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db

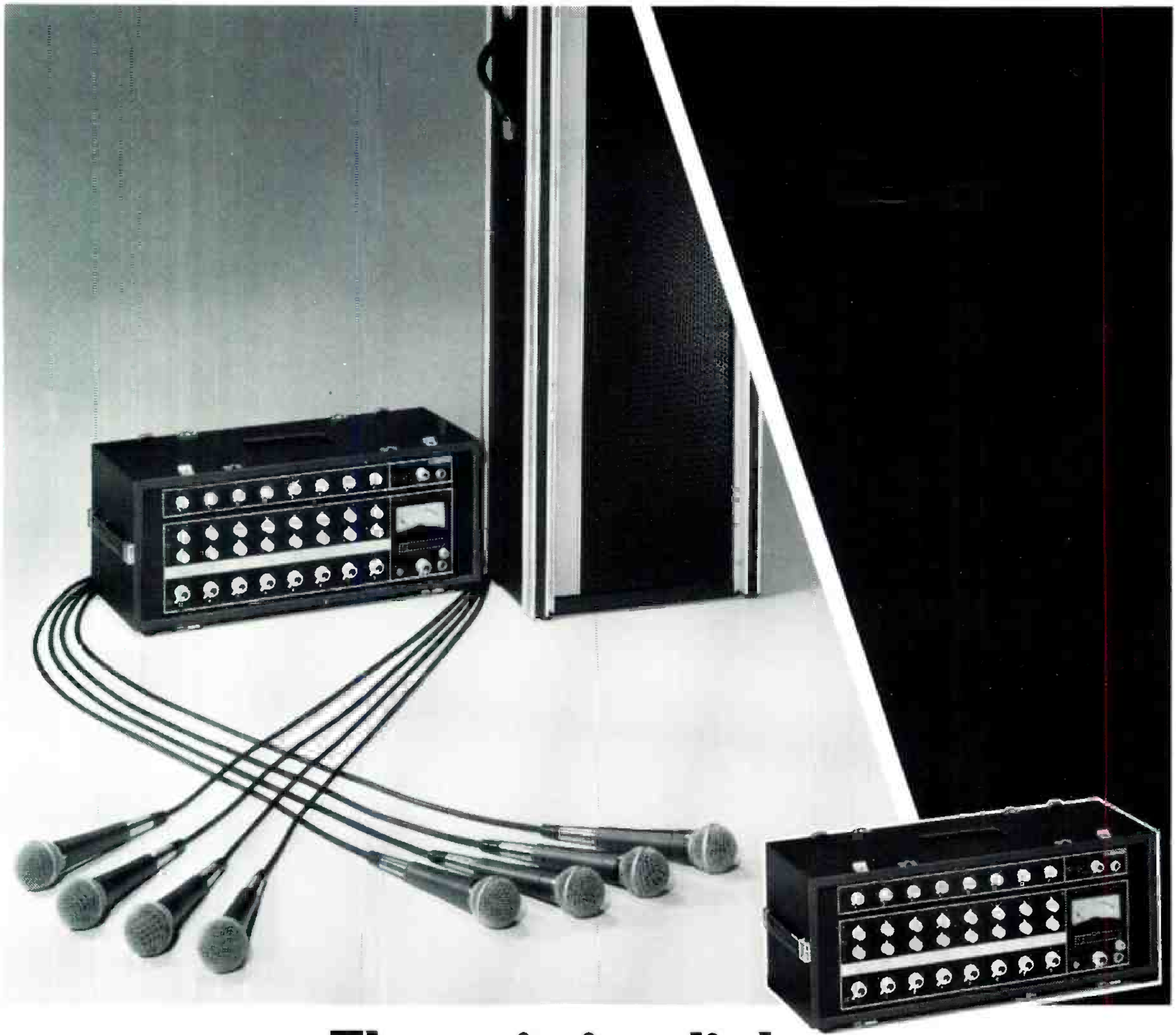
THE SOUND ENGINEERING MAGAZINE

DECEMBER 1976 $1.00

IS ISSUE:

rview: The Ampex ATR-100

- The Art of Tape Editing



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The SR110 features an eight-channel input /single output design–can be used as a single unit mixdown panel, or stacked for multi-channel recordings (use four for quadriphonic) or stereo broadcasts. Super space-saving–takes only 1¾" rack space. Both units are ideal for use with the SR101 Series 2 Console.

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SR109
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coming next month

- Part two of the Making of the Ampex ATR-100 will detail the development of the electronics and the achievement of phase coherence among channels of tape.
- David R. McLurg has written an article on the need and method of testing tape tension. As you might expect, this goes well beyond the use of spring type postage scales.
- Sidney L. Silver returns to our pages with The Application of Random Noise in Acoustic Measurement, the latest in his series of well-studied articles.

about the cover

- Brass is on our cover in all its brilliant glory.

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THE SOUND ENGINEERING MAGAZINE

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db is listed in **Current Contents: Engineering and Technology**

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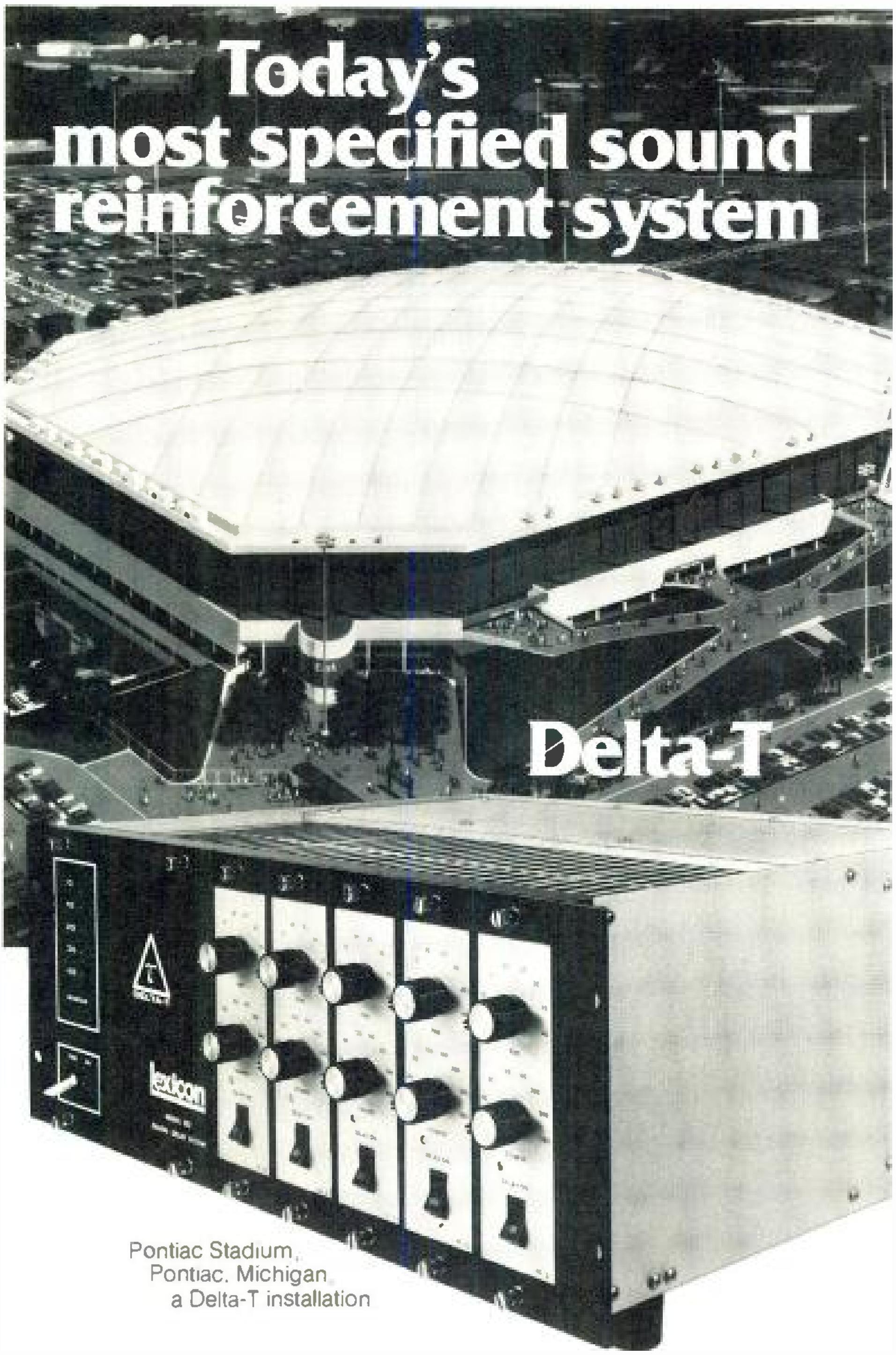

Pontiac Stadium, Pontiac, Michigan a Delta-T installation

CALENDAR

MARCH

14-17 **NOISEXPO '77,** the National Noise & Vibration Control Conference and Exhibition. Holiday Inn, O'Hare/Kennedy. Chicago. Contact: NOISEXPO '77, 27101 E. Oviatt Rd.. Bay Village, Ohio 44140. (216) 835-0101.

27-30 **NAB Convention Washington, D.C.** Contact: National Association of Broadcasters, 1771 N St., N.W., Washington, D.C. 20036. (202) 293-3500.

APRIL

19-24 **High Fidelity '77 Exhibition.** Heathrow Hotel. London. England. Contact: British Information Services, 845 Third Ave., New York. N.Y. 10022 (212) 752-8400.

25-28 **AUDEX, the International Audio Exposition,** trade show. Las Vegas Convention Center. Contact: Charles Snitow, 331 Madison Ave.. New York. N.Y. 10017. (212) 682-4802.

MAY

9-11 **International Conference on Acoustics, Speech, and Signal Processing,** Sheraton-Hartford Hotel, Hartford, Conn. Contact: Clifford Weinstein, B-345, Lincoln Laboratory, P.O. Box 73, Lexington, Mass. 02173. (617) 862-5500 X5465.

17-20 **London Electronic Component Show.** Olympia. London. England. Contact: British Information Services, 845 Third Ave.. New York, N.Y. 10022. (212) 752-8400.

NOTE: **db** is interested in acquainting our readership with all events that may be of interest. However, we frequently receive information too late to be of use. Kindly send all Calendar information to us at least three months before the event so it can be published in the issue appearing a month before the event is scheduled. Calendar items which are received well in advance are repeated for several months.

Model 1776 by Electro-Voice.

Getting that "big recording studio" sound out in the field can be tough. Because studios often depend on condenser microphones that are elaborate, delicate, and expensive.

What's needed is a new type of microphone that's strong, smooth, and moderately priced. Which exactly describes the new E-V 1776. A remarkable electret condenser cardioid uniquely suited to on-location recording and professional sound reinforcement.

Take a look. The no-nonsense diecast case fully protects the low-mass electret element and FET impedance matching circuit. Even temperatures as low as 0° or as high as 130°F don't bother the 1776. And, the 1776 shrugs off high humidity.

Yet the 1776 doesn't trade performance to gain strength. Its clean, open sound extends from 60 to 18,000 Hz, and its Single-D cardioid pattern offers the advantage of increasing bass as you move in ultra close. You can use this proximity effect to boost bass, or provide greater feedback reduction and performer isolation, or both.

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When you add up what the 1776 does—and how it does it—you'll find nothing else comes close to matching its unique design. Take studio quality wherever you go. Pick up a 1776 and hear the difference. Today.

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Model 1776 Single-D Cardioid Electret Condenser Microphone $99.00 suggested user net. Slightly higher in the West.

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Audio and the A.M. Process

● A.M. radio is our oldest form of broadcasting service. Despite the prophets of doom regarding the impact of f.m. and television, standard a.m. service is still going very strong. So much in fact, that the number of a.m. stations is equal to the number of f.m. and t.v. stations combined. A.M. broadcasting is a very important part of the American scene. Perhaps because a.m. has been around for so long. we may tend to take a patronizing attitude towards it. Although the system does have many natural as well as regulatory limitations placed upon it, when properly operated, a.m. is capable of producing a very good quality, medium fidelity signal.

AMPLITUDE MODULATION

The term a.m. means *amplitude modulation* of the rf carrier. When modulation is taking place, there are three elements present: the rf carrier, one upper, and one lower sideband. This is a full, two sideband system. The sidebands carry the audio modulation, and if the modulation is tone, then the power in each one of the sidebands will be equal to one-fourth the average power of the rf carrier (at 100 per cent modulation). The average power under program modulation will be much less.

The displacement of the sidebands from the carrier is determined solely by the frequency of the audio modulation. For example, if the modulation is 5 kHz tone, there will be a sideband at F_c + 5kHz. and another at F_c − 5 kHz. But if the audio were 10 kHz. then the sidebands would be: F_c + 10 kHz. and F_c − 10 kHz. The occupied bandwidth is twice the audio modulating frequency. Therefore, in the first case this would be 10 kHz. and in the second case. 20 kHz. FCC rules permit the station to occupy a bandwidth of 30 kHz: all emissions beyond this must be severely attenuated.

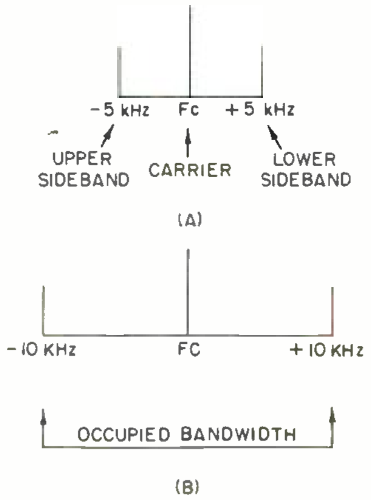
THE PLATE MODULATOR

Various types of modulators are in use—these may be at low or high level. The term *high level* means the transmitter p.a. stage is modulated. while *low level* refers to an earlier stage. One of the most common types is the high level plate modulator.

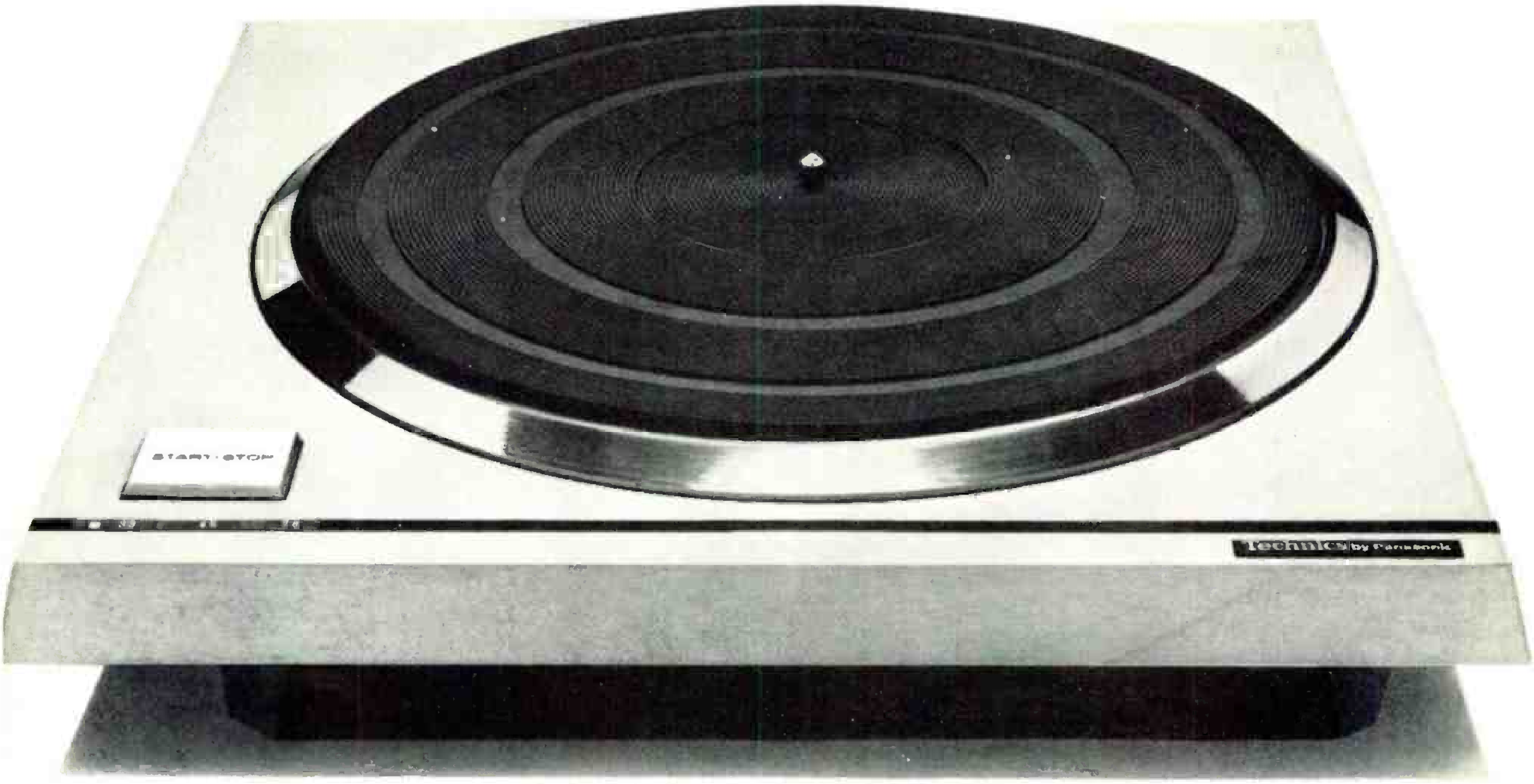
In a high level plate modulator. modulation takes place in the plate circuit of the p.a. stage. Since the sideband power is equal to 50 per cent of the average carrier power, the modulator must supply this power. To put it another way, the p.a. stage provides the rf carrier, and the modulator supplies the sideband power.

The output of the modulator stage is coupled to the d.c. plate voltage supply of the p.a. stage through a very large audio output transformer, called a *modulation transformer*. This unit is large and bulky because of the high

Figure 1. A.M. is a double sideband system. The displacement from the carrier is determined by the audio modulation signal frequency. (A) is the audio modulation at 5 kHz and (B) is the audio modulation at 10 kHz.



Your professional turntable cannot match the performance or reliability of this one.



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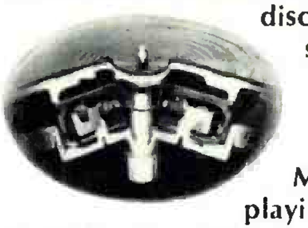

At 33⅓ RPM, the SP-10 MK II will reach the exact playing speed within 0.25 of a second. That's less than 1/12 of a turn. While it comes to a dead stop in only 0.3 of a second. And you don't have to worry about subtle slowdowns because a tracking force of even 1,000 grams won't noticeably affect its speed.

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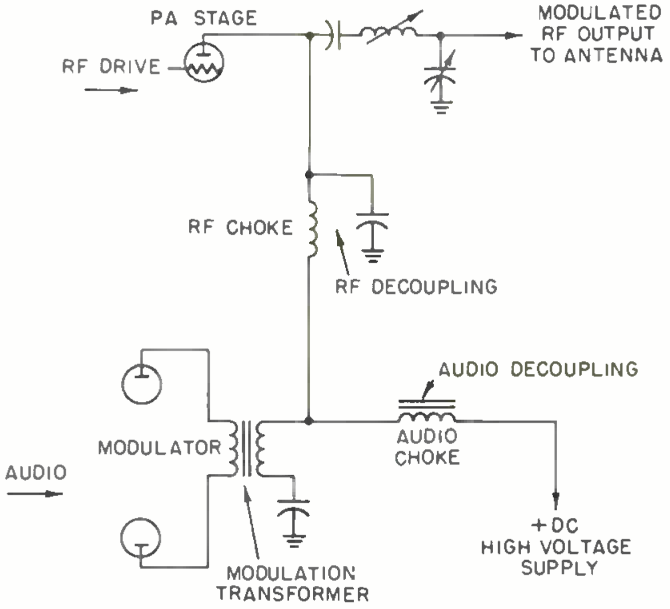

Figure 2. The high level plate modulator. Both rf and af must be decoupled from the power supply.

power audio it delivers and also because it is connected into the high voltage d.c. supply. The audio is coupled to the d.c. supply voltage in series with the supply voltage. Both the rf signal and the audio signal must be isolated from the high voltage supply itself, so rf chokes and by-pass capacitors are used for decoupling, and a large choke for the audio decoupling —this is usually called the *modulation choke.*

THE MODULATION PROCESS

Before audio is introduced, the p.a. amplifies the rf signal and supplies this to the antenna system. As soon as audio comes on (assume tone modulation for the discussion) the amplified audio at the output of the modulation transformer adds or subtracts from the d.c. supply voltage to the p.a. stage. When the audio is positive, it adds to the d.c. voltage, causing the p.a. output signal to increase; when the audio goes negative, the supply is reduced and the p.a. output decreases. The amplitude of the rf waveform is thus a replica of the audio signal and the carrier is modulated. When the audio positive peak causes the rf output voltage to double its unmodulated value, 100 per cent positive modulation is reached, and when the rf carrier reduces to zero on the negative audio peak, we have 100 per cent negative modulation. Since on the positive peak, the rf instantaneous peak voltage has doubled, the peak rf power output is four times its unmodulated power. $(P = E^2/R)$

SUPER MODULATION

The negative modulation peaks must be limited to 100 per cent according to the FCC rules. In recent times, the rules have placed a limit of 125 per cent on positive modulation peaks.

Many audio waveforms, especially those created by the human voice, are non-symmetrical, that is, positive and negative peaks are not equal. These peaks arrive at the modulator output in random fashion, so if a non-symmetrical peak should arrive in a polarity position with a negative high peak, the reproduced audio will sound lower than normal because the peak sets the percentage of modulation. To overcome this, some stations use electronic audio units that equalize the peaks, while others use an audio unit that senses the high peak and *always* makes this positive—switching polarity if necessary. This causes non-symmetrical modulation, allowing the positive modulation to be higher than the negative. There are other audio processors that will *force* the audio into deeper asymmetry, using clipping of the negative peak. This clipping creates an amount of distortion, but it can push the positive modulation to the full 125 per cent—if the transmitter can do it! If it can be accomplished, the sideband power will increase considerably.

Super modulation does place a heavy strain on the transmitter, the system components, and the cooling system. The high rf peaks can cause arc-overs and underrated components can break down. Most older transmitters were not designed with this type of operation in mind, so they may not be able to achieve it or they may break down.

AUDIO PROCESSING

Noise and interference are two of the limiting factors in a.m. Noise is a problem because it ordinarily amplitude-modulates the signal at the same time the program does, and interference occurs because of the low carrier frequencies used in the standard broadcast band. The skywave signals travel for hundreds of miles after sunset, so the majority of stations must reduce power, use directional antennas or both. In spite of these precautions, there are many interfering signals. Another factor creating interference is the growth of cities. They have expanded horizontally over considerable distances. While the term "fringe areas" is often used, conjuring an image of the boondocks, in many cases, especially for local stations, this is part of their main coverage at night! With the population spread out so far, they have difficulty providing an interference-free signal after sunset.

To overcome this, stations have resorted to heavy audio processing before modulation. This audio processing was used long before the supermodulation technique came into practice. The audio is run through AGC amplifiers that contain both signal expanders and compressors. The time constant is long so as to hold up the average of the program signal—changing the relationship between peak and average in the signal and holding the average modulation higher. Peak limiters are used to further compress the

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peaks, allowing a still higher average modulation. All this processing increases the average modulation and thus the sideband power so that the signal sounds louder in the fringe areas. As a tradeoff, however, the dynamic range of the audio is compressed.

LIMITATIONS

Modulating the transmitter with highly processed audio and on top of that, applying supermodulation techniques, cause many factors to become of greater importance than they are when lesser degrees of modulation are employed.

The high voltage power supply has greater demands placed upon it, since this is where all the power comes from in the first place. Consequently, it must be able to meet the new demands and its regulation must be good. If it cannot deliver, then there will be carrier shift. For example, at the positive modulation peak when the demand is greatest, the power supply d.c. voltage may *drop*, so consequently, the rf power output actually *decreases* instead of increasing! This is negative carrier shift and must be limited to 5 per cent according to FCC rules.

The p.a. and modulator tubes work especially hard and must be linear throughout the modulation amplitude swing. If the p.a. plate hits saturation, then the peak modulation softens and the signal is clipped. On negative peaks, the grid current of the p.a. increases and the driver must be able to supply the additional drive, or again the peak will soften. If the p.a. stage is a tetrode tube, the screen current will rise during negative peaks when plate voltage is zero or very low. This must be controlled, or distortion results. In all these cases, there can be carrier shift, soft modulation peaks, and various forms of distortion. Unless kept under control, these distortions can become very severe.

FIDELITY PROBLEMS

The mass of iron in the modulation transformer and the modulation choke will affect the very low audio frequency response (below 100 Hz). Besides poor response, this can introduce distortion and also set off parasitics at high modulation levels. Those stations on remote control which use over-the-air telemetry modulation of the carrier in the area of 22 to 28 Hz can experience problems with this function.

The upper end of the audio response curve can be rolled off by the capacities of the modulation transformer and the decoupling used at the output of the modulator. Perhaps the single largest limitation of audio response in the a.m. system is in its antenna system. Antennas have internal losses, and these increase as more towers are used. In the effort to get the most radiated rf power out of the antenna, the tuning results in a high Q system, with its attendant narrow bandpass. Coaxial transmission lines are used, so the antenna must match the characteristic impedance of the line *all across* the desired bandpass or there will be standing waves (reflections) where the match is not achieved. The match, then, should be for a bandwidth of 30 kHz. If it is not, the outer ends will be distorted or rolled off. Broadbanding will make the system less efficient, but this is the normal tradeoff in broadbanding -any tuned stage. The harmonic filters and matching of the transmitter to the line must also be broadbanded. If the system is not broadbanded, then it will simply act as a low-pass filter, and filter out the higher audio frequencies, limiting the bandwidth.

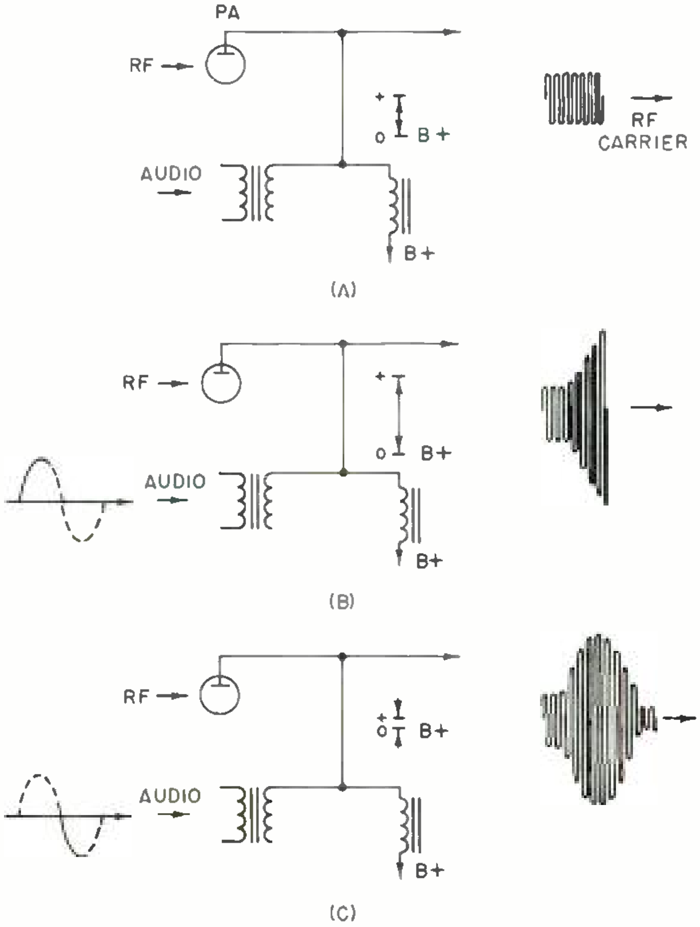


Figure 3. The A.M. modulation process: the audio causes the d.c. supply to p.a. stage to vary with the audio. This causes the p.a. rf output to vary and follow the audio. (A) indicates no audio input, (B) shows audio tone, positive peak and (C) shows audio tone, negative peak.

DISTORTION

Heavy audio processing before modulation, and then non-linear operation and clipping in the p.a. and modulator stages will introduce distortion elements, which must be carefully controlled. When audio clipping does occur, the transient response of the transmitter is very important. If it is poor, there will be transient overshoots at the leading and trailing edge of the flat part of the clipped signal, and this may be followed by *ringing*. These are essentially the same results you would observe on an oscilloscope when using square wave testing of an audio system or television system.

Clipping and non-linear operation also allows intermodulation to occur. This will produce sum and difference frequencies of the two intermodulating audio signals as well as harmonics of both. Many of the harmonics can be filtered out if they are high enough in frequency, but the sum and differences will remain in the audio. When these forms of distortion are allowed to occur, they muddy the reproduced audio so that it lacks brilliance and luster; it has a dead sound. If the amount of distortion becomes too high, then of course, the signal becomes unlistenable and the audience will switch the dial!

SUMMARY

The a.m. system, even with all its limitations, can produce a good quality, medium fidelity signal. But transmitter design as well as improper operation can cause serious fidelity deterioration, as can excessive use of audio processing. The largest single limiting factor to fidelity is the filtering action of the antenna system and its tuning, and the amount of broadbanding it has adjusted into it. ■

Each of these new Peavey Mixers costs less than $1,000.

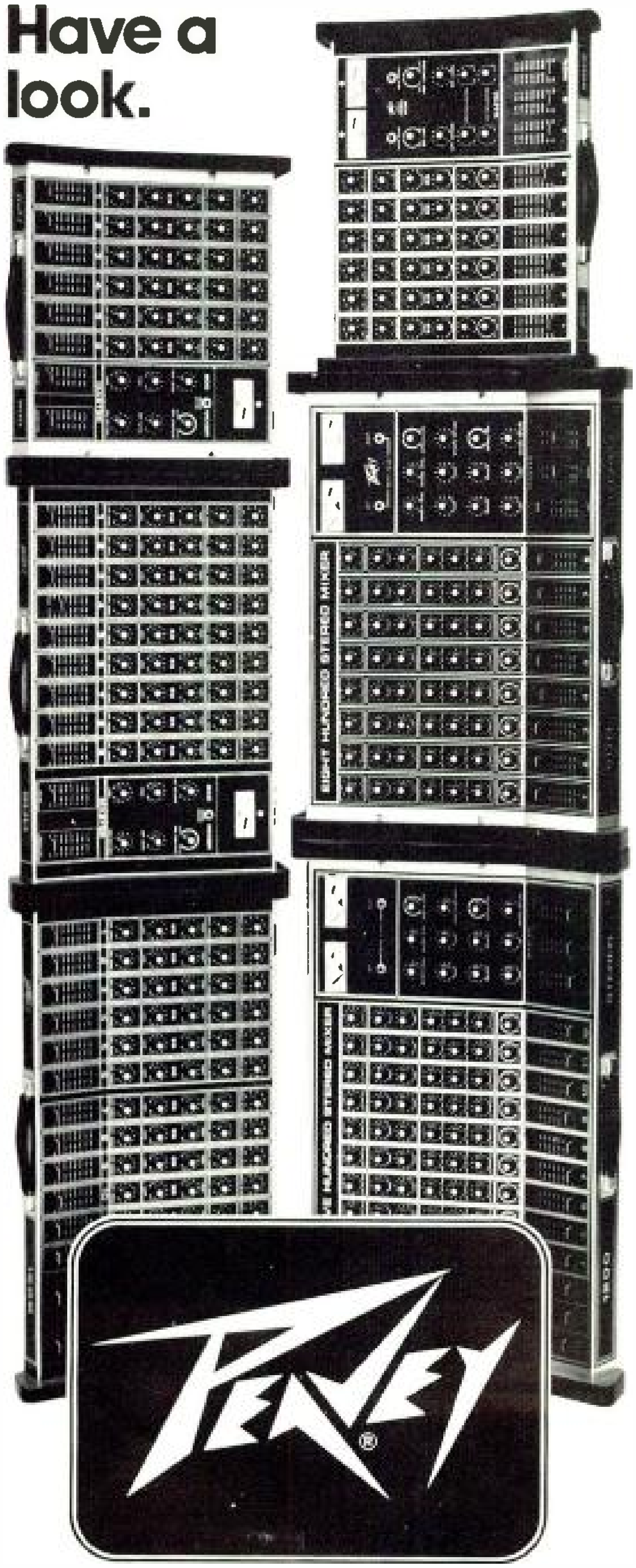
How do they stack up with the competition?

Have a look.

These new Peavey Mixers have been designed to satisfy the requirements of a continously variable sound reinforcement market. If your requirements are stereo, mono, high impedance, low impedance, balanced, or unbalanced,...we've got a mixer for you.

We offer as standard equipment such features as input attentuation variable to −40 dB, pre monitor, post effects with built-in reverb, effects return, and slide faders on even our most inexpensive mixer.

Comparison of distortion figures, frequency response, noise performance, and functional features will illustrate the tremendous value of these mixers.



600 Mixer

6 channel Mono, with 6 low Z, 6 high Z, & 3 auxiliary inputs
20 Hz to 20 kHz response with less that 0.1% THD
−123 dBV equivalent input noise
5 volts RMS out
Price: $349.50

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Stereo pan each channel
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Left & right Main and Monitor outputs transformer balanced
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900 Mixer

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−123 dBV equivalent input noise
5 volts RMS out
Price: $449.50

1200 Mixer

12 channel Mono, with 12 low Z, 12 high Z, & 4 auxiliary inputs
20 Hz to 20 kHz response with less than 0.1% THD
−123 dBV equivalent input noise
5 volts RMS out
Price: $549.50

1200S Mixer

12 channel Stereo, with 12 low Z, 12 high Z, & 5 auxiliary inputs
20 Hz to 20 kHz response with less than 0.1% THD
−126 dBV equivalent input noise
Stereo pan each channel
Effects & Stereo return with pan
Left & right Main and Monitor outputs transformer balanced
Transformer balanced inputs
5 volts RMS out
Price: $999.50

All prices are manufacturer's suggested retail.
Because of freight and duties, export prices are slightly higher.

For complete specs and information on any of the new Peavey professional Mixers write:

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db theory&practice

NORMAN H. CROWHURST

• My discussion of filters and crossovers has stimulated some two-way communication. Unfortunately, this kind of communication can be rather slow. In a classroom situation, if a student asks a question, even if I do not quite understand what he is getting at at first, a little interchange back and forth will quickly enable me to see how his thinking goes. Not so, when the interchange is slowed down by the "service" provided by the U.S. Postal Service!

One correspondent seems to think that a Butterworth characteristic is the best compromise, and on that premise, examines the first three orders briefly. Finally, after apparently concluding that the third order is about the best, giving a flat response and gradual phase shift, along with a steep roll-off on both outputs, he notes that, according to my definition in the September issue, its transfer function cannot be synthesized as a constant impedance network, and suggests that this would be a good argument for bi-amping.

Okay. I think I have the way he is thinking. Butterworh, Chebyshev, *et al.* are ways of arriving at certain response shapings in filters. When they are referred to as best, or a best compromise, those who make those claims, whether they realize it or not, are thinking in terms of the filter's attenuation/frequency response. If phase enters the picture at all, it is as a sort of afterthought: "Oh yes, when you change amplitude response, you do affect phase response, as well, don't you?"

So, my correspondent notes that a second-order Butterworth is non-minimum phase, by my definition, as are all even-order Butterworth designs, and for that reason moves on to third-order. Apparently, this transfer function appeals to him as ideal, or a best compromise thus far at least. So then he turns to the question of impedance, noting that it cannot as well provide a constant impedance input.

IMPEDANCE

Which piece do I tackle first? Perhaps I can best elaborate on my reason for introducing the constant *resistance* condition. Maybe I did not make it clear, but one of the properties of the constant resistance configurations is that, if both outputs are resistance loaded with their nominal value, the input of the crossover combination reflects a constant resistance, not merely a constant impedance.

But this configuration has other properties that are unique to it. And these are possibly more important. Both the attenuation and phase responses are completely complementary. This means that the phase difference between the two outputs is constant.

The 6 dB/octave roll-off is simple, of course, and has no need to be described as a Butterworth, or anything else—it is just a pair of simple r-c section combinations, one for high pass, one for low pass. The phase difference, when correctly aligned, is constant at 90 degrees.

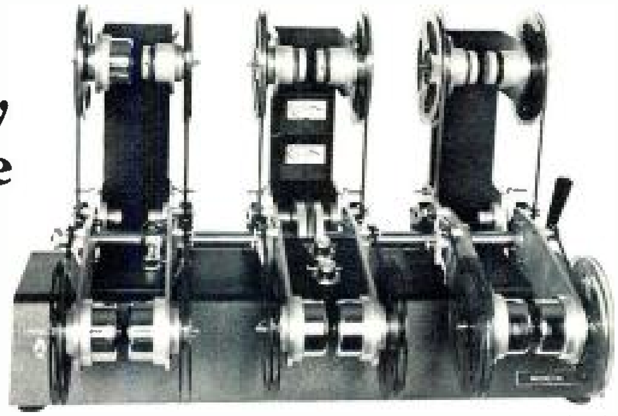
The 12 dB/octave variety has a 6 dB/octave slope at crossover, and a constant phase difference, between the two outputs, of 180 degrees. As you go more complex, each extra 6 dB/octave of ultimate slope adds 3 dB/octave of slope at crossover, and 90 degrees to the constant phase difference.

In that sense, I suppose one could say that any constant resistance crossover design is *not* a compromise, although Butterworth and Chebyshev shapings are compromises, in this sense. What this means is that the close-to-flat response is extended further, and the cut-off beyond crossover is made steeper by sacrificing something else.

But at the end of my column in the September issue, I said that the choice of crossover, and I was talking about constant resistance types, nothing else, at that point, is a compromise, so wasn't I contradicting myself? Just now I said any constant resistance type is *not* a compromise, and at the end of the September column I said choice of *slope*, by any method, is a compromise.

At the end of the September column, I was talking about whole electroacoustic systems, including the units you put together to make into a composite loudspeaker system. Just now, I am talking only about crossover filters themselves. There is a difference.

I am assuming for the moment that you make no compromises in the filter design which you achieve by using, of whatever slope you choose, a constant resistance configuration, or set of values. But you are still faced with a compromise, as to what slope to choose in designing your composite electroacoustic system. Is that clear?



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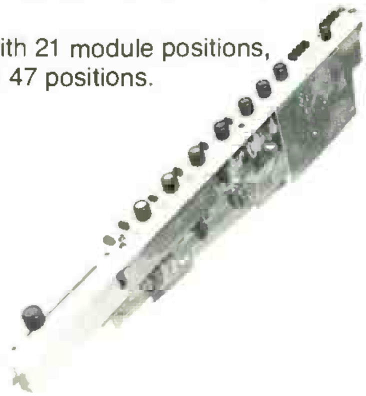

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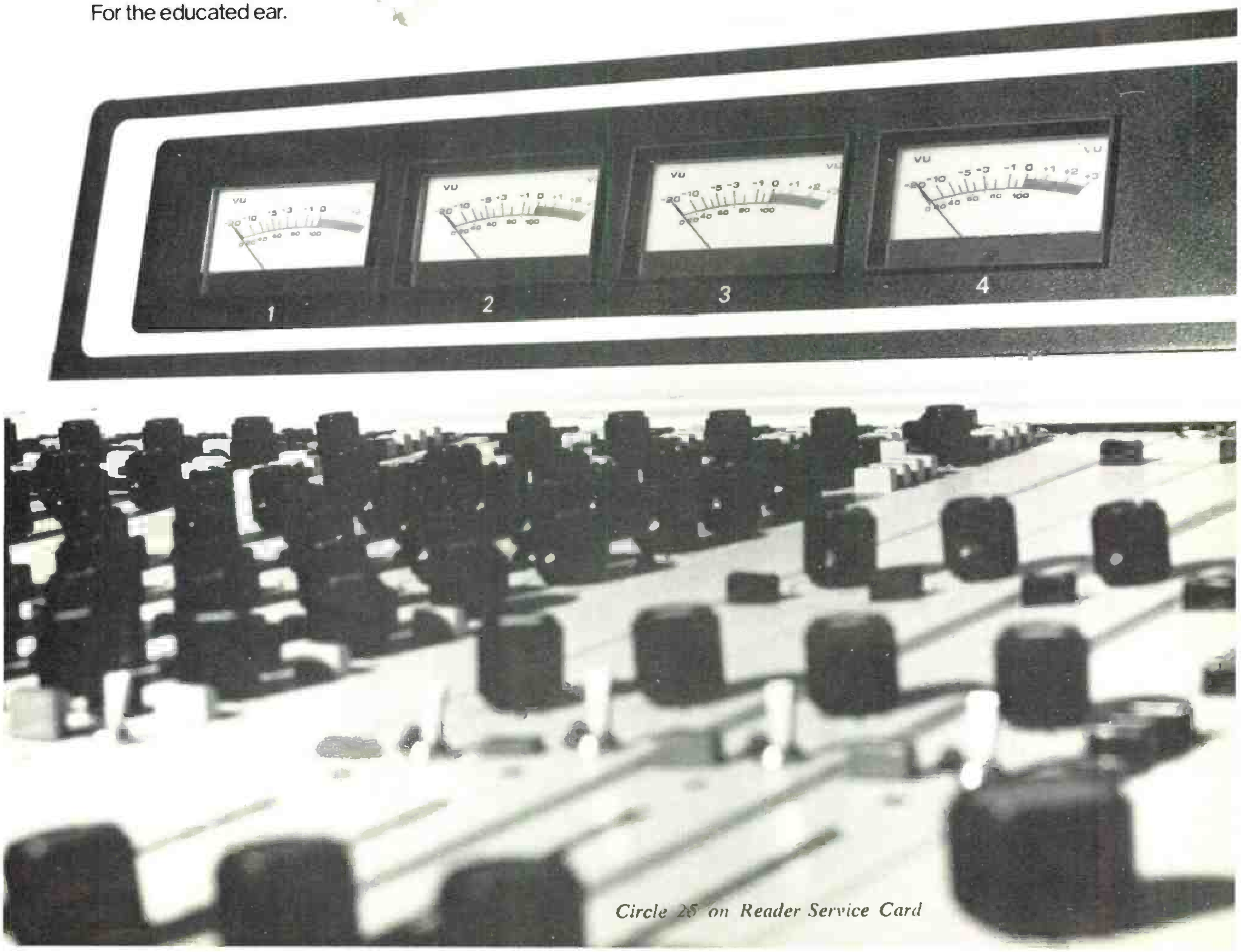

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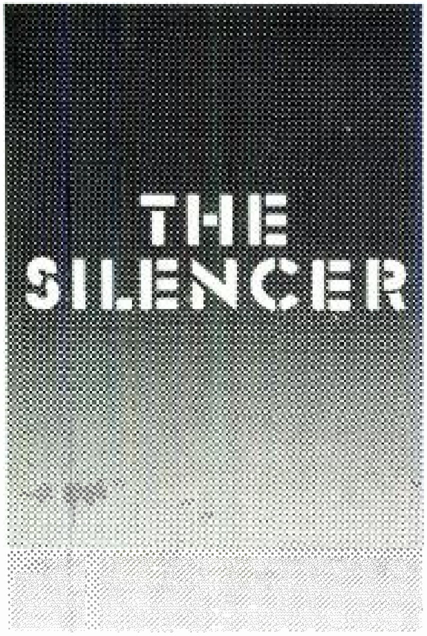

You see, I didn't say anything about Butterworth, Chebyshev, or any other fancy shaped response, thus far. So my correspondent's suggestion, "if you assume a Butterworth characteristic to be the best compromise," introduces a use of compromise that I had not even considered in the September column—a compromise *away from* the constant resistance basis.

My assumption of a constant resistance design tacitly assumes that we have perfect loudspeakers in the electrical sense, that their impedances are constant resistances, which is not so of any loudspeaker unit. Therefore my reasoning that enables me to use an uncompromised filter design is immediately compromised because I must use imperfect termination for my "perfect" filters! The question then arises, whether one form of compromise is any better than another. This is where it begins to get difficult to keep the variables, or parameters, separate in our minds, simply because they are not separate in the real world. But if we can use our pseudo-perfect discussion as a kick-off point from which to investigate the real world, perhaps we can begin to get our thinking straight.

The illustration I used in the September issue, synthesizing a 24 dB/octave response from two 12 dB/octave responses, was intended to help visualize why the rapid changes in phase response, that *inevitably* accompany the sharper roll-offs, produce effects on audio reproduction that are undesirable and thus constitute a factor that must enter the overall compromise. Using Butterworth, Chebyshev, m-derived, or any other design will not change this.

Because someone showed, in a certain context, quite different from this, that a Butterworth is *a* best compromise for that context, does not get out of the compromise we are discussing in any way whatever.

To illustrate how context changes requirements, we could move to the design of filters associated with multiplex demodulation. In that context, the ideal is not minimum phase, but linear phase, which means a phase delay that is proportionate to frequency. A phase delay proportionate to frequency is the same as a constant time delay. Achieving this objective makes the design of the rest of a multiplex system relatively easy.

PHASE ADVANCE

However, in crossover networks we are concerned not only with phase delay, but also with phase advance. Although phase advance involves difficulties in conceptualization, it's really quite simple, thinking in terms of phase. Phase can be delayed—why cannot it also be advanced? But if you are thinking in terms of *time*, then a phase advance represents a situation where the signal "gets there before it leaves." How can an electronic network possibly anticipate what is coming, before it arrives?

One way to clarify this is by analogy with waveguide technology. Inside a waveguide, there are three velocities of wave movement. Basic to the system is the propagation velocity. But then, due to the way the waves propagate, in a sort of criss-cross fashion, the wave *energy* does not progress down the guide quite as fast as the waves propagate. That speed is called the "group velocity."

When energy is flowing at that frequency continuously, another velocity is evident, called "phase velocity." Phase velocity is higher, or apparently faster, than propagation velocity, by the same ratio that propagation velocity exceeds group velocity. What does this mean, related to our multi-way speaker system? It means that phase advance, at the low frequency end of the high-pass response, occurs during steady transmission of a frequency in the vicinity of crossover. But at initial transients a different condition applies, where energy has to build up to that condition.

CROSSOVER FILTERS

Crossover filters are sharp cut-off, wide band filters. In having sharp cut-offs, where they do, they are quite similar to narrow-band filters with similarly sharp cut-offs. In fact the Butterworth and Chebyshev shapes apply the technology originally developed for narrow-band applications to wide band filters such as crossovers.

Now, if you have a single tuned, high-Q circuit, it will act as a narrow-band filter. Because of this, it will not respond immediately to just one cycle of its resonant frequency particularly more than it would respond to any other frequency. It takes time, if the Q is high—a great many cycles—for the filter to build up to its steady response.

After the signal is terminated, it takes an equal amount of time for the response to die away. In short, to transient signals, whose dominant frequency components happen to be in its resonant range, such a circuit rings. The same effect has been observed, to a lesser degree admittedly, with sharp cut-off crossovers, and for the same reason.

Now, to the question of bi-amping. Here we run into another popular misconception. It is true that bi-amping has advantages that we have covered

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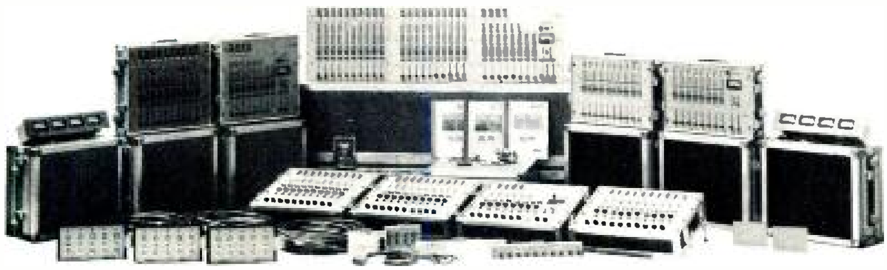

at other times, but creating a constant impedance network is not basically one of them. In making the suggestion, it is apparent that the intent is to achieve the same Butterworth shaping, in the signal fed to each amplifier, as that achieved in the prototype Butterworth filters.

What I have shown before is that when you do this, the phase response will at least duplicate the prototype filter responses. It may be worse, by being further from minimum-phase. But it cannot improve it. True, in terms of steady-state frequency response, phase compensations can be performed. But when these are applied to transients, the increasing difference between group and phase velocity, or in electronic circuitry, timing, shows up.

Putting the filters before the power amplifiers that feed the individual speakers will avoid any mismatch between speaker unit and amplifier that is caused by the crossover filters, when used, because the filters are eliminated in this method. But this assumes that the crossover mismatch is the major problem. Actually, the average loudspeaker unit shows more deviation in its impedance than crossover mismatch would add to it.

Perhaps this is an instance of something I have encountered many times, over the years. Often people ask me, "Which is best?" expecting an answer that suggests that one particular design, or approach will be named. When I ask "For what purpose?" or "In what context?" some seem to think I'm being evasive. In their view, there must be a best. So why cannot I tell them which it is?

CONTEXT

Possibly nowhere, to the extent found in audio, does context, or environment, affect the relative importance of the various parameters to the same degree. Why do some people find such difficulty in believing this? Some even add, "You can tell me, I won't quote you," as if the reason I will not tell is because I do not wish to offend advertisers who happen to sell products other than this mythical best.

Those who have followed my writings over the years know that I have done more than my share of keeping advertisers honest by examining false claims in my writings, published for all to read. If I have some information to convey, I will convey it, publicly. But where a question needs some qualification, such as in which context a system is to be used, I cannot pretend otherwise, that there is some best solution for everyone's problems. ■

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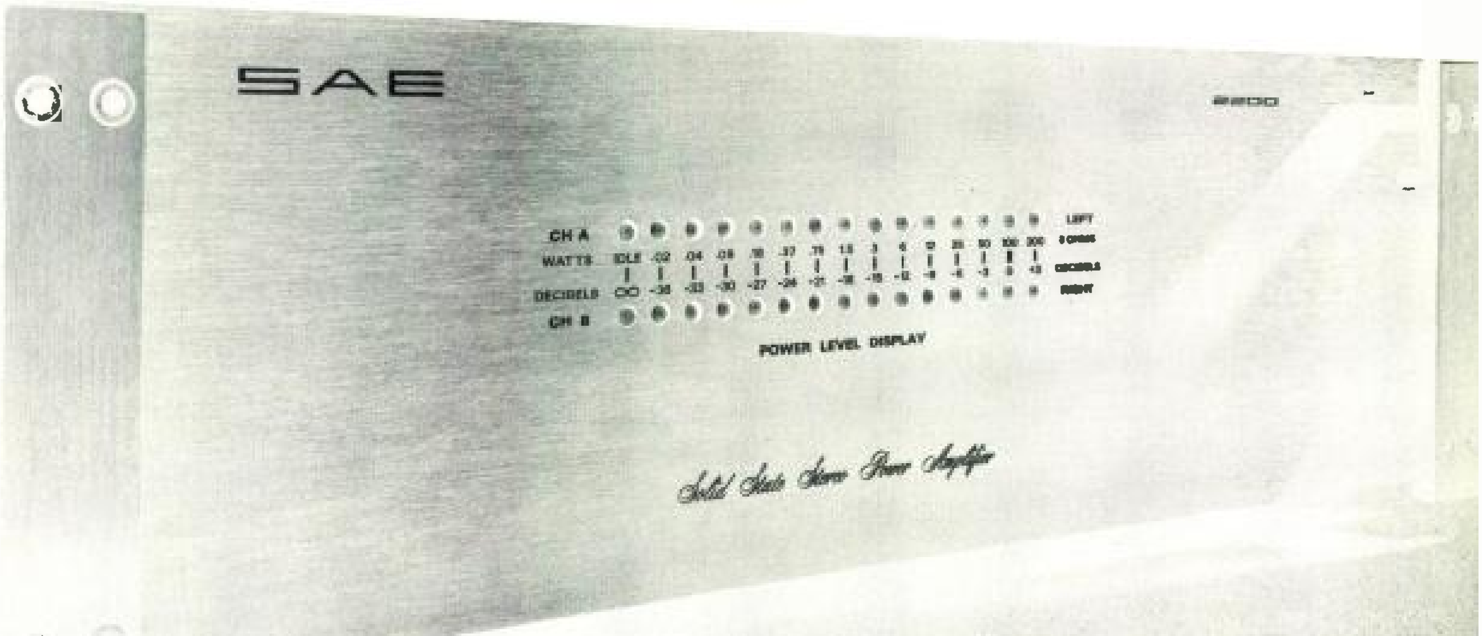

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JOHN M. WORAM

• *More* books on recording . . . Some years ago, when I first had the bright idea to write a book about recording studio technology, I was spurred on by the lack of literature on the subject. For although there were many texts around dealing with audio in general, there seemed to be few, if any, that dealt specifically with the recording studio.

My idea must have been a great one, for it seems to have occurred to just about everyone else, at just about the same time. Within the last few months, each morning's mail brings with it yet another book with the word "recording" in the title. Well, maybe not *every* morning, but it can get to seem that way to a new author who views with alarm these encroachments on what he had fantasized as virgin territory not so long ago.

It would be tempting to proclaim that all these books are just not worth reading, but the editor seems to have some sort of childish aversion to lawsuits. So, I'll make an attempt at being reasonably objective. That should please him, although it doesn't do much for my nerves.

SOUND RECORDING, by John Eargle. Van Nostrand Reinhold, 327 pages. $16.95.

Some years ago, author John Eargle and I briefly discussed the concept of working together on two separate, yet related, volumes. Although other commitments—and the fact that we live a continent apart—argued against the idea, I find his new book a perfect companion to mine, for which I am delighted, for Eargle's would be a tough act to follow if we were both using the same material. (I suppose that comments on my book while reviewing someone else's will raise a few literary eyebrows here and there.)

Sound Recording is divided into ten chapters of some twenty to forty pages each. Chapter titles are : 1. Physical Aspects of Sound 2. Psychoacoustics 3. Stereophonic Sound 4. Quadriphonic Sound 5. Microphones 6. Monitor Loudspeakers and the Monitoring Environment 7. Audio Control Systems 8. Magnetic Recording 9. Signal Processing Devices 10. Disc Recording and Reproduction.

Eargle's early chapters take us through a detailed analysis of sound and psychoacoustics. A fair amount of mathematics is involved, yet the presentation does not get bogged down in the usual "technicaleze." Therefore, the reader with little or no math background should be able to grasp the general concepts without too much trouble.

The basic principles of both stereophonic and quadriphonic sound are then covered at some length. In the latter chapter, Eargle presents the mathematical models for the SQ and QS matrix systems. Discussion of the CD-4 system is reserved for the book's final chapter, on disc recording.

Chapters 5 through 9 cover the basic recording studio hardware, from microphones to signal processing devices. Although these chapters come closest to my own opus, I am amazed (and certainly relieved) to discover remarkably little overlapping. For example, Eargle's analysis of the various microphone types is the more scientific, with a particularly detailed study of capacitor principles. Later on, in the chapter on Audio Control Systems, he covers the gain/loss structure of the signal path, and concludes with a review of the Allison and Quad-8 automation systems. A brief discussion of address codes and electronic editing follows, in the Magnetic Recording chapter.

The final chapter on Discs should be required reading for every engineer who thinks that the creative job is over when the master tape has been sent off for transfer to disc. Eargle begins with basic groove geometry, and takes us step-by-step through the complete disc cutting hardware and software systems, concluding with a brief description of record processing.

SOUND RECORDING PRACTICE, John Borwick, editor. Oxford University Press, 440 pages, $35.50.

Borwick's book is actually a series of twenty-five essays by leading specialists in various areas of audio engineering. The book is divided into eight sections: Technical Introduction, The Studio, The Equipment, Techniques, Manufacturing Processes, Allied Media, Appendices, and Glossary.

The amount of detail varies with each author and the space allotted to his topic. For example, Richard Swettenham's twenty-one page chapter on "Mixing Consoles" describes the typical multi-track console in great detail, with at least a paragraph or two devoted to each function. This is followed by Michael Beville on "Extra Facilities." In twenty five pages, Beville takes us through compressors, limiters, expanders, equalizers, noise reduction, echo and reverberation, delay lines, and phasing. The treatment is necessarily brief, yet gives the reader an excellent overview of the entire signal processing arsenal.

In a later chapter, John Culshaw describes "The Role of the Producer" in six pages, limiting himself to classical production. Peter Tattersall gets about twice as many pages for his essay on "Popular Music," where he describes a typical set-up for a small rhythm section.

Borwick himself contributes the introductory section, and a brief, yet informative, chapter on "The Microphone Circuits."

This is a book to be read from cover to cover, although not necessarily in the order in which it is presented. Since each chapter is from a different hand (actually, some authors have contributed several chapters), you may want to skip around. By all means do so, for Borwick has seen to it that each chapter is complete within itself. He has also managed to assemble a group of contributors who can discuss their specialties in a literate manner, so you needn't know as much as they do before beginning. Although not a technical text book by any means, *Sound Recording Practice* is an excellent survey of the recording industry. ■

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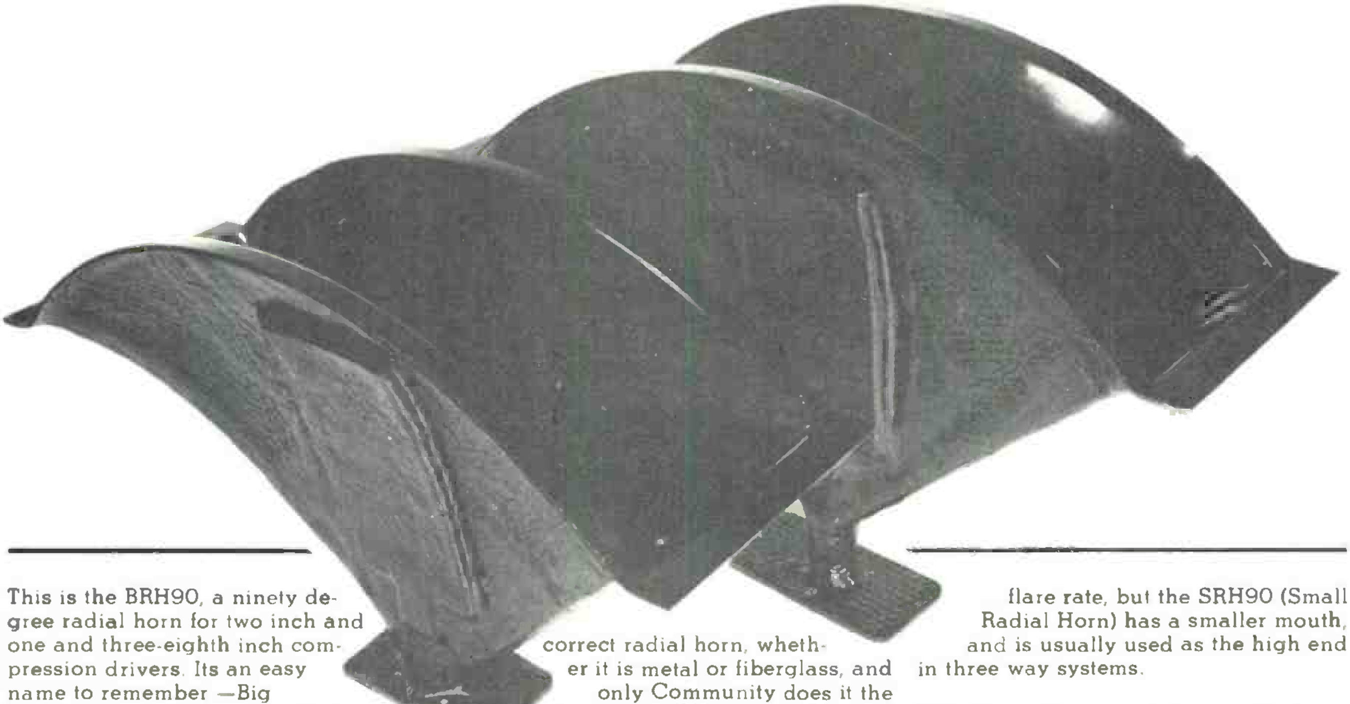

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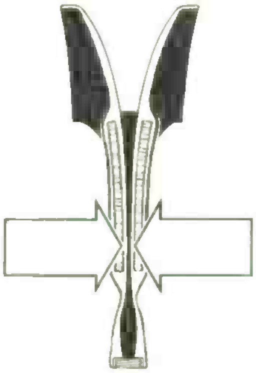

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D.	21"			20½"			18¾"		
Weight	25 LB.			20 LB.			12 LB.		
Finish	Black, High Gloss			"			"		
Horizontal Dispersion	KHz	-3dB	-6dB	KHz	-3dB	-6dB	KHz	-3dB	-6dB
	6	85	95	6	80	90	1 2	95	100
	2	90	90	2	90	100	3	90	95
	10	80	90	10	85	90	10	85	100
Vertical Dispersion	KHz	-3dB	-6dB	KHz	-3dB	-6dB	KHz	-3dB	-6dB
	6	50	90	6	55	100	1 2	50	70
	2	35	50	2	35	50	3	40	65
	10	20	35	10	20	35	10	20	30

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MARTIN DICKSTEIN

● For the past month, I have been traveling for a client, helping to get various regional sales meetings in shape audio-visually. It proved a fascinating experience, and some of the situations I found might be of interest to you.

The first city we went to was a large one in the not-too-far south. The site was a fairly large conference room in a hotel. The expected audience of about 125 was seated at tables, classroom style, with a raised platform for a stage at the end, opposite the rear doors. The room was over 50 feet long and about 40 feet wide.

Since the desire of the client was to have two lecterns, it was not possible to put the screen on the stage. When we arrived at the scene, we found that the company hired to install the equipment had beaten us to it and put the equipment up early. The screen was properly placed at a slight angle at the side of the raised platform, but both lecterns were widely spaced and facing into the room, toward the center. This meant that one of the speakers, the one nearer to the screen, actually had the slides behind his back. It might be okay to have this setup if someone else is actually advancing the slides while the speaker is talking from notes, a script, or ad lib, but not from the slides on the screen.

We had two alternatives. Either we set up the logistics of the speakers so that those who did not have to look at the screen worked from the position closer to it, or we moved something so that the people could speak from any spot they chose. The client decided to choose a third possibility, which I suggested, to make things a bit easier for all—eliminating the lectern near the screen so that all speakers had a clear view of the screen. This way, whether the slides were changed by someone else or the speaker, there was only one clicker needed at the front and another at the spot where the phantom slide changer was stationed.

CHANGING CABLES

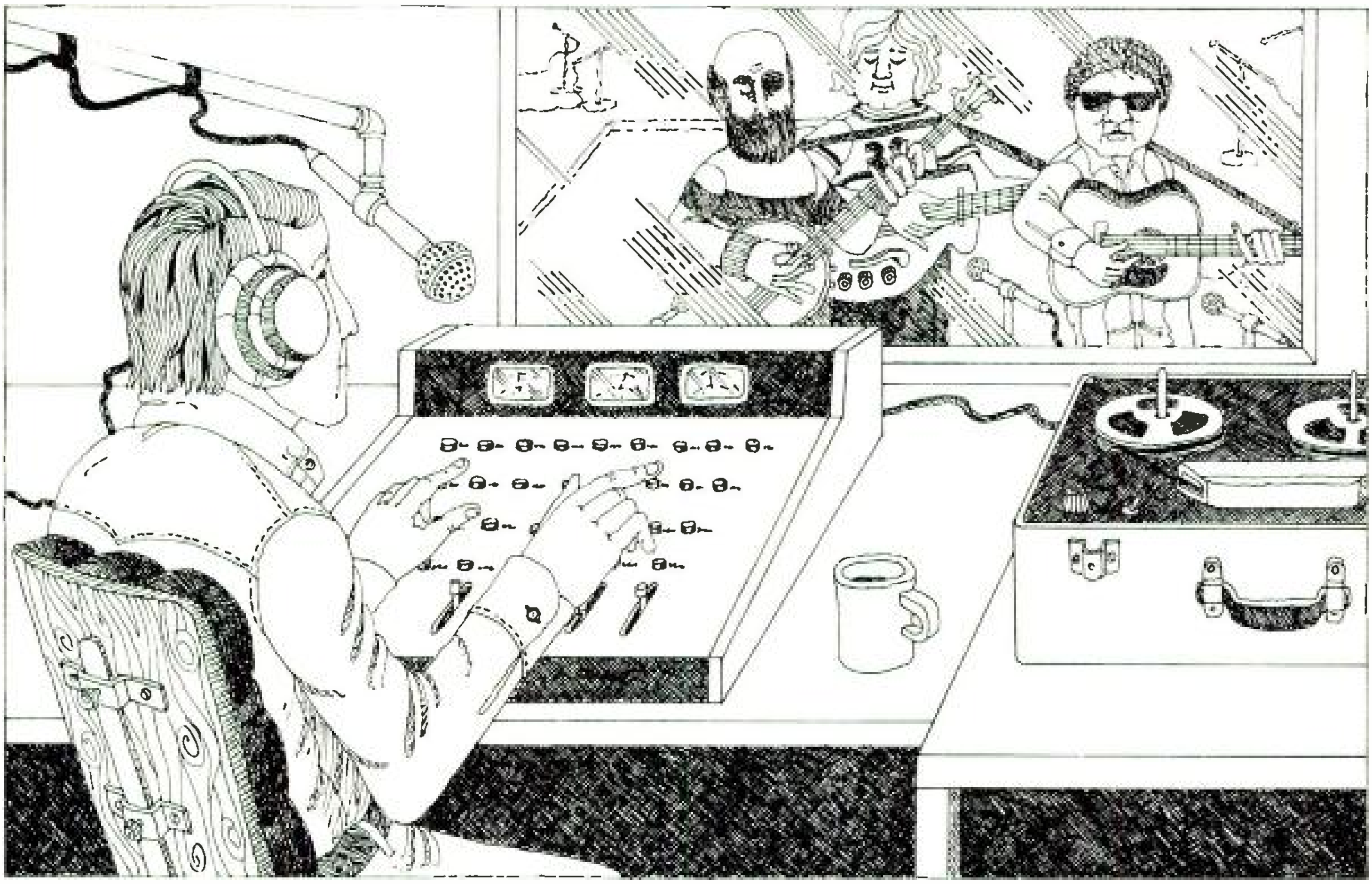
The changing of cables at the rear of the slide projector, incidentally, brings up another point. Since there was a change in cable necessitated by the fact that either the speaker or someone else was going to advance the slides, the alignment of the projector could be upset by pulling the control cable out and inserting another. Then the projector would have to be moved a bit to compensate. This setup, however, was slightly more complicated since there was also need for a dissolve with two projectors, and a hookup with a cassette player, which would also have to advance slides during one portion of the meeting in which the audio track was on tape. One way to do the setup was to have four slide units—one for the front activation, one for the phantom, and two for the dissolve slides, with a change in cable when the cassette's turn came up. The cost of renting the units was not large, in comparison with the cost of having all the regional managers fly in from various places and stay at the posh hotel. (Of course, five slide projectors would have eliminated all cable changes.)

One problem with having so many slide projectors was setting them up to provide a single image without keystone, and without taking up a lot of room with wide-legged projection stands. By using a small triangular stand which was placed over one of the projectors, a second unit could be placed over the lower one. This took less room than a side-by-side arrangement, but the upper unit was a bit unsteady and vibrated slightly at every slide change. Also, when a drum on the lower projector had to be changed, it was not easy because there was barely enough space under the three-legged stand to get a firm grip on the tray under it. Drum changes had to proceed carefully on the lower projector to avoid moving or upsetting the upper one.

We finally ended up with a three-projector setup. Two of the units were set for dissolve, using the triangular stand and one expanding leg table, with the third projector on a second projection stand. Since the program called also for the use of a 16mm film projector, a third stand was set up.

A small low table was also placed conveniently for the dissolver and the cassette player. (Incidentally, a large table was also located nearby for spreading out the multitude of soft-

You're sick of stereo recorders built for the home hobby market.



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But after your ATR-700 is on the job, producing recordings with a record/reproduce audio range of 40 Hz to 18 kHz, ±2 dB (at 15 ips), you'll enjoy one other advantage. This recorder is *really* rugged. It'll give years of reliable service.

ATR-700 is the professional stereo recorder from Ampex, built for professional engineers who can't take chances.

AMPEX

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Circle 23 on Reader Service Card

ware like about twelve slide drums, a script for each of the speakers whose slides had to be operated from a rear position, and some dirty coffee cups and ash trays which a few of the people left conveniently in the way.) In order to avoid a few of the cable changes, I had brought along, in my "emergency bag of tricks," a "Y" cable which I had made up previously, and hooked it up so that two remote controls could operate the same unit from different locations. (I had also brought the cable which allowed one person to activate two side-by-side projectors simultaneously, but did not have any use for it this time.)

THE DISSOLVER

The dissolver also proved interesting. When the script had been marked up for use with the overlapping slides, the unit used during rehearsal was a Kodak model. The way this one works is to advance the tray on the dark carousel, then turn on one lamp while the other lamp goes slowly to black. The drum on the dark projector does not advance until the next signal. This means that the cues in the script have to anticipate the actual moment when the slides overlap and the upcoming one is up to full brightness. At the meeting site, a different model was provided, which allowed for quicker action of the lamp because the drum on the dark unit advanced immediately after the lamp was out. This unit also permitted different rates of dissolve on fade-in and fade-out lamps.

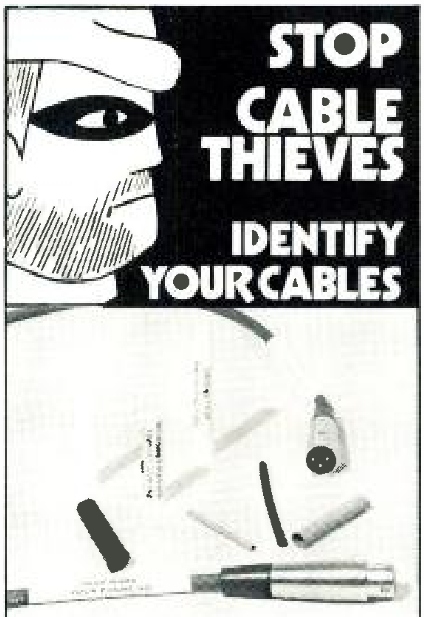
Although the slides themselves could not be switched any faster than the 1½ seconds it normally took the projectors to advance the drums, the cues on the script could be altered to fit the different unit, moved up closer to the actual points at which the slides would change on the screen. The increased rate of lapping that was possible with this model made for a more interesting effect on the screen. The show ran well, the client was pleased, and members of the audience remarked later to various upper echelons of the mother company that the presentation had a professional look. And the audio-visual company which had rented the equipment to the client was pleased with the speed with which the bill was approved.

CABLE JUMPROPE

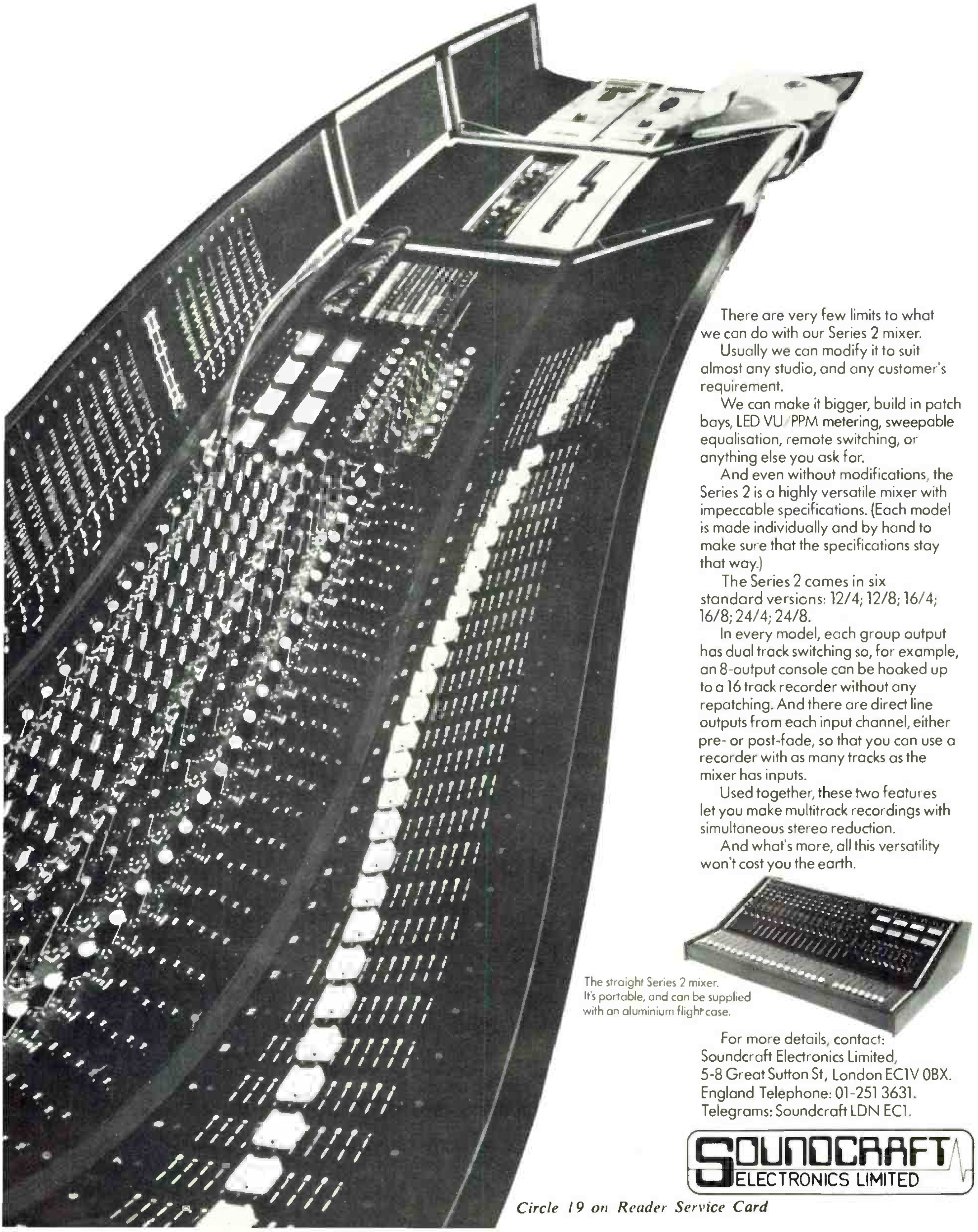
There was one thing that the a/v supplier did do that was a bit unnerving to the operator. The sound for the program, in addition to that issuing from the live microphone at the lectern, came from the 16mm film projector, the cassette player, and also a reel-to-reel tape unit. The operator was told to leave the cable plugged into the unit that was to be used next, then move it to the next one, and so on back and forth as required. This wasn't really a major tragedy, but it did give the operator one more thing to worry about, and also caused static in the sound system every time the plug was inserted.

The supplier was requested to bring in a mixer, or a matching box, so that all units could be left plugged in all the time. The latter alternative was used. No problem. (It might be of interest to those who had written to me when I did the column on putting in a pad from the speaker output of the 16mm projector to a line or mic input of a mixer, that the supplier did use a Shure A-15LA pad in the line. This network, complete like a line-matching transformer on a mic line, provided a 150 ohm output from the projector to the mixer on the room's speaker system.)

In one of the subsequent cities, the hotel had hired a local a/v company to furnish the rental equipment. The same line-up was requested. It was a good thing I called the day prior to arrival, to check. There was no 16mm projector on the list, and since the



Some of our customers' orders drive us completely round the bend.



There are very few limits to what we can do with our Series 2 mixer.

Usually we can modify it to suit almost any studio, and any customer's requirement.

We can make it bigger, build in patch bays, LED VU/PPM metering, sweepable equalisation, remote switching, or anything else you ask for.

And even without modifications, the Series 2 is a highly versatile mixer with impeccable specifications. (Each model is made individually and by hand to make sure that the specifications stay that way.)

The Series 2 cames in six standard versions: 12/4; 12/8; 16/4; 16/8; 24/4; 24/8.

In every model, each group output has dual track switching so, for example, an 8-output console can be hooked up to a 16 track recorder without any repatching. And there are direct line outputs from each input channel, either pre- or post-fade, so that you can use a recorder with as many tracks as the mixer has inputs.

Used together, these two features let you make multitrack recordings with simultaneous stereo reduction.

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England Telephone: 01-251 3631.
Telegrams: Soundcraft LDN EC1.

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Circle 19 on Reader Service Card

sound with images (cont.)

supplier's office was on the other side of this big midwest town, it would have been quite some time before he could get one over had we found out about the omission on the day we arrived.

FLUORESCENTS AND DIMMERS

At the hotel itself, the room this time was much larger than the other, combining two adjacent rooms. This meant divided light control for the dimmers, but not for the fluorescents; a micro-switch in the slide-track of the dividing doors tied together the fluorescents to either room's control panel, but not the dimmers. Each room had eight sets of incandescents, each with its own dimmer. A great setup if the operator could get two people to synchronize their movements. Anyway, the dimmers were preset with a minimum of light in the front, no light in the screen area, and enough illumination for the audience, in school-type seating, to take notes throughout. The fluorescents were left on only for the opening remarks and during the audience's entry and exit. That made it easy all around. (Incidentally, the lighting in the previous location did not allow for such fine adjustment, and a poor compromise was finally used.)

In this room, there was enough space to set up a riser platform in the rear of the room on which a long table could be placed for the projectors. This permitted the trio of slide units to stand side-by-side and allowed for quicker and smoother changes of slide trays. It also was a sturdier arrangement and had no jiggling of projectors or images.

It was interesting that the a/v supplier and hotel housemen had evidently worked together many times before and knew how they "always" set up the lectern and tables and screen and so on. But they had never used a side screen setup. The screen was also put along the front wall (opposite the entry doors). The "usual" way had been to place the screen between two pillars along one wall so that the audience entered from the seating left. This meant that the projectors had to be placed along the opposite wall. The throw distance was longer, the image larger (which was okay), but the image was also less bright, since incandescent sources had to be used for the dimmer effect of the dissolver.

The arrangement was changed to follow our suggestion, and the hotel was able to find a beaded screen in one of their other rooms. The image was larger and brighter, and they agreed afterward that the setup was better and they would recommend it to future clients. It also worked better since the ceiling in this room was only eight feet high (the other location had 14 feet ceilings), and the center chandeliers did not get in the way. (Both rooms discussed here had chandeliers which we were able to avoid in projection.)

The dissolver used here was different from the one used at the other site, but also allowed for the quicker effect of having the light change before the slides were advanced on the dark machine. Although the dissolve rates did not have the same variety of timings as were present in the other model, it did have a precise reverse in which the dark drum changed backwards before the lights changed. This kept the sequence correct in case a change had to be made in the slides during rehearsal.

All in all, it was a most interesting set of hops from city to city, and a great deal was learned by everyone. It was also enlightening in the matter of problem solving on the road. If everyone can learn from experiences and add it to the knowledge he already has, future presentations can be made bigger and better and with more confidence. ■

db new products&services

EIGHT-CHANNEL MIXER

- Each of the eight channels of Model 800S stereo mixer has separate low and high equalization as well as slide output level controls in the master section for left and right main and monitor functions. There is pre- and post-capability for monitor, reverb, and effects send controls, individual channel attenuation, and stereo pan. The rear panel features eight low (600 ohm) and eight high (50k ohm) inputs, left and right main and monitor outputs. An auxiliary input panel permits access to all mixing busses for compatibility with other consoles, tape machines, or other preamp signals to be mixed into the system. Included are two vu meters and a phone jack.

Mfr: Peavey Electronics Corp.
Price: $699.50.
Circle 50 on Reader Service Card

SUBMINIATURE AUDIO TRANSFORMERS

- A variety of miniature audio transformers are available for professional applications. All units feature semi-toroidal construction, low distortion, and wide frequency range. Non-standard transformers can be made to customer specifications.

Mfr: Beyer Dynamic (Revox)
Circle 53 on Reader Service Card

STEREO REVERB SYSTEM

- Dry/reverb output mixing, balanced line inputs, and line level drive into 600 ohms for studio, disco, and broadcast applications are offered in Model 242A stereo reverberation system. Other features include peak reading leds, input mixing for stereo return from a mono send, dual input level controls, and mic preamps. The unit can be used directly with a tape deck without the need for a mixing console or external mic preamps. The unit, which can be rack-mounted or used on a table top, is self-powered.

Mfr. Sound Workshop
Price: $450.00.
Circle 54 on Reader Service Card

LIGHTING EFFECTS PROJECTOR

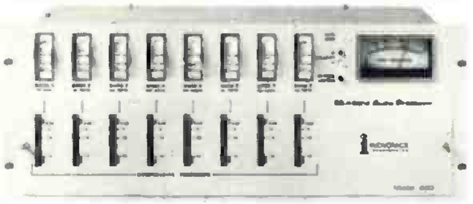
- Meteor 251 projector projects up to five effects simultaneously by means of a special accessory compartment located between the internal optics and the projection lens. All effects, operating from four low voltage sockets, are completely interchangeable. Other features include choice of 50, 300, or 2,000 hour bulbs, interchangeable lenses, squirrel cage cooling fan and high/low bulb switch. Additional attachments are available. The unit may be operated freestanding or mounted.

Mfr: Meteor Light & Sound (Revox)
Circle 52 on Reader Service Card

PORTABLE CONSOLE FRAME

- Front panel access in both horizontal and vertical positions and a rear shelf for power supplies are featured on RL500. The console accepts any 19 x 15¾ in. tape transport. The base section is 32 in. high, 25 in. wide and 29 in. deep; the instrumentation riser on top is 19 in. wide and adjustable to any height. The unit is constructed of 1/6 in. high pressure laminates.

Mfr: Rus Lang Corp.
Price: $350.00.
Circle 51 on Reader Service Card



Circle 45 on Reader Service Card

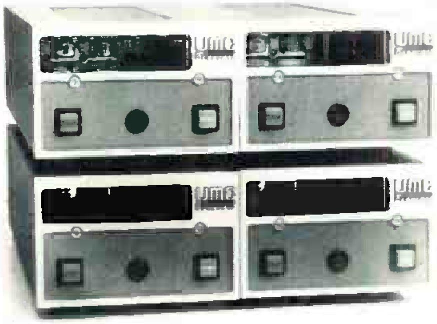
Circle 18 on Reader Service Card

7-PIN CONNECTOR

- Seven-pin Q-G (Quick Ground) connector series has a 7-circuit capability for multi-circuit interconnecting. The series includes 13 straight and right angle cord plugs and receptacles, as well as inserts for chassis and special purpose mounting. The connectors mate with all similar connectors; male and female inserts can be custom-mounted in panels to mate with plugs and receptacles with similar pin/contact arrangements. Features include a separate ground terminal electrically integrated with the housing, insert screw providing electrical continuity between the ground terminal and the ground contactor, one-piece high impact thermosetting plastic insert, positive polarization, and internal strain relief.

Mfr: Switchcraft, Inc.
Price: $4.85-$12.60.
Circle 61 on Reader Service Card

THREE-SLOT CARTRIDGE REPRODUCER

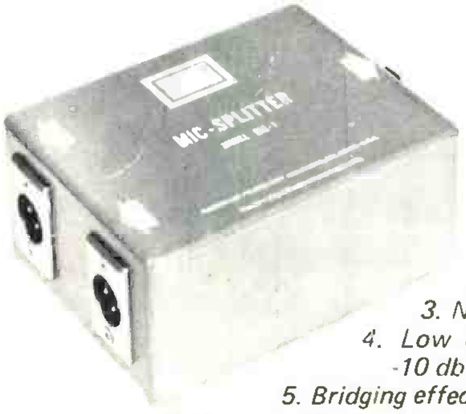


- Each of the four independent type 10 reproducers housed in Model 4D three-slot tape cartridge reproducer has an individual motor and power supply for each slot; if one unit develops trouble, the others are still operational. In addition, the units have independent tape drive systems. The ability to align each individual motor eliminates the shaft-to-deck perpendicularity problem, minimizing crosstalk often associated with this condition. 4D accommodates either four individual reproducers or two reproducers and one recorder/reproducer in the same desk or rack-mounted housing. Independent fast forward for each slot is also available.

Mfr: UMC Electronics Co. (Beaucart Div.)
Circle 60 on Reader Service Card

Sescom's "Mic·Spitter"

MODEL MS-1 LOW IMPEDANCE "MIC-SPLITTER"



1. *Used for separate monitor and P.A. mixing.*
2. *Keeps impedance constant.*
3. *No loss of signal (<1 db)*
4. *Low distortion — < .2% @ 30 Hz -10 db in.*
5. *Bridging effect on microphones.*
6. *Ground lift switch on right hand male XLR connector to eliminate ground loops.*
7. *Flat response ±1 db 20-20KHZ.*
8. *Rugged die cast housing.*

Pro Net: $38.15

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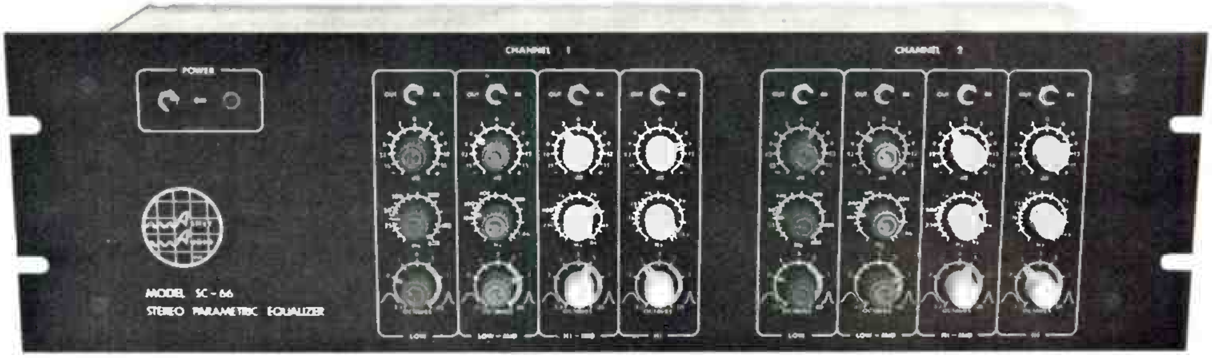
PARAMETRIC PRICE-PERFORMANCE *breakthrough*



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- All solid state M-241A led panel vu meter has no bearings or pivots, eliminating causes of overshoot. The meter contains 7 point red leds, vertical or horizontal, covering a range of −15 to +3 vu in 3 dB steps. Slow decay of the signal gives time for reading. The unit is made of shadow box bezel with hot stamped chrome scale on molded cycolac. The meter is inserted through a hole in the panel and clamped in the back.

Mfr: Pulse Dynamics Mfg. Corp.
Price: $30.00. (Quantity discounts available)
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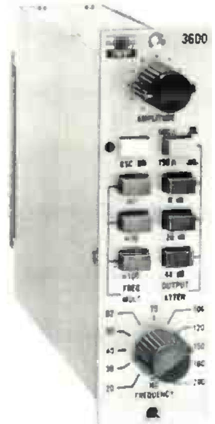
PRO LOUDSPEAKERS

- The crossover network designed into the speakers in this series is claimed to produce a flat frequency response with no identifiable characteristic. Pro-100 may be driven with a power amplifier up to 100 watts per channel. Midrange and tweeter drivers are in two planes, horizontal and vertical. Individual driver level controls are reached through an access panel which shows the frequency response curves for various settings of the midrange and tweeter controls. Frequency response is 35 to 20,000 Hz, ± 4 dB.

Mfr: H. H. Scott, Inc.
Price: $399.95.
Circle 58 on Reader Service Card

MODULAR PRECISION OSCILLATOR



- Accurate selection of 33 frequencies, covering the entire audio spectrum from 20 Hz to 20 kHz is provided by Model 3600 oscillator. Distortion claimed is typically 0.02 per cent, with stable output level over the full frequency range. The unit contains a built-in output transformer, selectable output, impedance of 600 or 150 ohms, and adjustable output level from −80 to +18 dBm. May be rack-mounted or installed in a console; a mating p.c. connector is supplied.

Mfr: Modular Audio Products
Price: $248.50.
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The Colorless Limiter.

What's a seemingly sane company like Orban/Parasound doing introducing yet another limiter? Well... for starters, because our new 418A Stereo Compressor/Limiter/HF Limiter is a direct descendant of our fabulously successful OPTIMOD-FM broadcast limiter—the one that's already been adopted by major groups and networks because of its unprecedentedly clean, natural, high-definition sound.

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In addition, the 418A incorporates a high frequency limiter with four user-selectable threshold time constants. As such, it's a natural for conditioning a signal to fit onto any consumer medium—like cassette—without high frequency overload distortion. Use it to mix through whenever time pressure is high—like demo sessions, or radio commercials. Use it in the broadcast production room to produce clean tape cartridges free from high frequency overload.

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STEREO EQUALIZER

• Each channel of Model 4100 equalizer has ten bands on I.S.O. octave centers from 31.5 Hz to 16 kHz. 10 dB of boost or cut is provided on continuously variable controls; all negative feedback produces equal Q in both boost and cut. Each channel has a variable low-cut control to provide 12 dB/octave of roll-off adjustable from 20 Hz to 160 Hz. Input level attenuators and overload indicators are provided on the front panel for each channel. An EQ IN-OUT switch and POWER switch control both channels simultaneously. The unit has an input impedance of greater than 40 kilohms and a recommended operating level of about zero dBm. Maximum output before clipping is +18dBm. Output impedance is about 100 ohms; the output circuits can drive loads of 600 ohms. The device incorporates low noise circuitry and magnetic shielding.
Mfr: White Instruments, Inc.
Price: $599.00.
Circle 55 on Reader Service Card

MODULAR MIXER CONSOLES

• Model 104-16X4A-16D is one of Series 104 and 108 modular mixer consoles, which include mainframes for eight, sixteen, twenty four, or thirty two inputs and four or eight stereo or mono mixes plus equalizers, panpot, four cue/echo sends, six step gain adjust with two input pad positions, and monitor-only solo buttons. Options include output mixdowns, talk-slate, graphic equalizers, and dual three-way tuneable crossover. Integrated circuits plug in; all sections are modular and plug in. Optional output transformers are available. All input modules have mic and line inputs plus module output and preslider breakin jacks.
Mfr: Interface Electronics
Price: (104-16X4A-16D) $3.900.00.
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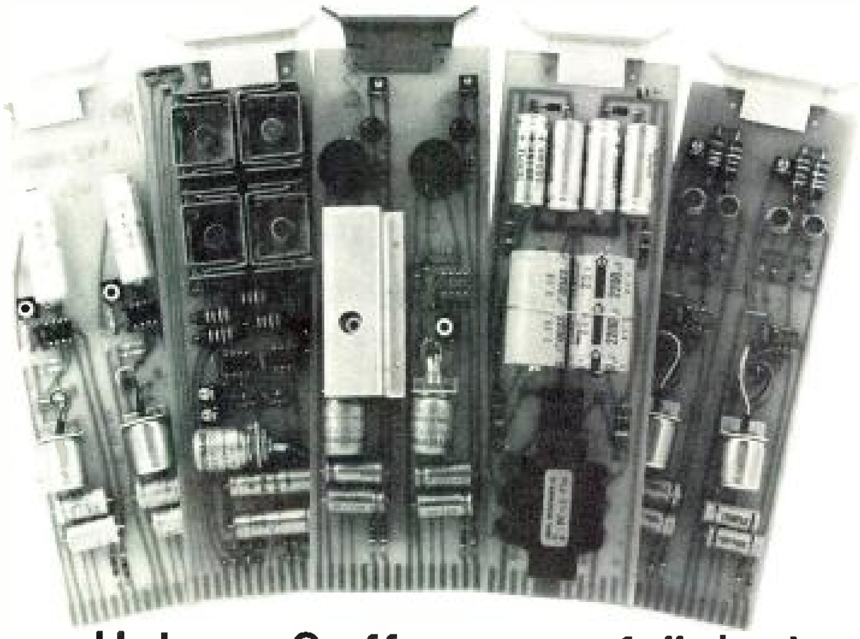
It's in the cards



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John Woram's The Recording Studio Handbook

FOR RECORDING ENGINEERS, TECHNICIANS AND AUDIOPHILES

The technique of creative sound recording has never been more complex than it is today. The proliferation of new devices and techniques require the recording engineer to operate on a level of creativity somewhere between a technical superman and a virtuoso knob-twirler. This is a difficult and challenging road. But John Woram's new book will chart the way.

The Recording Studio Handbook is an indispensable guide. It is the audio industry's first complete handbook that deals with every important aspect of recording technology.

Here are the eighteen chapters:

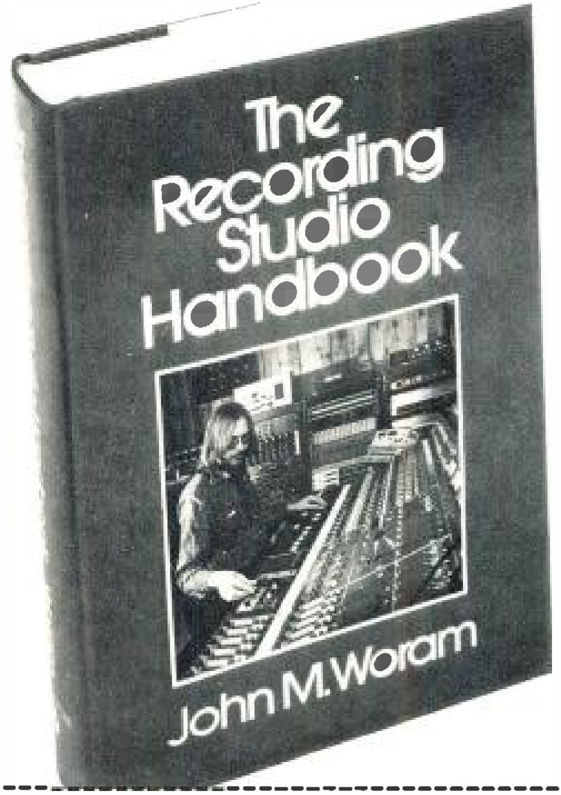
- The Decibel
- Sound
- Microphone Design
- Microphone Technique
- Loudspeakers
- Echo and Reverberation
- Equalizers
- Compressors, Limiters and Expanders
- Flanging and Phasing
- Tape and Tape Recorder Fundamentals
- Magnetic Recording Tape
- The Tape Recorder
- Tape Recorder Alignment
- Noise and Noise Reduction Principles
- Studio Noise Reduction Systems
- The Modern Recording Studio Console
- The Recording Session
- The Mixdown Session

In addition, there is a 36-page glossary, a bibliography and five other valuable appendices.

John Woram is the former Eastern vice president of the Audio Engineering Society, and was a recording engineer at RCA and Chief Engineer at Vanguard Recording Society. He is now president of Woram Audio Associates.

This hard cover text has been selected by several universities for their audio training programs. With 496 pages and hundreds of illustrations, photographs and drawings, it is an absolutely indispensable tool for anyone interested in the current state of the recording art.

Use the coupon at the right to order your copy of *The Recording Studio Handbook*. The price is only $35.00, sent to you with a 15-day money-back guarantee.



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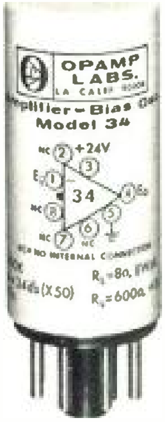

- Magnetic 100 kHz magnetic tape bias oscillator-buffer, designated model 34, is a dual purpose module used for general purpose signal processing, earphone-speaker power amplification, as a distribution amplifier, combine amplifier, microphone amplifier, or for magnetic tape erase-record sevice. The output is short-circuit proof. The device handles up to 1 uFd output capacitive loading, with 20V rms output at 1k ohm load as 100 kc oscillator.

Mfr: Opamp Labs Inc.
Price: $20.00.
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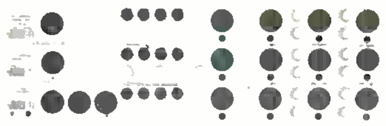
PANORAMIC MIXER



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LARRY ZIDE

The Making of the Ampex ATR-100

A visit to Ampex's Redwood City, California headquarters, and a talk with the development people, reveal the work that went into this state-of-the-art tape machine.

WE WERE SEATED in the office of Frank Santucci, the Ampex Audio Product Manager who, in many ways is the "godfather" of the ATR-100 project. Seated around the microphone of my cassette deck were Frank and the three engineers responsible for the development of this system: Robert P. Harshberger Jr., staff engineer who did the motors and control systems; Alastair M. Heaslett, senior staff engineer whose responsibility was the signal electronics; and Roger R. Sleger, senior engineer who created the mechanical system.

The resulting transcript of this talk took 31 typewritten double spaced pages but there was some doubling back and over-detailing that is not important. What follows will show the brain work that went into the ATR-100.

The first area of discussion centered on the use of ferrite tape heads on the machine. Ampex is not the first such user in pro audio, but ferrite has yet to be common as a replacement for classic metal laminated heads.

"The answer has to be multifold. The first reason is the very greatly improved longevity of the head. From a professional user's point of view, what this means is that adjustment of the machine to maintain a certain level of performance need not be made as frequently as with a metal head. It is no longer necessary to compensate for the fact that its performance as a transducer is changing during its life as the metal is worn away.

"The second reason is that it becomes possible to create a reproducer head where the noise and the high frequency region is not dominated by intrinsic noise generated by the head.

"The third reason is that because of this low loss, it becomes possible to use a biasing frequency which is sufficiently high, so that at the higher tape speeds (high in level or high in frequency), we can avoid or substantially reduce the effects of bias modulation noise at the higher tape speeds. Now, as tape's short wave length capability gets better, a greater and greater amount of bias signal is left recorded on the tape. In this respect, the bias is no different than any other type of signal that produces modulation noise.

"For example, at 30 inches per second, you can use a bias frequency of 150-250 kHz, which is common today (ATR-100 uses 432 kHz). When you take a tape you just recorded at 30 in./sec. with no signal going into it, rewind and then move it slowly past the head, you will actually hear a discrete signal coming back off the tape—recorded bias. True, when you play it at 30 in./sec. this frequency is 150-250 kHz, so one might say therefore that it doesn't matter. But the point is that the discrete signal having been recorded is saturating the tape. The effect on the over-all signal-to-noise ratio (the bias noise signal to noise ratio) at 30 in./sec. is really quite substantial.

If you attempt to produce a conventional laminated metal head with any kind of lifetime at all, that had to run with 400+ kHz bias frequency, you will find that you require an exceptionally large amount of bias power pumped into the head, most of which would be dissipated as heat losses in the head and very little of which would arrive at the gap as a useful plus. That's another reason why ferrite head technology was turned to. It permits us to realize significant performance improvements, particularly at the higher tape speeds."

TAPE AND MACHINE

One of the main points brought out in the discussion was the interrelationships that existed among divisions of the company. The development of the ATR-100 was interfaced with the construction and marketing of Ampex Grand Master tape—with the tape actually hitting the market place sooner.

Frank Santucci at his desk.

It came out that the tape division and this group worked closely over more than two years in which tape samples flowed in, were evaluated and changed, and a machine took shape as well.

It turns out that the machine did not really take final shape until the tape existed, then the final stages could be completed and an ATR-100 launched.

MECHANICAL CONSTRUCTION

The pinchroller-less design of this new machine is an obvious feature. It was revealed that this was a design feature that was sought right from the beginnings of design.

The ATR-100 is capable of handling 14 inch reels—though it is to be admitted that they are not yet readily available to studios. It is expected as more machines enter the field—as is now happening—these reels will come forth as well.

As for the mechanical construction, it was decided that a constant tension transport was needed, tension to be held constant regardless of reel size or position of tape on the reel.

"Once we made that decision, we looked at the tape transport and realized that the tension differential across the capstan is going to be constant since friction is relatively constant in tape. And once this is studied for a while, you become aware that a pinchroller isn't really needed because you can control that differential tension closely with pure electronics means by servo controls, so you can then do away with the pinchroller.

"Essentially this is what we did. We went a step further. Most machines would normally have a pinchroller covered with rubber, a high friction material. We don't do that, in fact, we use an anodized capstan which is relatively low friction. You can be running along in a play mode and grab the tape and the capstan will slip, even though the wrap is 135 degrees.

Alastair Heaslett, invariable cigarette in his hands, talks while Roger Slegel listens.

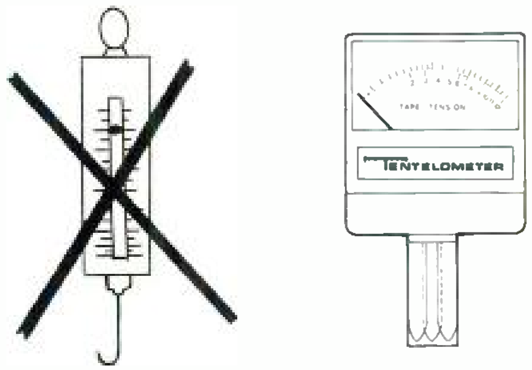
"This system, however, absolutely eliminates slip during normal motion. In fact, the grab from the capstan to the tape is so good that if you misadjust the tensions on the machine deliberately, you'll find that while the machine itself won't function suitably, it will still pull the tape. So there's a lot of margin."

FLUTTER

One of the demonstrations done at the recent AES show was to take the two reels and deliberately move them off center and run them, and still no wow or flutter was created.

"The servo is a lot more powerful than it need be for normal operations. The motors are a lot bigger than are really required to move that tape, they are rated at ¼ horsepower. It is a bi-direction servo in that you are never required to use the tape to pull the reels around. There is much less transient type of tension disturbance. The force that you're able to get with a reel servo is approximately 130 inch-ounces in either direction, which is twice as much as you can get on any other ½-inch mastering type of machine. Therefore, it can keep up with the offset reel.

"Another part of the servo is the way that the tension is sensed. The tension arms are actually driven magnetically, they are not spring-loaded. They are driven by a rotary solenoid and the force that they exert on the tape is controlled by the current throughout the solenoid. The force is also relatively independent of the position of the arm. So, while the position of the arm is used to sense the tension, it indirectly actually senses the actual position of the arm—which then controls the motor. The fact that the arm can move slightly does not affect the tension. It just controls the position of that arm and the reel will feed or take up in that position."



This time Heaslett, cigarette still there expounds while Bob Harshberger listens. But in spite of these appearances, both Harshberger and Slegel also contributed heavily to these discussions.

CAPSTAN SYSTEM

My next question had to do with whether the capstan itself enters into the servo operation.

"Yes. It is a phase lock capstan. The capstan moves—it is being directly driven—the reel servo control logic senses that the capstan is moving in a particular direction, programming these torques for the proper tensions. The tape will then follow by virtue of the errors created in the reel circle. It will just follow the capstan in either direction.

"On the fast modes (fast forward or rewind) the capstan functions as a velocity servo rather than a phase-lock servo. It goes to a certain velocity at a constant acceleration. When it reaches that velocity it stays at that velocity. The reel servos follow in the same way that they did in play modes. So that no matter which direction it is going, it creates the same error. In fast forward the tape simply accelerates to the maximum speed that the motor is capable of.

SERVO RELATIONSHIPS

"Now there is a control action as well. Under ordinary circumstancces (10 inch NAB reels), ¼ inch tape) it accelerates at a fixed acceleration to a top speed, stays there, decelerates if you press the stop button, at the deceleration that it accelerated at. However, there are conditions such as with a 14 inch reel with ½ inch tape, where it is not advisable to accelerate that big of a pack as fast as the capstan can accelerate. Remember that there is no control given to the reel servos, it's only applied to the capstan.

"So if the capstan is told to accelerate at this fixed rate, you do not want the reel to accelerate at that rate. It senses the extra large error in reel servos and controls the rate of acceleration of the capstan so the capstan will never accelerate faster than the reel can.

"You also have the opposite extreme where if you have a very tiny hub, or if you wound tape right on the spindle, the reel would be going much faster than the voltages in the reel servo would allow. So the capstan servo also senses that the voltages on the reels have risen to a high level and slows down the capstan. So if you have a very small reel on one end, as you are rewinding it actually does slow down."

MECHANICAL ACTION

One of the most impressive things about this new machine is its mechanically smooth and quiet operation. My next questions had to do with these aspects.

"We should talk about the guiding and the acoustic quietness of the machine. The tape lifters on the machine are conventional type solenoid operated lifters, but they are damped. So when you operate, you don't get any clunk

A closeup of the ATR-100 deck.

at all from the machine. In fact, the lifters are adjustable in the setup—so set them so you just don't get the clunk.

The capstan noise factor is down because the rpm of the motor is down. Note that the size of the capstan itself is about 2⅜ inches in diameter. It happens that one revolution of the capstan is exactly 7½ inches of tape. We wanted it to be intervals of time, 30 in/sec being four revolutions per second. The size of the capstan is significant for many other reasons. One is the area of the capstan's circumference. Because with the small diameter on most machines, you're very subject to run-out effects from the capstan. A run-out is when (as the shaft rotates) it is not really rotating around the center.

"It may be slightly bent so the effect of the driving radius to the tape varies. The shaft simply can't be made or supported in its bearing perfectly. That run-out translates directly to flutter. It becomes a velocity error around the tape. The larger you make the capstan radius, maintaining the same number for the run-out, say 1/1000 inch, the more you proportionately reduce your flutter."

Next month we will conclude this discussion, beginning with the four speed operation of the recorder. ■

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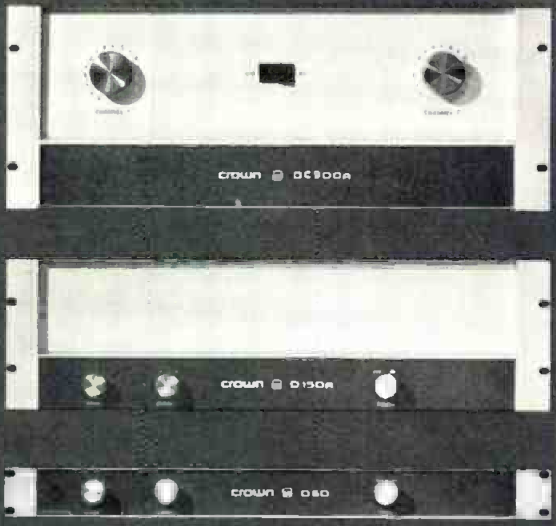

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MORTIMER GOLDBERG

The Art of Tape Editing

Smooth transitions and the undetectable deletion of unwanted material are achieved through expert maneuvering of tape and precise knowledge of just where to break.

THE DEFINITION of *edit* is to prepare for use by compiling, arranging, emending, etc. As applied to recording tape, this article will explore the techniques utilized in the assembling of material into a cohesive unit of *sound*, in the form of speech, music, or both in combination. Music has its rhythm, tempo, melodic variation, speech its cadence and inflections; the musical beat is synonymous with the pacing of speech. An orchestra conductor *edits* music by controlling the orchestra into a continuous and harmonious blend of instruments and a tape editor's function is to combine his material with like precision.

EDITING OF SPEECH

One of the initial aspects in the editing of speech is to correct the imperfections of speech delivery. These imperfections include, the mistake (fluff), natural hesitations, and the audible thought seeking sounds of "um" and "ah." These are most important in editing for broadcast, where the race against the clock requires maximum skill in editing to correlate all thoughts in an intelligible and natural sequence. The time factor doesn't necessarily apply to the editing speed, but more importantly, to the quantity of material information delivered in a prescribed time.

What is of prime importance in the natural sequence of speech is *pacing*. This naturally varies among individuals. Some speak rapidly, and some slowly. Some enunciate well, while others do not. Regardless of the type of speech pattern encountered, there is one thing that is common to all, and this is the *breath*. All words uttered are done so while exhaling; when the air supply is depleted, a breath has to be taken before continuing. This is what natural pacing is dependent upon in each individual speech pattern.

BLOCK CUTTING

For the moment, let us consider *block cutting*. This is the elimination of complete sentences, or thoughts, with no internal editing involved. In order to illustrate the cutting method necessary to preserve the natural pacing, refer to FIGURE 1.

A1 and Z1 are the first and last words, respectively, of sentence 1. B1 is the breath following sentence #1. The same configuration applies to sentences 2, 3, and 4. If we want to keep sentence 1, and cut to sentence 4, eliminating sentences 2 and 3, our sequence would be:

(1) Z1 + B1 + A4

This pseudo-algebraic formula indicates that the cut is made just prior to A2, maintaining the breath B1, and joining at the beginning of A4. Using the same method to indicate the *incorrect* means of making the splice, we might come up with

(2) Z1 + B3 + A4

In formula 2, the cut is made just after Z1, and joined to the breath B3, which precedes sentence 4.

You might ask what difference it makes; a breath is a breath. In some cases, this may be true, but usually, it is not. There are two distinct reasons for it *not* being the case.

Because sentences 2 and 3 have been eliminated, the joining of sentences 1 and 4 is the only thing discernible to the ear. The *breath* that followed sentence 1 is in cadence with the speech delivery to that point, and will flow smoothly to the start of sentence 4. Consider the alternative of using breath 3. Sentences 2 and 3 might have had an increase in verbal exuberance, and hence uttered at a much quicker pace than what preceded. As a result, breath 3 would not be in keeping with the pace of sentence 1, and an unnaturalness would be noticeable.

The second reason is more significant. There is a natural reverberation to the voice that is not readily noticeable

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Mortimer Goldberg is technical supervisor with CBS Radio, New York. His distinguished 25-year career includes the editing of numerous documentaries.

when listening to speech, either live or recorded. The degree of this reverberation is dependent upon the surrounding environs. There is less in acoustically treated rooms, such as broadcast or recording studios, and more in hard-wall or glass-enclosed areas. The voice of a public speaker in an auditorium, on a public address system, will have a great deal of echo as it reverberates though the auditorium from the loudspeakers. This, of course, is extreme reverberation. However, the sustenance of sound, from one syllable to the other, is ever present in normal conversation.

Referring again to Figure 1, Z1's final syllable reverberates into the breath, B1. The joining at the start of A4, as previously stated, will flow in natural sequence. Consider the other possible splice—Z1 + B3 + A4. Breath 3 (B3), contains the reverberation of Z3, and when spliced to combine with A4, in addition to the unnatural pace, will contain the extraneous sound of the Z3 reverberation content.

The difference between the two methods is that the first will be seemingly untouched, whereas the second will be unmistakably rcognized as altered or edited. The recognition is more acute to the trained ear, but still apparent nevertheless to the average listener.

FINDING THE EDIT POINT

The first requirement for locating the *edit point* is to have the ability to recognize the sound makeup of speech at very slow speed. This speed is not regulated by the electrical movement of the tape, but by manipulating the tape by hand, past the playback head. One method is to maneuver the feed and takeup reels in a to and fro fashion, with the mechanical brakes engaged. A fair amount of coordination is necessary for the rotation of each reel in unison. I have found that it is more effective to hold the takeup reel alone, and put the machine in *rewind* mode. The mechanical brakes are disengaged, and the hand pressure on the take-up reel prevents the tape from moving to the feed reel. This slight hand pressure enables you to move the tape freely, back and forth in a rocking motion, with one hand and with considerably more control. You are able to seek the identifying sound smoothly, without the squeak associated with the engaged mechanical brakes, when rotated in the *stop* mode. Release for cutting is made by merely pressing the *stop* button.

The identification of phonetic sounds, for the purpose of separating syllables, is the key to speech editing. The hard sounds of consonants are more easily recognized than the soft sounds of vowels. The sounds of B, K, and D are percussive in nature, because they are formed by contact—B with closed lips, K with the back of the tongue to the palate, and D with clenched teeth. The percussiveness, that makes the sound easily identifiable, is perceived by rocking the tape past the head at slow speed.

Let's use the word *corporal*, as an example. The start of the word will sound like a click. The P, in like manner, will sound like a puff. The softer intermediate segments of the word will not have as pronounced a definitive identification, as the C and P, but will be recognizable by tonal characteristics.

The vowels, A, E, I, O, U, are the most difficult to identify and separate because of their lack in percussiveness. Sounds of this nature are made with parted lips, and the character of the sound is produced by changing the form of the parted lips. Just voice aloud—a, e, i, o, u. Notice how each sound blends with the next, using limited mouth movement. It is helpful to listen to the material backwards, employing the rocking motion previously mentioned. The comparison of the sound, between forward and reversed motion, often clarifies the division between syllables. A vu meter is also often an invaluable aid. A separation, not discernible to the ear, is often indicated on the vu meter by a dip of the pointer.

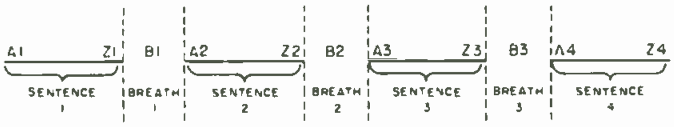


Figure 1. The cutting method used to preserve natural-sounding pacing.

It is essential to remember that when editing speech, it may be necessary to alter the *existing correct pronunciation*, for the sake of preserving "pace" and "naturalness." For example, *and* and *but* are two conjunctives used very frequently. In an original recording, the statement may go as follows:

> *Joe and David were going fishing and he decided, that is, David decided, to ask Bill to go along.*

The portion to be eliminated is *He decided, that is*, so that the statement would be:

> *Joe and David were going fishing and David decided to ask Bill to go along.*

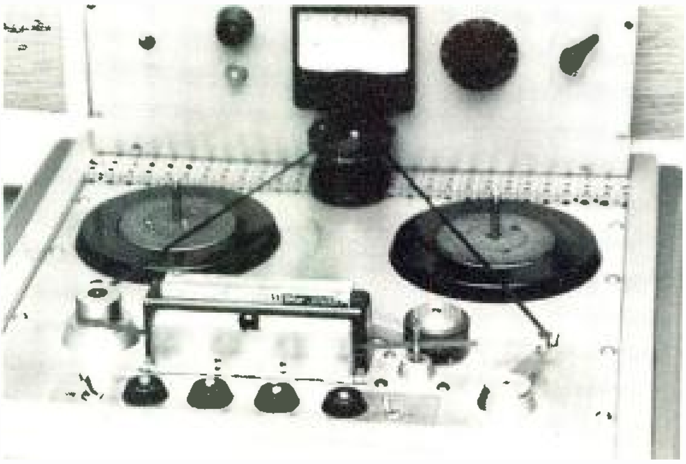
Cutting at the end of the complete word *and*, and splicing to *David* is a true representation of skillful cutting, which includes all components. Although *and he* flows perfectly, *and David* has the pacing destroyed by the stuttering effect of two consecutive *D* sounds. This is what is termed a *jump cut*. The *and* should be cut before the *d* as *an*, and spliced to *David*. The pacing is preserved and the result sounds unedited. This situation will always occur when *and* or *but* is cut from preceding a word beginning with a vowel, to before a word beginning with a consonant. The described editing techniques in these cases will always prove successful.

Everyone, whether speaking in conversation or being recorded, usually will never consciously mispronounce a word without immediately correcting the error. When editing this mistake, or fluff, using the corrected word completely usually results in an unnatural sound. Mistakes of this kind frequently occur in tongue twisting multi-syllable words, i.e.

> *After his application was received, he was given an applointment—appointment—for an interview.*

The corrected word is fine in its entirety, but the problem exists in the separation from *given an* to *appointment*. There is reverberation in the voice that carries through from syllable to syllable. In this case, the *N* sound from

Figure 2. Method of threading short loop on a recorder.



AN. blends directly to *AP* of *appointment*. Voicing this aloud will confirm the fact. Even a skillful separation after the *an*, to the corrected *appointment*, results in an abrupt interruption of the word *an*. This is due to the remnant of its final *N* sound, blending into the *A* of appointment. Leaving the *A*, of the first word and cutting to the *P* of the corrected word eliminates the problem. The advantage, in this instance is twofold. Cutting to the percussive *P* is easier than cutting on the vowel *A*.

INFLECTION

My experience as tape editor, on numerous CBS Radio documentaries, has been that recorded material destined for the cutting room floor holds the key to a successful editing session. Considerably more recording time is devoted to an interview than the actual program time allotted to it. Consequently, the initial editing consists of block cutting, for the purpose of extracting pertinent subject matter. (In this process, the unwanted material is often allowed to run off on the floor. I always wind this material on to a reel.) This tape is referred to as *out takes*. The remaining material must now be condensed to comply with the allotted allowable time. The correction of fluffs, hesitations, etc., is an aid to this time factor.

The constant use of parenthetical phrases are very time consuming, and do not contribute to the desired program subject matter. It is desirable to eliminate them, but in the process, additional problems are created. For instance:

> *and so the building was scheduled for demolition, because as I said before, the property was sold to the developer.*

In the interest of time, *because as I said* etc. is not necessary, since it was stated earlier. *Demolition* should end the statement, but the *inflection* is slightly up, due to the succeeding phrase which previously ended the sentence. The means for correcting the problem comes from an investigation of the *out take* reel. It is only necessary to locate a word with the same suffix *tion* (shun), with a *downward inflection*. It may seem to be a needle in the haystack situation, but not so. More often than not, the means of correcting mistakes or inflections by syllable substitution is available. The trick is to listen, recognize what is needed, and with practice, the surgery can be quite successful.

BACKGROUND SOUNDS

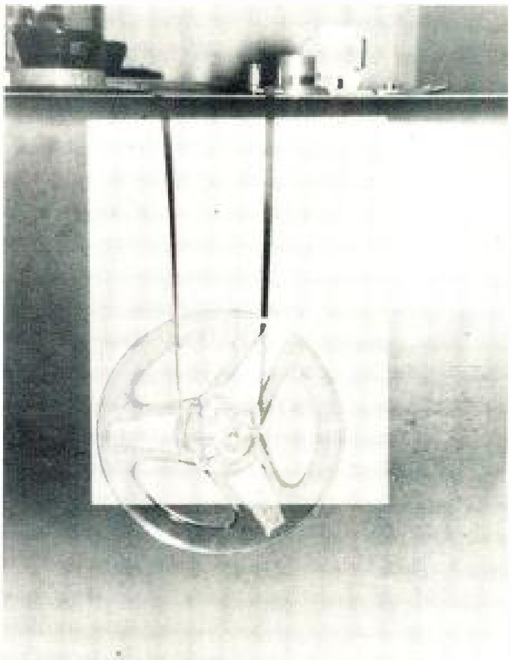
In terms of tape recording, the "background sound" is considered as some action that is occurring while a person is speaking. This could be music, street noise, office machines, etc. There is always background sound present, regardless of conditions, even in the ideally quiet setting of a broadcast studio. If you open one microphone in an empty studio and listen to the monitor speaker, a sound is immediately apparent. There is nothing of significant level indicated on the vu meter, but sound is readily discernible to the ear, even at the monitor's normal volume setting.

Consideration of the most subtle ambient sound is of substantial importance in tape editing. The use of a blank piece of tape for a pause instead of the "quiet studio sound" would result in loss of sound or *drop out*. In radio broadcasting, numerous recorded inserts are pesented during the course of a newscast. These inserts are termed *actualities*, which are on-the-spot statements by persons describing situations, or informing the interviewer on matters pertinent to the news item being reported. Invariably, there is a backgound sound behind the statement. Ideally, when the statement ends, the sound is faded out before the newscaster resumes speaking. This is particularly true for telephone recordings, which, because of the nature of the medium, have an inherent sound of "line noise." For radio listening, that fade gives a smooth transition from item to item.

Television production rarely bothers with this audio consideration, because the ear accepts an abrupt change when the eye views a new scene. In a t.v. newscast, there may be a picture of a battle going on with the associated sound, and at the conclusion of the segment, the scene switches

Figure 3. Extending the loop length and placing it through an empty 7-inch plastic reel.

Figure 4. A loop weighted down on one side of the machine through an empty plastic reel.



to the anchor man with a simultaneous absence of the background sound just heard. Seeing the anchor man seated before a microphone, in an entirely different environment, seems perfectly natural to the viewer and the abrupt removal of sound is not particularly noted. Radio listening is entirely different because the ear, when functioning without the eyes, is much more sensitive to audible change. This fact can be easily proven by listening to t.v. audio without watching the picture.

Due to the consolidation of material from numerous sources which are combined through narration, the production of radio documentaries require utmost consideration of background sounds. On-location recordings often are made in different areas for the convenience of persons being interviewed. Different interviews, film clips, etc., pertaining to the sequence, consolidated through editing, have varied background sounds, ranging from very quiet to very noisy. These variations occur not only because of change in ambient sound levels, but because of the editing out of verbal content. The informational material may flow very smoothly, but the sudden changes in ambient sound result in unnatural-sounding sequences. In order to remedy this situation, a common background sound, excerpted from original material, is *mixed* behind the entire sequence. The nonsequitar backgrounds are thereby melded into a common, and essentially integral part of the presentation.

This same augmented sound may be mixed behind the commentator as he bridges the various topics of the interview, despite the fact that he wasn't present during the original recording. The purpose is not to deceive the listener into the belief that what they are listening to occurred as presented. The commentator invariably qualifies the fact that he was not present. The augmented sound merely contributes to the cohesiveness of the overall product.

THE LOOP

The method used to accomplish this background is the *loop*. Refer to FIGURE 2, which pictures the closed end of tape threaded through a recorder. This particular-sized loop is adequate for a background that has a distinct pattern which, when repeated regularly, would be normal, such as the sound of a trip hammer or pile driver. Access to sound in the clear is often only available in short segments, extracted from pauses during the interview. In order to obtain as much tape as possible to form the loop, it is best to dub the sequence at the highest available speed. If the original recording was made at 7½ in./sec., then dubbing at 15 in./sec. will give twice as much tape to form an adequate loop size. Even at 15 in./sec., numerous dubbings may be necessary in order to obtain sufficient tape length to form a playable loop.

When sounds are very general, and not as particularly distinctive as the previously mentioned trip hammer, the completion of an adequate loop is much more difficult. The vague murmur of voices is immediately transformed into a constant annoying repetitive pattern when placed in short loop form. To avoid this effect, as many varied sections as possible must be dubbed and edited together, so that close repetitive sections are eliminated. Consequently, considerably more tape is involved to form the loop. FIGURES 3 and 4 indicate how this lengthened loop can be made to play effectively by the weighting action of an empty 7-inch reel. In the majority of cases, the lengthened loop will serve adequately as a background without noticeable repetition.

However, if the nature of the available sound is such that frequent repetition is unavoidable, then further corrective measures are indicated. The lengthened loop is recorded on another tape and then the two mixed together to form the final loop. The only thing necessary in this recording process is to have the two recordings displaced in order to avoid synchronism. This step is invariably adequate in obtaining a suitable background, of authentic origin, for use in a highly edited segment. In order to free a tape recorder for use in the final production, the loop is often transferred to the cartridge. The cartridge is made by deactivating the primary cue so that a 30 or 40 second cartridge will run continuously without stopping.

EQUALIZATION & FILTERING

The use of frequency discriminating equipment is generally an aid to the tape editor. On-location recordings are usually performed by reporters utilizing cassette machines. Frequently, in the haste of obtaining an interview, the proper microphone technique is neglected, and numerous sections of the recording have an "off mic" quality. The fact that the person is off mic also results in a voice level lower than normal.

Level adjustment can compensate for the loss, but additional noise is encountered. The use of an equalizer, with a boost in the 2000-3000 Hz range, does wonders for the off mic section. This frequency spectrum is in the voice mid-range, often referred to as the *presence area*. The effect is such that the voice is seemingly brought closer to the microphone, with increased clarity. The pre-emphasis of the equalized section causes its loudness to compare with other non-equalized normal sections of the tape. Invisible intercutting between these sections therefore becomes more feasible.

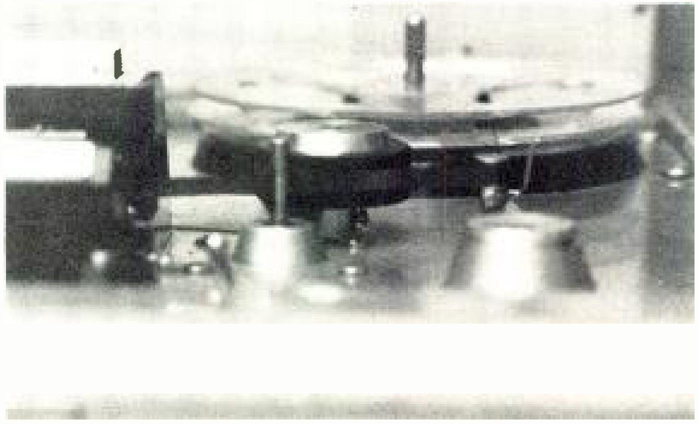
Quite frequently, due to a malfunctioning recorder, the resultant material sounds muddy or muffled. By *attenuating* the frequencies in the 100-200 Hz range, and again boosting the 2000-3000 Hz range, the clarity of the recording becomes considerably improved to acceptable intelligibility. The elimination of 60 Hz hum, or other steady tone oscillation is accomplished by the use of a notch filter. This device discriminates sharply between discrete frequencies; if it doesn't completely eliminate them, it attenuates them to a more acceptable degree.

MUSIC EDITING

The tape editing of music requires the additional skill of recognizing discrete tonal identification, commonly known as perfect pitch. This is not to say that this talent is absolutely essential, but an editor with perfect pitch is able to intercut or blend musical passages more accurately, with fewer attempts.

The needs vary to an extensive degree. In a professional recording session, the one-time recording performance is not unusual, but more often corrections are made without starting from the beginning. The pickup is recorded a few bars prior to where the error occurred. In editing, a convenient point is selected for ease of joining. This may

Figure 5. The normal threading of a tape machine.



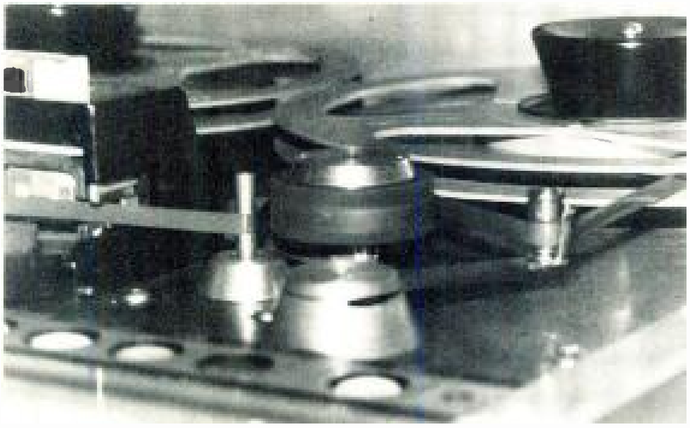


Figure 6. The reverse threading of a tape machine.

be a beat, a pause, or whatever effective area is convenient. The only requirement is that the joining be perfect.

More difficult situations occur when alterations of existing music recordings are necessary. This may involve the elimination of a chorus in order to shorten a selection, or excerpting passages for use as musical bridges in a dramatic sequence. Excerpted music is usually employed with a positive start, and then faded out when no longer needed. A more sophisticated treatment will have a natural ending at the desired time. This type of editing results in a newly created musical passage of specific time duration, to satisfy any and all needs.

Direct splicing within musical passages is the exception rather than the rule. A pick up on a "hold" note followed by, possibly, a drum beat, could readily be accomplished by a direct splice. More often, it is necessary to take the pickup point from one tape and blend, by mixing, from another tape to a third recording machine. This process requires two identical tapes of the original material, for the purpose of eliminating unwanted passages by selectively blending from one tape to the other. Unwanted portions of the first tape cannot be allowed to continue under the *blend point.* Blank tape is inserted at the end of the desired section, so that there is no possibility of unwanted sound. Very frequently, leader tape is used, so that there is visible indication of the end of the needed recorded portion.

The use of an echo chamber or a reverberation unit offers great advantage when musical separation of sounds is very short at the point of the desired blend. Cutting after the desired note will always sound abrupt, but re-recording this segment through an echo chamber will extend the note sufficiently in order to blend in *tempo* with the de-desired pickup point.

Precise timing is essential at the blend point in order to preserve the tempo. The tape must be in motion, at its proper playback speed, at the *mix point,* to avoid a pitch variation. Calculation of tape time to and from the point of mix can be accomplished by stopwatch timing. However, reaction time must also be considered and the entire process can become cumbersome. A very valuable technique is using the constant speed of the tape recorder transport to time the segment.

FIGURE 5 shows the normal threading of tape in between the capstan and pressure roller, and on to the takeup reel. The capstan revolves in a clockwise direction and controls not only the tape speed, but its direction. Examine the threading of the tape in FIGURE 6. The tape is placed up and around the pressure roller, and goes between it and the capstan from the opposite direction. The clockwise rotation of the capstan moves the tape *toward* the feed reel. In the play mode, with separate feed and takeup motors, there is no mechanical linkage and electrical braking causes each reel to accommodate the tape in whichever direction it moves. We therefore have a method of maintaining the normal position of the tape on the recorder, and playing it backwards at the operating speed.

Applying this technique to the mixing of two tapes requires the following procedure:

Tape 1 contains the first portion of the musical segment. At the point of blend, where no further sound is needed from tape 1, a length of leader tape is spliced in.

Tape 2 is prepared with leader tape prior to the pickup point for the mix with tape 1. Pot settings are determined in advance for proper level into the third tape recorder.

On Tape 1, the desired point of transition is set at the playback head. The pickup point on tape 2 is set at its playback head. Each tape is carefully threaded around the pressure roller and in reverse direction past the capstan, as previously described. Care must be taken that the desired blend point on each tape is at their respective playback heads. When all conditions are satisfied, the *play* button on each machine is pressed *simultaneously.* Both machines play backwards at their operating speed. The time duration of this *backward play* is arbitrary. The usual time is approximately ten to fifteen seconds, mainly for reaching a convenient spot for editing the new mix into the original material. When this is satisfied, both machines are stopped *simultaneously.* Carefully restore the normal threading of the tape without any change of position of the tape at the respective playback heads.

(a) The third machine is started in *record mode.*
(b) The mixer position for tape 1 is *open.*
(c) The mixer position for tape 2 is *closed.*
(d) Start tape 1 and tape 2 *simultaneously.*
(e) When the leader on tape 2 enters the head housing, *open* the tape 2 mixer position.
(f) When the blend occurs, tape 1 will be in leader tape. Close tape 1 mixer position to prevent any audio that is present on the tape after the leader portion from being recorded.

It is evident that the use of leader tape gives a convenient visible indication for the need to control the mixer positions. The most important characteristic in employing this technique is that aside from its placement accuracy, the tape recorders are amply stabilized in speed at the mix point.

VOICE & MUSIC SYNCHRONIZATION

This backtiming technique is also an invaluable aid in mixing voice with music, when it is desired to have the music accent on a particular word or phrase. The use is very significant in the announcement of a program with a musical signature or theme. For example, the theme starts with a trumpet fanfare, followed by a tympani and cymbal crash before continuing into the melody of the signature. The announcement is to start after the cymbal crash. The music tape is cued on its machine at that point, and the start of the announcement cued at the playback head of its machine. The simultaneous backtiming is as previously described. No internal editing is required because the whole musical sequence is required from the beginning. It is only necessary to have enough blank tape at the start of the reels so that the tape won't run off the reel before the completion of the backtiming process.

The applications of this technique are numerous, and any situation involving critical mixing of two tapes will be completed with very satisfactory results.

The recording and editing of tape is a fascinating thing, whether it be for vocation or avocation, but its degree of excellence is only attained through interest and practice. ■

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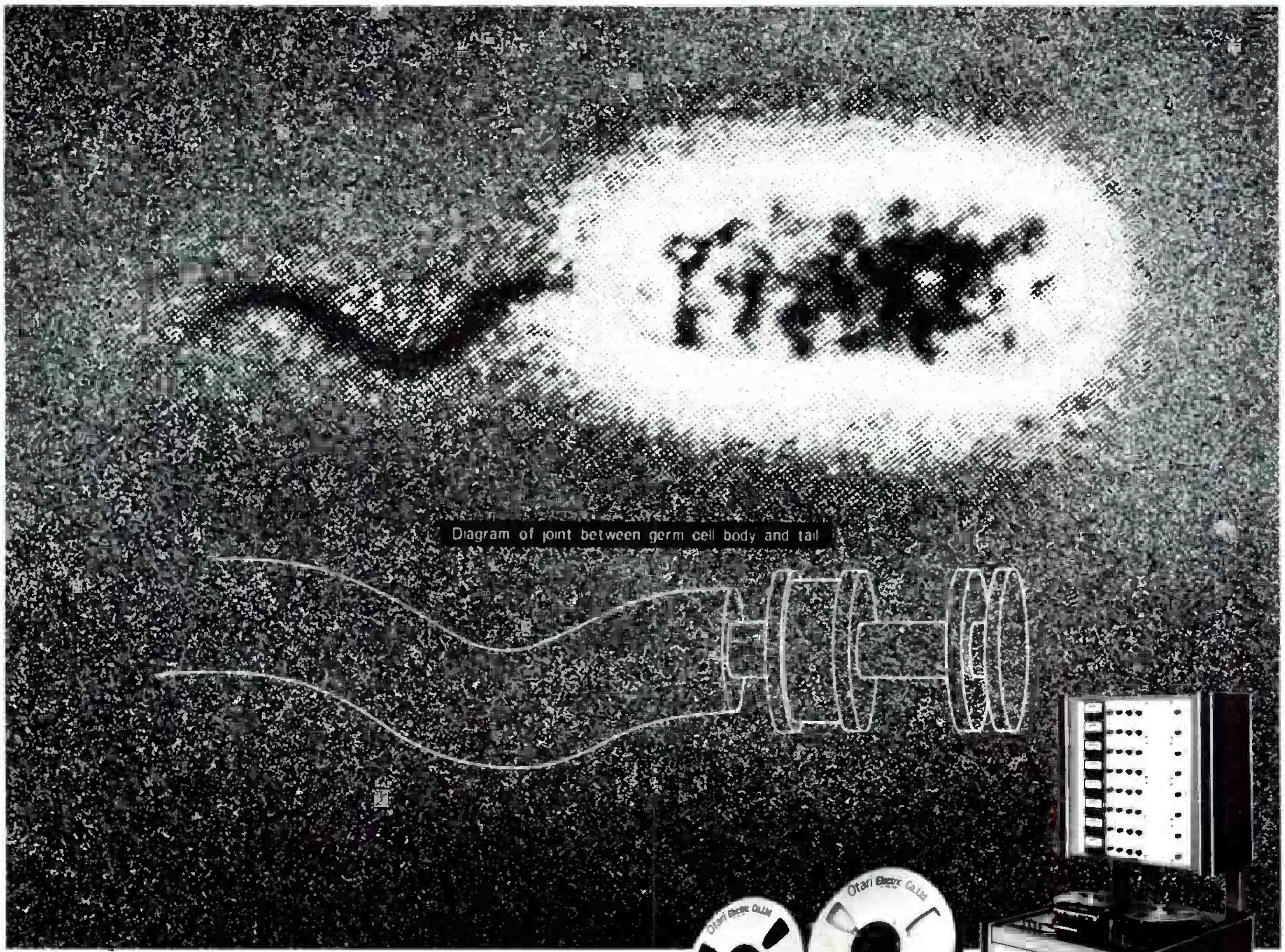

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