTINGS: Two adjustments need to be made, the subcarrier level, which is the deviation of the main carrier by the subcarrier, and the deviation of the subcarrier by the audio signal.

SUBCARRIER LEVEL: The recommended setting of the subcarrier levels is such that the main carrier deviation is equal to 1.38 MHz for each of the subcarriers.
As is shown in Figure 11 and derived in Appendix I, the level of the subcarriers as observed on a spectrum analyzer are 19.125 dB and 19.918 dB below the unmodulated carrier for a carrier deviation of 1.38 MHz. In practice, the subcarrier levels are adjusted for 19 dB and 20 dB below the modulated carrier.

SUBCARRIER DEVIATION BY THE AUDIO SIGNAL: When modulating the subcarrier with a tone of 400 Hz, the deviation is set for 185 kHz peak. This can be verified using the carrier Bessel Null method.

EXAMPLE: What is the level of the 400 Hz test tone for first carrier null so that the subcarrier deviation equals 185 kHz for 18 dBm (600 Ohm) peak program level?

m = 2.4048 for first Bessel Null.
ΔF = 2.4048 x 0.4 = 0.962 kHz for a 400 Hz test tone signal.
18 dBm equals 6.153 Volts RMS.

185 kHz deviation requires a test tone level of 6.153 Volts RMS.
1 kHz deviation requires a test tone level of: 6.153/185 = 33.26 mV RMS and 0.962 kHz deviation requires a (6.153 x 0.962)/185 = 32 mV RMS or -45.68 dBm test tone level.

Table 4 gives the required test tone levels for 75 kHz and 185 kHz peak deviation for peak program levels between 0 dBm and +24 dBm in 1 dB steps, using a 400 Hz and 1 kHz test tone.

EXAMPLE: What is the level of the 1 kHz test tone for first carrier Bessel Null so that the peak deviation of the sub-carrier is 185 kHz for an input level of 8 dBm?

From Table 4, the level of the 1 kHz test tone for first Bessel Null is -29.72 dBm or 25.29 mV RMS.
Figure 12 can be used for a quick but less accurate check.

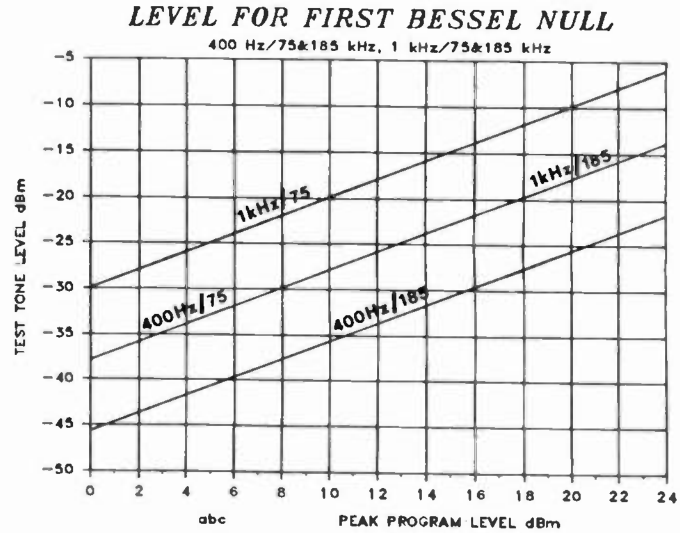


FIGURE 12

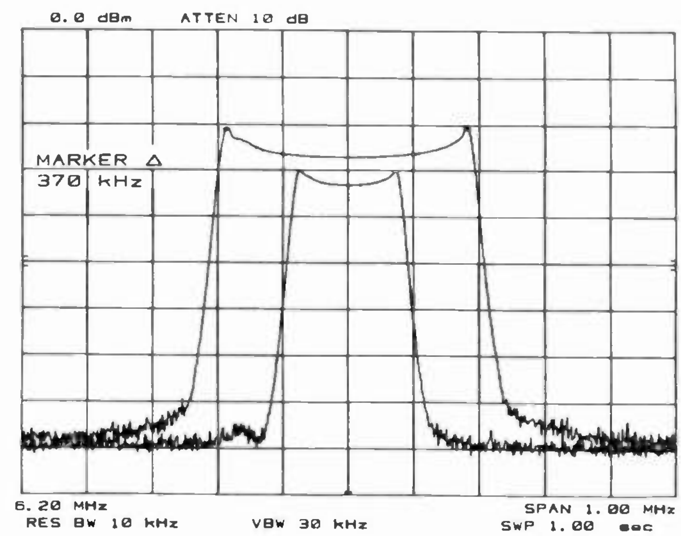


FIGURE 13

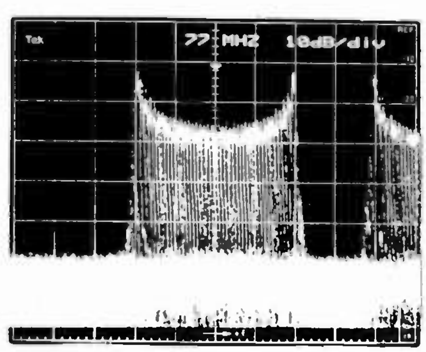


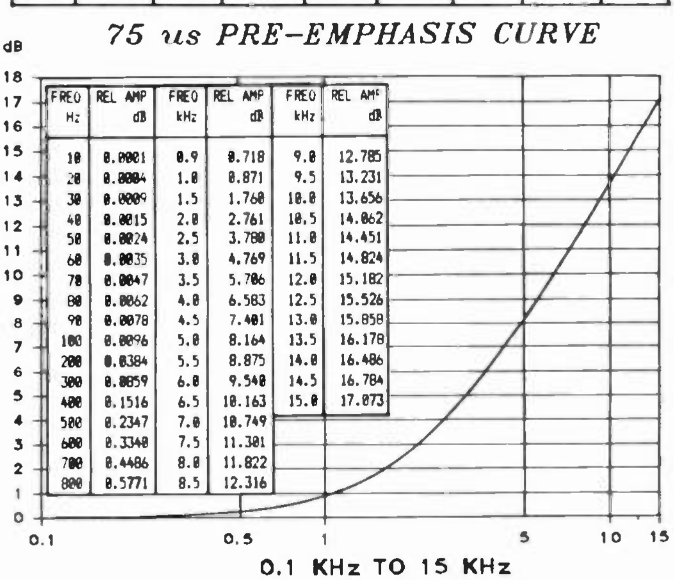
FIGURE 14

SIMPLIFIED METHOD: Using the spectrum analyzer the p-p deviation can be verified very quickly. Figure 13 shows the 6.2 MHz subcarrier modulated with a 400 Hz test tone to 150 kHz and 370 kHz p-p.

TEST TONE LEVEL FOR FIRST BESSEL NULL

TABLE 4

PRGR. LEVEL	FIRST BESSEL NULL FOR 75 kHz DEV.				FIRST BESSEL NULL FOR 185 kHz DEV.			
	400 Hz		1 kHz		400 Hz		1 kHz	
dBm	mV RMS	dBm	mV RMS	dBm	mV RMS	dBm	mV RMS	dBm
0	9.93	-37.8	24.84	-29.9	4.03	-45.7	10.07	-37.7
1	11.15	-36.8	27.87	-28.9	4.52	-44.7	11.30	-36.7
2	12.51	-35.8	31.27	-27.9	5.07	-43.7	12.68	-35.7
3	14.03	-34.8	35.08	-26.9	5.69	-42.7	14.22	-34.7
4	15.75	-33.8	39.36	-25.9	6.38	-41.7	15.96	-33.7
5	17.67	-32.8	44.17	-24.9	7.16	-40.7	17.91	-32.7
6	19.82	-31.8	49.56	-23.9	8.04	-39.7	20.09	-31.7
7	22.24	-30.8	55.60	-22.9	9.02	-38.7	22.54	-30.7
8	24.95	-29.8	62.39	-21.9	10.12	-37.7	25.29	-29.7
9	28.00	-28.8	70.00	-20.9	11.35	-36.7	28.38	-28.7
10	31.42	-27.8	78.54	-19.9	12.74	-35.7	31.84	-27.7
11	35.25	-26.8	88.12	-18.9	14.29	-34.7	35.73	-26.7
12	39.55	-25.8	98.88	-17.9	16.03	-33.7	40.09	-25.7
13	44.38	-24.8	110.94	-16.9	17.99	-32.7	44.98	-24.7
14	49.79	-23.8	124.48	-15.9	20.19	-31.7	50.46	-23.7
15	55.87	-22.8	139.67	-14.9	22.65	-30.7	56.62	-22.7
16	62.68	-21.8	156.71	-13.9	25.41	-29.7	63.53	-21.7
17	70.33	-20.8	175.83	-12.9	28.51	-28.7	71.28	-20.7
18	78.91	-19.8	197.28	-11.9	31.99	-27.7	79.98	-19.7
19	88.54	-18.8	221.36	-10.9	35.90	-26.7	89.74	-18.7
20	99.35	-17.8	248.37	-9.9	40.28	-25.7	100.69	-17.7
21	111.47	-16.8	278.67	-8.9	45.19	-24.7	112.98	-16.7
22	125.07	-15.8	312.68	-7.9	50.70	-23.7	126.76	-15.7
23	140.33	-14.8	350.83	-6.9	56.89	-22.7	142.23	-14.7
24	157.45	-13.8	393.63	-5.9	63.83	-21.7	159.58	-13.7



FREQ Hz	REL AMP dB	FREQ kHz	REL AMP dB	FREQ kHz	REL AMP dB
10	0.0001	0.9	0.718	9.0	12.785
20	0.0004	1.0	0.871	9.5	13.231
30	0.0009	1.5	1.760	10.0	13.656
40	0.0015	2.0	2.761	10.5	14.062
50	0.0024	2.5	3.780	11.0	14.451
60	0.0035	3.0	4.769	11.5	14.824
70	0.0047	3.5	5.706	12.0	15.182
80	0.0062	4.0	6.583	12.5	15.526
90	0.0078	4.5	7.401	13.0	15.858
100	0.0096	5.0	8.164	13.5	16.178
200	0.0384	5.5	8.875	14.0	16.486
300	0.0859	6.0	9.540	14.5	16.784
400	0.1516	6.5	10.163	15.0	17.073
500	0.2347	7.0	10.749		
600	0.3340	7.5	11.301		
700	0.4486	8.0	11.822		
800	0.5771	8.5	12.316		

FIGURE 15

Note again that the spectrum analyzer reads p-p deviation, not peak deviation. Figure 14 shows the 6.2 MHz subcarrier modulated with a 1 kHz tone as seen on the Tektronix 1705 spectrum monitor. The p-p deviation is approximately 400 kHz. With a 400 Hz tone, the p-p deviation is 370 kHz. With a 1 kHz tone, the deviation is slightly greater because of pre-emphasis. From Figure 15, the difference in gain between 400 Hz and 1 kHz is 0.72 dB or 1.086 times. Hence the deviation will be 370 x 1.086 = 402 kHz.

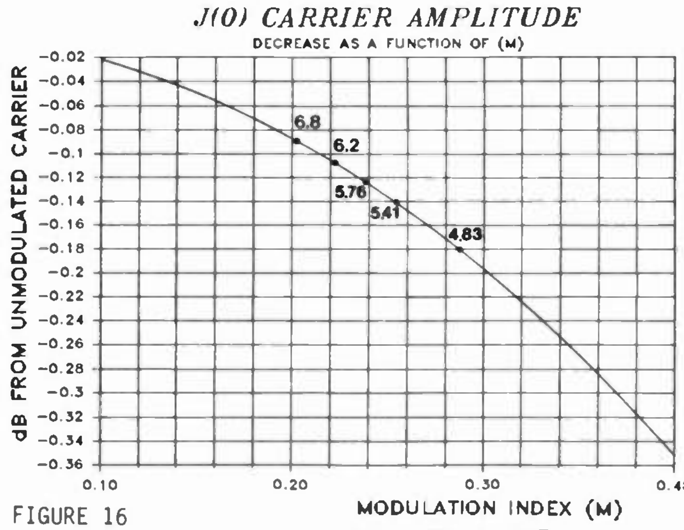


FIGURE 16

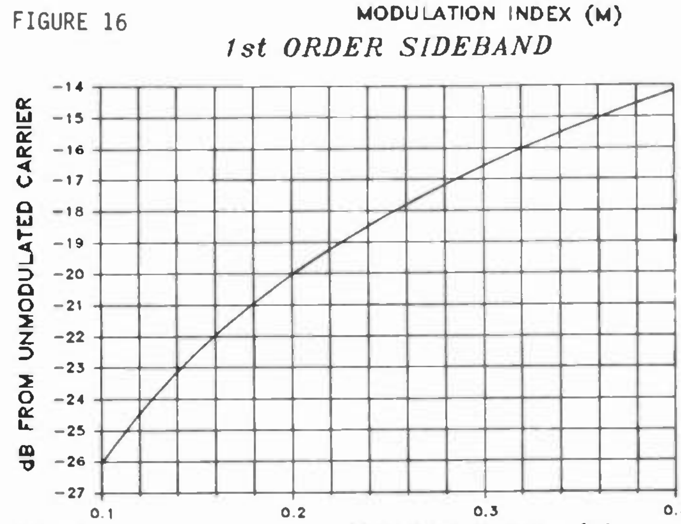


FIGURE 17

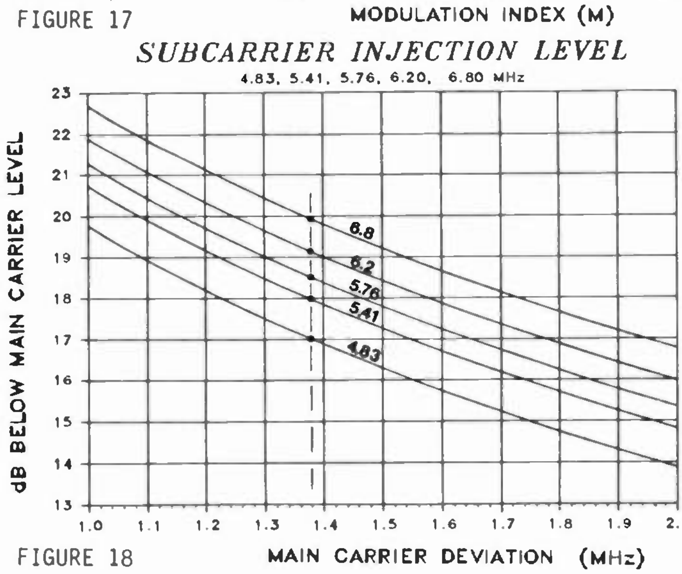


FIGURE 18

Figure 16 shows the reduction of the main carrier level and Figure 17 shows the level of the subcarrier below the level of the main carrier for any given value of (m) between 0.1 and 0.4. Figure 18 gives the level for the most commonly used subcarrier frequencies for a carrier deviation between 1 MHz and 2 MHz.

CONCLUSION: A method has been given which permits for quick and accurate verification of the carrier frequency deviation of FM type modulators using standard video test signals and a spectrum analyzer. The Tables and Charts provided should make it easy to determine the levels of test tones for any frequency deviation of the main carrier or subcarrier.

APPENDIX I.

FREQUENCY MODULATION: Frequency modulation is produced by varying the instantaneous frequency of the carrier by an amount that is proportional to the amplitude of the information to be transmitted, and at a rate given by the frequency of the modulating signal. The carrier amplitude remains constant in the process.

FREQUENCY DEVIATION: The maximum swing of the frequency from its mean value is called "Frequency Deviation" and is denoted by ΔF.

MODULATION INDEX: The ratio of the frequency deviation F, and the frequency of the modulating signal Fm, is called modulation index (m).

$$m = \frac{\Delta F}{Fm} \qquad (1)$$

m = Modulation Index.
ΔF = Frequency Deviation.
Fm = Frequency of the Modulating Signal.

SPECTRUM OF A FREQUENCY-MODULATED SIGNAL: The process of frequency modulation is inherently more complicated than that of AM and the mathematical messiness of FM does not lend itself to easy analysis. Theoretically, the information is spread about the RF carrier in an infinite number of sidebands whose amplitudes and phases are determined by the coefficients of the Bessel Functions. In practice, truncation of the insignificant sideband terms is necessary in order to make the system workable.

Figure 20 shows the spectrum of an FM modulated signal for m = 1. The magnitude of the carrier is no longer unity but is reduced by 2.32 dB. The first order sidebands J1 are 7.13 dB below the unmodulated carrier, the second order sidebands, J2 are 18.79 dB, and the third order sidebands, J3 are 34.17 dB below the unmodulated carrier. The sidebands are spaced from the carrier and each other by a frequency difference equal to the modulating frequency, and they are symmetrical in amplitude.

The average power in a frequency modulated wave is independent of the modulating signal. Increasing power in the sidebands is accompanied by a corresponding decrease in the power of the carrier.

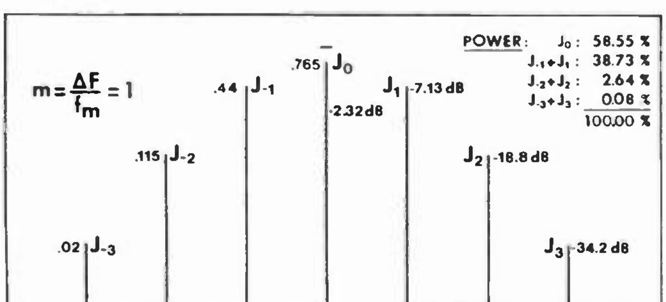


FIGURE 20 FM SPECTRUM

BESSEL FUNCTIONS: The general expression of the Bessel Functions of argument (m) is given by:

$$J_n(m) = \left(\frac{m}{2}\right)^n \sum_{p=0}^{\infty} \frac{(-1)^p \left(\frac{m}{2}\right)^{2p}}{p!\,(p+n)!} \qquad (2)$$

In which m = Modulation Index.
n = nth order Bessel Function.

From this expression, the magnitude of the carrier J0 and all significant sidebands, J1 through Jn, can be calculated for any value of the modulation index (m).

Figure 19 shows the relationship between the carrier amplitude and the amplitude of the first three sidebands as a function of the modulation index between 0 and 5.

Note that J0, the carrier component and Jn, the various sidebands go to zero amplitude at specific values of (m). The first carrier null J0 = 0 occurs for m = 2.4048, and for m = 3.8317 the first order sideband J1 goes to zero amplitude.

EXAMPLE: What is the magnitude of J0, J1, J2 etc. when modulating the visual carrier with the aural subcarrier for ΔF = 1.38 MHz and Fm = 6.2 MHz and 6.8 MHz respectively?

$$m = \frac{\Delta F}{Fm} = \frac{1.38}{6.20} \qquad m = \frac{\Delta F}{Fm} = \frac{1.38}{6.80} \qquad (3)$$

m = 0.22258 m = 0.20294

TABLE 5

	6.20 MHz	6.80 MHz
J0	- 0.1079 dB	- 0.0897 dB
J1	-19.1247 dB	-19.9180 dB
J2	-44.1982 dB	-45.7968 dB

These values cannot be read from the graphs of Figure 19, but the expanded graphs of Figures 16 and 17 offer adequate detail to determine the main carrier and subcarrier levels for practical purposes. J0 and J1 are expressed as functions of the unmodulated carrier which is assumed to be equal to 0 dB.

BESSEL FUNCTION CURVES

J0, J1, J2, J3.

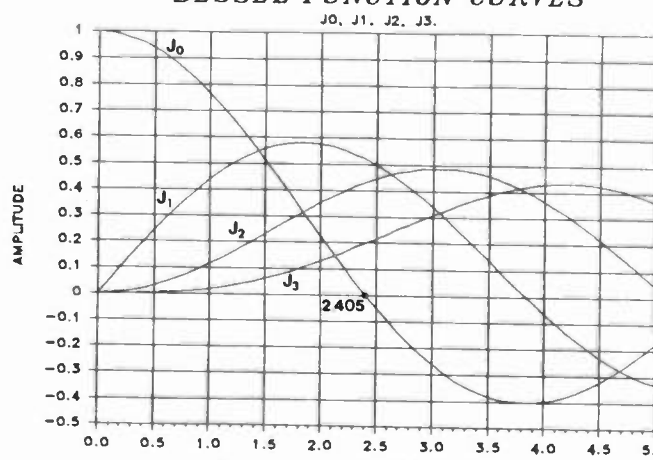


FIGURE 19

THE HOW AND WHY OF OPTICAL FIBER TRANSMISSION SYSTEMS

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Abstract

The fundamentals of optical fiber transmission, components, and systems are reviewed in this paper. Advantages of fiber systems are discussed, representative recent applications cited and possible additional applications suggested.

Introduction

During the past 20 years, optical fiber devices have progressed from research laboratory curiosities to a viable engineering technology. Practical optical fiber components and systems have been developed for applications in both data communications and control system sensing in a wide range of environments. Due to the recent development and demonstration of such systems specifically for local communication and instrumentation, their consideration by broadcast engineers in future installations is suggested.

The objective of this paper is to suggest such future applications to the reader by briefly reviewing optical fiber system device principles, fiber advantages, and typical current applications.

Background

Electromagnetic wave propagation along metallic waveguiding structures was demonstrated more than half a century ago and forms the basis of many communication systems operating in the radio frequency and microwave regions of the electromagnetic spectrum. The practicality of the similar propagation of light through dielectric waveguides was demonstrated during the mid-1960s by K. C. Kao, who first suggested the implementation of communication systems connected by clad dielectric optical fiber waveguides having reasonably low attenuation [1]. During the 20 years since Kao's first observations, numerous scientists and engineers have contributed to the development of practical low loss and large bandwidth optical fibers, fiber system components, active optical system devices, and total instrumentation systems which optimize the use of optical fiber techniques.

Fiber and Fiber Cable Construction

Optical fibers are constructed in the form of concentric cylindrical layers of dielectric materials as shown in Figure 1. The innermost of these layers, the core and cladding, have a combined diameter which is slightly more than 100 microns, typically or about the same size as a human hair. Although these layers may be either glass or plastic, the combined advantages of very low attenuation and thermal resistance make glass fibers most attractive for many applications. Surrounding the core and cladding are a series of additional concentric layers of protective material. The innermost of these are typically 100 micron-thick polymer buffer layers applied to the core and cladding during manufacturing to make the fiber pliable and to protect it during subsequent processing. Several such resulting buffered optical fibers are then packaged together in a cable assembly designed to protect the individual fibers from external stresses. Although the resulting cable which contains optical fiber components looks much like a cable containing conventional copper wire electrical conductors, its operation, methods for its handling, and its potential applications are very much different.

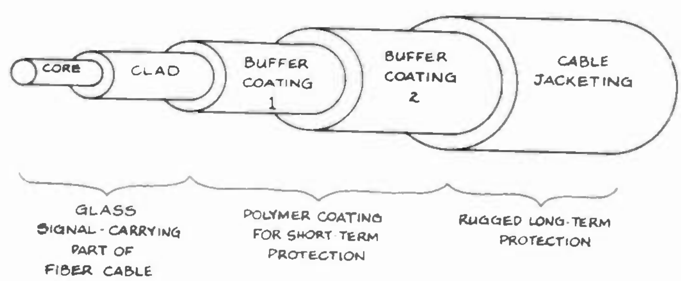

Figure 1. Basic optical fiber geometry.

Light Propagation in Optical Fibers

Light rays can travel down the core of an individual fiber much as a bullet shot into the open end of a pipe may travel down the length of the pipe. The sides of the pipe confine the path of the bullet similar to the way the cladding confines the path of the light. A number of different bullet trajectories are possible (along the axis of the pipe, simple back and forth reflections, corkscrew); similar light ray trajectories can be described [3].

Rays having different trajectories are termed different modes, and fibers may be classified according to the types of modes that they support. A sketch of the cores and claddings of fibers compared in this way is shown in Figure 2. The top two sketches in this figure represent cross sections of multimode fibers--fibers which contain many modes. The bottom sketch is a similar cross section of a single mode fiber--a fiber which contains only a single mode. The most important practical difference between these multimode and single mode fibers is the relative sizes of their cores. Multimode fiber core area is about 25 times larger than single mode fiber core area; this makes multimode fiber relatively easier to align and connect. Figure 2 also indicates that multimode graded index fiber may have a core diameter of 50 or 62.5 microns and 9 micron core diameter fiber is single mode at a 1300 nanometer operating wavelength.

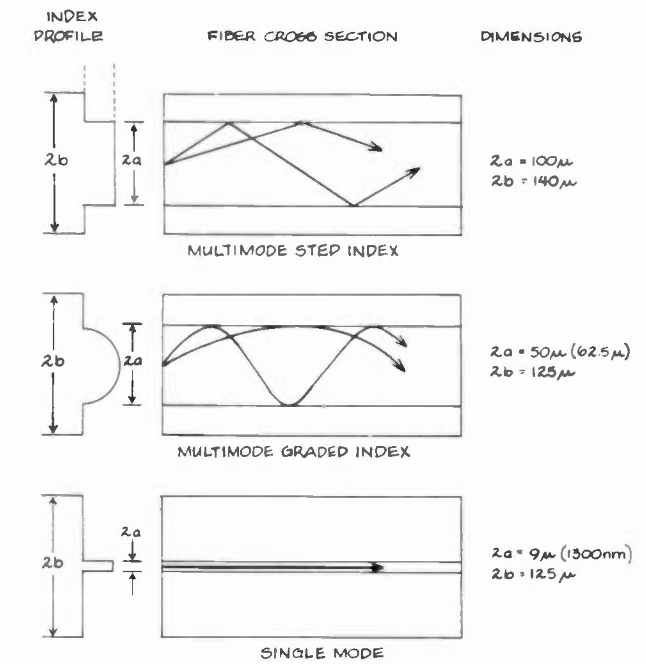

Figure 2. Sketches of geometries, index of refraction profiles, and light ray paths for step index and graded index multimode fibers, and for single mode fibers.

Practical Limitations on Ideal Performance

Practical optical fibers exhibit both attenuation and signal distortion because they are manufactured from real materials and because they operate with real input and output devices [4]. Optical attenuation in fibers is caused by the intrinsic absorption of the glass in the core and cladding, by the presence of small traces of impurity materials, and by the extrinsic bending of the fiber, especially in short period "microbending" geometries. This attenuation varies as a function of the wavelength of light transmitted through the fiber as shown in the sketch in Figure 3. This figure indicates that the attenuation of the fiber itself is on the order of 1 dB/km.

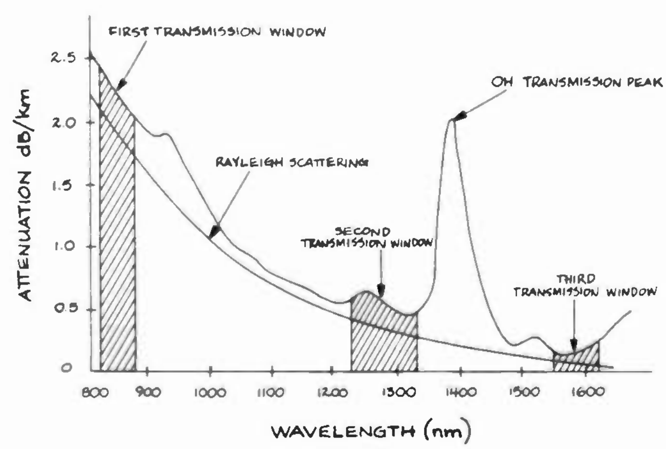

Figure 3. Sketch of optical fiber attenuation versus light wavelength.

Signal distortion in optical fibers, although small compared with that in copper wire cables, occurs due to several reasons. First, different light rays which travel along different trajectories such as those shown in Figure 2, introduce multipath distortion similar to that in radio frequency transmission systems. Second, different wavelengths of light travel with different speeds in glass. Thus, even slightly different optical wavelengths generated by the best currently available optical sources arrive at the destination end of the fiber at slightly different times. This distortion is greater, of course, for longer fiber lengths. Finally, the structure of the optical fiber itself produces a small additional amount of distortion. The total result of these three effects is to cause the distortion of transmitted digital or analog signals. This distortion may be expressed in terms of the bandwidth limitation of the fiber. Since signal distortion increases with the length of the fiber, fiber bandwidth is specified in terms of a product of bandwidth times distance.

Advantages of Optical Fiber Systems

Optical fiber systems are advantageous for applications in broadcasting for several reasons. This section reviews the major advantages.

1. Large bandwidth potential. Optical fiber communication systems may be thought of as radio frequency communication systems in which the carrier frequency, the frequency of the light, is on the order of 100 THz, approximately one million times higher than carrier frequencies of commercial television broadcasting. In principle, a modulation bandwidth increase by the same one million factor is possible. Currently available optical fiber systems have bandwidths about one thousand times greater than copper cable-connected systems.

   Such large bandwidth capability may be beyond current communication system needs or beyond the capability of existing data multiplexing and demultiplexing hardware. The additional communication capacity not used immediately is available to accommodate future system expansions.

2. Dielectric nature of optical fibers. Optical fiber waveguides are made of dielectric materials. The inherent properties of dielectrics are the bases of four additional properties of optical fiber systems.

   First, glass or plastic optical fibers are not electrical conductors so they do not act as antennas and pick up stray electromagnetic fields the way that copper cables do. Optical fibers and fiber systems are thus immune to noise effects induced by radio frequency interference (RFI), electromagnetic interference (EMI), and electromagnetic pulses (EMP).

   Second, optical fiber signals are light intensities measured with respect to darkness. Since darkness is the same everywhere, the "ground" connection required in electrical systems is not needed in optical systems. This results in total electrical isolation between ends of a fiber system and eliminates the possibility of ground loops.

   Third, since electrons are not transmitted in a fiber, the fiber cannot act as a radiator of electromagnetic fields and cause crosstalk between fibers in the same fiber cable or cable assembly.

   Finally, light in fibers cannot create electrical sparks, and individual fibers do not cause short circuits between transmission channels if they touch each other. This eliminates the possibility of fire hazard and allows fibers to be run safely through hazardous areas.

3. Small and lightweight components. Since a single optical fiber has a bandwidth equivalent to hundreds of twisted pairs of copper wire, a large twisted pair cable may be replaced by a single fiber cable. This results in a cross sectional space savings and a weight savings on the order of 90 percent or more. Such savings mean fewer and smaller cable trays and a resulting reduction in the size and cost of the physical structure needed to support the trays.

4. Low fiber attenuation. In addition to large bandwidth potential, fibers allow high frequency signals to propagate with very low attenuation with respect to that for conventional copper cables. For example, at 1 GHz the attenuation of ultra-low loss RG 19/U copper cable is about 100 dB/km; optical fiber cable attenuation at 1 GHz is about 5 dB/km. Such low loss permits extended spacings between repeaters in long, high data rate communication systems. In relatively short-distance systems such as those which could be implemented between locations in a broadcast facility.

5. Point-to-point transmission. Optical fibers permit point-to-point signal transmission the way copper cables do. Thus, fiber systems may use portions of the frequency spectrum which are restricted for radio frequency broadcasting. Additionally, optical fibers are difficult to tap and special designs may increase this difficulty, thus making such point-to-point links inherently secure.

6. Cost. Optical fiber components and systems have been developed rapidly during the past 10 years. Costs of optical fiber and system devices have decreased significantly during that time, making optical fiber systems comparable in cost with copper wire-connected systems in many cases. Although absolute installation costs of fiber systems may be higher in some cases, the above advantages suggest additional long term performance/cost tradeoffs that should be considered.

Simple Optical Fiber Devices

A simple optical fiber system consists of an optical source, a length of optical fiber cable, and an optical detector. As shown in Figure 4, such a system may also include fiber connectors, splices, couplers, and input and output electronic signal conditioners. During the past 10 years, the large demand for such system components by the long distance terrestrial communications market spurred competitive commercial component development. Similar components for use in the broadcasting industry have for the most part been modifications of such telco system devices. These devices are described in detail in references such as [5].

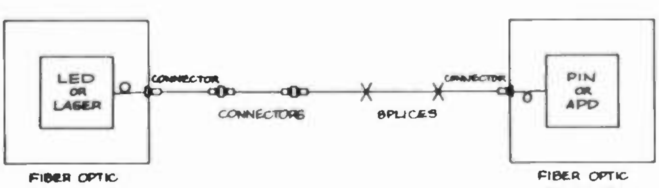

Figure 4. Typical simple optical fiber system design.

Optical Fiber Systems and Examples

Optical fiber components may be combined to create a wide variety of systems of potential use in broadcasting. Fiber system applications may be grouped into the areas of communications, sensing, illumination, and observation. Fiber communication systems use either analog or digital transmission techniques to transfer voice or data signals from sources to detectors. Such systems may be single point-to-point links or distributed multiple-source, multiple-receiver systems, requiring appropriate data formatting to achieve optimum system utilization.

Fiber sensor systems use the property that the light travelling in a fiber may be modulated in response to external effects if the fiber is suitably coupled to the external environment. Fiber sensors are usually classified according to which measureable property of the transmitted light is modulated. Hence, intensity, phase, polarization, wavelength, time domain, and modal domain sensors are available. In broadcasting facilities, such sensors may be used to monitor temperature, electric current, fire, or intruders.

Fiber systems may also be used for illumination by transmitting light from the location of a light source to the location where light is needed. Such systems may be advantageous due to convenience or safety reasons.

Finally, optical fiber systems may be used for observation. For example, a single fiber may be used to transmit spectral information concerning the nature of a fire to a fire control station; a coherent fiber bundle may be used to view obscure areas.

Examples of the current use of fiber systems in broadcasting include 1) the communication of digitized information from studio equipment, film chains, tape machines and special effects computers to control room electronics, 2) the similar communication of digital signals to remote transmitter facilities, 3) in-house local area networking of support computers, 4) limited area broadband communication via fiber link directly to homes, and 5) in surveillence systems.

Specific examples include those in cable broadcasting, local stations use, and special project applications. Examples in the cable broadcasting area are 1) the Southern Bell Hunters Creek system which is the first to provide digital programming to homes, these near Orlando, Florida, 2) the 38-mile Heritage Cablevision system in Dallas, and 3) planned or expanded systems by Ohio Bell and C & P Telephone in Washington, DC. Local station applications include WKFT-TV's 1 km studio to transmitter link, and WNDU-TV's EJ, earth station, and studio links. Special project uses of fiber in broadcasting are the U.S. Senate's new fiber video link, KTXH's new 72-channel Astrodome video feed, and ABC and NBC "Statute of Liberty" feeds which used tactical fiber cable.

Acknowledgement

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BROADCAST QUALITY TELEVISION CUSTOMER CONTROLLED 45 MB/S (DS3) DIGITAL NETWORK

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ABSTRACT

Today, network television broadcasters use satellites almost exclusively to distribute/collect program material to/from affiliated local TV stations. In the future, a terrestrial alternative or supplement, using DS3-rate digital channels on fiber optic and microwave transmission systems, may be available. Bellcore has issued proposed generic requirements to equipment suppliers and has proposed a trial of a nationwide, customer-controllable, tree-like network to act as a focal point for this purpose. The trial is gaining momentum. A paper presented March 31 at the 1987 NAB Conference described this activity in detail. This paper describes the considerable progress that has been made since that time.

BACKGROUND

Channels

DS3-rate (45 Mb/s) channels were chosen for this service capability proposal because it is becoming increasingly possible to obtain DS3-rate channels from one place in the country to any other place from the various local, regional and national common carriers and to request that the carriers connect the various segments together. Furthermore, a spare "protection" DS3 channel is automatically substituted for a "working" DS3 channel in the event of a channel failure. Also, an intricate web of digital facilities is being installed throughout the US that would provide alternate physical route protection if a route failure occurred.

Coders

DS3 rate video coders have been shown to be capable of providing broadcast television quality. ABC has used and continues to use 45 Mb/s coders for live television news feeds from its Washington, D.C. news bureau to New York on digital fiber optic facilities. ABC has shown that it is possible to use the feed from Washington, D.C., after it is decoded in New York, to chroma-key another signal into the background, such as when Ted Koppel at the Washington news bureau interviews someone in another city and the two pictures are combined in New York before being put on a satellite for distribution to affiliates.

However, although there are several suppliers of 45 Mb/s coders, there are no standards and therefore no two suppliers' products are compatible with each other. This is an impediment to the building and changing and interconnecting of large multipoint networks.

Networking

A television network broadcaster uses several (up to about 16) satellite transponders to transmit television signals for several reasons: 1) a back-up in case of failure of the primary signal; 2) to serve different time zones; 3) to broadcast several sporting events of regional interest; 4) to broadcast TV commercials of regional or even individual city interest; and 5) to broadcast promotional material or news segments for later use by affiliates. In addition, television broadcasters also have a need for collecting program material from network news bureaus and affiliates either for live use directly or, after further processing, such as when chroma-keying two signals together, or for recording and later broadcasting. Furthermore, network broadcasters have become accustomed to controlling or managing their own satellite network configurations autonomously. To be useful for the broadcasters, a terrestrial digital network must be equally flexible and easy to control.

PROGRESS

Coder Standards

Progress is being made in the area of standards for broadcast quality DS3-rate digital video coders so that products of different manufacturers will be end-to-end compatible. The Exchange Carrier Standards Association sponsors the T1 Committee of the American National Standards Institute. Two subcommittees, T1Q1.5 and T1Y1.1, are concerned with the analog performance and a standard algorithm for DS3-rate coders. These committees include representatives from local and interexchange carriers, equipment suppliers and broadcasters as well as Bellcore.

As of this writing, four proposals for a coding algorithm have been made to the T1Y1.1 group: from Bellcore, Bell Northern Research, NEC and Telettra. Only the Bellcore and Telettra proposals are complete; the others are partial proposals. An "experts group" has had several meetings to review these proposals. The Telettra proposal stands apart from the other three proposals because it is based upon a new technology called discrete cosine transforms (DCT), instead of differential pulse code modulation (PCM). Telettra, an Italian company, has offered to demonstrate its proposed algorithm in the U.S. in May to the standards body and to the U.S. and Canadian broadcasters. The Telettra proposal is attractive because it is actually a component coder although it is claimed that it can be used for composite signals without creating NTSC artifacts. But more importantly, if a network broadcaster wanted to send component video, or receive component video, from an affiliate or news bureau, as CBS has suggested, the same coder could be used and this could be accomplished on one DS3 channel instead of two or three as previously thought.

This looks so promising that a decision on the coding algorithm to be used in the trial has been deferred until after that demonstration. By that time, the "experts group" and the other standards members plus all the broadcasters will have more information available to consider. In any event, Bellcore will make its recommendation for the common algorithm to be used in the trial in May, weighing all the evidence that is available at that time. Suppliers can of course decide for themselves whether to begin exploratory development on a differential PCM or a DCT algorithm or both and need not wait until the May date to begin.

The Bellcore proposal is more than just a coding algorithm. It is also a complete plan for using other bits in the DS3 channel for television audio (multichannel television sound), for an order wire for talking purposes, and for creating an embedded channel for real time control and surveillance of the multipoint network elements including multipoint units located at the branch points of the tree-like network and codecs located at the end points of the network. However, the network plan is independent of the actual video coding algorithm that is in use. Therefore, the same networking plan can be accomodated using either a differential PCM algorithm or a DCT algorithm.

Networking

Considerable progress has been made in the area of networking. The basic 2-way, tree-like network architecture described at the 1987 NAB Conference has been preserved. The technical details (proposed generic requirements) needed for equipment suppliers to be able to build compatible multipoint units for creating the trees, and compatible video and audio terminals to permit remote surveillance and control of the network elements by a network broadcaster, have been documented, published, and discussed in detail with interested suppliers at a Technology Requirements Industry Forum presented by Bellcore in September 1987.

Furthermore, Bellcore has gained a better understanding of the needs of the network broadcasters and how these needs can be met by 2-way, tree-like networks. Just as today a broadcaster uses numerous full-time or part-time satellites to create separate networks for different time zones and for distributing programs targeted to different cities for regional sporting events or regional commercials, even targeted to a single city, numerous full-time or part-time trees could be used instead. Also, satellite networks could back-up terrestrial networks and vice-versa.

Figures 1 through 4 illustrate some of the possibilities for using terrestrial tree networks to serve the television network broadcaster/affiliates needs.

Figure 1 shows how separate independently controllable trees can be used for serving different time zones. With such an arrangement, the entire continental U.S. could be served as either three separate time zones with separate program material, or as a unified whole by feeding the same program material into each tree, if a national event needs to be broadcast live to all affiliates. In addition, back-up trees (not shown) could be provided on diverse routes with different interexchange carriers to help to assure survivability in the event of a route failure. These back-up trees could be used to distribute or collect less essential program material rather than being left idle.

Figure 2 shows how each of the time-zoned trees could be regionalized for sporting events or commercials when the source of the program material emanates from within the region. This is accomplished by commanding the multipoint unit at the entry to the region to loop-back the video and audio bits so that the program material is distributed to the selected affiliates within that region. This can be done temporarily, say during a local football game. The control always remains with the master station.
Figure 3 shows a new concept - a multicast switch. This is a device, that is only in the thinking stage now, that would permit program material to hop from one region to another. It would allow one or more regions to carry another region's program material and it would permit the master station to observe the program material in each region or to insert program material into a region. For example a football game between the Boston Patriots and the Miami Dolphins, emanating in Boston, could be watched in both the Boston region and the Miami region, or in just Miami for that matter. It is not mandatory to have a multicast switch for this purpose. The same thing could be accomplished with additional tree-like networks. For example, the back-up tree could be used to pick-up the Boston game and deliver it to New York and then New York could distribute that to Miami, all in real time via loopback of the video and audio bits at the New York location. Therefore, it is important to recognize that the basic tree-like network provides all the flexibility needed. A multicast switch might enable this to be done more economically but that remains to be seen.

Figure 4 shows even more flexibility obtained by interconnecting multicast switches together. Now a football game in Boston between the Boston Patriots and the San Diego Chargers can be hopped across the country to be observed in both areas. Again, the multicast switch is not mandatory for this purpose; another tree-like network could be used instead just as today another satellite transponder would be required to carry the Boston game over to San Diego.

The point of all this is that any number of basic tree-like networks could be concocted to suit the network broadcaster/affiliates needs for delivering and collecting both on-air and off-air program material between fixed locations. The flexibility is enormous.

Network Control and Surveillance

Progress has also been made in the area of network control and surveillance. The proposed generic requirements will help to assure compatibility of all the various network elements. They contain a complete proposal for an embedded control channel of about 112 kb/s that is contained within and is synchronous with the DS3 bit stream. The tree-like network can be reconfigured either by commanding each addressable multipoint unit one at a time or by sending commands ahead of time to be placed in a program store at the multipoint unit to be acted upon simultaneously in all multipoint units when an execute command is sent. Upstream commands are sent separately from downstream commands; however commands for multiple ports are combined to save time. It is estimated that a 200 point nationwide network could be reconfigured in about 6 seconds if each multipoint unit is commanded one-at-a-time and in only 25 milli-seconds if done via an execute command.

All the necessary commands and responses have been documented and a complete plan is included for detecting and dealing with bit errors so that the network will not be likely to respond erroneously. In addition, a background poll/response process for determining the status of various settings, or the contents of a program store, or the presence of any alarms, either in the multipoint units or the video or audio terminals, has been delineated. In addition, the multipoint unit will check its incoming port on the master station side for signal failure and will autonomously report that to an alternate master to take over control of the surviving network downstream of the failure if the failure is not restored within a reasonable time.

8-City Trial

The proposed trial is gaining considerable momentum with five U.S network broadcasters (ABC, CBS, NBC, Fox and PBS) agreeing to participate one-at-a-time in that order in an 8-city U.S. trial. Four interexchange carriers (Lightnet, Southernnet, Norlight and

Wiltel) have agreed to provide 2-way DS3 channels on fiber optic transmission facilities to connect the eight cities together in a tree-like fashion as shown in Figure 5. Twelve equipment suppliers have agreed to provide hardware for the trial. These include: ABL Engineering; AEG Bayly; Anritsu; Coastcom; Comlux; DSC Communications; NEC; Northern Telecom; RE Instruments; Tau-tron; Telettra; and Telinq. The seven Regional Companies will provide the 2-way DS3 access channels to the network broadcaster/affiliate locations on fiber optic cables. The eight cities are, from east to west: Boston and New York (NYNEX); Washington D.C. (Bell Atlantic); Atlanta (Bell South); Indianapolis (Ameritech); St. Louis (Southwestern Bell); Minneapolis (US West); and Los Angeles (Pacific Bell).

The broadcasters have each agreed to provide a report at the end of the trial concerning the quality of the television signal received on the real 8-city network. The signal can traverse the entire distance from Los Angeles to Boston and back again - a distance of perhaps 8000 miles on four different interexchange carrier's fiber optic transmission facilities. If there are errors in the DS3 bit stream after travelling all this distance, a forward error correction process will correct many if not all of those errors. In addition, a cyclic redundancy check process will determine if there are uncorrected errors and the video and audio decoders will attempt to mitigate the effects of those errors.

The broadcasters have also agreed to provide a report on the network surveillance and control aspects of the trial. The control scheme includes error detection in control commands and responses and ignores erroneous messages. This can be verified by the broadcaster. In addition, the flow of the video and audio payload bits can be controlled by the broadcaster to create a large variety of various combinations of which city can send to which other cities.

Suppose, for example, that New York is the master station for ABC. ABC could send program material from New York to any or all of the other seven cities while simultaneously receiving program material from any one (one-at-a-time) of the other seven cities. Or any city could send to any other city by sending to New York and having New York command the multipoint unit at New York to loop/through the signal back to the selected other city so that New York can view or record the signal while it goes on undisturbed and without being decoded to the other city.

The tree-like network can also be regionalized temporarily in a variety of different ways. Boston and New York can be thought of as a temporary region taking turns sending back and forth to each other. At the same time, Washington/Atlanta, Indianapolis/Minneapolis, and St. Louis/Los Angeles can do the same thing. Or, as another example, Boston/New York/Washington/Atlanta can be treated as a temporary region taking turns sending program material to any or all of the others in that region while at the same time Indianapolis/Minneapolis/St. Louis/Los Angeles do the same.

Canadian Trial

The Canadian Broadcasting Corporation and Bell Canada have agreed to conduct a separate trial between five Canadian cities: Montreal; Ottawa; Toronto; London; and Windsor. While it would be desireable to connect the U.S. and Canada together for a cross-border experiment, the necessary cross-border fiber optic capacity has not been made available as yet for a trial connection.

Trial Schedule

The trial schedule is driven by the ability of suppliers to develop products. It has been agreed that the trial could begin about July 1989. Allowing for three months for each of the five U.S. broadcasters, one-at-a-time,, the trial will last for about fifteen months. The Canadian trial may begin concurrently after about October 1989, when Bell Canada has the necessary fiber optic facilities in place and if the equipment suppliers have enough equipment to support both trials simultaneously.

CONCLUDING REMARKS

For those who are interested in more details, the list of references should be consulted. Any company that has not yet agreed to participate in the trials and may wish to may still be accomodated. However, it is not necessary to participate in the trial to become an equipment provider since all the necessary details (proposed generic requirements) have been published and will be kept up-to-date if the trial reveals that changes are needed.

We have come a long way toward the creation of a new commercial service capability. The outcome of the trials will be a final set of proposed generic requirements supported by reports from each of the participating network broadcasters as to the suitability of the requirements for satisfying their needs regarding signal quality and network surveillance and control.

REFERENCES

The following Technical Advisories (TA's), Technical References (TR's) and Special Report are available from Bellcore by writing to:

Bell Communications Research, Inc.
Document Registrar
Room 2J-125
435 South Street
Morristown, NJ 07960-1961

TA-TSY-000195, Issue 2 "Broadcast Quality Digital Television Terminals"

TA-TSY-000490, Issue 1 "Customer Controllable DS3 Multipoint Unit Requirements"

TA-TSY-000491, Issue 1 "DS3 Multipoint Unit Controller Requirements"

TR-TSY-000431, Issue 1 "15 kHz Digital Audio Terminal for Program or Television Requirements and Objectives"

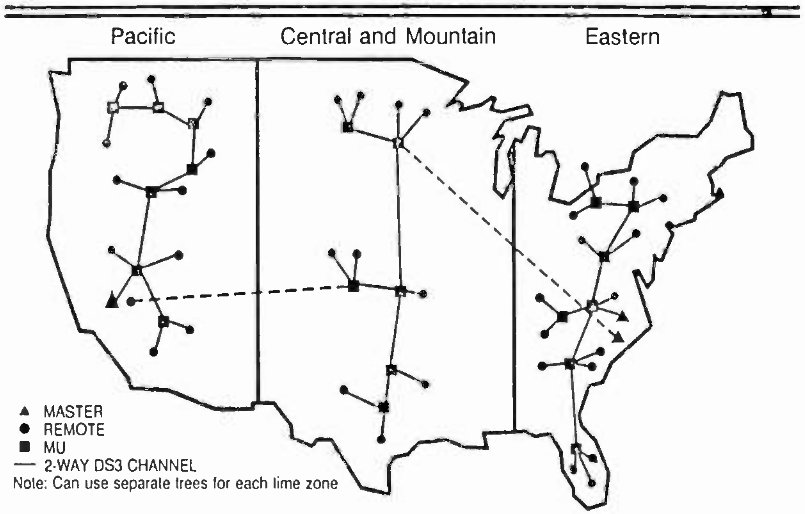
TA-TSY-000431, Issue 1 "Addendum to TR-TSY-000431 - 15 kHz Digital Audio Terminal for Program or Television - Requirements and Objectives (Additional Features for Use in Customer Controllable DS3 Multipoint Networks)"

Special Report SR-TSY-000687 "Nationally Compatible DS3 Rate (45Mb/s), Customer Controllable, Multipoint Networks for Broadcast Television Distribution: Possibilities and Challenges" This is a reprint of a paper presented at the National Association of Broadcasters Conference in 1987 in Dallas, Texas, March 31, 1987.

"DS3 Rate (45 Mb/s), Customer Controllable, Multipoint Networks for Broadcast Television Distribution/Collection" presented at the 129th Society of Motion Picture and Television Engineers (SMPTE) Technical Conference by Robert J. Blackburn and Paul Hessler, Bellcore, November 1, 1987

"Digital Broadcast Networks: A Down-to-Earth View" by R. J. Blackburn Telecommunications Magazine, January 1988

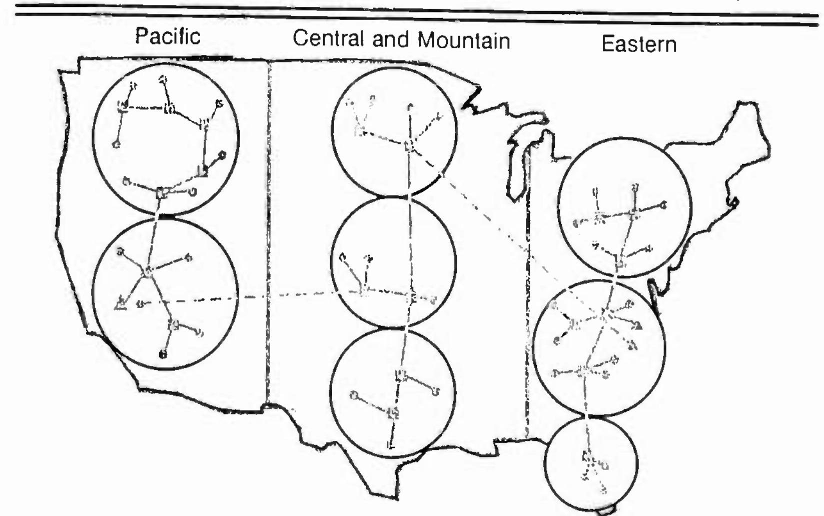
1. Time Zone Broadcasting And Collection



- Additional layer for back-up/other program material on diverse routes

FIGURE 1

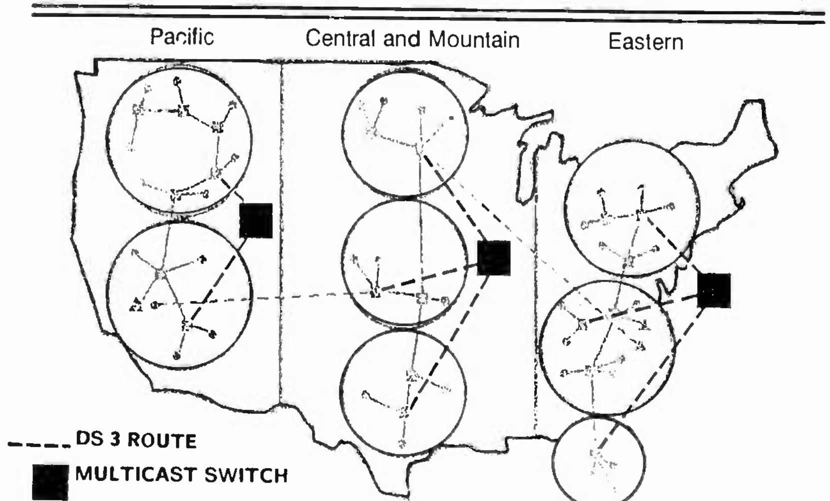
2. Temporary Regionalization Of Each Time Zone Via Loopbacks



- Backup does not have to be regionalized

FIGURE 2

3. Inter-regionalization Of Each Time Zone With Star Overlays



- Backup does not have to have star overlay

FIGURE 3

4. Full Inter-regional Capability (Non-Blocking)

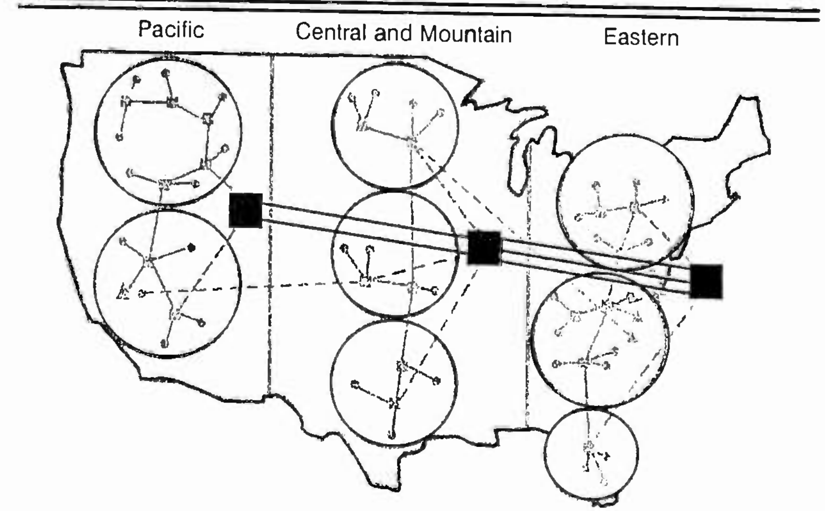


FIGURE 4

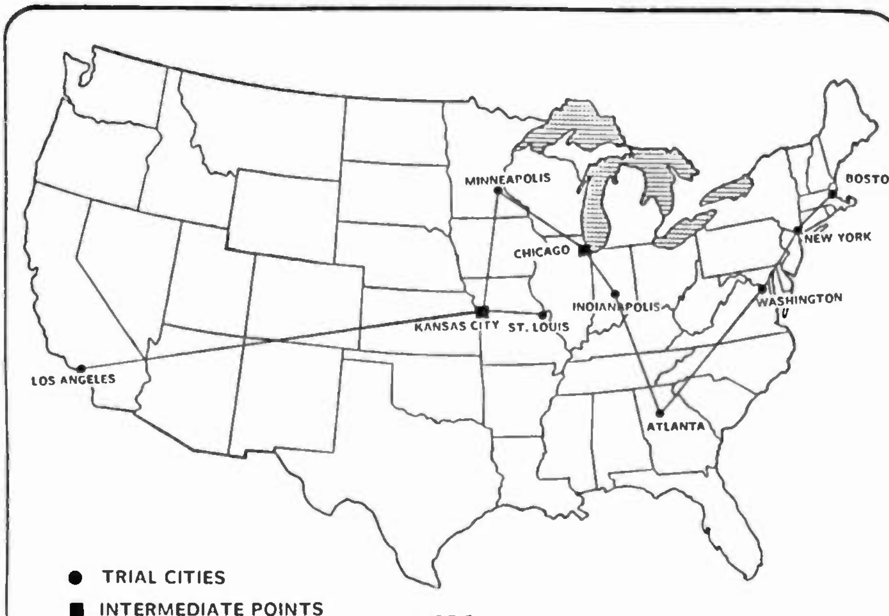

MINNEAPOLIS
BOSTON
CHICAGO
NEW YORK
INDIANAPOLIS
WASHINGTON
KANSAS CITY
ST. LOUIS
LOS ANGELES
ATLANTA
TRIAL CITIES
INTERMEDIATE POINTS

FIGURE 5

NAB
BROADCASTERS

NATIONAL ASSOCIATION OF BROADCASTERS