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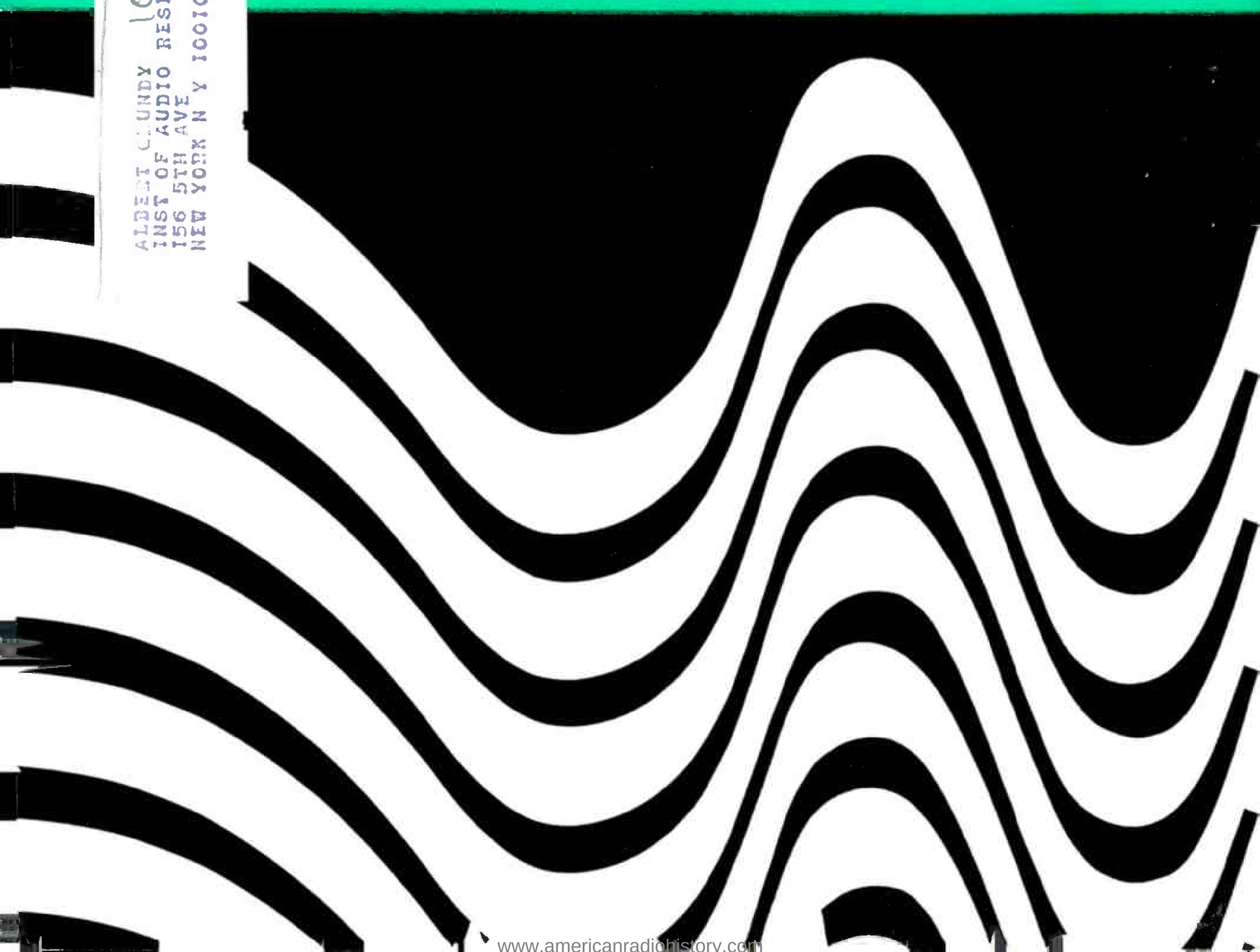
THE SOUND ENGINEERING MAGAZINE

March 1970 75c

A Primer on Noise Measurement, Part 3

A Switching/Summing Amplifier

Console Transformer Elimination



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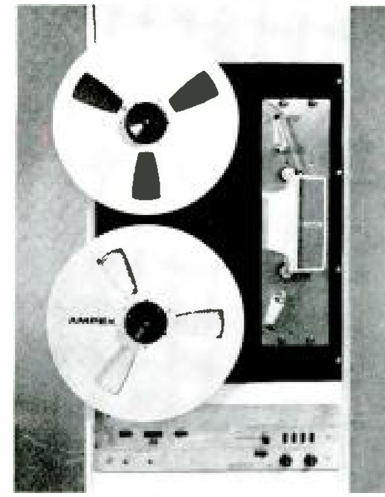

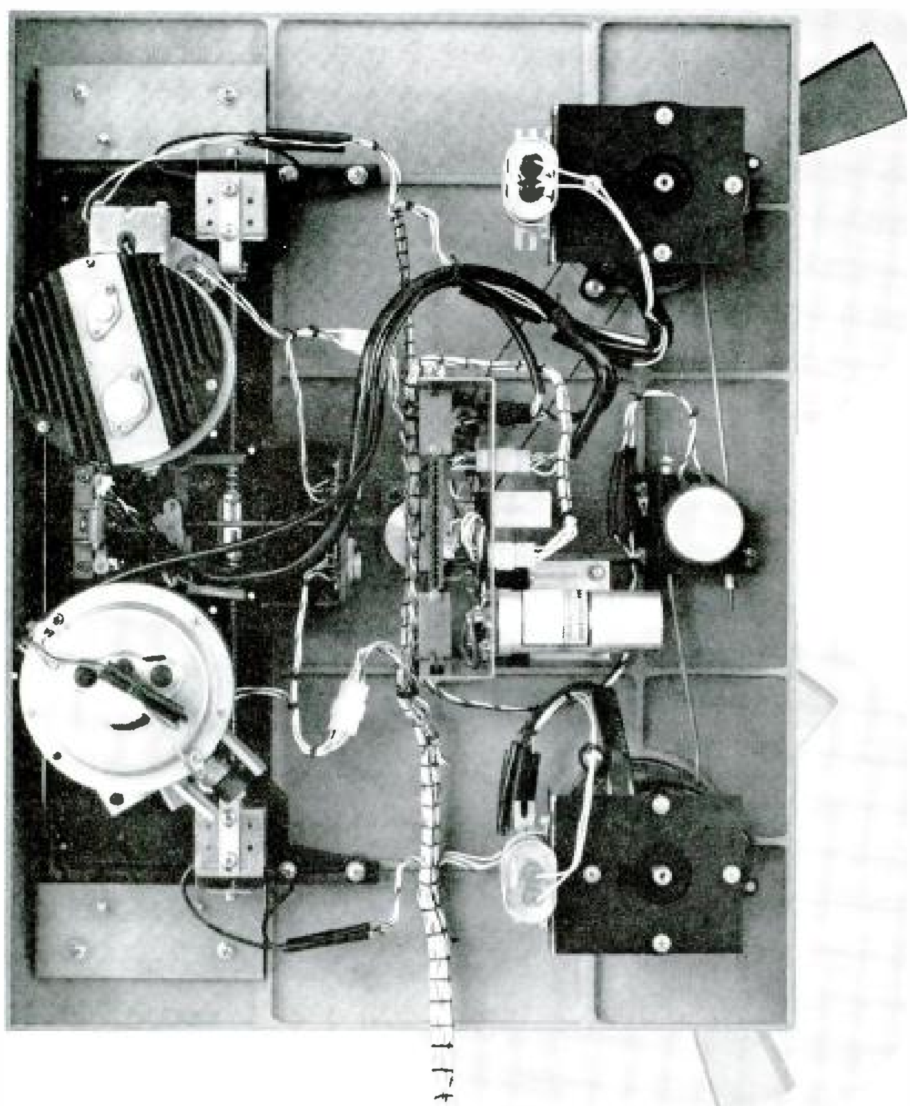

Announcing the backwards/forwards broadcast recorder, the Ampex ABR15 ...backwards

Coming Next Month

● Edward J. Gately, Jr., describes a new *building block* approach to the design and manufacture of recording consoles in his article MODULAR CONSOLE DESIGN.

MODERN STUDIO CONSTRUCTION TECHNIQUES by William R. Graham describes simple building procedures for the construction of studios by broadcasters and recordists on tight budgets.

LOW-FREQUENCY SOUND ABSORBERS by Michael Rettinger describes the control of low-frequency reverberation in the studio. Different materials are examined for their abilities to effect this control.

The April issue will also feature a complete guide and map to the exhibits to be seen at the forthcoming AES Convention in Los Angeles. In addition, there will be a listing of technical sessions and papers to be presented during the convention.

And there will be our regular monthly columnists, George Alexandrovich, Norman H. Crowhurst, Martin Dickstein, Arnold Schwartz, and John Woram. Coming in **db**, The Sound Engineering Magazine.

About the Cover

● A fanciful representation of sine waves without which there would be no audio. We salute the NAB Convention being held in Chicago, April 5th through the 8th. The place: the Conrad Hilton Hotel.

← Ampex Circle 10 on Reader Service Card

db

THE SOUND ENGINEERING MAGAZINE

MARCH 1970 • Volume 4, Number 3

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Letters

The Editor:
In reference to *Just Step Up To The Mic* in the November issue as well as the mention by the author Robert Hawkins in the January issue that he did not know the manufacturer of the velocity "Mystery Mic 2" I would like to shed the following information.

The microphone in question was a high-impedance velocity microphone manufactured in 1935 by the Amperite Company, at that time located at 561 Broadway, New York City.

I used several of these microphones in my sound amplification rentals at that time and found them quite satisfactory.

Edward R. Myrbeck
Acoustic Laboratory
Harvard University
Cambridge, Mass.

A goodly number of writers have agreed with Mr. Myrbeck in their identification of the mic shown in the November article and on its cover. We thank the many who did, and particularly Mr. Myrbeck who enclosed a xerox of an ad (which unfortunately we cannot reproduce because of the quality) that described the Amperex velocity as an "all around microphone" According to the ad it permitted 360-degree pickup when lowered and paralleled to the floor. it eliminated feedback in p.a., and the high-impedance model operated directly into a grid. *Ed.*

The Editor:
In your November issue, I was not in the least surprised (contrary to Mr. Burrough's letter; December issue) to find very little if any mention of Mr. Burrough's concern as the article dealt primarily with significant contributions to the development of microphones. It was sufficient to show that Electro-Voice microphones existed. If you are to mention every contribution to the progression of microphones, then you must of necessity give credit to the wiremaker, coremaker, etc.

It is important to note at the conclusion of this historical discussion of microphones that. in quality studio applications or critical situations, we find only one or two American microphones present. With few exceptions audio and recording engineers turn immediately to the European manufacturers for their microphone requirements.

If all the American manufacturers of microphones since 1927 had continued their efforts to produce quality instruments we, the professional users, would not be placed in a position of having to select foreign microphones. In fact, the names of Telefunken, Neumann, AKG, and others would, indeed, be foreign, instead of reverent trademarks of excellence within the American audio industry.

August H. Jones
Chief Engineer
Alteen Recording Corporation
Chicago, Illinois 60617

Is Mr. Jones correct in saying that few American-made microphones are used in quality studio applications? We have not made a study in depth. but we would venture to guess from observation in many recording and broadcast applications of the highest caliber that both American and European microphones exist side-by-side. In fact, in broadcast audio — where standards exist also — American dynamics seem to predominate. And we only refer to modern instruments; the many American ribbons still in quality use is impressive.

This does not put down European microphones (or Japanese ones either) in any way. Some exceptional microphones do indeed come from the manufacturers mentioned in the letter. But Electro-Voice, Shure, and others should not be placed in a secondary position — their products simply do not deserve it.—Ed.

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The Audio Engineer's Handbook

GEORGE ALEXANDROVICH

● We have set out to talk about filters and equalization. Now that we have been refreshed in our memory as to the basic qualities of resistors, capacitors and inductors, let us see what we can do with them. Let us assume, as a start, that we have two amplifiers, one feeding the other. In a typical installation one is a preamp (or the console output) and the other is a power amplifier. We shall also assume that the combination of these two units produces flat frequency response at the speaker terminals. But when we feed the signal generator output into the preamp and test frequency response through the speaker, we may discover that there is a considerable acoustical peak at the low frequencies as well as a few midrange peaks. Obviously, these inadequacies in response are stemming from speaker characteristics and the acoustics of the room. Our aim is to insert an equalizer into the electronic chain so that all frequencies fed into the system will be heard with the same loudness. This would hold true for the monitor/system, as for instance in the control room. But if this is a sound reinforcement system, installation problems multiply. We must now consider the response of the microphone as well, because any peaks in sensitivity (or response) of the microphone would, in a system adjusted for flat response, reduce maximum operating level.

If the system is for sound reinforcement and the requirements are such that only the midrange of the audio spectrum has to be reproduced, the use of a couple of resistors and capacitors can reduce frequency response at the spectrum, leaving only a restricted speech range. Although it may be not as effective as equalizing the entire range it will help. (See FIGURE 1 and FIGURE 2)

First, we must find out about the

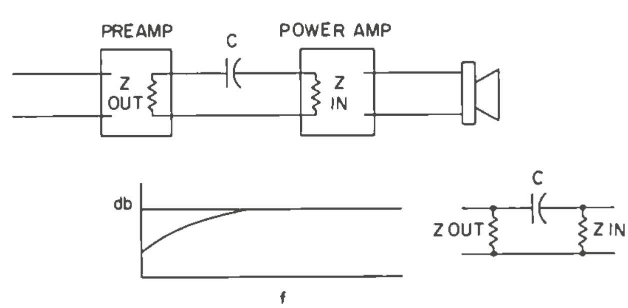

Figure 1. Attenuating the low frequencies.

Walter Carlos, creator of "Switched-On Bach" and "The Well-Tempered Synthesizer," uses the Dolby System.

Mr. Carlos says, "The raw materials of electronic music — the outputs of my Synthesizer, for example — are sounds which can be varied from striking purity to extreme complexity. After a desired sound is created, often with considerable effort, it must be preserved with care, to be combined later with others in a meticulous layer by layer process. The noises of magnetic recording are significant hazards in this regard, since they are particularly noticeable in electronic music. However, my experience confirms that the Dolby System effectively attenuates the noise build-up in electronic music synthesis. My studio at TEMPI is equipped with ten Dolby units, which I consider to be indespensible in my work."

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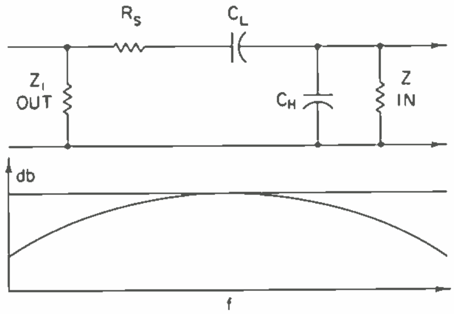

Figure 2. Rolling off the sides of the spectrum.

terminal characteristics of the two amplifiers. In particular, we must know the source impedance (preamp output) and the input impedance of the power amplifier. Considering the majority of the available power amplifiers we can assume input Z (impedance) to be on the order of 100k Ohms. Source impedance would most likely be 600 or 150 Ohms. Whether we want to just rolloff response, or insert an equalizer of any type between the two amplifiers, this information is vital. As mentioned before, a capacitor offers fairly high resistance to low frequencies and low resistance to the passage of highs. Inserting a capacitor of appropriate value between the two amplifiers will rolloff low frequencies at the rate of approximately 4-6 dB/octave (FIGURE 1). Rolloff will start at the frequency at which reactance of the capacitor will be great enough to produce an effect on the circuit. Since the capacitor is in series with the input impedance of the load, these two elements can be considered as a voltage divider. The capacitor being a series element, input Z to the power amp is a shunt element. Find out the attenuation at a specific frequency caused by this voltage divider use this formula.

$$\text{capacitive reactance } X_C = \frac{1}{2\pi fc}$$

$\pi = 3.14$
f = frequency in Hz
c = capacity in Farads

You can find what a resistance-capacitor produces at this frequency. Then substitute the value into the pad and figure the voltage loss. If you have a *Shure Reactance Slide Rule* (Shure Brothers Inc., 222 Hartrey Ave., Evanston, Ill., USA) you can find your answers in seconds.

Let us take a specific case where the input Z of the power amp is 100K and we want to roll off low frequencies so that at 100 Hz response is down by 5 dB. FIGURE 3 shows voltage ratios versus dB.

$$Z_1 = \frac{10^6}{2\pi fc} \qquad c = \text{in } \mu F \qquad (1)$$

Voltage V1 Ratio V2	dB
1.12	1
1.26	2
1.41	3
1.58	4
1.78	5
1.99	6
2.37	7
2.51	8
2.82	9
3.16	10
3.98	12
5.01	14
6.31	16
7.94	18
10.0	20
31.6	30
100.0	40
316	50
1000	60
10000	80
100000	100

Figure 3. The table will find the voltage ratio for the desired rolloff in dB.

Z_2 = Power amp input impedance

$$\frac{V_1}{V_2} = \frac{Z_1 + Z_2}{Z_2} \qquad (2)$$

From the above formula and the table you can find the voltage ratio for the desired rolloff in db. Substituting it into the formula (2) impedance Z_1 can be found. Using formula (1) we find C in μF. Then it is up to us to find the capacitor and insert it into the circuit.

$$\frac{V_1}{V_2} = 1.41 = \frac{Z_1 + Z_2}{Z_2};$$

$$Z_1 = 1.41 \times Z_2 - Z_2 = 1.41 \times 100k - 100k$$

$$Z_2 = 41k \text{ ohms}$$

$$Z_1 = \frac{10^6}{2\pi fc}; \quad 41k = \frac{10^6}{2 \times 3.14 \times 100 \times C};$$

$$C = \frac{10^6}{6.28 \times 4.1 \times 10^6} = \frac{1}{25.7} = 0.038\,\mu F$$

Identical formulae will be used for rolling off the high frequencies. We shall then substitute the capacitor for Z_2 (see FIGURE 2). Z_1 can be the impedance of the source or the additional resistor in series if the output impedance of the preamp is too low (several ohms) and it would require a huge capacitor to rolloff response. Besides the preamp am may not be capable of taking a purely capacitive load without going into distortion and overload. Addition of resistance in series (R_s) is a standard practice. The value of the resistor is generally chosen to be near recommended load resistance.

This method is a good approximation of the actual value and for all practical purposes is sufficiently accurate. If you were to consider all the losses and deviation from ideal, the obtained values of the capacitors would be most likely nonstandard anyway.

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The Feedback Loop

ARNOLD SCHWARTZ

● Last month we talked about phonograph cartridges and what compliance means in evaluating their performance. This month I would like to discuss how cartridge frequency response and bandwidth limitations are determined. Not many people have heard of Frank G. Miller, author of a brilliant study entitled *Stylus Groove Relations in Phonograph Records*. This study contains an analysis of the factors that determine the frequency response of a stylus in a record groove.

THEORETICAL

The basic elements of a high fidelity magnetic cartridge are shown in FIGURE 1. The stylus arm is assumed to be rigid, and is clamped at one end by a compliant mounting which has some mechanical resistance (damping). The transducing element is coupled to the stylus bar at this end. The stylus is mounted at the opposite end and is shown resting against a stereo groove wall (viewed 45 degrees to the record surface) with force F_Z. As the groove moves the stylus is deflected up and down about a mean position. The impedance to this motion is a mass, and a resistance. Miller derived a formula for the frequency response (H) of the stylus in the groove.

While this formula relates the factors that determine frequency response of the stylus driven by a record groove, it does not take into account the transducing mechanism that converts the stylus motion into an analog electrical signal.

RESONANT FREQUENCY

Miller's response formula is in terms of the resonant frequency, f_o, which is determined by the mass reflected at the stylus (m), the modulus of elasticity of the record material (E), the tip radius (R), and the tracking force (F_Z). The mass has the greatest effect on the resonant frequency. If we decrease the mass by one half, f_o and the damping factor (ϵ) increase by a factor of 1.4.

The modulus of elasticity represents

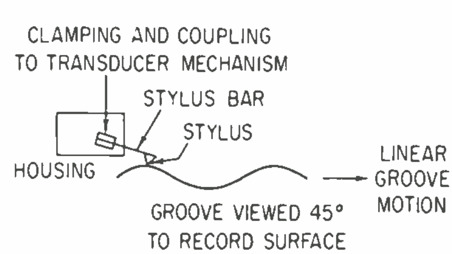

Figure 1. The playback stylus in a stereo groove.

the stiffness (reciprocal of compliance) of the record material, and for a given material such as vinyl, is a constant. Changes in both tracking force and tip radius have relatively little effect since f_0 varies as the 1/6 power of these quantities. In addition, increasing either of these quantities in order to raise the resonant frequency would be counter-productive to other cartridge performance characteristics.

We can now visualize how this stylus-groove resonance occurs. The effective mass at the tip resonates with the groove compliance and is damped by the resistance. The compliance, while it is determined mainly by the modulus of elasticity, is dependent to some extent on the tip radius and tracking force.

STYLUS-GROOVE RESPONSE

By using Miller's formula, response curves can be plotted (FIGURE 2) which show the relative velocity of the stylus when it is driven by a constant velocity recording. The frequency axis is displayed as the ratio $\frac{f}{f_0}$, as Miller's equation suggests, so that the shape of the stylus-groove response can be shown without reference to any specific resonant frequency. A group of response curves are shown for cartridges with selected damping factors from 0 to 1.4. In the theoretical case where the damping factor is zero, the response at f_0 goes to infinity. There is a resonant rise of about 3.5 dB for a damping factor of 0.7. The ideal damping factor is 1.0 where we have optimally flat response with the widest bandwidth for a given mass. Above the resonant frequency the response fall off 12 dB per octave. The useful cartridge output extends about one-half octave beyond resonance and can be considered the cartridge cut-off frequency. The frequency of the stylus-groove resonance determines cartridge bandwidth, and the damping factor determines the amount of the resonant rise.

The cartridge transducing mechanism must generate an electrical signal which is the analog of the stylus velocity. How closely does the cartridge output approximate the Miller equation ? At high frequencies, just below resonance, there are deviations which can be ascribed to problems in the transduction mechanism. At frequencies well below resonance we can expect to find close correspondence-which means flat frequency response. In the case of ceramic and crystal cartridges the piezoelectric element itself often has a resonance that is well below the stylus-groove resonance. This internal cartridge resonance is damped but rarely, if ever, can a flat response be maintained out to f_0.

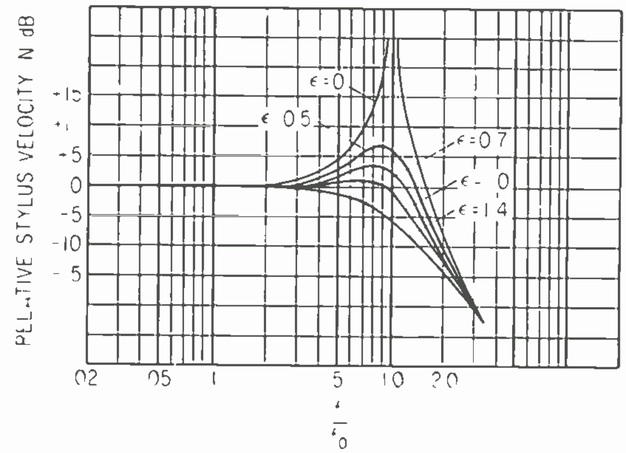

Figure 2. Stylus-groove response to constant-velocity recording according to Miller's formula.

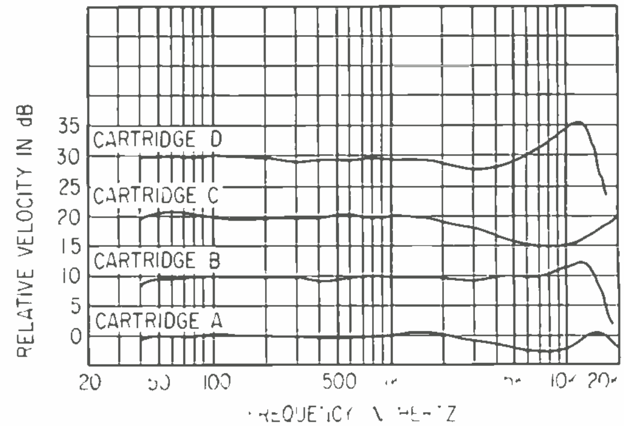

Figure 3. Response by some typical cartridges to constant-velocity recording.

CARTRIDGE RESPONSE

When we measure the response of a cartridge we are actually measuring the combined effect of two systems; the stylus-groove response, and the response of the transducing mechanism. How is this response measured? Before the advent of a recordable sweep frequency test record a laborious point-by-point plot of spot frequency bands had to be made. At CBS Laboratories when we were planning the STR-100, the first of the CBS test records, we decided to include a sweep frequency band synchronized with the General Radio level recorder. That this was a wise decision is shown by the fact that after almost ten years the STR-100 is probably the most widely used record for cartridge evaluation. The cartridge response curves of FIGURE 3 were derived from the STR-100, but for purposes of this article have been replotted to simulate a constant velocity recording.

Curve A (FIGURE 3) shows the response of one of the better, current magnetic cartridges. The stylus-groove resonance is approximately 16,000 Hz, and the droop in the 3,000 to 12,000 range is due to transducer response. Curve B shows the response of a high-quality magnetic pickup which was current about 8 years ago. Here we have a curve which comes very close to the Miller II function for a damping factor of 0.7, and with the resonance at about 13,000 Hz. Curve C shows another current, and highly regarded, cartridge with a resonance at about 20,000 Hz (if we extrapolate), but with a severe droop of over 5 dB in the high-frequency range caused by the transducer response. Curve D is a magnetic cartridge with a pronounced resonance at about 14,000 Hz.

RESPONSE AND RECORD WEAR

A stylus-groove resonance, such as we see in FIGURES 2 and 3, will cause the stylus to move with greater velocity than the groove itself. A 3 dB resonant rise will cause the stylus to move a distance that is 40 per cent greater than the modulation itself. When the stylus velocity is greater than the groove velocity, the tip will indent or emboss the groove and superimpose its response on the record. Last month I discussed compliance and record wear, and pointed out that there were other cartridge characteristics at least as important from the point of view of record deterioration. Compliance has no effect on the frequency or amplitude of the stylus-groove resonance as do the mass and damping. It seems, then, that tip mass and damping are extremely important cartridge characteristics which not only affect performance but can have important effects on the record itself.

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Theory and Practice

NORMAN H. CROWHURST

●Things happen in spurts. After I'd written the column that appeared in the December issue, where I promised something more about the design of different kinds of transistorized óscillator, I received some more pressing enquiries, so I didn't get to fulfill that promise till now.

But before I do that, something about non-audio fields. Recently, I sat down and wrote myself a document setting forth the legal precedence, based mainly on the Declaration of Independence (2,500 words), citing the relevant facts (2,000 words) and giving a concise statement of the resulting position 1,000 words), to be filed with my 1969 income-tax return.

A few people read this document prior to its submission, both to the IRS and to members of the Senate and Congress in Washington. Everyone who read it expressed the opinion that it should be published, but my primary interest is to render constructive service to my adopted country, of which my usual form of writing in this column is a sample. That I intend to continue doing, rather than trying to fight the income tax in the way that some good citizens occupy themselves.

Yet, as my friends point out, I owe it to my fellow citizens to let them have the benefit of this useful documentation, the compiling of which was virtually forced on me. So for anyone who would like a copy of this 10-page, close-typed document, I'll send a xerox for what it costs me to have it run off and mailed, which is about $2.

A more constructive activity, arising out of my concern for education, is a new kind of newsletter, titled SYNERGY, that interested teachers are working with me on getting published. If you know any teachers who might be interested, please send me their names and addresses at the same address, P.O. Box 651, Gold Beach, Oregon 97444. Now back to sound engineering theory and practice.

A quite convenient oscillator, for tube operation, used the circuit shown at Figure 1. This utilized half a 12AX7 (left) as a high-gain stage and half (right) as a phase splitter. The precise gain of the left half is controlled by the d.c. component of negative feedback from the cathode tap of the phase splitter, which in turn is derived from the bias at its grid, by rectifying signal and storing the d.-c. component on its side of the interstage coupling capacitor.

The pairs of elements labeled C and R control frequency. The upper pair provide positive feedback from the plate of the phase splitter, while the lower pair provide negative feedback from its cathode tap. At the frequency of oscillation, the signal at the gain stage grid is just in phase and of sufficient amplitude to maintain oscillation at the gain set by the d.-c. part of the negative feedback.

The values were selected to provide equal phase-inverted outputs from cathode and plate of the second stage.

Because this was a useful circuit (with minimum frequency-controlling components just 2 variables) we sought to design a transistorized version. The first try investigated the curvature of transistors that might be used in the same way tube curvature was used to automatically control the gain in Figure 1. Although the 12AX7 tube is not a vari-mu tube in the accepted sense, it possesses sufficient curavure to enable gain to be controlled over the small range necessary to maintain constant-amplitude oscillation.

This investigation yielded the fact that the most usable curvature occurs when the transistor uses no emitter resistor whatever, and is fed from a reasonably constant-current input source. The curvature is where the characteristics approach cut-off.

From this fact, the circuit of Figure 2 eventually evolved, which more or less translates the operation of Figure 1 for transistors. Q_2 is the gain stage, with zero emitter resistor, and Q_4 is the phase splitter.

Q_1 is an emitter follower that receives both parts of feedback, including the d.-c. part used to bias for automatic gain control.

Q_3 is another emitter follower, to stabilize the collector load of Q_2, and couple its output to the phase inverter stage, Q_4. Bias is effected by the diode in Q_4 base circuit. Before oscillation starts, Q_4 is biased by the 2.2k and 1.5k resistors, with the diode conducting. When signal builds up, the diode starts taking excursions into nonconduction, until the bias voltage on the output side of the double coupling capacitor sets the voltage fed back so as to just maintain oscillation.

This circuit uses a very short curvature of the transistor's characteristic, and is subject somewhat to transistor selection. Also, to operate the emitter followers properly, the arrangement requires a double power supply, with positive and negative parts. So why use

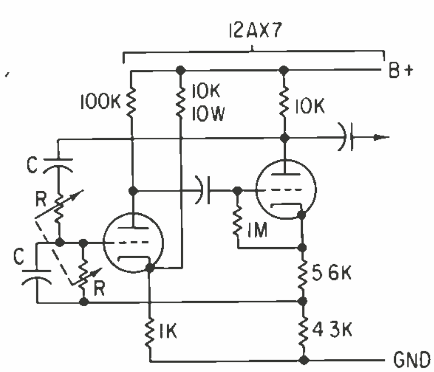

Figure 1. A basic half-bridge type oscillator, utilizing a 12AX7 to provide phase-split outputs.

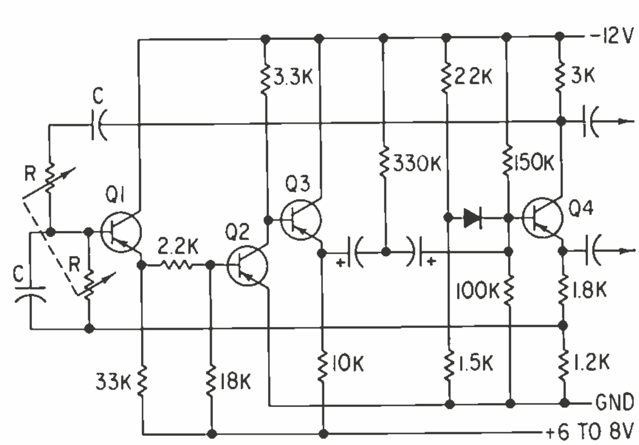

Figure 2. The closest possible translation of Figure 1, using transistors.

a curvature of the amplifying characteristic to very active gain? The next step was to use a quite different form of control.

Oscillation can also be controlled by varying the relative portions of positive and negative signal fed back, holding the gain stage at a reasonably constant, high value of gain. This approach finally led to the circuit shown at FIGURE 3. For this application, frequency was not required to vary over a very wide range, but we did want an output whose peak-to-peak value almost equalled the supply voltage, of good waveform.

Transistors Q_1 and Q_2 serve the same basic purpose as transistors Q_2 and Q_4 respectively) of the previous circuit, or as the two parts of the double triode in FIGURE 1. The values shown enable the frequency to be adjusted to precisely 1,000 Hz. Variation of positive feedback is achieved by making Q_3 part of the collector load resistor for Q_2, with a limiting value (62 ohms) in shunt.

Before oscillation starts, Q_3 is non-conducting. When the oscillator is running, Q_3 works in its saturation region, and reduces the collector resistor's effective value until oscillation is only just maintained.

Bias for this feedback-controlling transistor is obtained by isolating the oscillator output through Q_4, an upside-down emitter follower (which we can do with transistors by using one of opposite polarity), and rectifying its output with voltage-doubler diodes.

The 220-ohm resistor in supply negative to Q_1, Q_2, and Q_3 serves to provide a delay bias voltage. Q_3 and Q_5 are both cut off until oscillation starts. When the voltage developed across the 0.1 mFd capacitor feeding the 33k resistor to Q_5 base exceeds this delay bias, Q_5 starts to conduct, running Q_3 toward saturation, until oscillation is controlled.

The circuit enables the voltage at the collector of Q_2 to be controlled with quite close precision. Now we have to amplify this voltage to the magnitude of wave desired. This is achieved by Q_6, Q_7, and Q_8. Q_6 is an amplifier stage and Q_8 is an emitter follower directly coupled to it. Q_7 compares the d.-c. component of this output with a reference value at its base, to provide d.-c. bias for Q_6, and thus provides amplified d.-c. feedback to ensure the correct operating current for Q_6 to center the output voltage.

Q_6 works at maximum gain, current and voltage. But its base has two signal inputs: one through the 1.5k fixed resistor with the 1k variable in series; the other through the 6.8k resistor, which provides signal feedback from the output. The ratio between

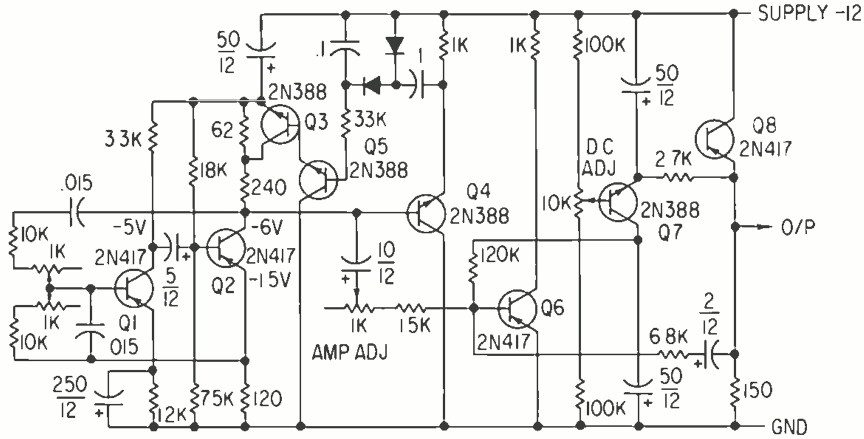

Figure 3. A final circuit, that uses a different principle to control oscillation amplitude, and provides a means to yield a signal output almost equal to the d.-c. supply, in peak-to-peak amplitude.

these resistors almost exactly sets the ratio between the input voltage at the slider of the 1k resistor and the output voltage.

In getting the full voltage swing, the emitter resistor of Q_8 sets one limit. If one extreme of swing cuts the current through Q_6 off, momentarily, the base of Q_8 rises close to negative supply voltage, and Q_8 virtually connects the output, momentarily, to supply negative—or very close to it. At the other extreme of signal swing, Q_6 passes enough current to connect the base of Q_8 virtually to supply positive, so Q_8 is momentarily cut off.

Because the emitter of Q_7 is held steady at mid-voltage, being a d.-c. control, when Q_8 cuts off, its emitter voltage is determined by the 2.7k resistor and the 150-ohm resistor, as a voltage divider across the lower half of supply voltage, which means the output will come within about 2.7 per cent of the supply voltage from supply positive.

In practice, the d.-c. output, and the amplitude adjustment can be set so these limits are not exceeded by the signal actually amplified, and the output voltage will hold itself quite stable.

The Sync Track

JOHN M. WORAM

THE OMNIDIRECTIONAL MICROPHONE

●Some time ago, I had the chance to sit in with a few engineers who were planning to open a new recording studio. When the subject got around to microphones, one of the men said that he was not considering the purchase of any omnidirectional mics, since he liked the advantages of the more popular cardioid types.

The advantages of the cardioid microphone are of course well known. In fact, perhaps a little too well known. For it seems that with the rising popularity of cardioid mics, the omni has been all but forgotten by many recording engineers. This is a great shame, since in some ways an omni is actually superior to a cardioid.

The most obvious advantage is financial. Omnidirectional microphones are generally easier to make, and are, therefore cheaper than cardioids of comparable quality. It takes considerable design ingenuity to develop and construct a directional microphone with good off-axis rejection. This additional work is passed on in the form of a higher price tag. However, people pay the additional money because of the importance of the word *separation* in today's studio operation.

Many producers become panicky if they hear anything other than the instrument right in front of it, picked up by the microphone. Consequently, microphones are often worked at very close distances, and in the search for separation, the cardioid has become king. Yet, in spite of some impressive polar patterns, it is often possible to get significantly more separation with the all-but-forgotten omni mic!

There are two reasons why this is so. First, consider the well-known proximity effects of the typical cardioid microphone. Especially at working distance, of less than two feet, there is usually a significant increase in low-end responses which gives that characteristic boomy sound associated with close-in miking.

Yet, for purposes of separation, most pop session instruments are miked within this critical 24-inch distance. Of course, there may be a low end cut-off switch on the mic, which will reduce — but not eliminate — the bass boost.

Now consider a worst case; miking an acoustic gut string guitar on a session which includes drums, electric guitars, and other loud sound sources. The acoustic guitar is a relatively low-level instrument playing within a high-level environment. A cardioid microphone placed two feet away will surely pick up too much unwanted sound from the other instruments. But, as the mic is moved in closer, the proximity effect spoils the sound of the guitar. By the time the mic is close enough to reduce the unwanted signals, not only will the bass boost be excessive, (see FIGURE 1) but the mic may actually be in the guitarist's way due to its physical bulk.

Here is where the omni comes to the rescue. There is no bass boost with an omni mic, so it can be worked at very close quarters without proximity-effect problems. And the typical omni mic

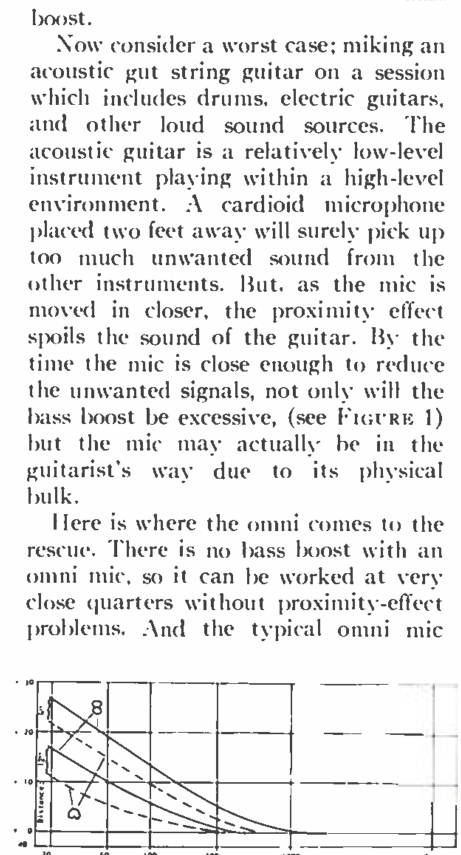

Figure 1. The transmission factor of a pressure gradient microphone without (figure 8) and with (cardioid) acoustic delay, in close proximity of a sound source that is non directional. (Graph taken from Gotham Audio Corp. literature on the Neumann U-67 microphone.

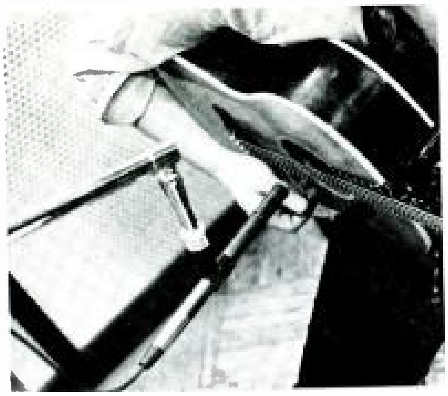

Figure 2. An omnidirectional mic such as the Shure SM76 can be used even this close to an instrument without proximity effects.

is often significantly smaller and slimmer than the cardioid, so at closer working distances, it is less likely to get in the musician's way. We find that at the extremely close working distances that the omni mic permits, the ratio of wanted to unwanted sound is so great, that the off-axis rejection of the cardioid type of pattern is no longer important. Remember, the cardioid microphone does not think. It has been programmed to reject certain sounds not because they are unwanted by you, but because they are off-axis. And you pay for the programming. With the omni mic, you don't get the off-axis rejection, but you don't need it either since the on-axis (desired) sound is now so loud at the close-in working distance that the off-axis signals are effectively masked.

And, of course, there are applications where separation is irrelevant, or where a wider acceptance angle is important. On vocal overdubs, the omnidirectional pattern is a big help if the singer tends to move about in front of the microphone. And of course, the omni is less sensitive to popping. You can demonstrate this with a microphone such as the Neumann type 86 or 87, which contain dual cardioid elements. The user can derive an omni pattern by means of a switch which puts the elements in phase with each other. (The figure 8 pattern is achieved by switching the elements out of phase, and the cardioid pattern uses only the front element.) In the omni position, any popping problem will be somewhat reduced, although for maximum effectiveness, a true omni-only microphone is recommended. As an additional bonus, the true omni microphone often has a smoother overall response and an extended high end.

For some weeks now I have been experimenting with a typical omni directional microphone, the Shure SM76. I chose this mic for a guinea pig for three reasons, none of which were terribly scientific. It was small—it was cheap—and I knew absolutely nothing about it, other than the first two reasons mentioned. On many occasions, it has bailed me out of some very difficult situations that could not be handled by a cardioid microphone. And fortunately, it also turns out to be an excellent general purpose mic, well worth a try in any modern studio.

As a postscript on the subject of close miking, some manufacturers recommend using a wind screen, or pop filter, on *all* close-in pickups, even if it is not acoustically required. Especially in studios located in smog centers (you know where they are) there is a certain amount of dirt in the air. Moisture from a too-close singer or brass instrument easily finds its way into the microphone, carrying with it any dust and grime in the area. The pop filter serves as an effective barrier for pollution as well as pops, and may keep the mic out of the repair shop for a long time.

And, in any discussion of close-miking, with or without omnis, some attention should be paid to the different *perspectives* one may achieve at various working distances. The classical repertoire is generally recorded at maximum working distances. Here, one argument against close miking is that the listener hears too many of the so-called production noises of the instrument. A violin which sounds pleasantly mellow may become more strident as the mic is moved in closer. Or a flute pick-up will become too breathy.

Of course, in the pop scene, distant miking is rarely considered, so differences are between close and closer, miking. As a mic is moved between a few feet and a few inches, there are apt to be significant changes in percussiveness, string noise, articulation, frequency balance within the instrument, and so on. This changing perspective must be considered, as most likely different settings of equalization, limiting, or echo will be suggested as working distance is varied. Ideally, the optimal working distance for the required degree of separation will give a suitable sonic perspective too. Practically, it may not, so one must expect to make adjustments at the console as working distances are varied. The equalization that sounds good at two feet may no longer be right at 6 inches.

Careful consideration of the many variables associated with good microphones placement is an important part of achieving the best sound possible. The omnidirectional microphone is no more the answer to every pick up problem than is the cardioid. However, every well-equipped studio should seriously evaluate the characteristics of the omni to see if its particular advantages cannot be put to good use.

Sound with Images

MARTIN DICKSTEIN

SLIDES

● After an audio-visual installation has been made, the client may begin to run into situations with which he is not familiar and which he did not expect because there was no mention made during the consulting stages prior to equipment purchase. It can be very helpful for the wise a/v supplier-installer to be familiar with several aspects of slides, the different mountings used for various reasons and equipment, and the effects that might be seen, as well as one or two difficulties.

The most common slide mounting is the cardboard holders supplied by almost all of the photo shops. This method is the simplest and is usually sufficient except that the slide must be handled carefully as the film is exposed on both sides and is thus vulnerable to fingerprints, scratches, and advise storage conditions. Heat and humidity play havoc with the emulsion and can ruin a slide in a short time. Also, projection of a slide, mounted in this fashion, with a particularly hot source could cause the slide to pop. This comes about as a result of the emulsion shrinking more than the base side of the film, causing a dishing out of the film with the concave side toward the projection lens. One method of preventing this is to hold the slide in the mount with a slight concavity on the side of the emulsion. Another is to mount the film by some other method.

Another mounting method provides glass to keep the slide from popping. One way is to laminate, or fasten, the film to one piece of glass. This is done on the emulsion side of the film with the base side of the film lest exposed. The film-glass laminate is now mounted in a holder. With this method, one side of the film is exposed so care must be taken in the handling of the slide to prevent scratches and finger prints. During projection, however, the film is kept flat by the glass yet is allowed to remain cool by having one side uncovered.

Still another method for mounting slides is in a metal frame or binder. For this process, the film is first put into a mask made to hold the film in small tabs. The mask is then placed between two pieces of glass cut to precise size, and then the entire assembly is slipped into the metal binder which has three holding/sides (channeled) and one half-open side. The fit is snug and some care must be taken to avoid breaking the glass. The top of the picture is inserted first with the aluminized side of the mask toward the three-channel side of the holder. This method protects both sides of the film from being marred by scratches or fingerprints, permits a little rougher handling, and allows the slides to be kept in various types of boxes or cabinets for easier reference.

There are certain types of slide projectors that provide random access to several trays (mounted one above the other in a special housing) with each tray holding up to a hundred slides. These trays provide a different type of metal slide holder and therefore require that the slide be mounted differently. (However, this does not mean that the slide mounted in this manner cannot be used in other types of trays or slide projectors. It can, but without the special metal binder.) This process also has the film mounted in a mask, and again between two pieces of glass, but, instead of now putting the assembly into a holder, the edges of the glass are bound in tape. The assembly is rolled firmly and evenly along the center of half-inch tape until all sides are enclosed. Extreme care must be taken to be sure that the tape is even all around (no extra thickness due to overlap) and that the corners are mitered to prevent loose edges from sticking out and catching in the projector causing a jam-up. This mount allows freer handling of the slide without damage to the film itself.

As a result of the different types of available slide mounting, certain precautions must now be taken to assure proper operation during projection. The simple cardboard mounting ends up with a slide that is thin in its profile. If thin glass is used in the metal-binder process, the profile of this finished slide will also be thin. However, where thicker glass is used, or where the tape is put around the slide-glass assembly, the final profile is thicker. Care must be taken in the tray used to hold the slides during the projection or trouble will result. Certain trays (the black ones among others by different manufacturers) are made to take only the thinner slides. The gray trays are called Universals and are made to operate with the thick as well as the thinner slides.

Certain phenomena will be noticed during projection of a slide which may appear only interesting to the casual observer, quite disturbing to the perfectionists, perhaps embarrassing to the speaker giving the presentation, and of importance to the person in charge of the slides and equipment of the facility. One such occurrance is the appearance of *water spots*.

When glass-bound slides are projected under high heat sources, there is a certain amount of moisture evaporated from the emulsion of the film. This moisture condenses on the inside of the glass and on the film. Sometimes, the spot will disappear when the slide heats through and the condensate re-evaporates. However, there will be occasions when the moisture will remain in one spot. This spot will swell and spread out as the heat increases during projection, and finally remain as what appears as a dirt-spot on whatever part of the picture it happens to have formed. Continuous presentation of slides with condensation taking place may eventually appear clouded or smokey. All such slides should be cleaned and re-mounted periodically to extend their useful life.

Another visual phenomenon seen with glass mounted slides is known as *Newton rings*. These appear as oddly shaped rounded rainbow patterns which expand and move as they heat up during projection. These are caused by uneven contact between two adjacent extremely smooth surfaces causing light interference patterns at the points of contact. The name comes from the studies made by Sir Newton of the effects produced by a thin film of air between the convex surface of a lens and the flat surface of a plane piece of glass. The effect is similar to a thin layer of oil on water. The thickness of the layer will determine the color seen, due to the frequency of the interference patterns caused by reflection from the top of the layer and the bottom, or water-oil surface. Where reflected rays are in phase they will reinforce. When they are out of phase they will tend to cancel out. With monochromatic light of a particular frequency incident on the layer, certain effects will result. Different frequencies of incident light have different effects. White light, a mixture of all the colors, has still another effect. This is the one seen in projectors.

When these rings prove annoying, there are certain preventive measures that can be taken. In the mounting process, if the slide must be masked off for any reason to avoid showing part of the film during projection, the masking must be done carefully without buckling the film. Also, the film should not be fastened down firmly (with tape, for instance) on two opposite sides to allow for some expansion of the film during heating. Also, there is available an anti-Newton-ring glass which can be used for mounting. This glass is made with a slightly roughened surface to prevent the rings from forming. (It is possible, however, that in rare cases, when the illuminating source is of high intensity and the screen image is very large that a little of this roughness will appear in bright areas to observant viewers sitting too close to the screen.) For those who wish to, they can purchase a very fine abrasive powder and apply this carefully and lightly to the inside surfaces of the mounting glass, making sure all excess is carefully and completely washed and wiped off before mounting. The minute scratches on the glass may help alleviate the ring situation.

Under all circumstances, however, the storage of the slides is of utmost importance. Slides should be kept in a cool, dry place and in the dark. Cracked glass should be changed as soon as possible, and all slides should be wiped clean (with a white lintless cloth or cloth) before placing them into a tray, or storage cabinet. Tape should be checked regularly to prevent jamming. Moisture-absorbing materials should be used in the slide storage area to prevent excess humidity from creating problems unseen until the slide is being presented, because it is then too late.

Console Transformer Elimination

ALLAN P. SMITH

Are transformers necessary in audio control consoles? The author makes a case for their elimination and details the advantages that this elimination can bring.

Transformers have been utilized in professional audio equipment to such an extent that they have become an integral part of preamplifier subassemblies. Typical microphone preamplifier or line amplifier subassemblies have both an input and an output transformer. Often there are as many as four amplifier subassemblies in tandem in an audio control console. This means that the audio signal must go through as many as

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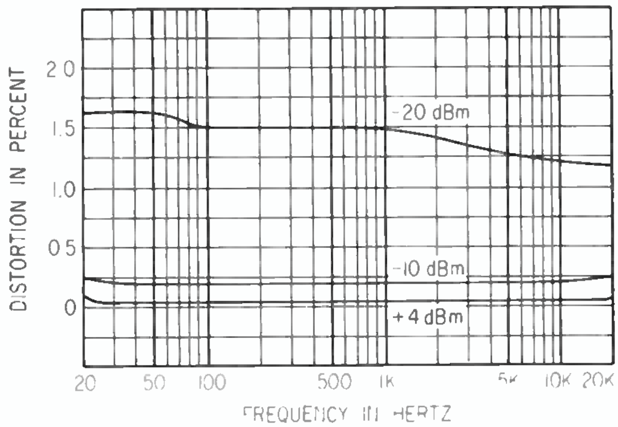


Figure 1. The total harmonic distortion of a typical 600:600 ohm high-level transformer.

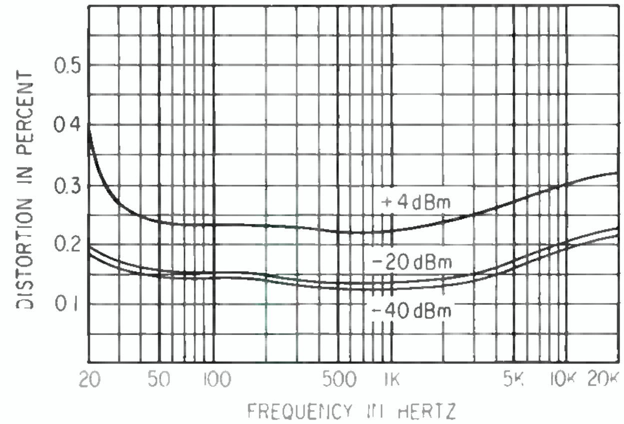


Figure 2. The t.h.d. of a typical 600:600 low-level transformer.

eight transformers as the signal passes through the console. Why do we need this many transformers and how do they affect console performance?

The typical microphone/line preamplifier used today contains a step-up transformer at the input which matches impedances of 50,150 or 600 ohms to a secondary impedance of 15k ohms. There is an output transformer which steps down from 15k ohms to 600 ohms. When vacuum-tube circuitry was in use, high impedances were necessary to match optimum operating points of the tube circuitry. Transistors are low-impedance devices, yet designers still insist on design techniques that were valid for vacuum tubes.

Why are these transformers necessary? First of all, the typical microphone preamplifier must also serve as a line amplifier so that the same amplifier subassembly may be used in any part of the console. This requires that input impedances of 50 and 150 ohms are necessary for microphone inputs and a 600 ohm input is necessary for the line input. A 600 ohm output is necessary to match the 600 ohm input of the next amplifier. Also typical audio control consoles are so arranged that the operator may patch in or out of any point in the console for use with auxilliary equipment.

The functions performed by a transformer in an audio control console are as follows:

1. Impedance matching through step-up or step-down configurations.
2. Elimination of radio frequency interference pickup on input lines.
3. Isolation of ground potentials between pieces of equipment.

Impedance matching of input and output circuits is critical in many of today's transistorized console preamplifiers. Input circuits of these amplifiers usually use a common-emitter circuit configuration in which the lowest noise figure for the preamplifier is reached when the input impedance is between 10k and 15k ohms. Thus, in order to achieve the lowest noise figure for the preamplifier, a step-up transformer is required for proper impedance matching. Step-down transformers are used at the output of the preamplifier to match a 15k ohm output to a 600 ohm line impedance. This type of design clearly indicates that many circuit designers are still thinking in terms of vacuum tubes.

American console designers are accustomed to thinking in terms of an exact match of 600-ohm impedances because they think in terms of dBm and maximum power transfer. In audio control consoles, preamplifiers are voltage amplifiers and power transfer is of no concern. Generations of misunderstanding have perpetuated the myth of dBm and power transfer in audio consoles to such a degree that many console designers cannot differentiate between science and witchcraft. If two amplifying units are to be connected together and the input or load impedance of the second unit is equal to or greater than the source or output impedance of the first unit, full voltage transfer will result. However, if a circuit can be designed so that the load impedance is at least 10 times the source impedance, any variations in the load impedance will not affect the voltage transferred. This holds true all the way from microphones to line inputs. If an amplifier were to be designed with a 600-ohm input impedance, any source impedance from a 50 ohm microphone to a 600 ohm line could be matched without a transformer unless the voltage gain obtained through a step-up transformer was required. The same holds true for the output impedance of an amplifier. If the load is to be 600 ohms, an amplifier output impedance of 60 ohms or lower will assure full voltage transfer if the 600 ohm load should vary.

Radio-frequency interference pickup on input lines poses another problem. Many audio control consoles are operated in the middle of horrendous fields of radio-frequency interference. A properly designed radio-frequency interference filter installed at the input of each console line should eliminate this problem. However, if balanced lines are used to null out interference pickup, transformers must be used.

Isolation of ground potentials between pieces of equipment requires considerable measurement and investigation. If each piece of equipment to be used can be checked out and potentials between signal output minus terminal and the equipment chassis are eliminated, a common grounding system for all equipment can relieve the requirement for transformers between equipments for this purpose. One common but often overlooked source of a ground loop between equipments is the three-wire power cord.

Ideally, if an amplifier could be designed that would decouple grounds with the facility of a transformer but without the compromises associated with a transformer, the transformer would not be necessary. The design of such an amplifier would eliminate the following deficiencies of transformers:[1]

1. Elimination of cost.
2. Elimination of bulk.
3. Elimination of weight.
4. Elimination of distortion.
5. Elimination of frequency discrimination.

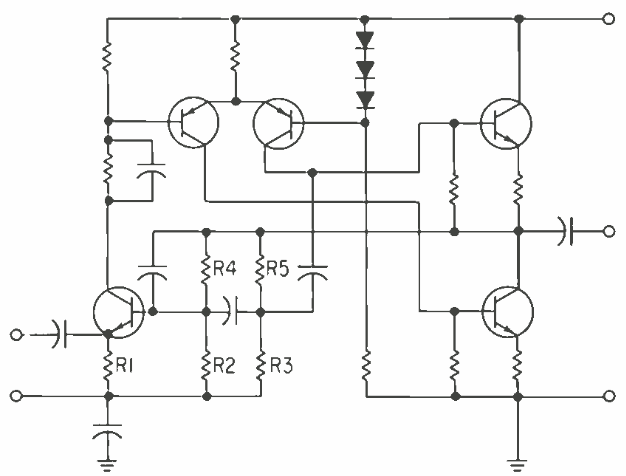


Figure 3. The amplifier as described in the text, including the modifications.

6. Elimination of phase shift.
7. Elimination of impedance limitations.
8. Elimination of ambient pickup.

Distortion in a transformer? That question was raised by an editorial reviewer for a leading professional engineering society journal. It is obvious that the gentleman had never bothered to measure a transformer. The results are startling. First of all, almost all professional audio control consoles utilize a 600:600 ohm isolation transformer for high level inputs. Figure 1 shows the total harmonic distortion measurements made on a typical 600:600 ohm high level audio transformer. The manufacturer claims 10 to 50,000 Hz response and a maximum level of +15 dBm. Measurements were made on this transformer at +4 dBm, −10 dBm, and −20 dBm. It is obvious from this information, that the total harmonic distortion generated by a typical high level transformer operating at a signal level of −10 to −20 dBm. is sufficient to mask the lack of distortion in the rest of the console. For comparison, a low-level 600:600 ohm transformer which might be used at microphone levels was measured. The measurement results are shown in Figure 2. It is obvious that low-level transformers do not generate the distortion associated with passing low levels through a high level transformer.

How do we eliminate transformers in a professional audio control console? An electronic device that would accept input and output grounds of differing voltage potential, and would provide a signal output that would track the input signal voltage changes on a 1:1 ratio, could properly be defined as an *active isolation transformer*. The described device would fulfill all isolation needs within a complex control system, thereby eliminating the requirement for transformers—and since no impedance transformation is involved with low-impedance devices in the described application, a solid-state amplifier would only require one active isolation transformer to satisfy the grounding decoupling requirement. However, an active isolation transformer cannot be expected to isolate potentials of 120 volts between pieces of equipment. Only a transformer will perform this function.

The amplifier shown in Figure 3 is the basic amplifier previously described in the October 1965 issue of the Journal of the Audio Engineering Society[2], with modification. This modification as described herein, deteriorates in no way, the performance or characteristics previously described.

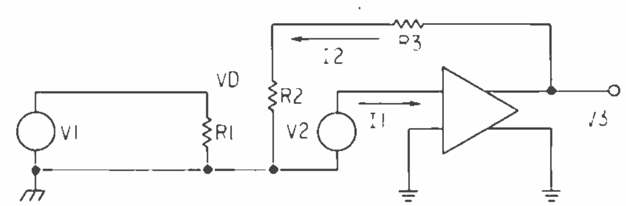


Figure 4. The equivalent circuit of Figure 3 viewed as a signal analysis.

Rearrangement of the fundamental circuitry allows the amplifier to perform as if coupled by an external transformer at the input (with regard to low-frequency ground decoupling) with no other change in the operation of the amplifier.

In order to achieve grounding isolation between input and output, the signal minus is severed, and a capacitor is inserted between the two. The capacitor maintains the previous configuration at high frequencies, but leaves the input isolated at low frequencies and d.c. (Complete isolation at all frequencies was not attempted because of the interelectrode capacitances involved). This circuitry allows the input section to be grounded through the signal source without causing the normal ground loops.

As illustrated in Figure 4 (the equivalent circuit of Figure 3), the signal source (V_1) is connected to the amplifier input loading resistor (R_1) with both + and − leads, with the grounding being accomplished at the source location. The equivalent resistor (R_2) represents the parallel combination of R_2 and R_3 of Figure 3, R_3 is the equivalent of R_4 and R_5 of Figure 3, and the current source I_1, represents the input transistor of Figure 3.

The equivalent circuit of Figure 4 clearly indicates only two connections between the input and output stages: first stage output and feedback path.

Voltage V_2 is a division of the output voltage V_3:

$$I_2 = \frac{V_3}{R_3 + R_2} \qquad V_2 = I_2 R_2 \qquad \text{Equation 1}$$

$$V_2 = V_3 \left(\frac{R_2}{R_3 + R_2}\right) \qquad \text{Equation 2}$$

The voltage V_D is shown as the difference between the input voltage V_1 and the feedback voltage V_2.

$$V_D = V_2 - V_1 \qquad \text{Equation 3}$$

The current source I_1 is shown as a function of this voltage V_D.

$$I_1 = f(V_D) \qquad \text{Equation 4}$$

The current, in actuality, is an exponential function of the voltage, but for this discussion, remains purely academic, since V_D *must* be zero for stabilized operation, as will be shown next.

Current I_1, is fed directly into the output stage, which for derivation purposes, is assumed to possess infinite gain. (The actual gain is sufficiently large to allow this assumption without causing significant inaccuracies in calculation).

For any value of V_D, the output voltage (V_3) is infinite. Therefore, V_D must equal zero.

From equation 3:

$$V_D = 0 = V_1 - V_2$$

$$V_1 = V_2 \qquad \text{Equation 5}$$

Substituting into equation 2:

$$V_1 = V_2 = V_3 \left(\frac{R_2}{R_3 + R_2}\right) \qquad \text{Equation 6}$$

Rearranging equation 6:

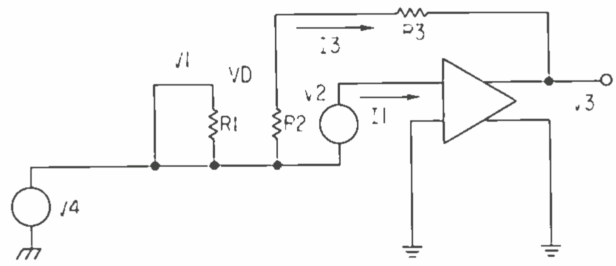


Figure 5. The equivalent circuit of Figure 3 viewed as ground potential analysis.

$$V_1 = V_3 \left(\frac{R_2}{R_3 + R_2}\right)$$

$$\frac{V_3}{V_1} = \frac{R_3 + R_2}{R_2} = \text{Gain of amplifier} \qquad \text{Equation 7}$$

This equation describes the voltage gain of the amplifier and empirical verification has been established with over 500 amplifiers (equation versus empirical varies less than 1 per cent).

Therefore, all assumptions, indicated previously, do not affect the equation sufficiently to be empirically measured, and the validity of the equivalent circuit has been established for the purpose intended. Setting the input voltage (V_1) equal to zero, the circuit of FIGURE 5 results.

The new signal generator (V_4) represents the unwanted signal introduced (by long input leads).

From equation 3:

$$V_D = V_2 - O$$

$$V_D = V_2 = O$$

$$I_2 = O \text{ and } V_3 = O \text{ (with respect to } V_1) \qquad \text{Equation 8}$$

$$I_3 = \frac{V_4}{R_2 + R_3} \qquad \text{Equation 9}$$

To satisfy equation 8, the output must increase to compensate for V_2 which is:

$$V_2 = I_3 \times R_2$$

$$V_2 = V_4 \left(\frac{R_2}{R_2 + R_3}\right) \qquad \text{Equation 10}$$

Substituting the feedback equation (Equation 2):

$$V_2 = V_4 \left(\frac{R_2}{R_2 + R_3}\right) = V_3 \left(\frac{R_2}{R_2 + R_3}\right)$$

$$V_3 = V_4 \qquad \text{Equation 11}$$

Therefore, the output voltage (V_3) follows the *unwanted* input signal (V_4) on a one to one basis, while at the same time, the output voltage (V_3) follows the input signal (V_1) on a ratio determined by equation 7.

$$V_3 = V_1 \left(\frac{R_3 + R_2}{R_2}\right) = V_4 \qquad \text{Equation 12}$$

Rearranging equation 12, the ratio of the input signal to the unwanted signal is:

$$\frac{V_1}{V_4} = \frac{R_2}{R_2 + R_3} \qquad \text{Equation 13}$$

The equation indicates that the ratio of these signals, or the *isolation of the amplifier*, is identical to the voltage gain of the amplifier (equation 7).

Therefore, any unwanted signal, as a result of differences in ground potential between input and output, is attenuated by an amount equal to the voltage gain of the amplifier.

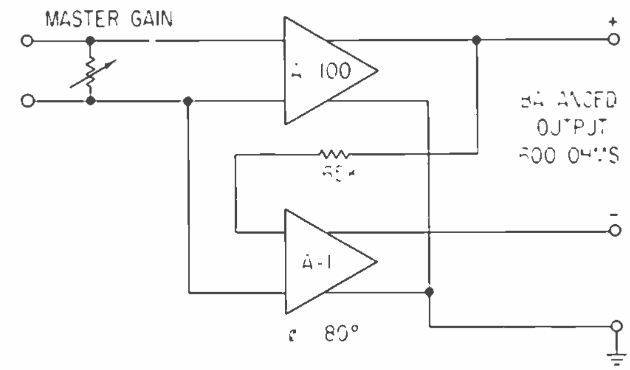


Figure 6. A functional diagram of a balanced console output without transformers.

Wherever design considerations normally would require the use of a transformer for ground decoupling purposes, the employment of this amplifier performs the decoupling function while simultaneously performing an amplifier function. For example a bridging function can be accomplished, with grounding isolation, merely by inserting a series resistance between the amplifier and the source to be bridged. All input impedance restrictions are, of course, eliminated with the elimination of all transformer requirements in this function.

Branching circuits, mixing circuits, and mix-down circuits, all can be independently isolated without the employment of a single transformer.[3]

The amplifier described here has an output impedance of 6 ohms and may be loaded by 600 ohms or higher. A variation of this amplifier is called a *Zero Amp* mixing amplifier. Its operating characteristics are identical to the standard amplifier except that it provides 180 degree phase reversal and when used with a series 10.7k ohm resistor provides phase reversal at unity gain. The combination of the standard amplifier and the Zero Amp mixing amplifier, as shown in FIGURE 6 will provide a balanced console output without the use of transformers. The undistorted power output of this combination is +24 dBm into a 600-ohm load.

The following performance measurements indicate the performance capabilities of this transformerless console concept.[4]

Signal to noise ratio (−50dBm. input for +4 dBm out)	80 dB.
Frequency response (20 Hz to 20kHz)	±0.1 dB.
Total harmonic distortion (20 Hz to 20kHz)	Unmeasurable
Intermodulation distortion	.05%

REFERENCES

[1]William G. Dilley, *An Integral Active Isolation "Transformer"/Audio Amplifier"* Audio Engineering Society Preprint No. 566.

[2]William G. Dilley, *A New Solid-State Amplifier for Studio and Broadcast Use,* Journal of the Audio Engineering Society, Vol. 13, No. 4 (October, 1965) p. 332.

[3]*Ibid,* Reference 1.

[4]Allan P. Smith, *A High Performance Control Console With Flexibility,* Audio Engineering Society Preprint No. 533.

A Switching/Summing Amplifier

WALTER JUNG

The component described has many uses, paramount among them is its function as part of an electronic music synthesizer. A complete parts list is included.

With the increasing popularity of electronically generated or controlled waveforms, be it music, sound or whatever, a definite necessity arises to have complete capability of systematic programming, switching and otherwise manipulating various voltage sources. As an example, if electronic musical expression is to develop to its logical end the ultimate hardware involved will out of sheer necessity probably involve some degree of computer control—otherwise the array of controls would be staggering for even the simplest tones. A realistic degree would use the computer as an interface element between artist and tone generators, etc. The performer could program his desires into the machine and the computer arrange the correct combination of oscillators, modulators, gates, shapers, and so forth to generate the desired output. Thus, the menial switching and so forth can be done on a large scale—a scale which would be inconceivable for a man to accomplish, and the artist is free to perform creatively without the cumbersome burden of the machine's mechanics.

A basic piece of hardware to such a system is a waveform gate, switcher, or sequencer with capability of control by standard digital logic levels. This item would take any number of input channels and select by digital command any or all of the various inputs into a common output channel. By appropriate time sharing techniques various kinds of modulation can be easily performed, with the mechanics of the modulation being done at the digital level where a large variety of standard, inexpensive logic modules are available. A system of this sort theoretically has no limitations as to degree of complexity or versatility—the constraints that do exist are only practical ones.

A few brief points summarize the functional characteristics of this module:

Inputs: Any number of input channels, 1 to *n*, each standard level, 600-ohm line impedance.

Output: One, standard level, 600-ohm source impedance

Control inputs: One for each input line. A logical *one* ≧ plus 5 volts) inhibits signal transmission. A logical *zero* (ground level, current sinking) enables signal transfer.

Switching time: For high speed modulation or gating, switching time should be about 10 µs for either *on* or *off* transition.

Isolation, insertion loss: An *off* channel should exhibit ≧60 dB of attenuation to an input signal while an on channel should have a 0 dB insertion loss. As an option, *on* insertion loss should be capable of being trimmed ±a few dB to match channels.

Frequency response, distortion: In *on* state, frequency response and distortion characteristics of input signal should remain unaltered. No switching transients of control signal should be transmitted to output.

Other Objectives: Use of inexpensive, readily available components in an easily repeatable circuit configuration is highly desirable. Eventual conversion to IC format would be useful.

OUTLINE OF APPROACH

With the previously listed objectives in mind, the block diagram of Figure 1 depicts the various components interconnected to fulfill these goals. Each input line is buffered by a high-input/low-output impedance emitter follower which allows bridging of lines (if desirable) and provide a low source impedance to the switching element(s).

The digitally controlled switches are series operated transistors which are effectively either *on* or *off*, the state being under control of the digital control lines.

The output of all switches feed a common current summing bus and summing amplifier. At the output of this amplifier appear all input lines which happen to be *on* at a given instant in time. In a case such as two or more *on* channels, the output is the algebraic sum of the inputs with a scale factor of +1 applied to each (each input is 0 dB insertion gain, non-inverting).

Thus, this device can either sum any or all input channels, select a single channel, sequentially switch a number of them or even provide random access programming. And all of this is continuously under control of the digital lines—all of the channels have a common control voltage sense, 0 volts d.c. or ground equals a switch *on*, and a +5-volt level is a switch *off*. Since these two voltages are standard TTL logic levels, this allows direct interfacing with any common computer-type hardware—flip-flop, gates, even high order functions such as MSI or LSI complex arrays. Although this article cannot begin to illustrate the various and extensive applications that are possible using logic control, a few of the basics will be shown with the hopes that it might serve as a starting point.

CIRCUITRY

The circuitry involved in this device is shown in Figure 2. The three basic parts of a single switcher/summer are shown

Walter Jung has contributed before to db's pages.

here; input buffer, digitally controlled switch, and summing amplifier. The summing amplifier is used once per system and the other two circuits repeated as many times as there are channels in the system.

The input buffer is merely a simple emitter follower, long-tailed to the negative supply to provide optimum linearity. The variable bias connection shown (R_3) is provided to adjust Q_1's emitter potential to exactly match Q_5. With all input channels at the same static potential, there will be no switching thump as various channels are switched in and out.

The digitally controlled series switch uses the variable resistance properties of a field effect transistor (FET) as the control element, and translates the input voltage into a proportional current to be summed in the following summing amplifier. Since the *on* state resistance of common FETS is quite variable (both initially as purchased due to production tolerances and as a temperature drift) a fairly high swamping resistance (R_6 & $_7$ is used in series with the FET to minimize these variations. For the device shown, R_{ON} max is 500 ohms, so the amount of resistance shown provides at least a 10/1 reduction in this parameter's sensitivity. And by making a portion of this resistance variable, the individual device variations from FET to FET can be adjusted out and each channel set to exactly 0 insertion gain in the *on* state. The control voltage to drive the FET is by necessity bipolar, so the driver circuit of Q_3-Q_4 and their associated components is used to translate the 0- to +5-volt computer talk into FET language. [1]This effectively links the switch's *on* state to the sense of the digital control line. A 0-volt signal at the control line robs current from Q_3, and Q_4 switches off, driving the gate of Q_2 positive. The positive level on the gate turns the channel resistance *on*, passing current through the switch and into the summing junction of the next stage. For the opposite state (*off*) the sequence is reversed, Q_3 on, Q_4 on, and Q_2 off inhibiting current transfer into the summing amplifier. Since both the input and output circuits of

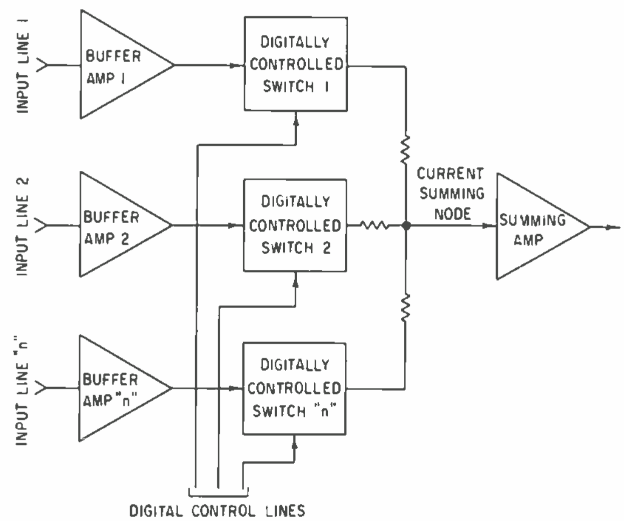


Figure 1. A block diagram of the system needed to fulfill the goals set in the article.

this stage are d.c. coupled via the emitters of Q_1 and Q_5 these d.c. potentials must be equal, which is the reasoning behind the aforementioned R_3. As a result there is no transient thump or bounce when channels are switched in and out and all input channels are transmitted to the output without a d.c. pedestal.

The summing amplifier is in reality an extension of a technique previously described [2, 3, 4, 5] with an output buffer amplifier and 600-ohm termination. The grounded base transistor Q_5 serves as the summing junction for all incoming switched signals and translates the composite sum of these input currents into a voltage across the 5.1k collector load. Since this value of resistance is equal to the total value of the switch resistance, the input current provided by each input will develop the same voltage across this common resistance, resulting in a 0 dB net gain. The Darlington emitter followers

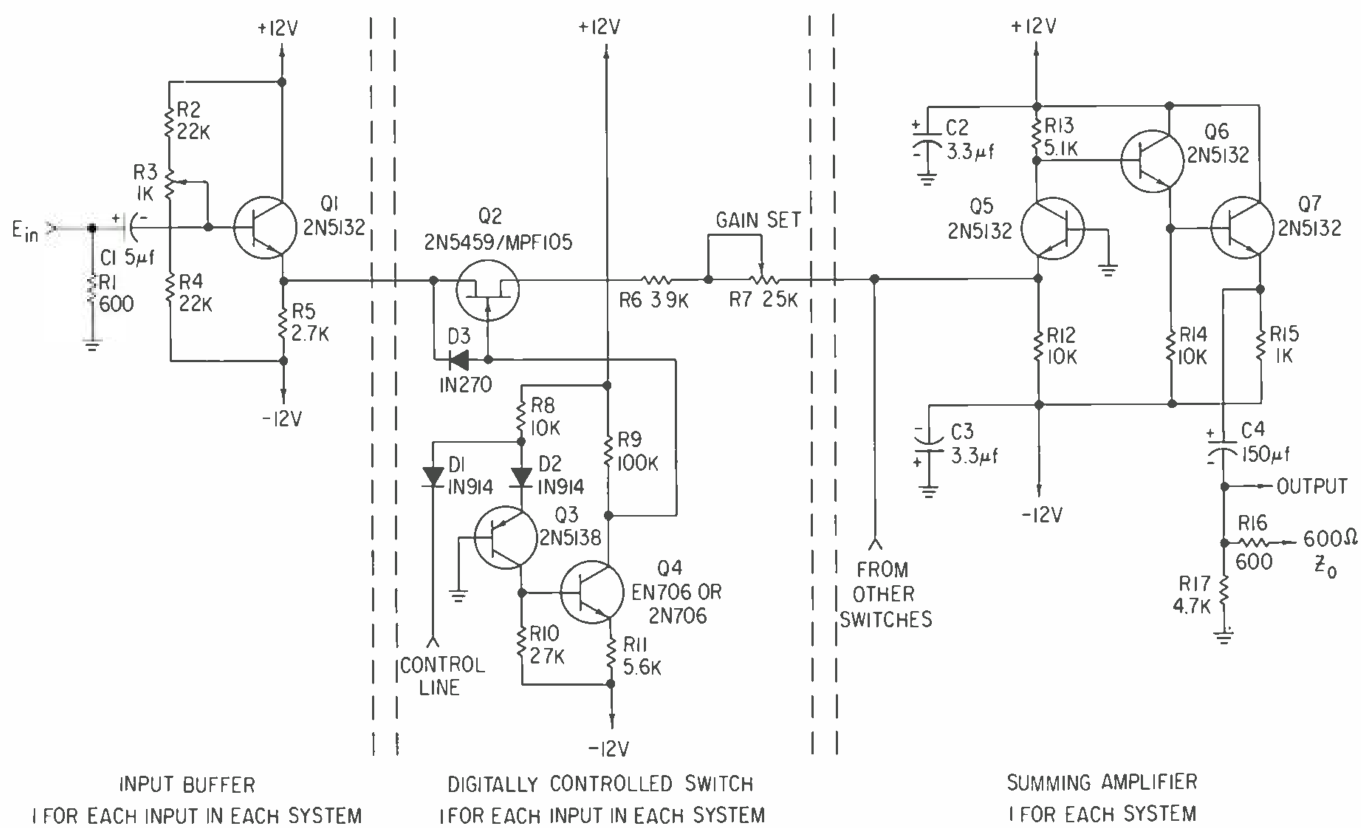


Figure 2. The schematic of the three parts that make up the system.

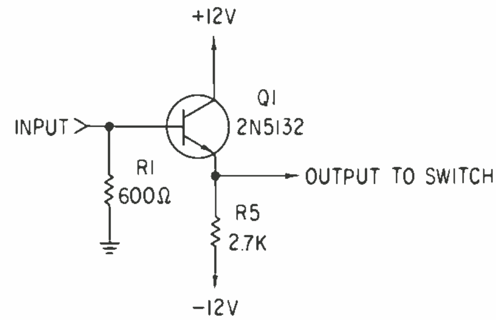


Figure 3. An input buffer with simplified biasing and line termination. Note that to use this version delete C_1, R_2, R_3, R_4 from the parts list at the end of the article.

Q_6 and Q_7 are straightforward and the essentially zero impedance output of Q_7 is source terminated for a 600-ohm *Zo*. The direct output is available for applications not requiring a source termination.

A power supply for the system is illustrated FIGURE 4 which will suffice for small scale applications. To plan large systems the current requirements are: buffer amplifiers—5 mA, switches—3mA for high level input, 1 mA for low input, summers—15 mA. Regulation is not extremely critical, but ripple should be low, preferably 1 mV or less for good s/n and the two levels would like to be balanced (+v = −v) for the biasing shown.

APPLICATIONS

To illustrate the performance of this circuitry the following section is devoted to actual measured performance data on a two-channel version of a switcher/summer. Later on some suggested ideas will be shown for other possible applications.

Perhaps the most stringent performance parameter that a switching device must meet is its isolation in the *off* state. FIGURE 5 shows a curve of isolation *versus* frequency in the range of 10 Hz to 100 kHz. Below 10 kHz the isolation exceeds 60 dB which is quite respectable for a plastic economy FET. The loss in isolation above 10 kHz is due to capacitive feed thru across the switching device. FIGURES 6 and 7 also illustrate *on/off* ratio, as they show two waveform photos of alternate cycles of full *on* to full *off*, with FIGURE 7 showing an expanded scale of the residual feedthrough.

To show the fast switching capability, FIGURE 8 shows a composite output waveform of alternate sine and triangle waves of two sources with different amplitudes. For the purposes of demonstration, the source used was a dual-output function generator, but it should be noted that any sources can be used, either harmonically related or asynchronous.

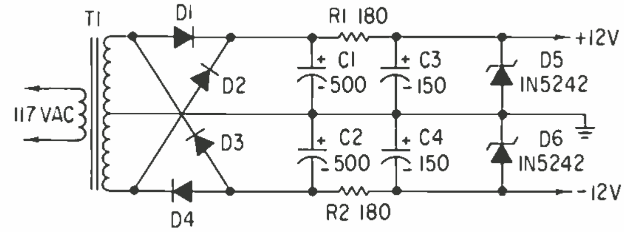


Figure 4. The power supply.

FIGURE 9 shows a single function with another function being gated in summation periodically. This illustrates both summing and gating.

While these waveforms do illustrate the capability of switching between waveforms, the real versatility of a device such as this comes to bear when a few applications are envisioned. One of the first to come to mind is an multiplexer which sequentially scans various inputs and delivers a serial output waveform. Or, a simplified version might be a two-input sampler driven by a flip-flop, which would cause the output to toggle back and forth between the two sources.

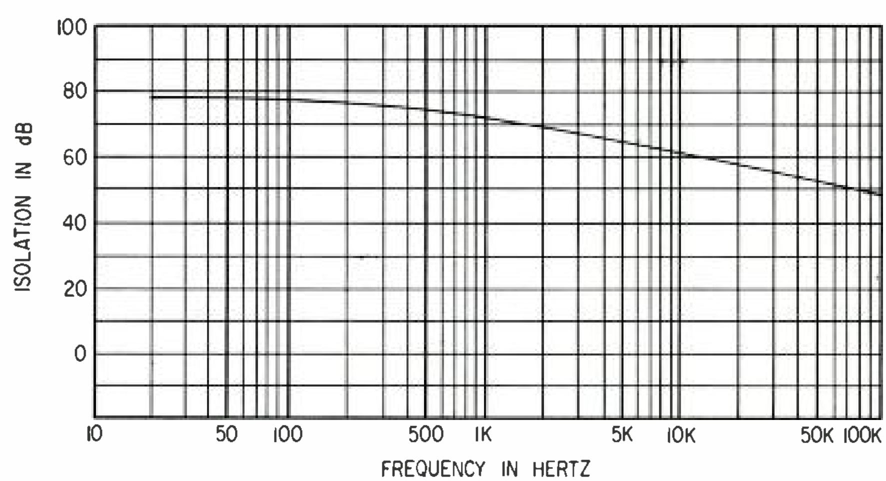
An all-electronic fader, control could be realized by controlling the switch with an ultrasonic frequency, the duty cycle of which is variable. As the duty cycle increases, the output would increase proportionally. As the duty cycle is lowered, effective output would be reduced. By driving a second channel with a complementary signal, a cross fade would be made between two different inputs, since one signal would be increasing as the other was decreasing and *vice versa*.

If special waveform processing techniques are applied to an input signal and this processed output used as the gating drive, some unusual responses can be effected. Threshold and peak detectors, waveform shapers, etc. can be used to implement a variety of special effects. One rather simple example would be a burst generator, which could be realized by using the zero-axis crossings of the input waveform to clock a counter and deliver a predetermined number of output cycles or burst.

SUMMARY

This article has attempted to describe a practical approach to a combination audio waveform switching/summing device which can be readily interfaced with common digital logic hardware. This brings about a marriage between two systems: the analog control of a signal line and the digital elements which command this control. By defining the control sense of this communication link in digital terms, large scale manipulation and synthesis can take place on the digital level, where it is best afforded.

Figure 5. A graph showing switch isolation plotted against frequency.



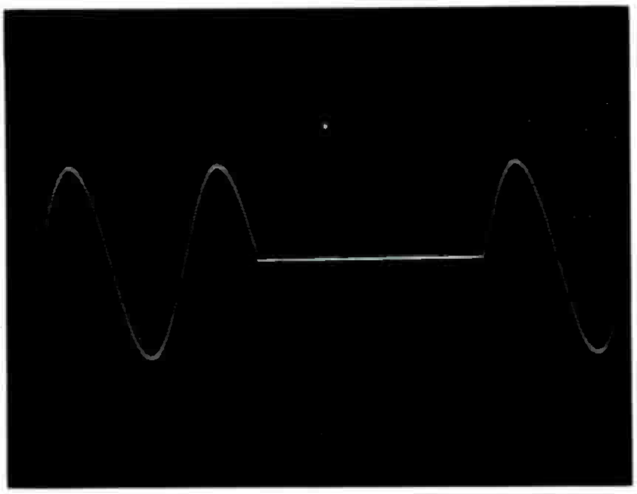


Figure 6. Gated sine wave source, 2 cycles on, 2 cycles off. 0.2 milliseconds/cm horizontal, 1 volt/cm vertical.

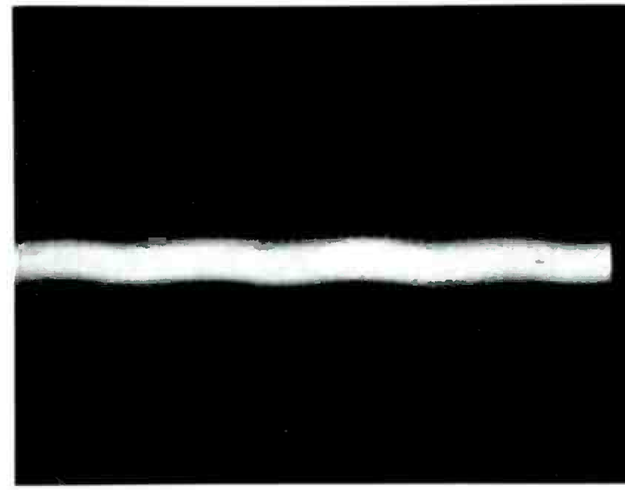


Figure 7. Expanded scale to show switch isolation—input to switch is 6 V p-p sine wave and switch is in off state, 0.2 milliseconds/cm horizontal, 5 millivolts/cm vertical.

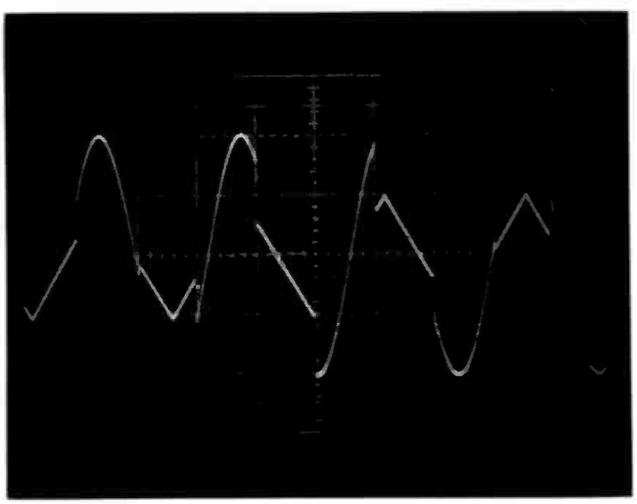


Figure 8. Alternate switching between two sources—sine and triangular waveforms, 0.5 millisecond/cm horizontal, 0.5 volt/cm vertical

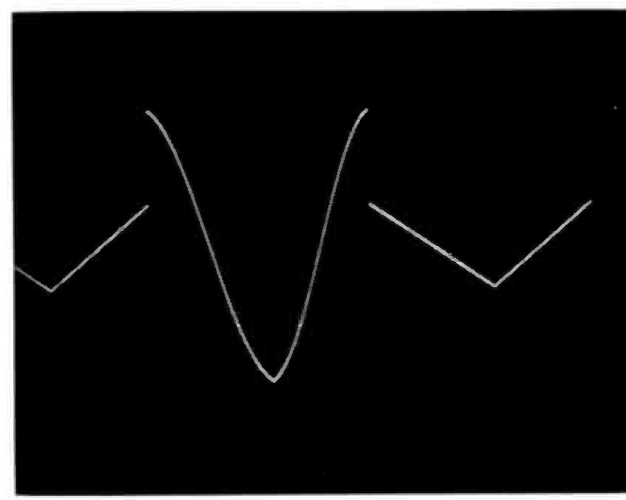


Figure 9. A continuous triangular waveform with higher amplitude sine wave gated in summation for 1 cycle. 0.2 milliseconds/cm horizontal, 2 volt/cm vertical.

REFERENCES

1. Walter G. Jung, *TTL/DTL Interface to FET Analog Switch*, EEE, July 1969. Page 107.

2. Earl E. Swartzlander, Jr., and Charles H. Downey, *Two Transistor Summer Boasts Stability*. EDN. December, 1968. Page 85.

3. Walter G. Jung, *A Common-Bass Mixer-Filter Amplifier*, db, August 1969. Page 26.

4. Walter G. Jung, *Gain Control Revisited*, Broadcast Engineering, July 1969. Page 40.

5. Walter G. Jung, *Grounded Base Transistor Acts as Current Summer*, Electro-Technology, November, 1969. Page 46.

BIBLIOGRAPHY

James S. Sherwin, *FET's as Voltage—Controlled Resistors*, Silconix Application Note, copyright 1964.

Mark Shipley, Sr., *Analog Switching Circuits Use Field-Effect Devices*, Electronics, December 28, 1964.

Arthur D. Evans, *IC's End the Driver Gap* Electronic Design, November 22, 1966.

James S. Sherwin, *Take the Fog Out of Field Effect Design*, Electronic Design, May 24, 1966.

James S. Sherwin, *How, Why and Where to Use FETs*, Electronic Design, May 17, 1966.

Thor B. Bergerson, *Using FETs in Chopper Circuits*, EEE, September, 1966.

Carl David Todd, *FETs as Voltage—Variable Resistors*, Electronic Design, September 13, 1965.

The FET as A Voltage Variable Resistor, Dickson Application Note, Vol. 1, Number 4 copyright 1965.

The FET as A Chopper Switch, Dickson Application Note, Vol. 1, Number 7 copyright 1965.

Thor B. Bergerson, *Field Effect Transistor Circuits in Chopper and Analog Switching Circuits*. Motorola Application Note AN-220, February, 1966.

In addition to the above articles, two very extensive bibliographies are available from both Siliconix and Dickson. Write the addresses below and request:

Siliconix Application Tip
A Select List of FET Articles
from
Siliconix Incorporated
1160 West Evelyn Avenue
Sunnyvale, California

Dickson Application Note
Volume 1, Number 1
An Annotated Bibliography on Field-Effect Transistors
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Dickson Electronics Corp.
P.O. Box 1390
Scottsdale, Arizona 85252

PARTS LIST

Input Buffer: multiply the below quantities by the number of channels required per system—all reference designations pertain to FIGURE 2.

Item	Description	Quantity	Price
R_1	Termination resistor, 500 or 600 ohms	1	$ .10
R_2,R_4	22k, ½w, 10%, carbon composition	2 × $.10 =	$ .20
R_5	2.7k, ½w, 10%, carbon composition	1	.10
R_3	1k trimmer pot, IRC-CTS X201R102B	1	.39
C_1	5μf, 50 v, electrolytic, Mallory MTV5CB50	1	.40
Q_1	2N5132 npn Silicon transistor—Fairchild or National	1	.17
	subtotal component cost		$1.36

Digitally Controlled Switch: multiply the below quantities by the number of channels required (as in above). All reference designations pertain to FIGURE 2.

Item	Description	Quantity	Price
R_7	2.5k trimmer pot, IRC-CTS X201R252B	1	$ .39
R_6	3.9k, ½w, ±10%, carbon composition	1	.10
R_8	10k, ½w, ±10%, carbon composition	1	.10
R_9	100k, ½w, ±10%, carbon composition	1	.10
R_{10}	2.7k, ½w, ±10%, carbon composition	1	.10
R_{11}	5.6k, ½w, ±10%, carbon composition	1	.10
D_1,D_2	1N914 or 1N414B Silicon diode GE or ITT	2 × .30	.60
D_3	1N270 Germanium diode—GE or ITT	1	.18
Q_2	2N5459/MPF105 FET—Motorola	1	.90
Q_3	2N5138 pnp Silicon transistor Fairchild or National	1	.19
Q_4	EN706 or 2N706 npn Silicon transistor Fairchild	1	.35
	Sub total component cost		$3.11

Summing Amplifier: All reference designations pertain to FIGURE 2.

Item	Description	Quantity	Price
C_2,C_3	3.3 μf, 15V, tantalum-Mallory TAC 335K015P03	2 × .63	$1.26
C_4	150 μf, 15V, electrolytic-Mallory MTV150CK15	1	.48
R_{13}	5.1k, ½w, ±10% carbon composition	1	.10
R_{12},R_{14}	10k, ½w, ±10% carbon composition	2 × .10	.20
R_{15}	1k, ½w, ±10% carbon composition	1	.10
R_{17}	4.7k, ½w, ±10% carbon composition	1	.10
R_{16}	500 or 600 ohms to suit line impedance	1	.10
$Q_5Q_6Q_7$	2N5132 npn Silicon transistor Fairchild or National	3 × .17	.51
	Sub total component cost		$2.85

Power Supply: All reference designations pertain to FIGURE 4.

Item	Description	Quantity	Price
T_1	117Va.c./24Va.c. CT @ 1A Triad F-45X	1	$3.77
D_1-D_4	1 Amp, 100 PIV Silicon Rectifier 1N400Z or equivalent—ITT, Motorola	4 × .36	1.44
D_5,D_6	12V ±20% zener, 500 milliwatt 1N5242 or equivalent—Motorola	2 × .67	1.34
C_1,C_2	500 μf, 75V, electrolytic-Mallory MTV 500CD25	2 × .42	.84
C_3,C_4	150 μf, 15V, electrolytic Mallory MTV 150CK15	2 × .48	.96
R_1,R_2	180Ω, ½W, ±10%-carbon composition	2 × .10	.20
	Sub total component cost		$8.55

Summary of costs **Input Buffer**	$1.36
Digital Switch	$3.11
Summing Amp	$2.85
Power Supply	$8.55
Total	$15.97

The above costs reflect the electronic component cost required to build one single-channel system. The price of a d.c. board, chassis, or package and miscellaneous hardware is excluded, since it can vary widely, depending upon the number of channels. To calculate additional cost of electronics per channel, add $4.47 per channel. Since the above prices are single piece catalog prices, volume buying or judicious selection can materially reduce the costs.

Listed below are several distributors which carry the items necessary to build the unit.—

Cramer/Baltimore
922-2A Patapsco Avenue
Baltimore, Md. 21230

Newark Electronics
500 N. Pulaski Road
Chicago, Illinois 60624

Allied Radio Corporation
100N. Western Avenue
Chicago, Illinois 60680

Hamilton Electro Sales
8809 Satyr Hill Road
Baltimore, Md. 21234

Primer on Methods and Scales of Noise Measurement, Part 3

WAYNE RUDMOSE

This installment concludes the discussion of sound measurement equipment and goes on to cover physical measurements in noise control.

To measure noises which change rapidly with time (impulse noises) special attention must be given to the entire system, i.e., microphone, amplifier, and output indicator. Consider two types of impulse sounds, the first being the noise generated by a rifle being fired and the second being the noise generated by a drop-forge hammer. These are both classified as impulse sounds, but clearly the duration of the noise from the rifle will be very much shorter than the duration of noise from the forge. Because of this, the instrumentation which would be acceptable for measuring the forge noise will not necessarily be acceptable for measuring the gun noise.

The last statement needs an explanation. The customary way of specifying the performance of a measurement system is to state the frequency range for which the response is uniform. Seldom is equipment rated for its range of rise times; however, a simple engineering rule of thumb can be explained which relates rise time to frequency response. Consider an acoustic system responding to a 1000-Hz signal (1000 Hz is chosen only to make the arithmetic easy). The sound pressure fluctuates about the normal atmospheric pressure in such a manner as to produce a compression and a rarefaction in 1/1000 sec. (i.e., the wave has a period of one msec). Remembering the shape of a sine wave, one can see that the system responds to a change in sound pressure from atmospheric to maximum compression in ¼th of the period or in ¼th msec. This can also be thought of as the "rise time" of the sine wave. Put another way, a 1000-Hz signal might be thought of as having a rise time of ¼the msec.

Suppose a measuring system was quoted as having a uniform frequency response up to 1000 Hz. (This upper frequency of response is commonly referred to as the "cutoff frequency.") If the upper limiting frequency of this system is 1000 Hz, it is evident that the system could not respond correctly to changes that take place in less time than ¼th msec. Thus frequency response can be related to rise time response.

Wayne Rudmose is a group vice president of Tracor, Inc., located in Austin, Texas. The material of this series originally appeared in the American Speech and Hearing Association Reports 4 (February 1969).

Since impulse sounds from guns have rise times of the order of a few µsec, proper measurement requires a system able to respond to rise times even shorter than a few msec. Suppose the required rise time response is one µsec, which corresponds to the rise time of a pure tone of 250,000 Hz. The upper cutoff frequency of the system then would have to be at least as high as 250,000 Hz. If the rise time to be measured were shorter, the upper cutoff frequency would have to be still higher.

It should be evident by now that to measure gun noise requires special equipment. Microphones that are most suitable for this type of system are the very small (1/8-in. diameter) condenser types or ceramic types. High quality amplifiers and oscilloscopes with microsecond sweep rates make up the remainder of the system. Photographic recording of the oscilloscope wave pattern is the usual form of output registration.

Returning to our comparison of gun and drop-forge noise, the measurement of drop-forge noise does not require a system with the extended high-frequency response required to measure gun noise since the rise time of the pressure wave for the forge noise is much longer than is the case of gun noise. For the drop-forge measurement, a high quality microphone and amplifier (response uniform to 15 or 20 kHz) would be satisfactory, and all that would be necessary would be to use an oscilloscope as the output indicating device. The microphones and amplifiers of most high-quality sound level meters would be acceptable, but their indicating meters would be unacceptable since they cannot respond to such rapid changes, as was pointed out earlier.

When measuring gun noises where the frequency response of the system must extend to such very high frequencies, it should be remembered that the wave lengths of these acoustical signals are very short. Since this is the case, many subtleties arise in positioning the microphone and in making the measurements. It is sufficient to say that the measurement of such impulse noises should not be undertaken by someone unless he is very well grounded in physical acoustics.

Recently the sonic boom has become a topic of national interest. Since this sound falls within the definition of an impulse sound, it follows that instrumentation for the measurement of sonic booms is sophisticated and in certain cases quite complex and so the comments concerning gun noise measurement also apply to the measurement of sonic boom noise.

It may not have been obvious, but the discussion on method of data display changed significantly when the topic shifted from the measurement of steady-state noise to impulse noise. Measurements of steady-state noise normally yield amplitudes as a function of frequencies. Mathematical techniques developed by Fourier and Laplace show that it is possible to represent a time-varying signal such as sound in at least two ways. Amplitudes may be shown as a function of frequency or as a function of time. It is possible to compute one relationship if the other relationship is known. The form of representation is normally dictated by the sound under investigation. In the case of impulse sounds, it is more convenient to represent the amplitude as a function of time rather than to show the amplitude as a function of frequency. The impulse noise generated by rifle fire usually produces a very rapid compressional rise to a peak, followed by a relatively slow rarefaction wave. The most relevant pressure value in this type of noise is probably the peak pressure of the initial compression. On the other hand, a sonic boom has a time pattern with more of the appearance of a stretched out "N." In this case, the pressure value most commonly referred to represents the pressure difference between the compressional peak and the rarefaction peak, a measure sometimes referred to as peak-to-peak sound pressure. As the various speakers present their papers, I am sure you will observe the many forms of presentation of physical data associated with the noises under discussion.

PHYSICAL MEASUREMENTS IN NOISE CONTROL

Concerning the physical measurements related to noise control, there are four major concepts I would like to define, namely: *absorption coefficient, reverberation time, transmission loss*, and *noise reduction*.

The absorption coefficient of any material represents the ratio of absorbed energy to the energy incident upon the material. If the material is a perfect reflector, the absorption coefficient is zero; and if the material is a perfect absorber, the absorption coefficient is one. Absorption of sound is accomplished by generation of heat which results from the movement of air particles against the surface of the absorbing material. In any material, the more surface exposed to air movement the greater will be the absorption. From a materials point of view, the design objective for high absorption is to produce a fiber of very small diameter so that within a given volume many more fibers can be packed and hence much more rubbing surface area can be exposed to the air.

Thus the two principal ingredients are (1) the exposure of a large amount of surface area to the sound; and (2) the movement of air within the volume occupied by the material when the sound wave strikes the material. This latter requirement is often forgotten, and it accounts for the fact that even though a material exposes a large amount of surface area, it may not be an efficient absorber. This is especially the case when 1 or 2 in. of absorbing material is attached directly to a massive wall. As sound strikes the wall it is almost totally reflected due to the large difference between the speed of propagation of sound in the massive wall material compared with the speed of sound in air. As a result of this very efficient reflection, a pressure maximum is associated with the surface of the wall. Also since the wall is massive and does not move, the particle velocity at the wall is minimum. The nature of compressional wave motion is that a velocity maximum occurs at a location one-fourth of a wave-length removed from a velocity minimum. If a low frequency sound, say 125 Hz, strikes the wall, the particle velocity at the wall will be essentially zero. Since the wavelength of a 125 Hz signal is approximately 10 ft., the maximum particle velocity will occur at a position approximately 2½ ft. (one-fourth of a wavelength) away from the wall. Thus the particle velocity of the air within a distance of 2 in. of the wall will of necessity be very small; hence the absorption will be low. If this same 2 in. of material is spaced out away from the wall the particle velocity will be greater and thus the absorption by the material will be greater. If one wishes to have a high absorption coefficient for low-frequency sounds the depth of the absorptive material must be great or the material must be spaced out away from the wall. This is one of the major problems in absorbing low frequency sound.

Reverberation is a term which the layman intuitively feels he understands. Technically, the reverberation time of a room is the time required for any sound to decrease in amplitude by 60 dB if the sound source is turned off and the sound allowed to decrease by virtue of absorption within the room. If the room has very little absorption (if all of the surfaces are acoustically hard), a greater length of time will be required for the sound to dissipate in the form of heat and hence the longer will be the reverberation time; if the absorption within the room is great, the reverberation time is shorter. From the standpoint of noise control, the problem is usually not how long a time is required for the sound to diminish 60 dB after the source is turned off, but what is the sound pressure in the room while the source is operating. This is similar to the concept of power available from a source, discussed earlier. As the source continues to operate and radiate sound, part of this radiated sound will be absorbed and eventually a condition will exist where the amount of energy radiated equals the amount of energy absorbed and the sound pressure within the room will reach its steady-state value. It follows, therefore, that the greater the sound absorption within the room the lower will be the steady-state sound pressure with a sound source such as a machine operating. Thus the reverberation time of a room is measured not for the purposes of decay of sound but for the purpose of determining the average absorption characteristics of the room. Once these are known and once the sound power radiated by the source is known, the sound pressure levels generated in the room by the source can be calculated. Incidentally, these sound pressure levels cannot be calculated from pressure measurements made using the same machine operating in another room unless the absorption characteristics of this second room are also known.

Normally part of the problem of noise control is how to prohibit sound from traveling from one room to another. A wall separating a room containing a noise source from a room desired to be quiet can be characterized by its transmission loss (TL). Transmission loss is a property of the wall itself, and is defined as the ratio of the energy transmitted through the wall to the energy striking the wall. To measure the transmission loss of a wall, a noise source is placed in a reverberant or semireverberant room, one wall of which is the wall under test. On the other side of the wall under test (the receiving room) there may be complete absorption by all of the other wall surfaces (an anechoic chamber) or there may be another semireverberant room. Since sound pressure levels rather than energy levels are measured in both rooms the data must be corrected on the basis of the size of the panel under test and upon the absorption characteristics of the receiving room. These corrections are necessary for the determination of the transmission loss of a wall, and it must be kept in mind that the TL values listed for a wall structure relate to the transmission of energy. Energy transmission and sound pressure transmission are not the same thing.

In actual problems of noise control, the sound pressure levels are of much greater interest than are energy levels. The "real world" problem is to reduce the SPL of the noise in the receiving room. The difference in the sound pressure levels in two rooms separated by a wall is defined as the noise reduction (NR) in decibels of the wall. Thus a particular wall will have TL values that differ from NR values because the NR values are functions of the wall and the room. The analogy to electrical transmission problems is that sound pressure corresponds to voltage and in the transmission of electrical energy voltage may increase or decrease depending upon the impedance across which the voltage is measured. Voltage and power do not always change in the same manner. Thus NR values may be greater or less than TL values depending upon the wall size and the absorption in the receiving room.

If one knows the TL values of the wall and the reverberation time of the receiving room, the sound pressure levels in the receiving room can be calculated provided the sound pressure levels in the source room are known. The problem of noise control resolves itself into knowing the sound power of the source and the reverberation time of the source room in order to calculate the sound pressure level in the source room, and knowing the TL values of the wall and the reverberation time of the receiving room the sound pressure levels in the receiving room can be determined. One normally increases the efficiency of a wall as a sound barrier by making its TL value smaller and smaller, i.e., the less energy which gets through the wall the smaller the TL value. This implies that more sound is reflected by the wall back into the source room, and to achieve this phenomenon there must be significant discontinuities of speed propagation in the wall with respect to the speed of propagation of sound in air as well as large density discontinuities. Since TL values are less than unity, the dB values of TL are negative; however; it is common practice to omit the negative sign and refer to the dB values as the amount of wall attenuation.

Recently it has been agreed that a single rating of walls can be stated in such a way as to make it meaningful. It is obviously of little use to know the average value of TLs measured at frequencies covering the range of interest since the average value provides no insight as to whether the wall is "good" at certain frequencies and "bad" at other frequencies. Just as knowledge of a sound spectrum (amplitude vs frequency) is important, so is the knowledge of TL versus frequency for a wall important. The new single classification of rating walls is called "sound transmission class" (STC) and relates to the frequency characteristics of a wall. Frequency contours of TL have been agreed upon and in general these TL values, in decibels, rise from 125 to about 350 Hz at a rate of approximately 10 dB/octave and then change to a rate of about 3 dB/octave for frequencies of 350 Hz to 1500 Hz and remain uniform with frequency above 1500 Hz. The STC rating of a wall is given as the uniform (flat) value of this frequency contour for the higher frequencies. Thus a wall with an STC value of 45 dB has transmission loss values (actually attenuation values) which equal or exceed the frequency characteristics of the standard contour which carries the "45 dB" rating. Measurements of TL values are made using one-third octave bands of noise rather than pure tones.

(*Concluded next month*)

How to Mic a Sports Broadcast

ELLIOTT FULL

Perhaps this could also be called "How Not to Mic a Sports Broadcast". The author has been involved in a multitude of sports broadcasts with and without the benefit of an engineer—this over many years. He has thus solved a few problems and left a few more unsolved.

Radio station KXIC AM/FM is the only commercial station in this University city (Iowa City) with 20,000 students. Since Iowa is a big ten school, the ol' gridiron is of prime importance. On the days teams meet, that are considered good at whatever game they are playing, we often have three separate feeds. One for us and two for other stations. Our own game broadcasts are fed to other stations around the state. With bad teams, seasons, and times, we may have a 3 station network. With winners this number may get up to 20. The number of stations is also dependent on whether the contestants are putting the ball over goal lines or just through hoops!

Technically, a sports feed involves: A—1 to 4 microphones; B—an "engineer" (sometimes optional); C—a remote amplifier, earphones, pencils etc.; D—a broadcast loop and a (spare) talk loop.

We have found that inserting an equalizer in the incoming loop from the more distant stadiums noticeably improves the broadcast sounds. Here are the usual ways to mic a sports broadcaster, none very successful:

1. Mic stand and mic on the table. He is often off mic and covered by the crowd.

2. A very light mic mounted on his earphone band. Too heavy, baddish quality, popping, and if it breaks, it's broken.

3. Mic on neck harness. Touchdowns and baskets are made to the left and to the right where our hero is off mic and competing unfavorably with the crowd.

4. Hand held microphone. It doesn't work if he tries to throw it at the ref!

Into this miasma has come the saving graces of the pop singer and the microphone designer. The pop singer faces an audience similar to the sports audience, also often out of their minds. The singer is usually paid better, and his living depends on being heard and having good audio. Somehow this mixture has come up with a new breed of microphone that actually allows the sportscaster to remain in contact with his fans, most of the time.

The new mics differ one from the other, but the small, switchable, pop proof, hand held, cardioid must have come from heaven! Just tell jocko to stay in front of it and start talking when the second hand reaches straight up. The fans may sound like they are on a bad trip and the score 20-20, with the bad guys five feet from pay dirt. If your instructions are not grossly disobeyed, the listening audience may still be with it and treated to a nearly complete game. How does all this add up to a good sports setup? Put the great little cardioid on a goose neck instead of a mic stand. Fasten the goose neck fitting to the table so the mic will swivel right, left and center. Put a drop of oil on the fitting threads so they don't wear out or squeak. Give the color man or men a similar mic with an on-off switch so that they don't catch the engineer with his pot down. Some of these neat mics have silent switches so audio from the crowd noise mic easily covers up any little thunk from the switch. Finally, the crowd noise mic, no problem, this can be any of the many microphones that failed the "sports test" in the last 10 years. Cover it up though, so that any slight breeze outside doesn't cause the vu to go off scale.

The amplifier—why it's a 10-year old home built job that has it's own limiting. It is equipped with 6AL5, 12AX7's, 6386 etc., *remember?* Naturally there is a battery, solid-state standby unit available, just in case. Why hasn't a manufacturer come up with a solid-state, limiting-remote amplifier. This type of amplifier does cover for the engineer when the score is 20-20 and the ref. . . . .!

Elliott Full is vice-president and chief engineer of KXIC AM-FM in Iowa City, Iowa.

New Products and Services

MIXING CONSOLE

●The Studer consoles are now available from U. S. stock. The model 089 is a compact twelve input, three channel studio and broadcast unit with complete equalization, echo, filter, and panpot facilities. The design is unusually complete including talkback with speaker and amplifier, complete monitoring including 20-watt amplification and speakers, a 20 to 20,000 Hz test oscillator, remote control and signalling facilities—and even a built-in ashtray. The console is collapsible and comes in a fitted shipping case and may be powered from power lines of any standard available voltage or batteries.

Mfr.: Studer (dist. by Gotham Audio Corp.)
Circle 54 on Reader Service Card

CASSETTE TAPE

●A new tape development is claimed to make it possible to record extended high frequencies at cassette speed. A new gamma ferric oxide is used. Extreme polishing of the tape surface improves head contact and reduces head wear. The new formulation utilizes a needle-like shape as compared to the rectangular shape of standard oxides. This is states to permit eight times greater density of magnetic particles, improving resolution and reducing distortion and print through.

Mfr.: TDK Electronics Corp.
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COLUMN SPEAKER

●High power capability in excess of 250 watts is claimed for this six eight-inch speaker system. Designated the model PL-250 Galaxy, the column also features a "sparkle producer" horn mounted on top. This is a high-power voice speaker designed to carry vocals above the tremendous amplification levels put out by electric guitars. The system is 64 inches high and has a two-conductor phone jack input. Impedance of the column is 8 ohms.

Mfr.: Atlas Sound
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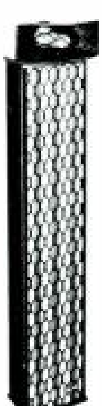

PROJECTOR CONTROL

●Will Szabo of the audio visual engineering firm bearing his name is here presenting a new control consolette to C. D. "Pete" Peterson, Manager of audio/visual and photography for Xerox Corp. The unique circuit permits the Kodak RA950 random access slide projector to be used as a remotely controlled sequential projector. It was designed for the Xerox Corporation Board Room in Rochester, N. Y. Providing for two side-by-side images the black box permits advance, reverse, and focus functions to be controlled by pushbuttons on the executive control consolette located at a lecturn or broad table. A RA950 controller is built into the consolette with all its functions retained, the single controller having access to both or either projectors. When a sequential program is interrupted for random access to any of the 162 slides, a memory restores both projectors to their proper sequential position when the program is resumed.

Mfr.: Will Szabo Associates
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POWER AMPLIFIER

• 120 watts continuous are offered by the model SE460 amplifier. This is actually 60 watts per channel as this is a stereo amplifier. Custom equalizer boards are available that tailor the signal for the specific speaker system connected to it. Distortion is claimed to be so low as to be the residual of the test instruments. Construction is all solid-state.
Mfr.: James B. Lansing Sound
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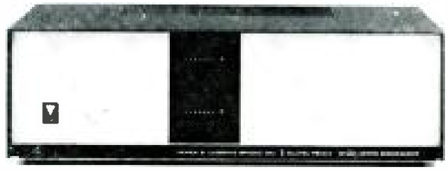

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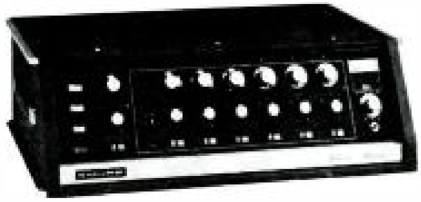
PACKAGED SOUND SYSTEM



● Model VA302 is designed to meet the most exacting requirements in either a portable or permanently installed sound system. The package consists of a console/amplifier and two speaker columns. The console has six input channels, each channel with its own volume, treble, bass, and reverb in-out controls. Each channel is further adjustable by its own attenuator switch for optimized amplification characteristics of various program material. Master controls are also provided. Four anti-feedback switches are included. The system delivers 100 watts RMS Continuous. The speakers each contain two 10-inch drivers, and four eight-inch units for at total speaker cone area of 358.14 square inches. Low impedance microphones are to be used with the system.

Mfr.: Shure Brothers, Inc.
Price: $1095.00
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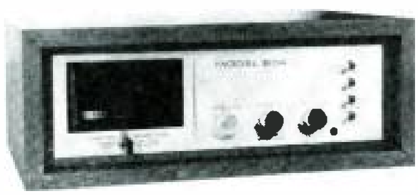
DIGITAL FM TUNER



● Model 804 uses digital readout to replace conventional f.m. tuning. The frequency of the station tuned is flashed on readout tubes. Tuning is in single-step moves upward along the f.m. dial with a center-channel stop made at every frequency for which there could be a carrier. Tuning is crystal accomplished. The unit contains a crystal frequency sysnthesizer that produces 100 separate and stable crystal-controlled frequencies that exactly conform to the 100 possible f.m. station positions between the band extremes. The tuner can be manually or rapidly swept upwards with stops possible every 0.2 mHz. The full 100 channels of the f.m. band can be swept in about five seconds. Tuning is always thus precisely on the center of every channel—even without a carrier present to guide the tuner—and may be remote controlled or clock programmed. Construction is all solid-state (except for the readout tubes) and features circuit designs for the highest state-of-the-art performance characteristics, in both regular and stereo f.m.

Mfr: C/M Laboratories
Price: approximately $1000 (avail. spring 1970)
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SCULLY TAPE RECORDERS — one to twenty-four track and model 270 auto players, many models in stock for immediate delivery. SCULLY LATHES — Previously owned and rebuilt. Variable or automatic pitch. Complete cutting systems with Westrex heads. MIXING CONSOLES — Custom designed using Wiegand Audio Lab modules. From $7,000.00. **Wiegand Audio Laboratories, 3402 Windsor Road, Wall, New Jersey 07719. Phone: 201 681-6443.**

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TWO TAPATHON 702 "Librarian" background music tape players with interspersing feature. Used, good condition. **(801) 278-6242.**

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AT LAST! A headset designed specifically for studio use. Single phone with adjustable headband and vinyl cushion has 2000 ohm impedance. Rugged 20 foot cable is super flexible. $9.95 ea. Quantity discount available. **Studio Engineering Consultants, 19123 Castlebay Ln., Northridge, California 91324**

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PROFESSIONAL RECORDING PERSONNEL SPECIALISTS. A selective service for employers and job seekers: engineers, tape editors, production and studio mgrs, traffic assts, etc. Call us today! **Smith's Personnel Service, 1457 Broadway, N.Y.C. 10036. Alayne Spertell 212 WI 7-3806.**

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SERVICES

WHATEVER YOUR EQUIPMENT NEEDS — new or used — check us first. Trade your used equipment for new. Write for our complete listings. **Broadcast Equipment & Supply Co., Box 3141, Bristol, Tenn. 37620.**

CUSTOM STYLUS — cartridge re-tipping, re-building, replacements. **International Audio Stylus Corp., 111-D Lake Ave., Tuckahoe, New York, 10707 (Telephone: (914) SP9-1297.**

People, Places, Happenings

● Elsewhere in this issue are dates, times, and the place for the **West Coast AES Convention.** Full details, show map, paper, etc. will be in the April issue.

● The **Midwest Acoustics Conference** will be held on Thursday, April 16th at the auditorium of Illinois Institute of Technology Research Institute. The Institute is located in Chicago, Illinois. The Midwest Acoustic Conference is sponsored annually by the **IEEE Chicago Acoustics and Audio Group,** the **AES,** and the **Acoustical Society of America.** Contact **Dan Queen, Perma-Power Company,** 5740 North Tripp, Chicago. Illinois 60646 for additional information.

● Final plans have been announced for the tenth annual **United States Institute for Theatre Technology (USITT) Conference** to be held April 15-18 at the Barbizon-Plaza Hotel, New York City. Panels at the conference will discuss sound reproduction and reinforcement for theater staging, for dance, a review of the National Electrical Code as it affects theaters, new equipment and procedures in technical production, and the architecture and psychology of aesthetic distance. There will be tours of a major scenic studio, theater facilities at the new Julliard complex at Lincoln Center, and a Broadway theater. Additional information is available by writing USITT Conference, 245 West 52nd Street, New York, N. Y. 10019.

● **Sidney R. Goldstein** has joined **Parasound, Inc.** as national manager in charge of Parasound's products and services. Specifically, he will be in charge of promoting **Paul Beaver** and **Bernard L. Krause** Moog Synthesizer recording and broadcast commercial services and the complete line of **Orban/Parasound** Stereo Synthesizer products.

● Production and plant management at the speaker manufacturing plant of **EPI** is now the responsibility of **James Gorman** according to an announcement by company president **Winslow N. Burhoe.** Mr. Gorman comes to EPI with twelve years experience at **Acoustic Research, Inc.,** the last four of which were in the position of assistant production manager.

● **Don Davis** has been promoted to director of commercial sound products for **Altec Lansing** of Anaheim, California. Prior to this appointment he was most recently manager of Altec's Acousta-Voicing process of which he is one of the co-inventors. Don Davis has been with Altec since 1959 and has served as a regional sales manager, national sales manager of audio controls, and manager of Acousta-Voicing. He is also a past western-vice-president of the AES, and has been a western convention chairman. He is currently on the board of governors of the AES. He is also a member of the SMPTE and the Acoustical Society of America. In his new position he will coordinate and direct all marketing and sales activities for all audio controls, commercial sound, and Acousta-Voicing products.

● **Sharyl Story** has joined **Audio Magnetics Corporation** as manager of marketing services, **Ray Allen,** vice-president/sales, announced today.

Miss Story will coordinate the sales department and promotional activities for Audio Magnetics, one of the nation's largest manufacturers of cassettes.

Prior to joining the company, Miss Story was marketing services manager for **Allen Jones Electronics, Inc.** in Gardena.

● **Peter McDonald,** president of **Universal Education and Visual Arts,** recently announced two key organizational appointments. **Leo B. Guelpa** has been promoted to assistant vice president for research and production, and **Dr. Alfred Roman** has been promoted to director of research and development, the post formerly held by Mr. Guelpa.

In his new position Mr. Guelpa will work closely with members of the company's expanding film production and research and development departments in order to provide the necessary continuing liaison between these two operations. He will coordinate the activities of the two departments, especially in the areas of curriculum trends and forecasts, educational film acquisitions, and supplemental film materials.

Dr. Roman is now responsible for the direction and evaluation of all educational film research and development programs, including product planning, product development, and product acquisition. This includes supervision of the company's advisory committee of leading educational consultants located in schools and universities throughout the world.

● **Shelby F. Young** has been named president of **Allied Radio Corporation of** Chicago. Mr. Young, who is 35 succeeds William E. Cowan, 58, who resigned as president of Allied and as a director of **LTV Ling Altec, Inc.** of Dallas, Allied's parent organization. Although Allied Radio has been in business for 49 years, Mr. Young is only the third man to hold the presidential title. Prior to this promotion, he was general manager of Allied's consumer division which included both mail order and retail store sales. Allied now has 36 retail outlets, mostly in the midwest, though two are in Texas.

In just a few short months the Electro-Voice Model 676 has gained quite a reputation as a problem solver—no matter what the odds. Now the 676 has a teammate. The Model 674 has the same unique backbone that rejects unwanted sound... an exclusive with Continuously Variable-D (CV-D)™ microphones from Electro-Voice. And the improvement in performance is dramatic.

Troubled with feedback or interfering noise pickup? Most cardioid microphones cancel best at only one frequency—but CV-D* insures a useful cardioid pattern over the entire response range. And its small size means the pickup is symmetrical on any axis.

Bothered by rumble, reverberation, or loss of presence? A recessed switch lets you attenuate bass (by 5 or 10 db at 100 Hz) to stop problems at their source. And there's no unwanted bass boost when performers work ultra-close. CV-D eliminates this "proximity effect" so common to other cardioids.

Wind and shock noise are almost completely shut out by the CV-D design. Efficient screening protects against damaging dust and magnetic particles, and guards against annoying "pops".

As for overall sound quality, only expensive professional models compare with the 676 and 674. The exclusive Acoustalloy® diaphragm gets the credit. It's indestructible—yet low in mass to give you smooth, peak-free, wide-range response with high output.

The Model 676 slips easily into its 1" stand clamp for quick, positive mounting. The fine balance and shorter length of the 676, and absence of an on-off switch makes it ideal for hand-held or suspended applications.

The Model 674 offers identical performance but is provided with a standard mounting stud and on-off switch. Either high-or balanced low-impedance output can be selected at the cable of both microphones.

Choose the 676 or 674 in satin chrome or non-reflecting gray finish for just $89.00. Gold finish can be ordered for $5.50 more (list prices less normal trade discounts). There is no better way to stand up to your toughest sound pickup problems. Proof is waiting at your nearby E-V sound specialist's. Or write for free catalog of Electro-Voice microphones today.

An important footnote: There is no time limit to our warranty! If an E-V microphone should fail, just send it to us. If there's even a hint that our workmanship or materials weren't up to par, the repair is no charge—even decades from now! Fair enough?

*Patent No. 3,115,207

Electro-Voice®

SETTING NEW STANDARDS IN SOUND

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