THE SOUND ENGINEERING MAGAZINE

db

MAY 1977
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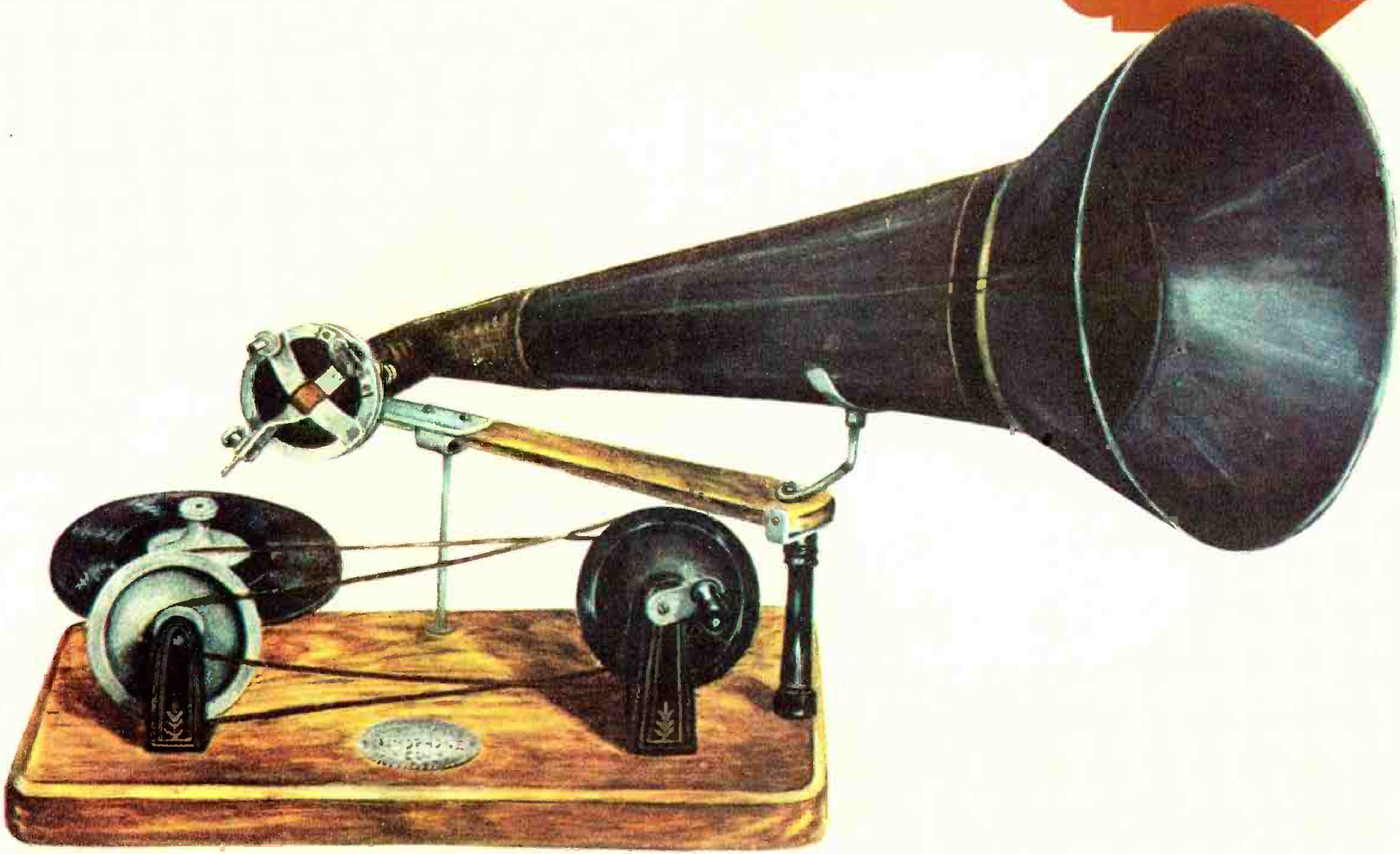

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coming next month

- Alan Fierstein tells us how to construct A PORTABLE OSCILLATOR FOR AUDIO TESTING.
- Wisdom regarding CUSTOM MASTERING is shared by Glenn Snoddy.
- Results of **db**'s test of the SHURE ROOM EQUALIZATION SYSTEM will be reported. This will be a full evaluation of both the generator system and equalizer recently introduced by Shure as a low-cost method of balancing speaker systems against a room. It can do more than that, as will be shown.
- More on the PARIS AES show will be on hand in Part 2 of our report, including more pictures and descriptions of new products shown.
- A **db**-eye view of the NAB bash recently concluded in Washington, D.C. will keep you up to date on what the broadcasters are up to.

Department of Correction

- The February cover carried a wide angle photograph of an attractive studio operation and failed in the page on ABOUT THE COVER blurb to detail anything about it. The photograph is of the Poison Oak Studio in Hollywood, California at 6100 Primrose. The zip code is 90068 and the phone number is 213 462-2698. And let's not forget the handsome acoustic design created by Jeff Cooper.

about the cover

- This fascinating drawing was found in the N.Y. Public Library's ubiquitous picture collection. We quote the obscure accompanying caption:

GADGETS
. . . are not beyond the gambit of the *Index*. This 1887 hand-wound tinhorn gramophone with the leather belt was the first of its kind made in America. It looks neither like a console, an old Dutch sideboard, nor a case of fancy books, but it has an honest comeliness of its own, notwithstanding. Artist: Charles Bowman.

From an unknown publication, dated June, 1937.

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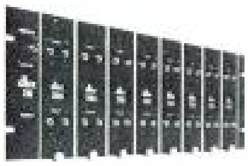

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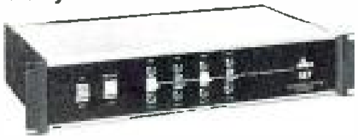

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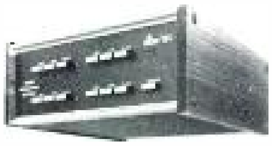

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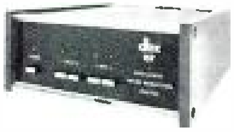

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The latest edition of this popular catalog of tools includes 138 tightly packed pages listing almost any tool you might need. Mfr: Jensen Tools & Alloys.

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PRODUCTS HISTORY

Tracking down the year in which a model was manufactured is simple with this chart of the "Evolution of Ampex Products and Technology." Interesting historical information, going back to 1947, for students. Mfr: Ampex Corp.

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HI-D SWITCH

A product bulletin describes the Hi-D Switch, a miniaturized momentary action pushbutton switch with molded box construction and p.c. terminals for high density mounting on printed circuits or flat, flexible cable. Mfr: Switchcraft.

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Portable and stationary testing equipment, including potentiometers, cable testers, vlf testing, a.c. and d.c. high voltage detectors, and noise isolation devices are covered in a 12-page brochure. Mfr: James G. Biddle Co.

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CALENDAR

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10-13 **A.E.S. Show.** Los Angeles, Ca. Hilton Hotel. Contact: AES., 60 E. 42nd St., NYC 10017. (212) 661-8528.

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

18-20 **Synergetic Audio Concepts Seminar.** Los Angeles. Contact: Don Davis, Synergetic Audio Concepts, P.O. Box 1134, Tustin, Ca. 92680. (714) 838-2288.

20-22 **Consumer Hi Fi Show.** Statler-Hilton, New York City. Contact: Charles Ray, Audio/Communications Show Corp., 30 E. 42nd St., Suite 1620, New York, N.Y. 10017. (212) 986-7592.

23 **National Video Production Workshop. Videomed,** 4878 Ronson Ct., Suite A, San Diego, Ca. 92111. (714) 560-4454.

JUNE

1-3 **Synergetic Audio Concepts Seminar.** Dallas, Texas. Contact: Don Davis, Synergetic Audio Concepts, P.O. Box 1134, Tustin, Ca. 92680. (714) 838-2288.

6-7 **Spring Consumer Electronics Conference.** Chicago, Ill. Contact: Dr. Walter Ciciora, Zenith Radio Corp., 6001 W. Dickens Ave., Chicago, Ill. 60632. (312) 745-4898.

8-10 **Synergetic Seminar.** Indianapolis, Ind. Contact: See above.

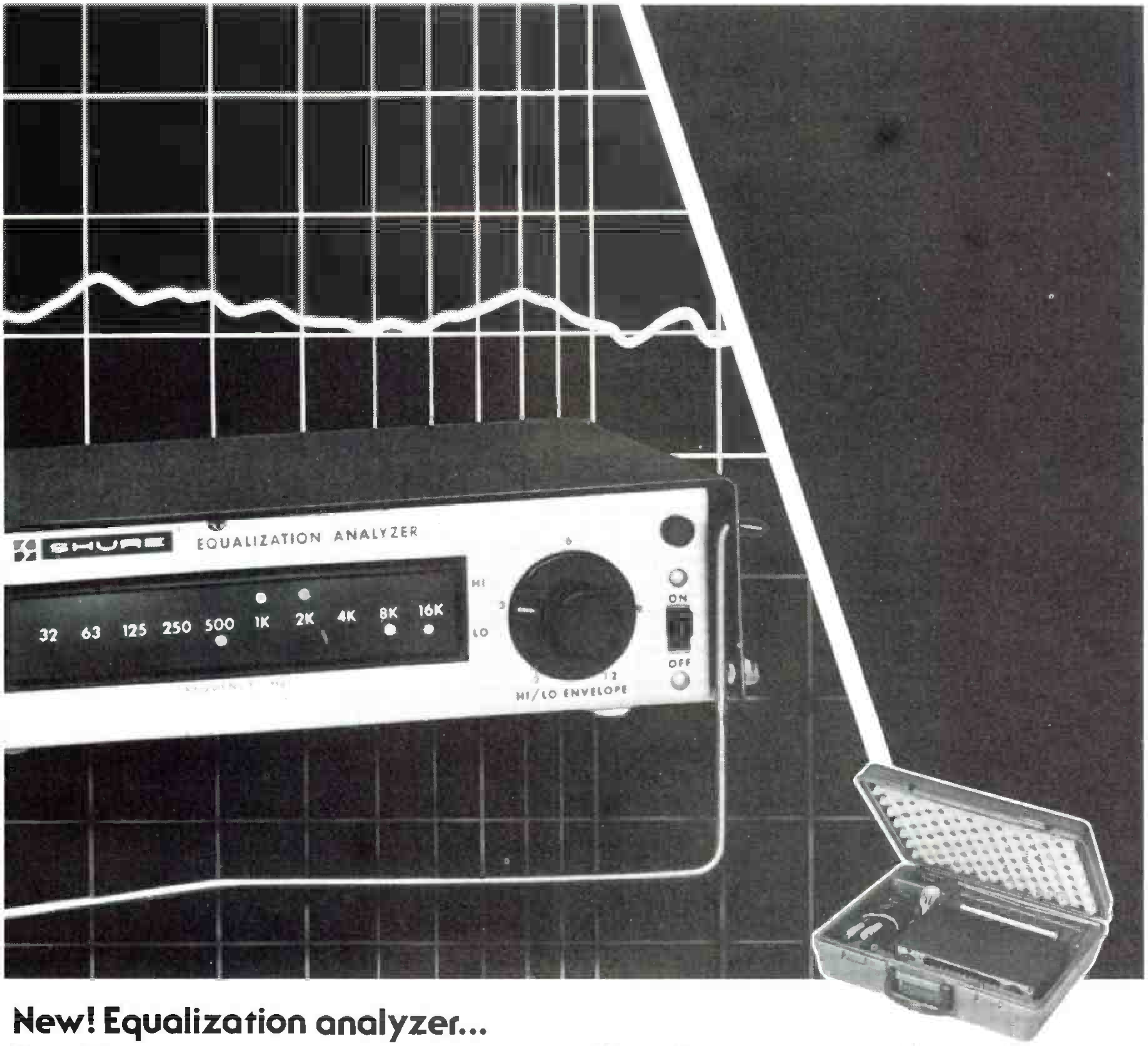
13-14 **Worcester Polytechnic Institute Project Management Seminar. Boston, Mass.** Contact: Peggy Kilburn, Center for Management Research, Executive Plaza, 850 Boylston St., Chestnut Hill, Mass. 02167. (617) 738-5021.

15-17 **Synergetic Seminar.** Chicago, Ill. Contact: See above.

22-24 **Synergetic Seminar.** Minneapolis, Minn. Contact: See above.

27-28 **Worcester Polytechnic Seminar.** (See above) Worcester, Mass.

27-Aug 5 **New England Conservatory of Music,** audio courses and seminars, including electronic music. Contact: Bob Annis, New England Conservatory, 290 Huntington Ave., Boston, Mass. 02115. (617) 262-1120.



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The FM Modulation Monitor

● Every f.m. station must monitor the modulation of its carrier with an FCC-type approved modulation monitor. This function is just as important to the f.m. station as it is to the a.m. station, although some different circuits and techniques are required because of the vhf carrier and the character of modulation. We only have space to discuss a few of the most important circuits.

THE SIGNAL

The f.m. modulation process converts audio amplitude variations into corresponding carrier frequency changes. When the audio signal amplitude peak has caused the carrier to deviate 75 kHz higher or lower than its resting frequency, this is 100 per cent modulation in f.m. broadcasting. The deviation rate of the carrier is determined by the frequency of the audio signal.

The voltage amplitude of the carrier will continually change with modulation as it gives up power to the creation of the many sideband pairs that occur, and its amplitude will actually drop to zero at several points along the way, to full 75 kHz deviation. Each of the sidebands will have an amplitude and phase relationship to the carrier at every instant. These relationships must be maintained throughout the transmission paths or the recovered audio will suffer in some manner. But even though the carrier amplitude changes, so does the amplitude of each of the sidebands. At any instant, the amplitude of the carrier and the amplitudes of all the sidebands present will total up to a peak value that is equal to the peak amplitude of the unmodulated carrier. Thus, the peak amplitude of the f.m. modulated wave remains constant.

THE FRONT END

Direct demodulation of the vhf carrier could be done but this would require a very critical design and operation of the demodulator. The design and operation of a demodulator at low frequencies is far less critical than at vhf, so common practice in monitors is to reduce the carrier frequency to a low i.f. frequency by standard heterodyne techniques. The monitor front end thus becomes a single channel, crystal controlled receiver—even though it is coupled directly to the transmitter output circuits by coaxial cable.

The designer will select an i.f. frequency that is in the neighborhood of 1 MHz or less. At these low frequencies, the signal can easily be amplified and processed by broadband, un-tuned, solid stage stages. Because of the high input signal available, very few stages of i.f. amplification are required. Two or three stages may be used, but one or two of these will act as a limiter rather than an amplifier. The bandwidth of this i.f. section must be the same as that required for the f.m. carrier since it will also deviate plus and minus 75 kHz.

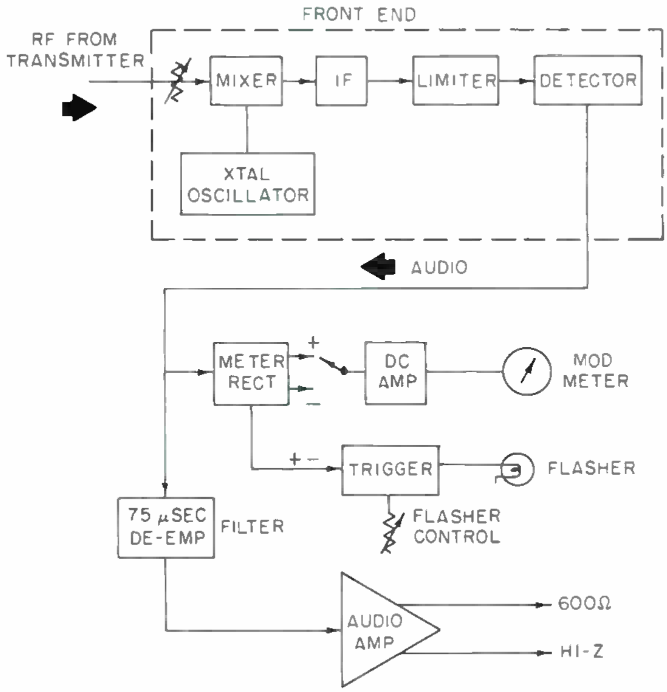

Figure 1. The basic monitor circuit.

DEMODULATION

Since demodulation is the reverse process of modulation, the demodulator's function is to change carrier frequency deviations back into audio amplitude variations. There are a number of different circuits which can be used for demodulating the f.m. carrier and with the advance of solid state technology, the number is increasing. All types do not yield the same degree of accuracy, but can be suitable for a particular application. The monitor is a test instrument, so the demodulator used must yield a high degree of accuracy.

One type of demodulator which is popular in monitors is the *pulse counter discriminator*. This is a very broadband demodulator that requires few components, produces low noise and distortion in itself, and requires no tuning whatever. Two versions can be used: a balanced arrangement with four diodes in a quad, or a single-ended version with only two diodes. We will discuss the single-ended version.

LIMITING

The i.f. carrier will pass through one or two limiter stages to remove any amplitude modulation that may be present on the carrier. The i.f. signal is sine wave in shape although it is constantly changing frequency during modulation. The limiters will simply clip the positive and negative peaks of these sine waves so that they now appear as square waves. Out of the limiter, the i.f. signal will thus be one positive and one negative square wave pulse for each cycle of i.f. To make certain the peak to peak amplitude of these pulses remain constant, the output limiter is operated from a well regulated d.c. power supply. The limited i.f. signal is fed to the demodulator.

THE DISCRIMINATOR

The complete discriminator circuit is composed of a series coupling ca-

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But the real mark of Community radial isn't apparent unless you cut it in half as we have done here. If you look closely at the cutaway, you will see that the corridor in the horn just past the throat gets extremely narrow before it flares out. That pinch in the horn is absolutely necessary to any mathmatically and acoustically correct radial horn, whether it is metal or fiberglass, and only Community does it the way it should be done. And not only do we do it right, we do it cheaper.

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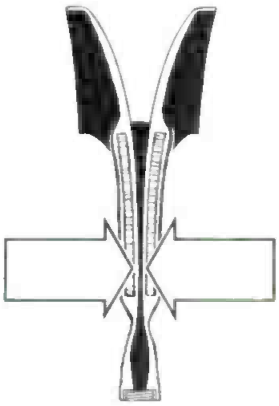

	BRH90			RH90			SRH90		
Flare Rate	240Hz			345Hz			345Hz		
Operating Range	500Hz up			600Hz up			1,000Hz up		
Size: H	11 1/8"			12¼"			6½"		
W.	33⅝"			30⅜"			24¾"		
D.	21"			20½"			18¾"		
Weight	25 LB			20 LB.			12 LB.		
Finish	Black, High Gloss			"			"		
Horizontal Dispersion	KHz	-3dB	-6dB	KHz	-3dB	-6dB	KHz	-3dB	-6dB
	6	85	95	6	80	90	1.2	95	100
	2	90	90	2	90	100	3	90	95
	10	80	90	10	85	90	10	85	100
Vertical Dispersion	KHz	-3dB	-6dB	KHz	-3dB	-6dB	KHz	-3dB	-6dB
	6	50	90	6	55	100	1.2	50	70
	2	35	50	2	35	50	3	40	65
	10	20	35	10	20	35	10	20	30

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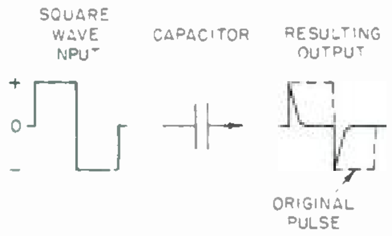

Figure 2. The coupling capacitor differentiates the square wave pulses into very narrow spike-like pulses.

pacitor, two detector diodes, a shunt filter capacitor and a load resistor. The coupling capacitor must be considered part of the design since it will shape and affect the square wave signal applied to the demodulator, so its value is somewhat critical and must also be of close tolerance.

A capacitor has a reactance value that is an inverse function of frequency, so this coupling capacitor can become discriminatory in itself and thus control the current flowing through it. This control is not in the resistive sense but more in affecting the shape of the pulse that it will pass. A value must be selected that will not change the peak amplitudes of the pulses that pass.

A capacitor in series with a square wave will not pass the d.c. component (flat top) of the pulse. Only the a.c. component (leading and trailing edges of pulse) will pass, so the resulting pulse is a very narrow 'spike" which represents the leading edge and a negative overshoot indicating the trailing edge. Since the i.f. signal is a series of positive and negative going square wave pulses, the negative overshoot combines with the leading edge of the negative pulse. The output side of the capacitor then. is a series of very narrow, positive and negative going, spike-like pulses, but they all have the same amplitude. The frequency of the i.f. signal will constantly change during modulation, so when it goes higher in frequency, there are more cycles in the signal according to the frequency at that instant. The output side of the capacitor will have a corresponding number of positive and negative spikes. When the i.f. deviates lower in frequency, then there are fewer cycles and less spikes passing. The amplitude of all these pulses must be equal in amplitude; the only change is the number of them. This is the signal presented to the detector.

THE DETECTOR

The detector is essentially the same as the a.m. detector and works in the same manner. A second diode is used to improve the action. The only real difference is the input signal. In a.m.. the input signal is a constant frequency r.f. that varies in amplitude, while in the f.m. detector the input is an rf signal of constant amplitude that varies in frequency.

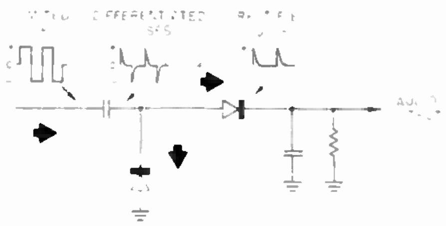

Figure 3. The detector action. Pulses passed by series diode are counted by the filter capacitor.

The diodes are placed in polarities to the input signal so that the shunt diode conducts on every negative pulse and shorts this pulse to ground. This diode simply acts as a clamp. preventing any voltage buildup on the capacitor during negative pulses. and sets a ground reference to the circuit during those times. The series diode is polarized so that it conducts on each positive pulse and passes these on to the filter capacitor and load resistor. With-

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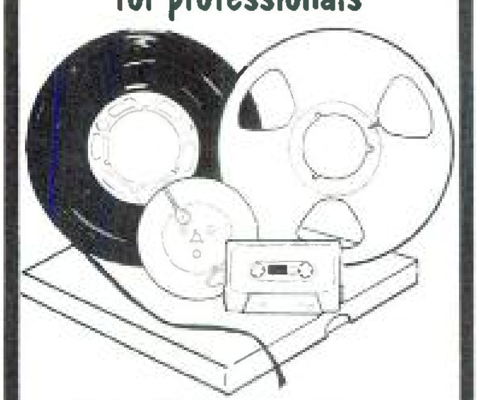

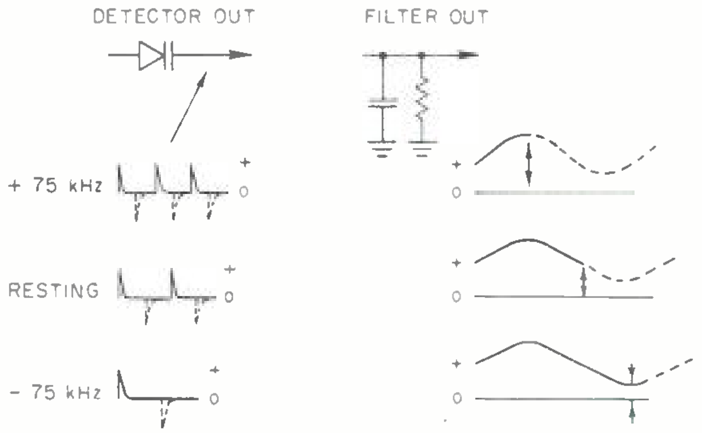

Figure 4. The number of positive pulses out of the detector cause the d.c. voltage amplitude out of the filter to trace out the original audio sine wave. (Note: the number of pulses shown only indicate a relative number.)

out modulation, the steady i.f. signal develops a steady d.c. voltage across the capacitor and load resistor. With sine wave modulation, as the positive half cycle of the audio builds up to its peak, the carrier deviates progressively higher in frequency a corresponding amount. The *number* of pulses flowing to the filter capacitor increases.

The capacitor integrates, or "counts" the number integrates, or "counts" the number of pulses. As the number of equal amplitude pulses flow onto the capacitor at an accelerating rate, the capacitor charge buildup in amplitude corresponds to the amplitude of the positive half cycle of the original audio. As the negative half-cycle of audio causes the carrier to deviate lower in frequency, the number of pulses flowing into the capacitor becomes progressively less. The capacitor loses charge between pulses a corresponding amount. The output voltage across the load resistor is a varying amplitude d.c. voltage, the variation in amplitude a replica of the original audio signal.

PEAK MODULATION

The percentage of modulation is measured with a high impedance voltmeter circuit which incorporates the required FCC damping characteristics. Audio from the detector is fed to a meter rectifier which provides a varying d.c. output that contains the required damping. The varying d.c. voltage is amplified by a d.c. amplifier to drive the front panel modulation meter. The rectifier incorporates a phase splitter and combiner so that the plus and minus modulation peaks may be selected for the meter circuit. Selection of the plus and minus modulation peaks does not have the same significance in f.m. as it does in a.m. However, when tone modulation is used, these peaks will indicate linearity of the modulator and system and they should agree.

Since the meter circuit cannot follow fast modulation peaks, an electronic peak indicator circuit is used. A section of the phase splitter combines the positive and negative peaks in full wave fashion so that both peaks are positive and in tandem. The full wave audio peaks are fed to an electronic trigger circuit whose sensitivity is preset by the front panel flasher control. When either a positive or a negative audio peak exceeds the preset threshold of the trigger, it will operate and turn on a flasher light.

AUDIO

Aural monitoring is provided by a high quality audio amplifier. FCC rules require that the audio be passed through a 75 μsec pre-emphasis filter before modulation, so if we are to listen to the recovered audio, it must be restored to its proper response curve. The recovered audio will pass through a 75 μsec de-emphasis filter before feeding the audio amplifier. We should point out that the modulation meter and flasher circuits do not use the de-emphasis filter since they must monitor the actual carrier modulation—and that is pre-emphasized audio.

The audio amplifier will provide a 600 ohm balanced output for external monitor amplifiers, and it may or may not also provide a high impedance drive for a distortion analyzer. If the distortion drive is a separate amplifier, this will also use the 75 μsec de-emphasis filter.

OTHERS

Monitors will contain many more circuits and functions than we have been able to discuss here, and some will use a different demodulator. The important element in any monitor is the accuracy it yields when measuring the modulation or making tests on the transmission system. Next month we will discuss some of the problems of monitoring the f.m. signal. ■

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db theory & practice

NORMAN H. CROWHURST

• Periodically, I get inquiries about the use of spectrum analyzers for various testing purposes. Sometimes the questioner is asking if it cannot be used for a purpose that is not feasible. But there are many ways that the spectrum analyzer can be used, performing better, as well as faster, than alternative methods. A spectrum analyzer is a useful item of equipment that can replace quite a battery of formerly needed instruments, an economy for those setting up shop.

LOUDSPEAKERS

I have discussed this before. Properly used, loudspeakers can be tested on-site, as installed, so you get the response of the building as well as that of the equipment. To do this, you need a pink noise source—that was what I discussed before—and enough microphones to obtain good averaging of what the system distributes so that people will hear well, wherever they may be sitting.

The advantage of using noise is that it eliminates the major basis for invalidation by standing waves because noise is aperiodic. But noise can still respond to resonator effects and, if the auditorium has major resonances, which can also be viewed as a build-up of standing waves, this method of testing will show them.

ELECTRONIC COMPONENTS

The spectrum analyzer really can perform a number of functions at once, relative to electronic components, by using sinusoidal inputs of various kinds. For measuring frequency response, power, and various kinds of distortion related to power, the analyzer performs a number of functions simultaneously, as well as performing some of them better than equipment designed specifically for the purpose, such as a harmonic distortion meter, or one of the intermodulation distortion testers.

For checking frequency response and power output, the spectrum analyzer merely serves as a rather sophisticated output voltmeter that tells you frequency, or frequency content, as well as just the voltage; its function is to obviate false readings. If you measure frequency response without looking at what you are measuring on a 'scope, you could be measuring either hum or noise instead of the sound in question. A spectrum analyzer cannot make that mistake.

HARMONIC DISTORTION

The conventional harmonic distortion meter merely nulls out the fundamental and measures everything else that is left. If you change frequency, ever so little, you have to renull the meter. It never gives you any indication of what the residue is that you are measuring, although most meters provide output terminals or a jack that enables you to look at the residue on a 'scope.

There is one disadvantage of both the harmonic distortion meter and the spectrum analyzer that a quite simple tester can overcome, the fact that both of them look at frequency content of the output without reference to input. You assume that the sine wave input is perfect. The simple tester overcomes this disadvantage, where it is significant.

The spectrum analyzer gives you an instant display of information that a wave analyzer would take you much longer to explore. By expanding the display scale, you put the fundamental off the left hand side of the 'scope, so you look only at the harmonic components after you have calibrated your readings for level on the fundamental.

INTERMODULATION DISTORTION: SMPTE

For both forms of intermodulation distortion, you get more information or more accurate information with a spectrum analyzer than with the older equipment designed for just that test. The input signals, of course, must be obtained in just the same way. For the SMPTE test you need one low frequency and one high frequency and, to use the standardized test, you set the levels so the lower frequency is exactly 12 dB higher in level than the higher frequency.

Now, if you are looking at the output, distortion shows up as lower level sideband lines on either side of the bar that represents the higher frequency. As with the distortion meter, the conventional IMD meter for this measurement merely eliminates the two input frequencies, and sums these sidebands.

To get a reading you need to evaluate the rms sum of the sideband components. Thus if there is 0.8 per cent second and 0.6 per cent third, the total is 1 per cent. But I feel it is more important to see what you have than to get an accurate single number that represents all of it.

The real advantage of the spectrum analyzer is that it catches some of these sideband components that the regular, old-fashioned meter misses. The old meters merely measure residual amplitude modulation of the higher frequency tone by the lower. If any frequency or phase modulation of the higher frequency tone by the lower occurs, the old meter does not see it.

As far as effect on reproduction is concerned, if the low frequency tone modulates the high frequency one, in either way, it is equally audible as a form of distortion. Once it is released into the air to become an acoustic wave pattern, it is difficult, if not impossible, to tell which form of modulation it is.

Frequency, or phase modulation, and amplitude modulation, when looked at the way in which a spectrum analyzer does, produce the same spurious components. So the readings you see on the spectrum analyzer screen are more representative of what you hear than the reading obtained from an IMD meter.

INTERMODULATION DISTORTION: CCIF

This test uses two higher frequencies and, normally speaking, produces a lower tone as the spurious byproduct. This is how the conventional tester for this type detects the lower tone. For example, if you put in 5000 hertz and 5100 hertz, the instrument looks for 100 hertz as the distortion byproduct.

This assumes only one kind of distortion—asymmetrical, or second harmonic, as referred to a single sine waveform. Second order curvature of the transfer characteristic will produce, in addition to second harmonics of the input frequencies, which would be 10,000 and 10,200 hertz, the sum and difference frequencies, which are 10,100 and 100 hertz.

Now suppose the main distortion is due to third order curvature. The distortion products will be third harmonics, which are 15,000 and 15,300 hertz, two sum frequencies, which are 15,100 and 15,200 hertz, and two difference frequencies, each obtained by subtracting one frequency from twice the other, thus equalling 4900 and 5200 hertz.

Obviously a test circuit that looks only for 100 hertz will find none of these frequencies. But those within the audio range will show up on a spectrum analyzer. Perhaps you should spread the original frequencies a little, say to 5000 and 5500 hertz. Then the second order components would be 10,000 and 11,000 harmonic, and

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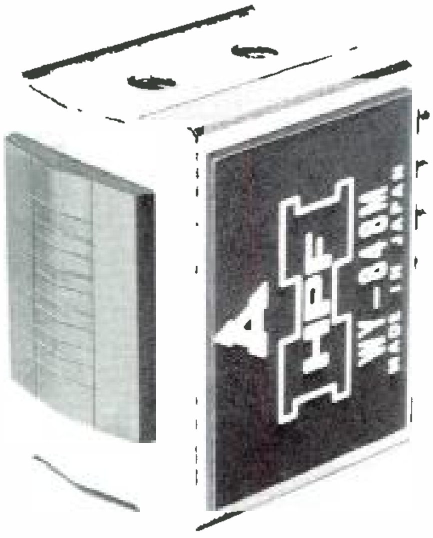

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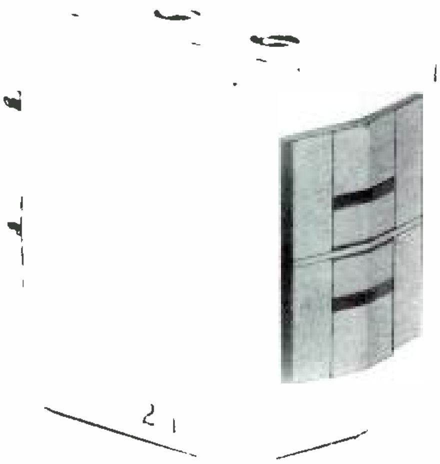

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10,500 with 500 as sum and difference.

The third order components would be 15,000 and 16.500 for harmonics, with 15.500 and 16.000 for sum and 4500 and 6000 for difference frequencies. Using the same input frequencies, for fourth order curvature, there would be 20,000 and 22,000 for harmonics, and three sum frequencies, three difference frequencies, found by combining twice and twice again the input frequencies, and three times one with once the other, each way. Thus the sum products will be 20.500, 21,000 and 21.500 hertz. And the difference products will be 1000, 9,500 and 11.500 hertz.

Every higher order curvature increases the number of additional distortion products. Going to fifth order, the harmonics are 25.000 and 27,500 hertz. The sum frequencies are 25,500, 26.000, 26.500 and 27.000 hertz, while the difference frequencies are 4,000, 6.500, 14.500 and 17,000 hertz.

Note that asymmetrical distortions, second and fourth, produce a component that is much lower, 500 or 1000 hertz, while the symmetrical distortions, third and fifth, produce everything higher than the input frequencies, except one of them.

If you want a distortion component from these curvatures that is lower in frequency, you must change the test frequencies, based on a ratio. To pick out third order curvature, one frequency must differ from twice the other one by the desired difference. Thus you could use 5000 and 9500 hertz, yielding harmonics 15.000 and 28,500 hertz, sum frequencies 19.500 and 24,000 hertz, and difference frequencies 500 and 14.000 hertz.

Figuring out ratios and differences can get complicated but with a spectrum analyzer and two oscillators, of which at least one has variable frequency, you can change the ratio and watch the various byproducts "crawl" across the screen in different directions, which can be quite instructive, to help determine what each one represents.

POWER VERSUS DISTORTION

This is another thing that a spectrum analyzer makes more meaningful. As you increase input level, the height of the bars representing different distortion products varies differently. The most meaningful way to observe this is to use a single frequency input and watch the harmonics. For example, the first order of harmonic to appear may be second, due to asymmetry in the way an amplifier begins to overload.

But then as input is further increased, the second order of harmonic may not grow more than a certain amount before third and fifth virtually shoot up to overtake it, meaning that overload is now more nearly symmetrical.

There are more intriguing things you can do with a spectrum analyzer, that I will get into some other time. If you want more information about spectrum analyzers, try contacting Kay Elemetrics Corp. 12 Maple Avenue. Pine Brook, NJ 07058, or Tektronix, Service Instruments Division, P.O. Box 500, Beaverton, OR 97077.

There may be other sources; those are companies that have sent me information about their products within the last year. Not so long ago, a spectrum analyzer would have been prohibitive in cost as a test or measurement tool. A research or production company wanting to perfect a product might use such a tool, but for evaluating the usual equipment, it would have been too expensive to justify.

But that is changing. Now you can get a spectrum analyzer for less than the combined cost of all the other items that it will enable you to do without, even enabling you to make better measurements. So I would say it is a worthwhile device to own. ■

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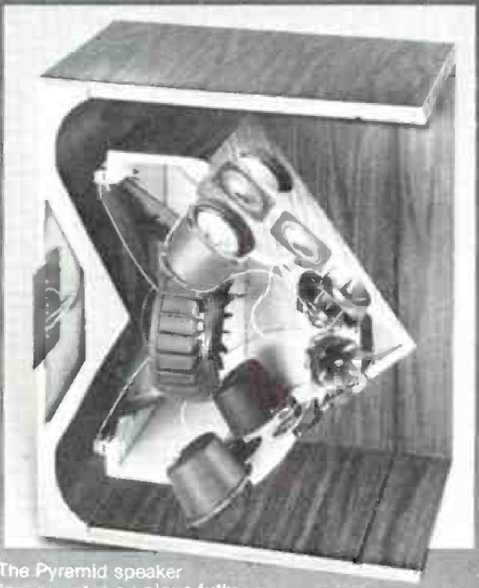

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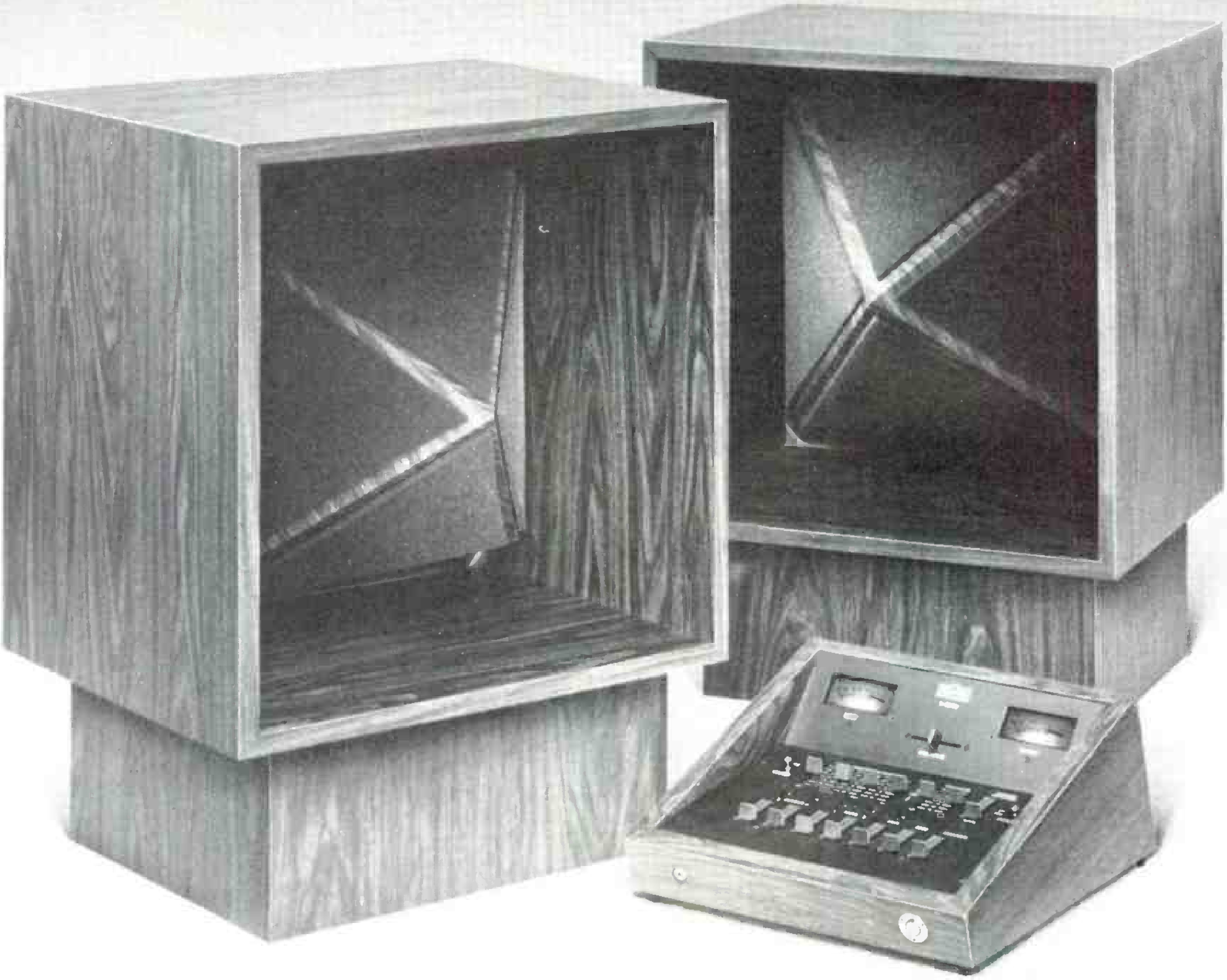

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db sound with images

MARTIN DICKSTEIN

• In the more usual projection installations, there is a great deal of thought given to getting the largest image on the screen with the greatest amount of brightness, and rightly so. Where the image has to fill a screen large enough to be seen at the rear of a room in which the last row is more than, say, six times the width of the screen, a smaller image would be lost. And brightness, of course, drops off as the image size increases. The philosophy is, therefore, think big and bright. What happens, however, when the situation calls for a smaller image? Any problems? There sure are a few, anyway.

In a recent sales presentation with which I travelled, slides were used to supplement a large video projection image. The design of the program called for a three-image lineup vertically of the slides alongside the large video screen. The size of the slides had to be slightly less than one third that of the height of the video image. This allowed for a small space between the slide images. From the front view, the total height of the slides was the same height as the whole video screen.

Since the show travelled, one of the prerequisites was to be able to assemble and dismantle equipment as quickly as possible with as few pieces, all easily manageable, as would fit into as few small (quantity as well as size) crates as possible. For the three screens, a frame was made with rectangular cutouts in the proper slide proportion, which would be placed alongside the frame of the video screen. This meant that for a six-foot wide screen for t.v. the slides had to be 20 in. high, thus limiting the width to 30 in. With this kind of arrangement, the image could either be projected to the front or rear screen, depending on the location of the setup.

LONG NARROW ROOM

In one location, the room was long and narrow. The set, including the t.v. screen and the slide screens, were spread across almost all of the front of the room. The slide projectors were set up along the side wall and hidden from the view of the audience, both side and rear, by putting up a wood frame and draping black cloth all around. The material was quite heavy to shield the light and keep the sound of the motors and fans and slide tray rotation from getting to the audience. The area directly behind the slide projectors was, of course, not suitable for seating because the slide screens would be blocked from view and the images keystoned slightly because of the very small angle. Masking of the screens hid the keystoning from the front view.

First, we found that all slide projectors are not created equal. A minor difference, but a difference nevertheless, was the time it took to change slides, and then to auto-focus. When the three units switched simultaneously, two were in precise step, but one was slow. The show's creator deemed it essential that the three slides come up exactly at the same time, and since the image was a 3-slide high picture, he could expect this effect to occur as planned. After a change of units, it worked out okay.

Then we found that the slides did not line up in a 3-high image. Although the projectors were positioned carefully, and adjusted, the image never quite looked as it should. The top of the building or statue was off to one side from the middle. Having the slides mounted in plastic holders made it easy to make this adjustment. But we had to keep in mind how the slide was cut. In one case, moving the slide inside the mount required masking the side that became exposed.

An interesting situation arose when it came time to balance the image intensity of the slides compared to the large screen video image. Since the t.v. image was projected onto a curved screen the audience saw what closely resembled a t.v. screen. The brightness and colors were good when seen alone, but alongside the slides, the little images were bright enough to distract from the main image. The original matté white material was replaced with white cardboard. There was still not enough of a drop in brightness, so grey cardboard was tried. This was much better. An experiment was also made with black cardboard, and although the image brightness of the slides was quite acceptable in relation to the video image, because some slides were quite dark, the decision was to work with the grey reflective surface. Incidentally, since the distance from the projectors to the screens was quite short, zoom lenses were used, which decreased some of the light emitted out of the projectors. Machines with incandescent bulbs were utilized to provide lower output than possible with the newer halogen bulbs.

WORKING REAR SCREEN

When the whole show was set up out of town, it was found that front projection for the slides would not be feasible. The room to be used was wider than the original site, and seating could not be limited to just the center of the room. To shoot from the rear of the audience would have required auditorium lenses of about 13 in. focal length, and these were not readily available. To avoid setting up slide projectors within the audience area, it was decided to work with a rear screen for the slides. The setup was made away from the wall by enough distance to allow the projectors to fill the open areas provided, the same 30 x 20 in.

There were several ways to go on this. One was to use rear projection lenses, the ones with 90-degree angles and a mirror internally set to round the bend, or we could use straight lenses projecting into mirrors which could reverse the images for proper viewing from the front. Another possibility was to project directly onto the screens and reverse the slides in the tray. We had tried to provide for all possibilities before hitting the road. Six slide projectors were rented by phone, days in advance of arrival, to permit a full hundred percent backup. With these, the longest throw lenses that could be gotten were requested, and the shortest rear projection lenses that were available. The order also in-

cluded the fact that incandescent lamp projectors were required. (This last request received lifted eyebrows that were audible on the phone!)

When we found that rear projection lenses were not available, we packed our own. The longest lenses possible, according to the rental supplier, were 11 in., not enough for our needs. Hence, the decision to work rear screen. To help us there, the a/v supplier had provided front surface mirrors and straight 2 in. lenses as we requested (we asked for the straight lenses so that we could shoot straight for the screen with reversed slides as the third alternative). It was found to be quicker and easier to work with the last alternative, so the slides were turned 180 degrees.

As everyone knows, for the operator to work behind the rear projection screen when he is lining up and focusing the slides, it is necessary for the viewing room, as well as the area behind the screen, to be dark. (Our setup put the projectors about 5 feet behind the screens.) The images were finally set up properly with no keystoning and good vertical lineup, but it was noticed that the slides seemed very bright. Well, rear screen images are usually brighter than those reflected from a matté white front screen, but these were even brighter than usual. The projectors provided contained halogen lamps. Units with incandescent lamps were not available. (We had even packed the 500-watt incandescents in the spare kit in case the units were provided with the high intensity four-pin bulbs. Now, even these could not be used.) The supplier thought, even though we had tried to explain the problem on the phone during ordering, that we would be more pleased with the high intensity lamps since that's what everyone else always asked for.

There were also a couple of other problems that became visible while the room and rear projection areas were totally dark. First, while the projector lamps were on, one of the screens appeared to be brighter than the others, even with a blank slide in the aperture. Second, with the projection bulbs off, there appeared on each of the screens a small dim light right in the center of the screens, with the projectors in the "fan" position. The first problem was easily solved. Although there is some advantage in using the new translucent 80-slide drums in longer throw front or rear projection, close up there is enough light coming through to illuminate the screen as compared with the earlier grey or black trays. When all screens are the same, that is, with white drums behind them, such a situation might be acceptable, but when one screen is white and the others are opaque, the difference is very noticeable. In this case, black tape was run around the white drum and the problem was solved. A transfer of slides to another drum, a grey or black one, would also have worked, but only white drums were provided with the rented projectors, and only two dark trays of slides were sent with the show, along with one white one.

DIM WHITE LIGHT

The second problem, the small dim white light in the middle of the screen, was found to be from the auto-focus light inside the projectors. In anticipation of some sort of light source problem, we had packed a small case containing a variety of neutral density filters, the same kind or similar to those used in photography. In combining these into equal values for all projectors to cut way down on the light output of the sources, and putting the

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filters in front of the lenses, we not only were able to achieve dimmer slide images to match the t.v. screen, but we also decreased the auto-focus light to a low enough level so that it was not noticeable.

Now that the screen was dark grey in appearance, it was noticed that there was a very slight illumination on the side of each of the small screens, sort of a faint triangular shape. Turning off the projectors solved the problem, but even in the "fan" setting the light still showed. Obviously it came from the auto-focus light, but how? It turned out that enough light came through the slot of the front elevation knob to show on the screen. A small piece of black tape over the slot solved that one. Since the projectors were set accurately and would not have to be raised or lowered, the taping of the front knob did not bother the operation of the projector.

FOCUSING

Just a tip or two more on small-image projection. In a rear screen set-up, you're probably aware that the image is seen best from directly in front and that as you move around to one side or the other, the intensity seems to drop off on the side of the screen farther away from you. The same is also true when a rear screen image is high compared to the eye level of a seated audience. Toward the rear of the room, the viewer's angle of observation may be equal to the top and bottom images of a 3-image high arrangement, but at the front of the audience the viewer has a much higher angle to the top image and it may seem much dimmer to him than the bottom or midddle ones. This should be considered when setting up the seats for viewing.

Try also not to arrange the seating at too wide an angle as slides will lose legibility to those people in the far out seats. You may also find with small images that the center of the slides may appear too bright or washed out when viewed from directly in front. The density of slides does vary and the background colors will make a difference also. A marked shifting may also be noticed with auto-focus projectors when slides that are mounted differently are used in the same drum. Cardboard slides will shift more than plastic mounts because the latter have glass to keep the film from buckling or bending. Also, a shift will be noticed when slides are put into the tray left-to-right reversed because the emulsion side will be toward the lamp in one slot away from the source in the next. (Incidentally, the latest projectors for home use from Kodak come with a special lens made specifically to compensate for cardboard-mounted slides where the film has a slight curvature within the mount to keep them from buckling while in the heat of the lamp.) Great care must be taken when setting up rear screen projection that focusing shifts will not move the image up and down or left to right on the screen. This will happen if the projector using a 90-degree lens or a straight lens and mirror is not aimed in a precisely horizontal and vertical position at the center of the screen area. Shifts like this are very noticeable when the images have to line up exactly.

Well, there are some thoughts on projection for a small screen rather than a big one. Maybe the older, lower power, incandescent-lamp projectors had their advantages after all. And maybe thinking small, rather than big, isn't as easy as it seems. ■

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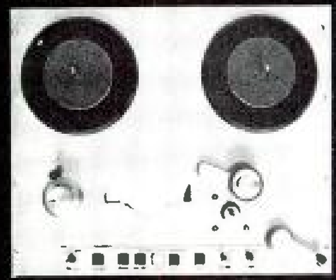

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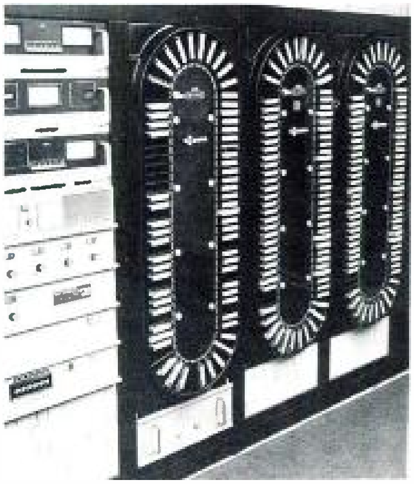
CARTRIDGE GO-CART



• 78 cartridges, borne on carrier trays linked in a flexible chain over an oval belt can be accommodated on this go-cart. Controls direct the desired cartridge to the play table, where it is removed to a stationary cartridge deck for stable play and accurate tape-to-head alignment. When play is completed, the cartridge is rewound and automatically replaced in its carrier, the carrier responding to instructions governing the next cartridge scheduled. Maximum access time is 8 seconds. The unit swings out on hinges, remains operational even while opened. Compatible with most existing systems and control devices, its micro-processor computer-based control logic is flexible for various usages. The manufacturer also offers a 42-cartridge unit.

Mfr: IGM

Circle 51 on Reader Service Card

DIGITAL TIME DELAY

• Priced under $2,000, Delta-T Model 92 is designed for smaller, lower-budgeted installations. Claimed dynamic range is better than 90 dB: noise and distortion claimed less than 0.1 per cent. The unit provides two adjustable audio signal delays of up to 120 ms, each controlled by a single front panel knob. Also included are audio and input and output transformers, an automatic, fail-safe, audio bypass feature, silent power up/power down circuitry, and rear mounted XLR-3 type audio connectors. A five-position led headroom indicator is calibrated in 10 dB increments below limiting. All units have universal power compatibility, 115/230, 50/60 Hz, with international connectors and detachable power cords. Designed for rack mounting in 3½ in. of space, the unit has plug-in modules for its memory and audio subsystems. Construction is all solid state, employing mos-ram memory and low power Schottky i.c. logic.

Mfr: Lexicon, Inc.

Price: Under $2,000.00

Circle 52 on Reader Service Card

CORRECTION

An error was made in reporting on the SM11 Lavalier Microphone from Shure Brothers, in our March, 1977 New Product section. The diameter was incorrectly reported as 1-5/16 in. The diameter of the microphone is 9/16 in. Its length is 1-5/16 in.

CASSETTE DUPLICATOR SYSTEM

• A reel-to-reel bidirectional master and cassette slave loaders are used on this modular automated duplicator system. The cassette tape is duplicated while it is being loaded into the cassettes, speeding up reproduction time to 32 to 64 times normal. The system has a frequency response to 15,000 Hz; there is a claimed flutter of less than 0.1 per cent. An optional automatic cassette feeder automatically performs the tasks of cassette insertion, leader extraction and threading of the leader into the splicing platform. Then the automatic operation of the cassette slave loader takes over to cut the leader, splice the tape to the leader and load the cassette with tape while recording the signals from the master. It then cuts tape and splices the leader at the end of the program, and winds the remaining leader into the cassette, ejecting the finished product from the loader. The feeder can be ordered with the original installation or installed later. The manufacturer has an extensive operator's training plan, as well as a tryout period when a prospective customer can use the facilities, on a service basis, to have cassettes duplicated before actually ordering the equipment. The systems can be used for volumes of from 10,000 to 1,000,000 cassettes per month, adaptable to long or short runs, 2- or 4-track formats.

Mfr: Recortec

Circle 53 on Reader Service Card

STEREO PREAMP

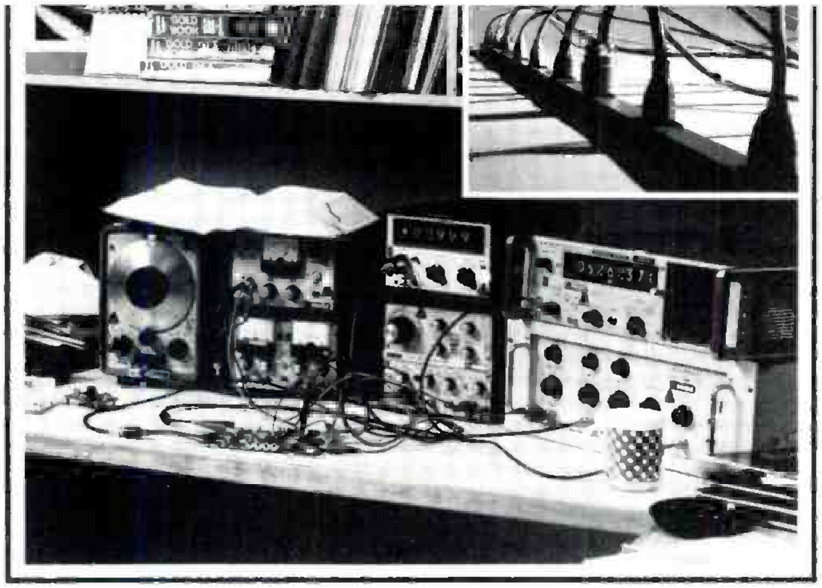
• Low-noise monolithic operational amplifiers on the M-200 stereo preamp are claimed to have less than 0.01 per cent distortion and i.m. from 20 to 20,000 Hz with a s/n ratio of −82 dB (phono). There are separate bass and treble controls: power and volume controls operate independently with tone controls switched in and out of the circuit by means of a separate tone control switch. The phono stage has low cartridged interaction, and claimed low distortion with good square wave response and low noise while using a cartridged input. On the rear panel there are two switches and one unswitched a.c. convenience outlets.
Mfr: Bauman Research Instruments Company
Price: $350.00.
Circle 54 on Reader Service Card

PORTABLE SOUND BAFFLES

• A series of three interlocking studio "gobos" in widths of 3 ft. or 3 ft. 6 in. offer claimed separation of better than 0.85 NRC. Smaller models designed for drum platforms and seated instrument positions have sloping plastic tops which maintain visibility and produce variable sound reflection. All models have brass casters and utilize an interlocking hinging system for connection and support. The baffles are covered with washable fabric in a variety of colors, fireproofed on request.
Mfr: Sugarloaf View
Circle 55 on Reader Service Card

PHONO PREAMPLIFIER

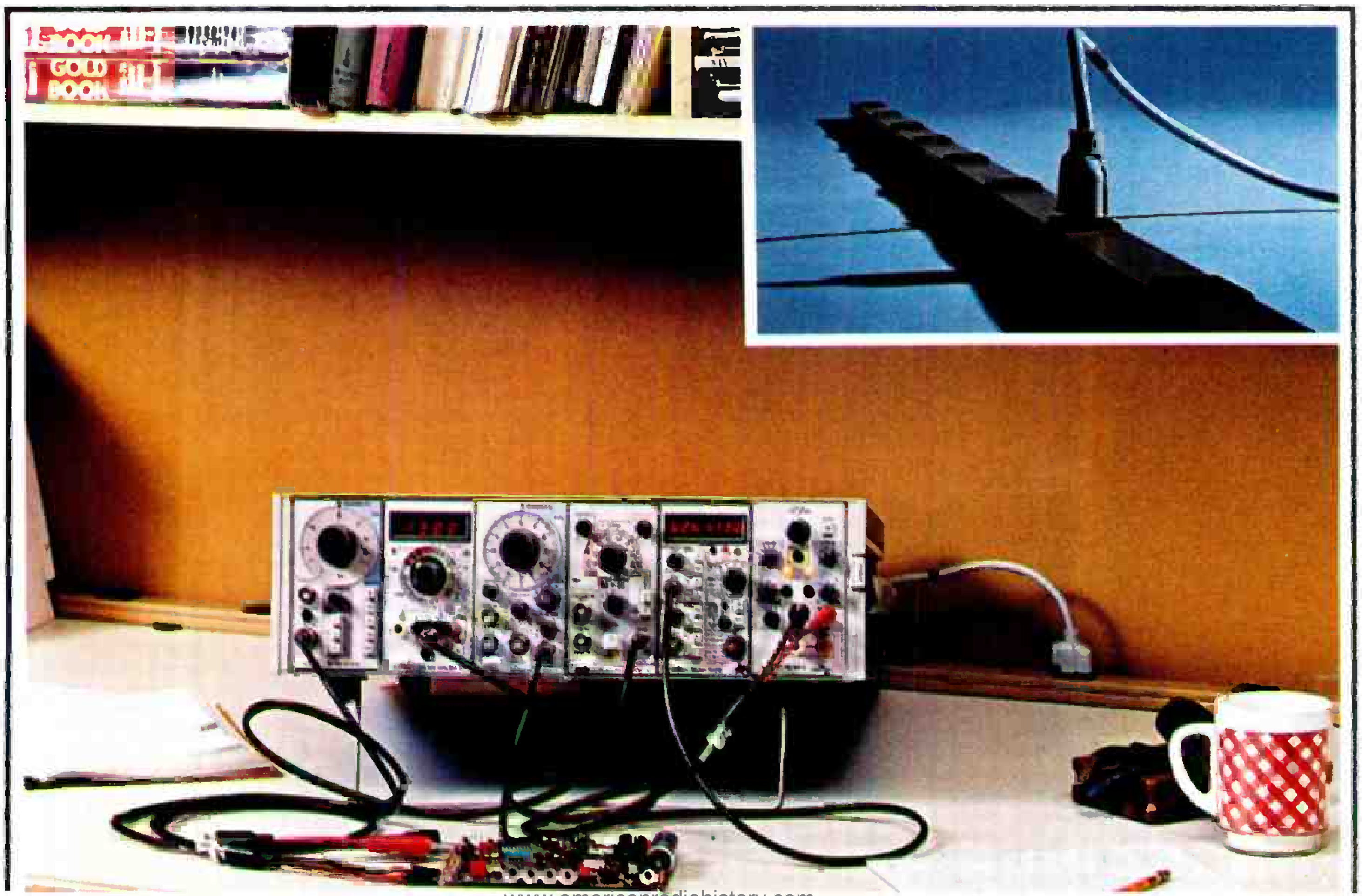
• This RIAA equalized amplifier is connected through auxiliary or spare high level inputs, to improve record playback performance. Although low in price, it delivers Class A operation, with low distortion claimed; ten day money-back trial performance is offered. Of solid state construction, the unit measures 9.5 x 5.2 x 2.5 in.
Mfr: PS Audio
Price: $59.95.
Circle 56 on Reader Service Card

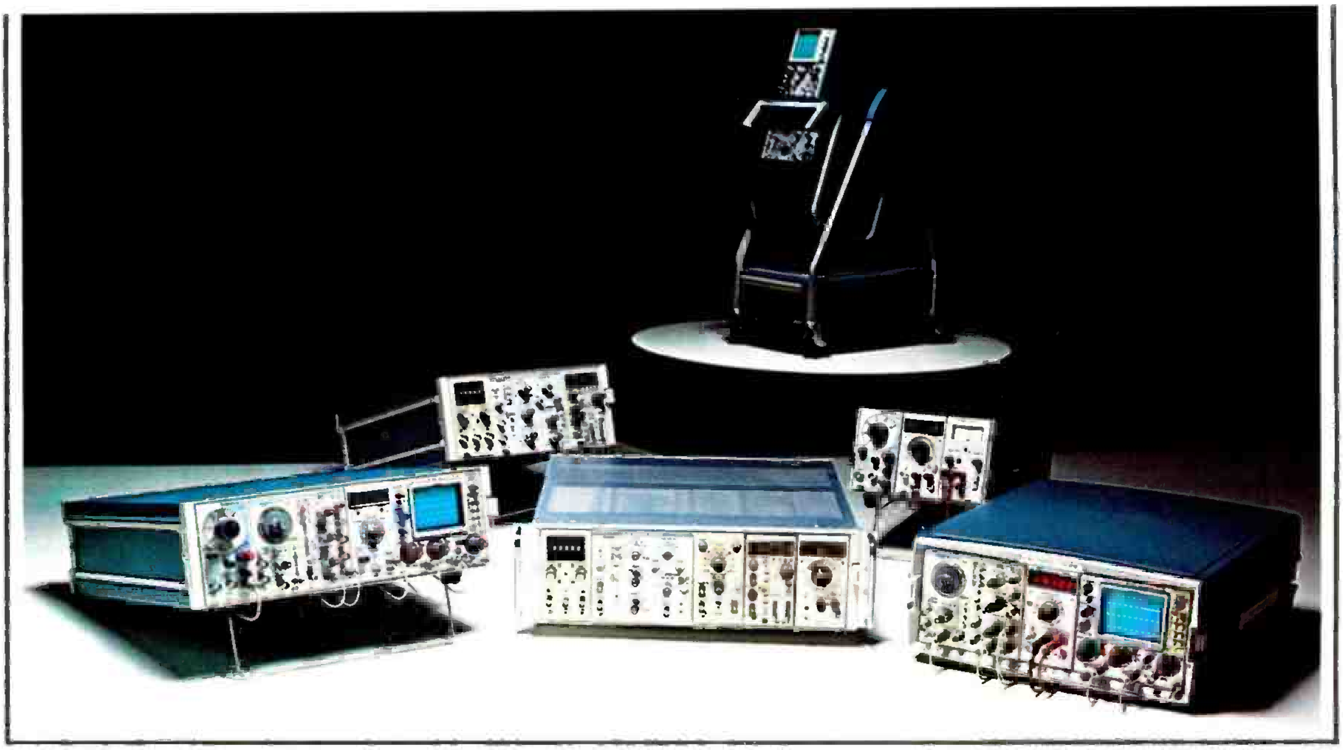


Here's an idea that could change your thinking about test equipment.

A complete test station doesn't have to be an assortment of special-function instruments. A working workbench doesn't have to be crowded and unhandy. And a truly portable test lab doesn't have to be out of reach.

TM 500 offers you an alternative: a modular line of compact, interchangeable plug-ins and mainframes. Multiconfigurable both in packaging and in performance, TEKTRONIX TM 500 is designed around the idea that test equipment which is compatible in every respect can, in fact, cover a broader range of functions and meet a wider variety of measurement situations. To say nothing of its ability to adapt more readily to new challenges.





Modular DMMs, counters, generators, amplifiers, power supplies, oscilloscopes, logic analyzers, and word recognizers can be interfaced electrically. Signals

can be routed from one plug-in board to another via the mainframe mother board, thus enabling you to build a test instrument that's more powerful than the sum of its parts.

The plug-ins can be configured together in six widths of mainframe and four types of packages, depending on where and how you'll use them. Select the plug-in modules you need just as you would monolithic instruments, and then combine them in the appropriate mainframe for a convenient, uncluttered bench setup; mount up to 6 functions

side-by-side in a 5¼-inch rack; build a rollabout station that "follows" you around the lab. Or

pack a portable test station in the small-as-a-suitcase TM 515 Traveler Mainframe, which carries up to 5 modules and typically weighs less than 35 pounds, including the modules.

The result is a total test system that looks like a unit . . . works as a unit . . . yet is still configurable to new or changing measurement requirements. So the next time you're looking into test instrumentation, specify the one product line that's designed for configurability.

For further information or a demonstration of TM 500 Instrumentation, write or phone: Tektronix, Inc., P.O. Box 500, Beaverton, Oregon 97077, (503) 644-0161 ext. 5283. In Europe: Tektronix Limited, P.O. Box 36, St. Peter Port, Guernsey, Channel Islands.

Tektronix
COMMITTED TO EXCELLENCE

TM 500...designed for configurability.

Circle 21 on Reader Service Card

FUNCTION GENERATOR

● External frequency control vco d.c. to 1 MHz. is offered on solid state Model 12 function generator. Front panel outputs produce 1 ttl compatible square wave, one triangle wave 4 Vpp. and a switch-selectable square, sine, or triangular wave with 0 to 20 Vpp amplitude. The selectable output has continuously adjustable d.c. offset and will drive 5 Vpp into 50 ohms. Frequency range is 1 Hz to 1 MHz with 200 PPM/°C frequency stability. Claimed sine wave distortion is less than 2 per cent from 1 Hz to 100 kHz. External frequency sweeping permits f.m. control of all waveforms. Also featured are frequency shift keying FSK and remote tone burst RTB.
Mfr: A.E. Corp.
Price: Kit: $79.95.
Wired: $124.95.
Circle 57 on Reader Service Card

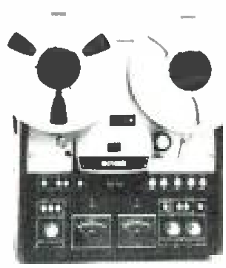
INDUSTRIAL VIDEOCASSETTE

● Sixty minutes of visual is delivered from a compact 7.5 oz. cassette, using an azimuth recording system which operates at the slow speed of 1.57 in/sec. on Betamax SLO-260 ½-in. videocassette recorder-player. The unit includes camera input, manual audio level control, audio dubbing, pause mode and rf capability to receive signals from a television monitor/receiver. The videocassette is backed up by the D-500 duplicator which has the mass production capability to produce the cassettes.
Mfr: Sony Corp.
Price: SLO-280: $1,445.00
Player only SLP-1100: $1,150.00.
Circle 58 on Reader Service Card

STEREO PHONO CARTRIDGES

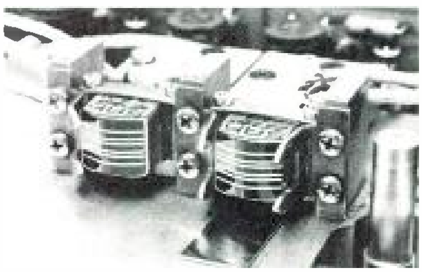
• The 0.2 x 0.7-mil elliptical stylus of the AT15XE cartridge is mounted directly to a tapered stylus cantilever to reduce tip mass, especially effective with high frequency requirements. Frequency response is from 5 Hz to 30,000 Hz, with a channel balance of 0.75 dB, channel separation 28 dB at 1 kHz and 23 dB at 10 kHz. Tracking force is from ¾ to one and ¾ grams.
Mfr: Audio-Technica Inc.
Circle 59 on Reader Service Card

TAPE DECK



• Four-head TT-1000 tape deck has three motors, full-logic solenoid-actuated tape motion, 10 in. reel capacity, mic and line inputs, vu meters, a digital counter and auto-reverse. The transport operates at 7½ or 3¾ in./sec.; frequency response is 30 Hz 20 kHz ± 3 dB.
Mfr: Bohsei Enterprise Co.
Circle 60 on Reader Service Card

REPLACEMENT HEADS



• A life expectancy of more than ten times that of metal heads is predicted by the manufacturer of these hot pressed ferrite heads with glass-bonded gaps. The head is plug-to-plug compatible with the original metal heads used in the manufacturer's cart machines.
Mfr: Saki Magnetics
Circle 61 on Reader Service Card

AMPLIFIER/EQUALIZER

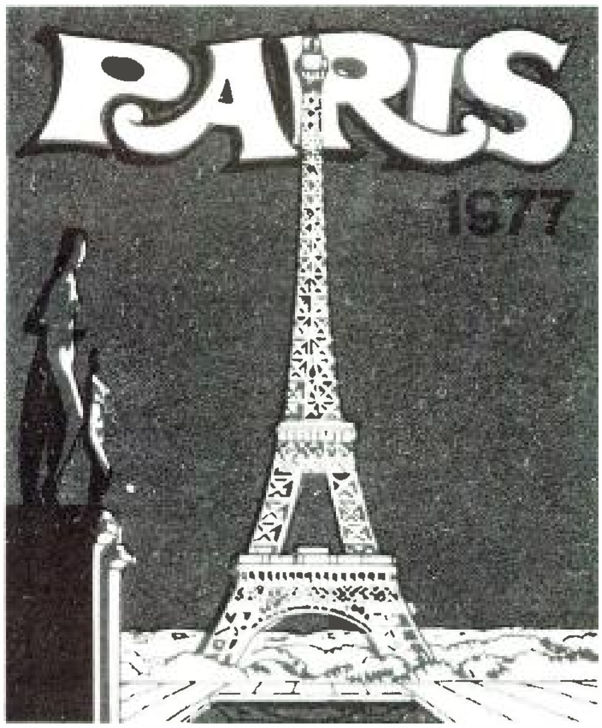
• Three units stack in a metal rack mount cabinet—Model 217R straight-line preamplifier, Model 210R graphic equalizer, and Model 202C, 100 watt per channel power amplifier. Model 217R preamp has front panel accessible phono stage cartridge loading, gain selector, and subsonic filter. A source selector handles two phonos, tuner, auxiliary and two tape circuits with by-directional bypass copying. Model 210R ten-band octave equalizer per channel has ± 15 dB boost or cut per octave, with synthesized inductors for the elimination of distortion and noise. Line or tape equalization includes a tape monitor and equalizer defeat. Model 202C power amplifier has modular construction and class AB design with ± 70V power supply for 200 watt per channel peaks at clipping on program material.
Mfr: Spectro Acoustics
Price: $1,100.
Circle 62 on Reader Service Card

CLIP-ON REMOTE INTERCOM

• A steel belt clip mounts lightweight (16 oz.) RS-100A intercom station to clothing. The unit contains a power amp that is current limiting and short-circuit-proof. Controls consist of a volume control, call light, mic off-on call switch, and a 4-pin XL connector for the headset. In addition, the unit features an adjustable side-tone function. Two XL connectors tie into the main station or other remotes. The compact unit measures 2¾ x 4⅞ x 1⅝ in.
Mfr: Clear-Com (Lumiere)
Circle 63 on Reader Service Card

JOHN WORAM

The 56th Convention of the Audio Engineering Society



Paris, France, 1-4 March, 1977

PARIS!!? There was a time when "April in Paris" was the dream of just about all of middle America. Many years ago, Lafayette came to the aid of Washington, and Americans never forgot their debt to France. But then Eisenhower came to the aid of De Gaulle, and the French have never forgiven us for it. In fact, the average Parisienne did little to conceal his animosity towards the average American, and in time the message filtered back; "Yankee, stay home!" As French satirist Pierre Deninos put it, "When France no longer has hereditary enemies, it chooses one from among its friends." After awhile, the $5.00 exit fee imposed at Paris's Orly Airport got to be the biggest bargain in all of France. After a day or two there, most Americans would have gladly paid ten times that amount, just to get out of the country.

Well, a few years ago, the pendulum may have begun swinging back towards normalcy. A few enlightened Parisiennes are beginning to consider that perhaps after all the universe does not rotate about the Eiffel Tower. Tourists—even Americans—are beginning to be tolerated and, *sacre bleu!*, a few locals will even make an attemp to understand languages other than French. (Formerly, those who were unable to converse in fluent French were treated as social lepers.)

And so, against this backdrop of slowly thawing coldness, the Audio Engineering Society ventured into Paris for its 56th convention, held at the Hotel Meridien. Perhaps not quite in the shadow of Eiffel, but at least a few minutes stroll to the Arc de Triumphe.

The show got off to an ominous start when the French authorities decided, on opening day no less, to close down all the small demonstration rooms scattered about six floors of the hotel. It seems that a tragic fire at another hotel in another country had finally alerted them to some sort of potential hazard that had not been considered previously! And so, in a flurry of packing crates, an extra exhibit area was hastily assembled on the hotel's mezzanine floor. And a special award to the folks at L. J. Scully for not destroying the hotel when they were told to pack up their just-assembled disc cutting lathe and move it to the new area. (By the way, L. J. Scully is now the US representative for Denmark's Ortofon Company, manufacturers of cutterheads, amplifiers and other tape-to-disc hardware.)

The show closed on a somewhat more whimsical note, at the awards banquet, held at another hotel. A post-banquet champagne party had been planned, to toast the awards winners. However, the banquet ran a trifle late, and so the hotel staff, in cooperation with a French honor guard, decided to toast themselves instead. When the honored guests eventually arrived, they found a very happy staff, and no bubbly! And now that I've done my bit for French-American relations, let's look at:

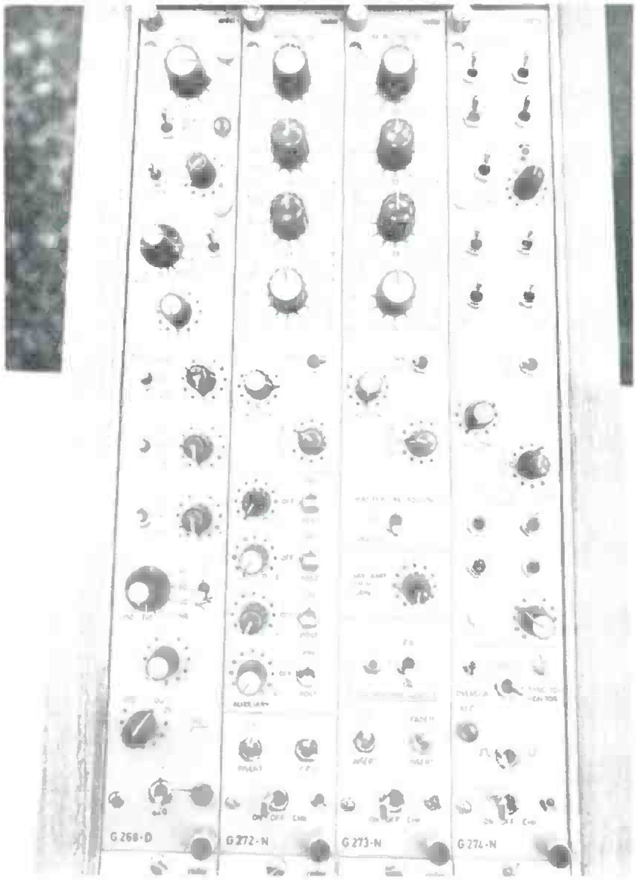
TAPE RECORDERS

Several tape recorder manufacturers were on hand with machines seen rarely—if ever—in this country. Leevers-Rich, a firm well-known in Great Britain, showed their pro line 1000 and 2000 series machines, and we may get a further glimpse of these at a future American convention.

Telefunken's Magnetophon 15A transport. Note the vari-speed and auto locator modules on the right.

Telefunken Magnetophon 15A drew a lot of interest, and was more often than not surrounded by a crowd of admiring button-pushers. The 15A comes in one- or two-inch formats, with up to 32 tracks of electronics available. Space below the electronics panel may be used to house up to 24 tracks-worth of Telefunken's new c4 noise reduction system. (The c4 was also shown in a 24 track rack mounting system.) To the right of the 15A's transport, space is provided for housing various auto-locator and vari-speed modules. The SZ15A module offers continuously variable speed control over a ±50 percent range. An led readout indicates the percentage of nominal tape speed, and an engraved chart on the module face plate lists the percentage required for various musical intervals. (For example, one half tone = 94 per cent or 106 per cent, etc.)

But the ultimate tape recorder toy must surely be Lyrec's Tape Position Controller, with 16 memory positions for storage and recall of tape positions. In addition to readouts of tape position and keyboard, there are A and B displays. With two different locations entered in these displays, you may instruct the machine to search for either one, or to play the section between A and B continuously. Either display may be loaded from the keyboard, or from the tape position readout. And once you've tired of this exercise, you can store all four displays in memory, move on to some other area, and return later on to the positions stored in the memory.

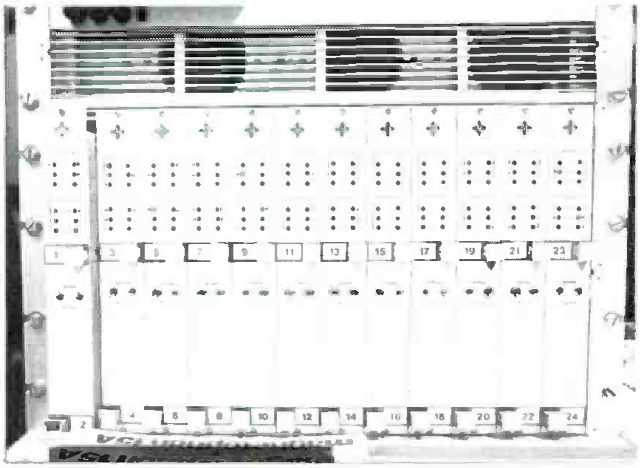


A close-up on some of Cadac's console modules.

Tweed Audio's 22 in/4 out console comes from Scotland.

CONSOLES

At Helios Electronics' booth, director Dick Swettenham presided over one of their super boards, to be delivered to Maison Rouge in London. An interesting feature is a noise reduction interface system built into the console's front panel. (Relax, it's not another non-compatible compander!) The idea is that you'll tuck your noise reduction cards (dbx, Dolby or whatever) into an out-of-the-way



The Telcom c4 Noise Reduction System from Telefunken.

The Helios Electronic console for Maison Rouge Studio, London. The noise reduction panel is to the left of the meters.

From Britain, Audio Development's line of battery operated consoles.

This is Calrec's line of condenser microphones. They are not yet found in the U.S.

The Proline 2000TC, from Leevers-Rich. Note the dual capstan drive, and the tape position indicator to the left.

card frame within the console, and yet retain finger-tip control over their functions. Presumably, the system could be adapted to accommodate several sets of cards, all of which may be powered from the console's own supply.

Audio Developments showed off their line of mini-, micro-, and pico-mixers, plus a "super system." These range from 6 in/2 out to 24 in/4 out. All systems are designed for battery operation, and appear to be sturdy enough to drop off a truck—at least a few times. The company has sold several systems here in the colonies, and is looking for American representation, as are many of the other British firms seen at the show.

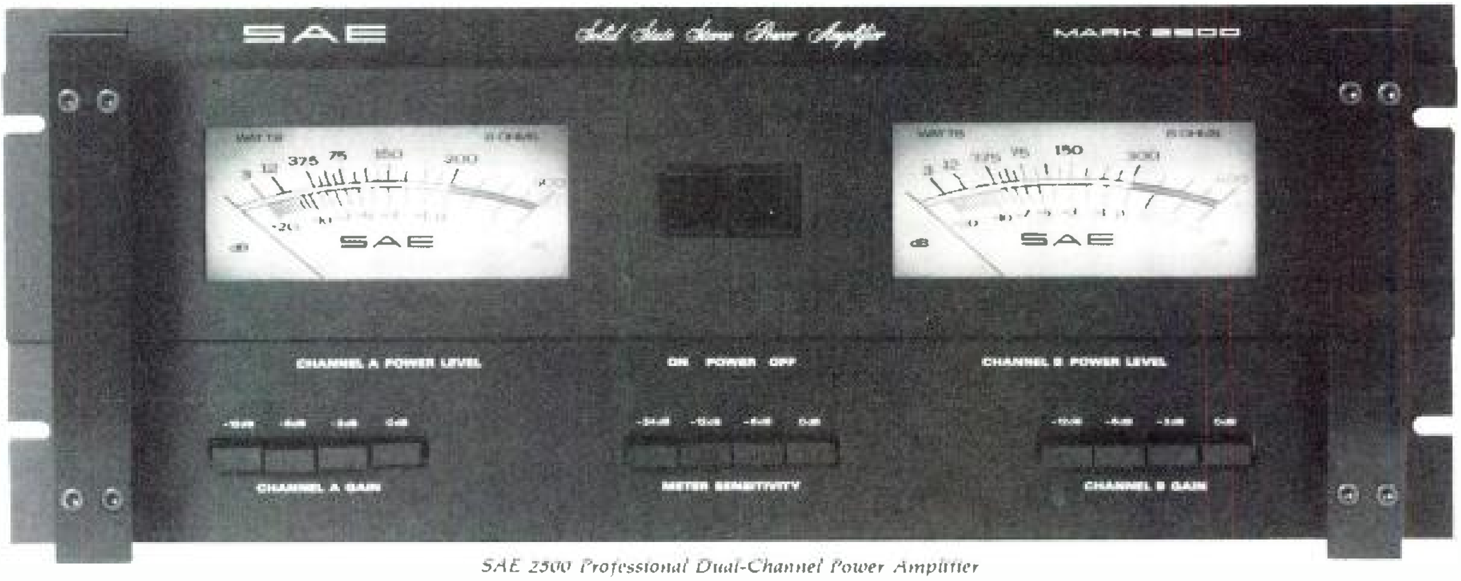
After a brief skirmish with French customs officials, Harrison Systems got their elegant 2824 console set up on the second day of the show. (At least I think it's an elegant console: I couldn't fight my way through the mob that flocked around it. Judging from the descriptive literature, it's got enough features to warrant a full-blown story. (Do you read me, Dave Harrison?)

ODDS 'N ENDS

JBL introduced their new 4301 two way speaker system, comprising an 8-inch woofer and a 2-inch tweeter, mounted in a 1½ cubic foot enclosure. The system is designed for the small control room, and carries a $159 price tag.

JVC was one of the victims of the last-minute closing of the demo rooms. Fortunately, they maintain a perma-

Powerful alternative.



SAE 2500 Professional Dual-Channel Power Amplifier

When you compare power amplifiers, you have to look at the hard facts. The SAE 2500 Professional Dual-Channel Power Amplifier has them—top power, specifications, reliability and features that make it the most "powerful alternative."

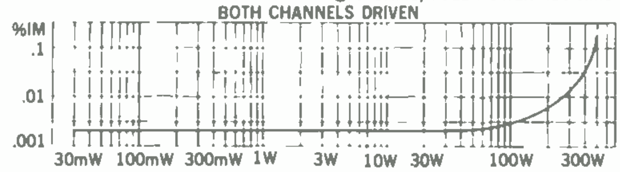
Power. 450 Watts RMS per channel, both channels driven into 4 Ohms from 20Hz to 20kHz at no more than 0.1% total harmonic distortion. Or, 300 Watts RMS per channel, both channels driven into 8 Ohms from 20Hz to 20kHz at no more than 0.05% total harmonic distortion.* Plus, a new, smaller wide-channel power transformer coupled to 4 computer-grade capacitors for a power supply that varies no more than 10% from no load to full load. (For extra protection, there are relay and thermal cut-out devices.)

Other Specifications:

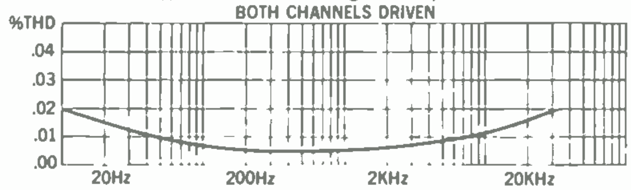
IM Distortion from 250mW to rated power at 4 or 8 Ohms with any 2 mixed frequencies between 20Hz and 20kHz at 4/1 voltage ratio..........0.05 Max.
Frequency Response at rated power................ ±0.25dB, 20Hz to 20kHz
Noise ....Greater than −100dB below rated power
Transient Response of any Square Wave ...... 2.5μsec. rise and fall time
Slew Rate ....40V/μsec.
Dimensions.......................... Front panel 19" Wx7"H, Chassis: 15¾"D (excluding handles, controls and connections)

*These specifications comply with FTC requirements for power amplifiers

2500 INTERMODULATION DISTORTION @ 8 OHMS, FULL POWER 49V RMS BOTH CHANNELS DRIVEN



2500 TOTAL HARMONIC DISTORTION @ 8 OHMS, FULL POWER 49V RMS BOTH CHANNELS DRIVEN



Reliability. The SAE 2500 gives you high current capability with Parallel-Series-Output Circuitry (PSO)—without loss of wide power bandwidth, low leakage current or super-high slew rate. Sixteen triple-diffused output transistors have an electrical and thermal SOA 50% higher than maximum design requirements for reliable high demand capability. This configuration can handle anything from continuous full signals to highly reactive surge loads—all day long without failure or overheating. Dual relay disconnect circuits and plug-in board design further assure reliable performance.

Features. Feedback level controls assure a constant input impedance of 50k Ohms and reduce the noise figure to more than 100dB below rated output in all positions. Loudspeaker protection relay-activated circuit automatically disconnects speakers in case of ±DC outputs. Plus, direct power reading VU meters and forced air cooling.

The SAE 2500 Professional Power Amplifier weighs only 58 lbs. making it practical for portable sound reinforcement, public address, communications and recording applications.

The professional alternative.

SAE

Scientific Audio Electronics, Inc.
P.O. Box 60271
Terminal Annex
Los Angeles, California 90060

Please send me the reasons (including available literature) why the SAE 2500 Professional Dual-Channel Power Amplifier is the "Powerful Alternative"

Name ______________________

Address ______________________

City ______________ State ______ Zip ______

Circle 37 on Reader Service Card

A 24 in/4 out console from ABE. Note the unusual position of the 24 x 7 patch bay just left of the center top.

nent display room just down the block from the hotel, and their exhibit was quickly set up there. Their Q.B. Phonic system—still under development—adds a startling touch of realism to quad (Q) and stereo (B, for binaural) speaker systems. The trouble with conventional speaker systems is that they are unable to convey that sense of being in the room with the musicians that one experiences when listening to a binaural recording over headphones. The Q.B. Phonic system attempts to create that effect, and although listener positioning remains critical, an illusion is created that sounds originate from locations much closer to the listener than the actual speaker placement would indicate. Thus, in one demonstration, the narrator seems to walk right up to you and whispers in your ear. It's a dramatic effect, which must be heard to be believed. Even then, it's unbelievable!

On the other hand, would you rather have your headphones sound like loudspeakers? Eugen Beyer introduced their LSE 1, a loudspeaker-simulating earphone system. As most listeners know, a conventional stereo program sounds a lot different over headphones than over speakers. With headphones, left and right information is accentuated, while center-placed soloists seem to be located somewhere within the listener's head. Although this may be acceptable (interesting, even) for casual music listening, it rules out the practicality of headphone monitoring during recording. On the other hand, the nature of some on-location recording work rules out the use of speakers.

How's this for an auto locator? Lyrec's Tape Position Controller, with a 24 track machine in the background.

Beyer explains that with the LSE system, "... signals are changed by a very complicated electronic set-up in a way that sound waves, originating from the headphone system and addressed to the ear drum, are identical with these sound waves which the ears would have reached when listening to the same recording through speakers."

Does your studio need a *goniometer?* Finland's Kajaani Oy manufactures one suitable for mounting in your console. On the off-chance that you may be uncertain just what a goniometer is, it's simply a small cathode ray oscilloscope, set up for X-Y monitoring of phase relationships between any two audio signals. ■

STEPHAN PEUS

Microphones and Transients

THE TRANSMISSION CHARACTERISTICS of microphones during transient investigations are readable directly from the oscilloscope screen or 'scope photo without the necessity of complicated devices for transformation into the frequency domain. Rise and decay time behavior of speech and music is to be found in the 10-100 ms area, while the decay phenomenon in high quality microphones may end in less than 1 ms. That is how such microphones manage to grasp the fine structure of sound. Many microphones constructed solely for linear frequency response add their rise time behavior to the sound being picked up, which then appears as amplitude modulation (a.m.) and gives the microphone its typical "sound." For this reason, the following investigation of the transient behavior of microphones should be taken very seriously, even if this is only one small part of the development of a new microphone.

IMPULSE MEASUREMENTS AND ELECTROACOUSTICS

The transient behavior of electro-acoustic transmission systems has been the subject of discussions for some time. It is at times measured in amplifiers using square waves and is sometimes even published. Publications about transsient measurements in microphones are largely limited to the recognition that the transient response, i.e. the result of tests in the time domain after transformation into the frequency domain, contains the transmission functions: transmission factor and transmission angle. The measurement is a very rapid one and one doesn't need an anechoic chamber since the influences of the test room only reach the microphone outside the measurement time. This advantage must be weighed against the expense of a computer which puts out the transformations in graphic form in real time directly from the electrical microphone signal.

Whatever measurement system is used, our purpose is to find out why microphones with nearly identical frequency response curves can have such vastly differing sound quality.

TRANSIENT BEHAVIOR

The frequency response in microphones is measured in their stationary, or static, condition at moderate velocities, while a microphone's "daily bread" is the reproduction of predominantly impulse-type sound occurrences. The first two illustrations show the order of magnitude involved. FIGURE 1 shows the transient behavior of the individual harmonics which make up the syllable "ke." Characteristic for this guttural sound is the impulse rise time pattern and the relatively long decay time of about 120 ms. It is more typical for speech sounds to have a shorter build-up time which is also more gradual, rather than impulse rich. Singly pronounced vowels have the shortest rise time: e.g. "e" only 6 ms. What I am referring to is the time during which there is a significant change in the amplitudes of the harmonics. One can speak of a fully built-up state only for singly pronounced speech sounds intended for research purposes, but never for continuous speech.

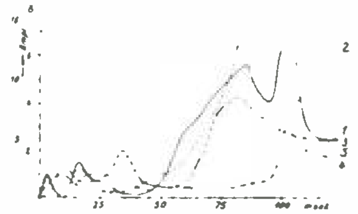


Figure 1. Development of the harmonics of the syllable ke.

Characteristic of musical instruments is not only their longer rise time, but also the shape of this phenomenon. FIGURE 2 shows an example of an especially short transient rise time of a trumpet, again dissected into its harmonics.

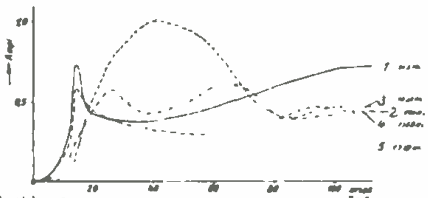


Figure 2. Development of the harmonics of a B-flat trumpet with a fundamental frequency of 340 Hz.

Much shorter times are encountered when the instrument is played since only the change from one *quasi stable* condition to the next is involved.

These two examples with particularly short rise times were selected because in publications, in which the transient behavior study of electro-acoustic devices is recommended, the impression has been given that the impulse content during the rise time period is so short, that a transducer—in this case a microphone—is too slow to be able to transmit the fine impulse structure. This paper is

Stephan Peus is a German engineer who works for George Neumann GmbH in Berlin. This article appeared in the German publication, radio mentor, *based on a paper given at the Tonmeister Convention in Cologne, Germany in November 1975. It was translated by Stephen Temmer, president of Gotham Audio in New York City.*

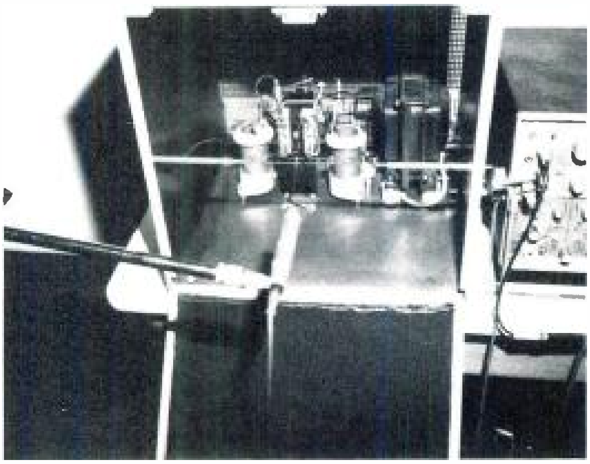


Figure 3. Construction of the spark discharge apparatus.

intended to show that things are actually somewhat different.

THE TEST PULSE

For the measurement of the transient behavior of microphones, one needs a very short, reproduceable signal such as that produced by a pistol shot or the spark discharge of a condenser.

Some years ago, Dr. G. Boré of the Georg Neumann Co. constructed a spark-discharge apparatus in which the power line voltage was raised to 5 kV and after rectification, was used to charge a 600 pF capacitor. A spark gap parallel to this capacitor was adjustable in width to control the energy of discharge, while an r-c network set the repetition rate. The entire apparatus was enclosed in a shielding metal box. To protect the electrodes and to filter out those frequencies far above the threshold of hearing, the cover was lined with felt (shown hinged down in FIGURE 3).

It is not acoustically possible to produce a one-sided pulse such as the square wave used in electrical measurements, since the air together with the discharged capacitor forms a resonant circuit through which the greatly heated and compressed air radiates as a spherical wave, first for a short time at super sonic speed, and then with the well-known differential equations of a sound field. The high degree of attenuation of this resonant circuit resulting from the friction resistance of air only permits one period close to the aperiodic boundary.

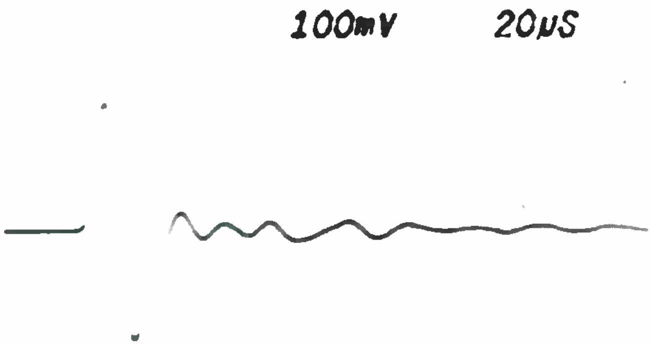


Figure 4. Sound pressure diagram of a spark discharge recorded using a measurement microphone. The time axis is divided into 20μs sections.

The oscilloscope photo in FIGURE 4, made using a very high quality measurement microphone, shows the sound pressure pattern of such a discharge. The slender positive portion, followed by the somewhat flatter and wider negative one resulting from the friction resistance, is typical for such a discharge.

TRANSIENT RESPONSE OF THE MICROPHONE

Every microphone is a transmission system of limited band width. Put in simple terms, it is a low pass filter with an upper boundary frequency and a substantially flat response up to that frequency. This boundary frequency causes a widening of the positive part of the pulse; i.e. the positive part of the transient response. The mean duration and therefore the median pulse width as shown in the illustration, is inversely proportional to twice the boundary frequency.

$$t = \frac{1}{2f_{boundary}}$$

The transition from the pass band to the cut-off area is visible in the response figure as an eigen-oscillation at the boundary frequency. Duration and amplitude of this oscillation are interrelated as follows: a steep cut-off at the upper end of the pass-band results in a short but high damped decay wave, while a flat cut-off produces a longer damped wave with smaller amplitude.

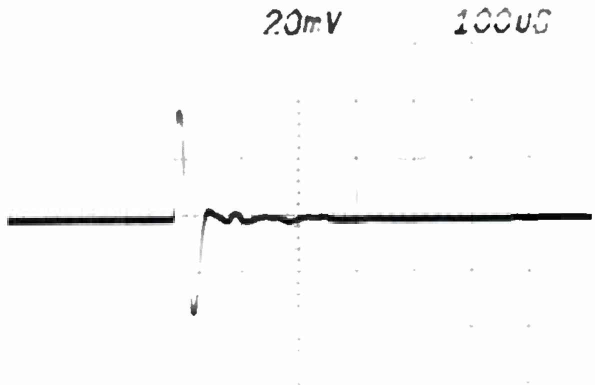


Figure 5. Transient response of a studio quality microphone.

It is, therefore, not to the point to place the boundary frequency as high as possible for the application at hand, but rather to make the transition if it falls within the audio range, such that the relationship of oscillation amplitude and duration of the damped decay wave is optimum for the intended use.

This was visible in the transient response of the measurement microphone where the boundary frequency of 60 kHz is noticeable both in the particular pulse width of about 8 μs and the damped decay wave at this frequency.

CONDENSER MICROPHONES

FIGURE 5 shows the transient response of a studio quality microphone which reproduces the pulse correctly within its physical limitations. The boundary frequency of over 20 kHz shows itself again in the corresponding width of the pulse, but almost not at all in the eigen-oscillation at that frequency, due to the proper choice of slope in the transition range. This microphone produces no sound coloration whatsoever.

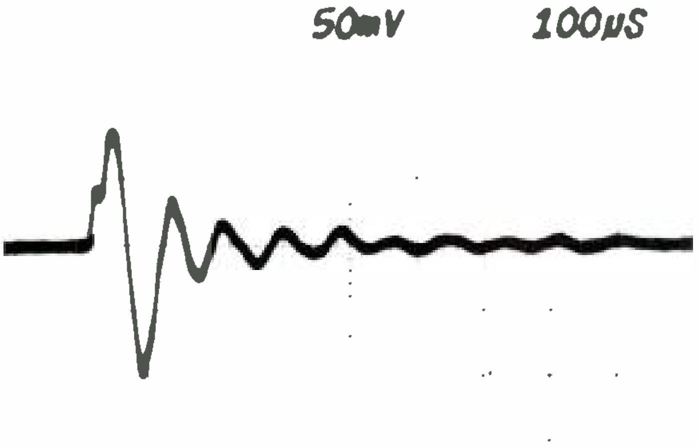


Figure 6. Frequency response of a cardioid condenser microphone with a 16 mm membrane diameter.

By contrast there are microphones which have built into them oscillation-prone components, such as small chambers accessible to the sound to boost a certain frequency for the sake of increased brilliance or to expand a frequency response of insufficient width.

FIGURE 6, for example, shows a response curve with a boost of only 2 dB at about 8 kHz and an early cut-off. In FIGURE 7, we see the transient response showing these properties in the width of the pulse and the resonance of the built-in chamber at about 8 kHz.

This type of microphone, therefore, boosts frequencies in the 8 kHz range by some 2 dB and additionally may excite this oscillation-prone system at its resonance. This leads to an intensification of the effect. These response-boosting means must be used with great care, since acoustic resonators are very narrow-band in nature and are readily recognized by experienced listeners.

Figure 7. Transient response diagram for the frequency response in Fig. 6.

FIGURE 8 shows a condenser microphone which has been constructed solely to produce a "pretty" response curve and proves that the choice of a condenser microphone transducer with good frequency response does not necessarily guaranty a good sounding microphone.

At this point it is important to point out one of the properties of the human ear as a receiver—the so-called *psychological rise time*. Investigations have shown that two signals which have differing transient response characteristics will only then be judged by the ear as having different sound quality if their rise time is longer than 250 µs.

This condition was not met by the microphone with the resonant circuit (FIGURE 7), so that single pulses recorded with it will likely not ring at the 8 kHz frequency. However, it should be noted that this transducer will be constantly excited by the impulse character of speech and music, which will result in a quasi-stationary oscillation at this frequency, undoubtedly leading to coloration of the resulting sound. This is reputed to be desirable at times.

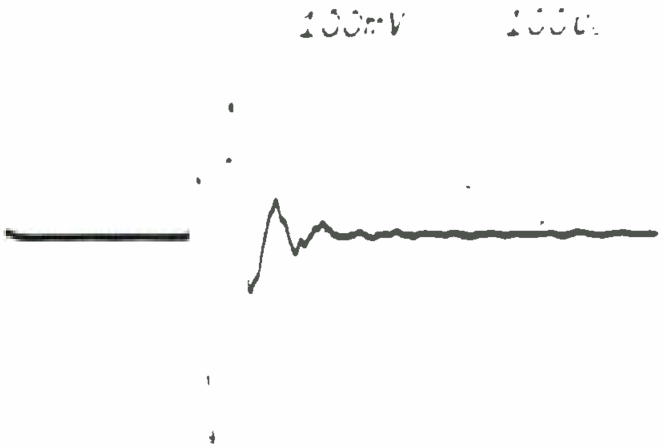


Figure 8. Transient response of a condenser microphone designed for a "beautiful" frequency response.

Even if no resonant systems with a frequency in the audible range are built into a microphone, the membrane with its mass and the stiffness of its suspension forms an oscillation-prone system in which the not-to-be-neglected vibrating air mass must be added to that of the membrane.

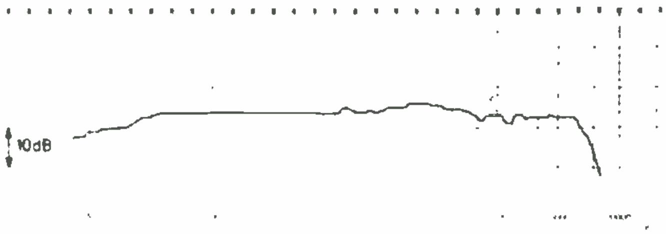


Figure 9. Frequency response of a high quality dynamic studio microphone.

DYNAMIC MICROPHONES

Both gradient condenser microphones and dynamic microphones generally have a membrane self-resonance within the audible range in common. This resonance, depending on the type of microphone, is damped either in a frequency independent (friction controlled), or frequency dependent (mass controlled) way. Due to their transducer principle, dynamic microphones usually have larger membranes with correspondingly greater mass, and must therefore have more compliant suspension to achieve the resonant frequency. As a result, such systems have a much longer decay than microphones with smaller effective membrane mass, and with it less compliant suspensions, such

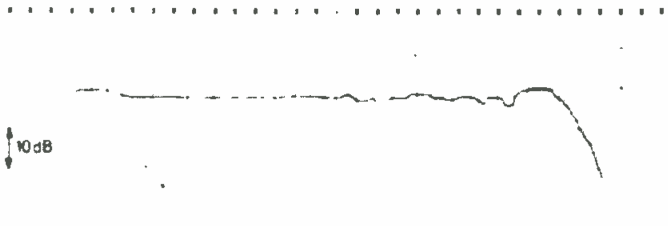


Figure 10. Transient response diagram for the frequency response in Fig. 9.

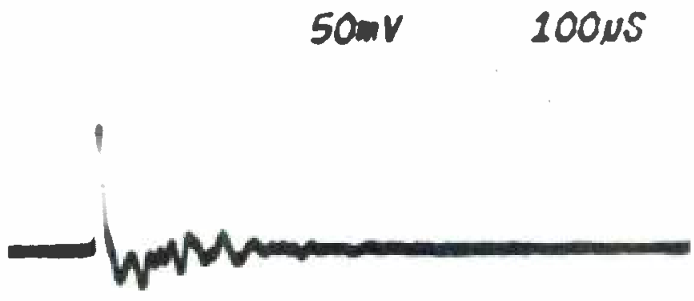


Figure 11. Transient response of a double ribbon microphone.

as condenser or ribbon microphones. Two illustrations show the effect of this on the transient response of these microphone types: FIGURE 9 shows the frequency response of a dynamic studio quality microphone, which is linear throughout the pass band but shows the typical steep cut-off at 13 kHz.

In the corresponding transient diagram, FIGURE 10, this boundary frequency is readily recognizable in the pulse width and the slow decay of the damped oscillation. There is a 20 μs pulse in the leading flank which cannot be explained here but which will not be of interest due to its very short duration.

RIBBON MICROPHONE

FIGURE 11, our last example, shows the transient response of a double ribbon microphone. The small mass is reflected both in the pulse width and the eigen-oscillation at a very high frequency. One can recognize 25 kHz and 50 kHz, the latter being confirmed as the boundary frequency from the pulse width. FIGURE 10, on the

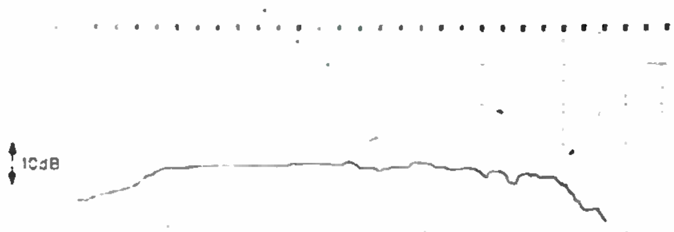


Figure 12. Frequency response of the double ribbon microphone.

other hand, clearly shows that the response ends at about 10 kHz, so that one must assume that only a portion of the double ribbon is modulated at this high velocity and actually misrepresents such a high boundary frequency.

There are several transmission characteristics, therefore, which may be read directly off the oscilloscope screen without the necessity of transforming into the frequency domain. The fine structure of the signal is visible and recognizable, and one can see how many a microphone built solely for good frequency response, gets its characteristic "sound." ■

MICHAEL RETTINGER

Low Frequency Sound Reproduction

Picking up elusive low frequency tones requires an understanding of loudspeaker cone diameter and cone travel.

It is sometimes claimed, in control rooms, in recording studios, at outdoor concerts, and other areas employed for the faithful reproduction of music, that "the real lows" are missing in the program. Specifically, tones below 50 hertz are barely audible, it is claimed, and tones of 30 hertz are non-existent. On careful investigation it is generally learned that one or the other requirement for extreme low-frequency reproduction on the part of the loudspeaker is not met: (1) the resonance frequencies of the woofer and cabinet are considerably above the lowest frequency that is intended to be generated by the unit, and (2) the cone diameter is too small for the necessary excursion.

We are, here, not concerned with item 1, assuming that the loudspeaker manufacturer has done his part in providing a radiator capable of generating the extreme low frequencies advertised in his brochure and stated in his specifications. Item 2, however, is another matter, and one not usually considered in greater detail. It is the purpose of the following to describe the cone travel and cone diameter of a loudspeaker intended to radiate an appreciable amount of low-frequency power.

To come to grips with the problem, consider Figure 1, the so-called ISO R-226 set of equal loudness contours. For a 30-hertz tone to sound as loud as a 1000-hertz tone which has, say, a sound-pressure level SPL of 90 dB relative to 0.00002 newtons/m², which represents a fairly loud signal, the 30-hertz tone has to have an SPL of 110 dB, as shown by a dot on Figure 1.

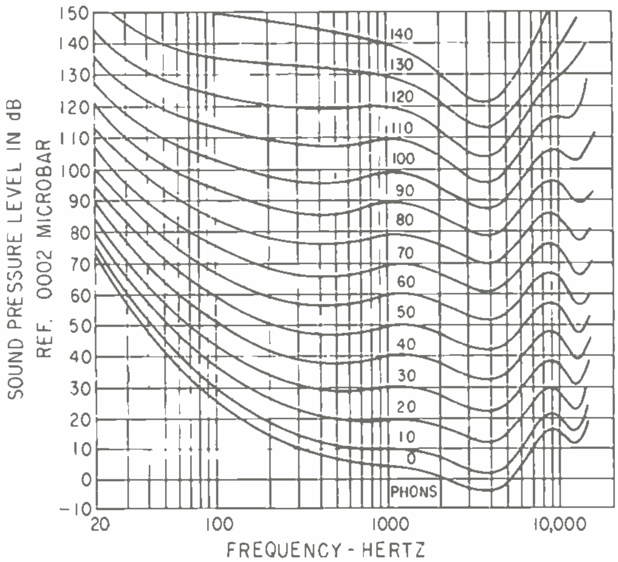


Fig. 1. Equal loudness contours (ISO R-226).

REQUIRED ACOUSTIC POWER

Next, assume that the 30-hertz tone is to exhibit this 110 dB level in the open at a distance R of 32.8 ft. (10 meters), from the loudspeaker placed in a large flat baffle or in a large cabinet to prevent radiations from the rear of the cone to interfere objectionably with those departing from the front. What is the acoustic power required to achieve this level when the woofer radiates its output into a hemisphere, so that its directivity factor Q comes to 2? The required acoustic power (not the electric power, which may be 20 times as large if the efficiency of the loudspeaker is 5 per cent), or power level WL referred to 10^{-12} watts is given by

$$\begin{aligned} WL &= SPL + 20 \log R - 10 \log Q + 0.7 \\ &= 110 + 20 \log 32.8 - 10 \log 2 + 0.7 \\ &= 138 \text{ dBp} \end{aligned}$$

where the "p" of dBp refers to the picowatts reference level. Such a power level corresponds to a power of 63 acoustic watts.

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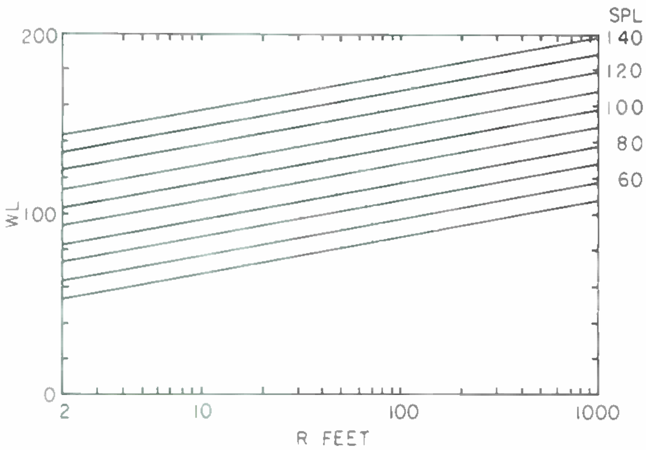


Fig. 2. Acoustic power level relative to 1 picowatt (10^{-12} watts) of a hemispherically radiating sound source as a function of its (outdoor) distance R in feet for various sound-pressure levels, SPLs, relative to 0.0002 microbars or 0.00002 newtons per square meter.

The next equation to be considered concerns the relationship between a vibrating baffle-mounted circular piston of diameter D inches, its displacement amplitude d in inches from its rest position at a frequency f, and the resulting acoustic power w. It is

$$d = \frac{117000(W)^{1/2}}{f^2D^2} \qquad (1)$$

For f = 30 hertz, D = 16″, W = 63 watts

$$d = \frac{117000(63)^{1/2}}{30^2 16^2}$$

$$= 4 \text{ inches}$$

It is obvious that no 16 in. diameter loudspeaker cone of standard construction can travel 4 in. from rest posi-

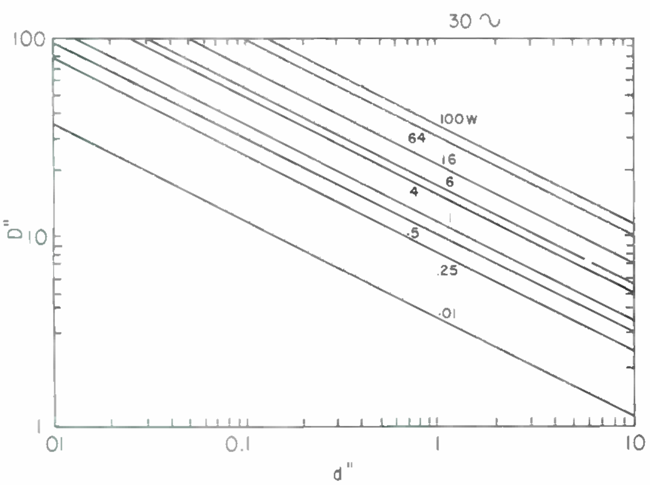


Fig. 3. Diameter D of a baffled circular piston in inches as a function of its excursion d in inches from its rest position at 30 hertz for various required acoustic powers in watts.

tion. Assuming that it may move as far as 0.25 in., it takes 16 units to achieve the required volume displacement, that is, displacement times cone area, or

$$V = dA$$
$$= 4 \times 8^2 \pi$$
$$= 16 \times 0.25 \times 8^2 \pi$$

FIGURE 2 shows the power level WL of a hemispherically radiating sound source as a function of its outdoor distance R in feet for various SPLs. Thus for an SPL of 110 dB at 32.8 ft., the WL is 138 dBp, marked by a dot on the figure.

FIGURE 3 shows the diameter of a baffled piston D in inches as a function of its excursion d in inches from its rest position at 30 hertz for various required acoustic powers. Thus, for W = 63 watts, D = 16 inches, d equals 4 inches, also marked by a dot.

When the loudspeaker is indoors, the cone excursion requirements are not quite so severe as they are outdoors because the reflected sound provides some sound reinforcement, particularly in the reverberant field of the enclosure. Assume a room 25 x 60 x 90 ft. with a volume V

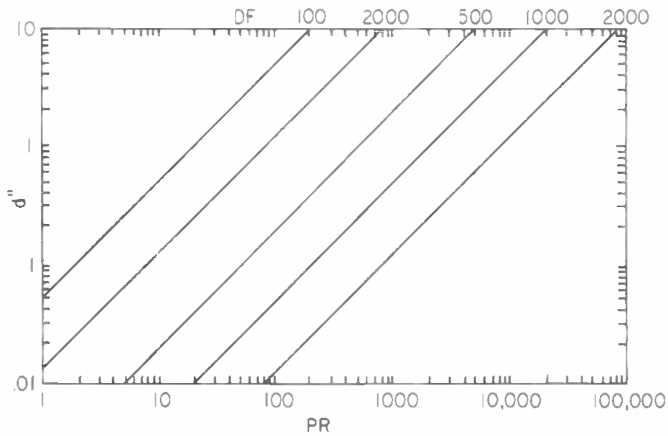


Fig. 4. The excursion d in inches of a circular baffled piston from its rest position as a function of the product of sound pressure P in microbars and the source distance R in feet, for various products of piston diameter D in inches and the frequency f in hertz.

of 135,000 cu.ft., a total interior surface S of 18,300 sq.ft., a reverberation time of 1 second at 30 hertz, and a total absorption A = 0.05 V/T = 0.05 x 135,000/1 = 6750 sabins at 30 hertz. For this condition we have

$$WL = SPL - 10\log\left(\frac{Q}{4\pi R^2} + \frac{4}{A}\right) - 10.3$$

$$= 110 - 10\log\left(\frac{2}{12.6 \times 32^2} + \frac{4}{6750}\right) - 10.3$$

$$= 131.47 \text{ dBp}$$

This power level is equivalent to 14 watts. For such an acoustic power, the 16 in. diameter cone displacement d comes to 2 inches (see FIGURE 3) which is still too large for an ordinary loudspeaker diaphragm. Allowing, as before, a cone excursion of 0.25 in., at 30 hertz, it requires 8 loudspeakers to achieve the desired acoustic output at 30 hertz.

It is seen, therefore, that loud low-frequency tone reproduction generally takes a battery of loudspeakers. Many recording studios are simply not equipped for the purpose. Equalizing the amplifier for increased low-frequency radiation on part of the loudspeaker is inadvisable, since it leads either to high distortion, cone break-up, or voice-coil burn-out.

SIXTY INSTEAD OF THIRTY HERTZ

If the mixer is satisfied with the reproduction of low frequencies above 60 hertz, the situation is much less critical. From equation (1) we see that when 60 hertz instead of 30 hertz is agreed upon as the lowest note to be reproduced with adequate loudness, the cone excursion needs, for the same SPL, to be only a quarter that necessary for 30 hertz. Hence only two, instead of eight, emitters would be necessary for the room with its 135,000 cubic foot volume discussed above.

Refuge may also be taken in the use of horns to achieve better low-frequency reproduction in interior rooms as well as outdoors. The "problem" here consists in the ungainly large acoustic "transformers" necessary to accomplish this. Such large horns are generally not available on the open market, and even if they were, their lengths (not infrequently 10 feet or more depending on the throat diameter of the horn) often look out of place in an auditorium.

PEAK AMPLITUDE

FIGURE 4, an all-inclusive chart, shows the peak amplitude d in inches of a baffled circular piston as a function of the product of the pressure P in microbars desired at a source distance R in feet, for various products of piston diameter D in inches and the frequency f in hertz. As an example, the excursion of a hemispherically radiating 16-inch diameter piston at 30 hertz ($Df = 16 \times 30 = 480$) at a distance of 32.8 ft. for a sound pressure of 63 microbars ($PR = 63 \times 32.8 = 2066$) comes to 4 inches, marked by a dot on the figure.

To convert sound pressure levels to sound pressure in either microbars (dynes/sq.cm.) or newtons per square meter, employ the chart below. It should be noted that the pressure levels referred to as 0.0002 microbars (with the pressure in microbars) are identical to those referred to as 0.00002 newtons/sq.m. or pascals (with the pressure also in pascals).

SPL	P (microbars)	P(pascals)
0	0.0002	0.00002
10	0.000631	0.0000631
20	0.002	0.0002
30	0.00631	0.000631
40	0.02	0.002
50	0.0631	0.00631
60	0.2	0.02
70	0.631	0.0631
80	2.0	0.2
90	6.31	0.631
100	20.0	2.0
110	63.1	6.31
120	200.0	20.0
130	631.0	63.1
140	2000.0	200.0
150	6310.0	631.0
160	20000.0	200.0

Note: In the MKS system, the reference pressure of 0.00002 newtons/sq.m. is often stated as 20 micronewtons sq.m., or 20 micropascals. ■

JOHN MAXWELL

Noise of Sources

A true appraisal of a preamplifier's signal-to-noise characteristic is derived from calculations made during actual operation.

THE ELIMINATION or minimization of noise is a perplexing problem for audio engineers. Many preamplifiers and components come with outstanding noise specifications, only to disappoint the user. The difference betwen specification and application arises because the amplifiers are specified under ideal, not real, conditions, i.e. a transducer connected to the input. Often the transducer noise is as large or even greater than the amplifier noise, degrading the signal-to-noise ratio. Before amplifier or component noise can be considered, familiarity with the source noise is essential.

REVIEW OF NOISE BASICS

There are three type of transducers: Resistive, capacitive, and inductive. The noise of a passive network is thermal noise, generated by the real part of the complex impedance, as given by Nyquist's equation:

$V_n^2 = 4kTRe(Z)\,\Delta f$

V_n^2 = Mean square noise voltage (V^2)
k = Boltzmann's constant (1.38×10^{-23} VAS/°k)
T = Absolute temperature (°k)
Re(Z) = Real part of complex impedance (ohms)
Δf = Noise bandwidth (Hz)

The noise may be represented as a spectral density (V^2/Hz) or more commonly in $\mu V/\sqrt{Hz}$ or $nV/\sqrt{Hz}$ and is given by:

$e_n^2 = V_n^2/\Delta f$

The total noise voltage in a frequency band can be readily calculated if it is white noise [i.e. Re(Z) is frequency independent]. This is not the case for capacitive or inductive sources, or most ordinary noise.

Rapidly changing network impedance and amplifier gain equalization combine to complicate the issue. The total source noise in a non-ideal case can be calculated by breaking the noise spectrum into several small bands where the noise |Re(Z)| is nearly white and calculating the noise of each band. The total source noise is the rms sum of the noise in each of the bands N_1—N_n.

$V_{noise} = (V_{N_1}^2 + V_{N_2}^2 + — + V_n^2)^{1/2}$

John Maxwell is a senior engineer at National Semiconductor in Santa Clara, Ca.

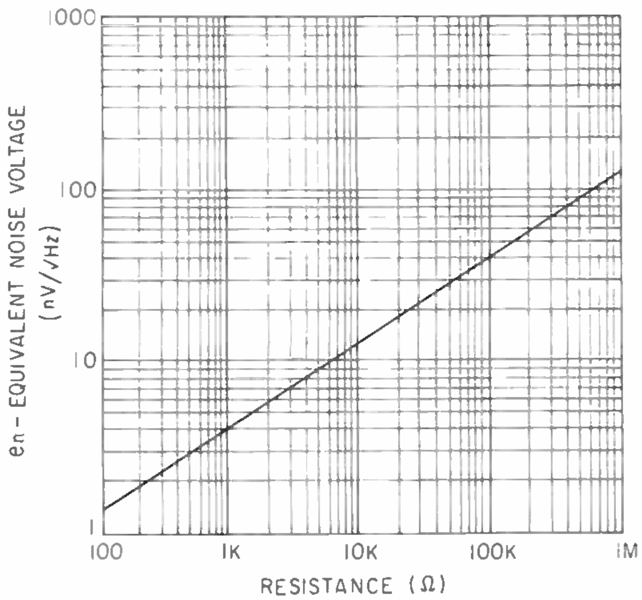


Figure 1. Thermal noise voltage versus resistance.

The expression does not take amplifier gain equalization (like RIAA) into account, which will change the character of the noise at the amplifier output. By reflecting the gain equalization to the amplifier input and normalizing the gain, to 0 dB at 1 kHz, the equalized source noise may then be calculated.

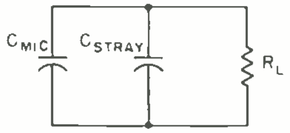
$V_{EQ} = ([A_1]^2 V_{N_1}^2 + [A_2]^2 V_{N_2}^2 + — + |A_n|^2 V_{N_n}^2)^{1/2}$

Where V_{EQ} = equalized source noise (μV) and
$|A_n|$ = magnitude of the equalized gain at the center of each noise band (V/V).

SOURCE NOISE

Models are needed for capacitive and inductive systems so that noise calculations can be made. Namely, the real part of the impedance needs to be determined.

A lumped model of a capacitive source, such as a condenser or electret microphone, consists of the microphone and stray capacitance shunted by a load resistance.



$$Z = Re(Z) + jIM(Z)$$

$$Re(Z) = \frac{R}{1 + \omega^2R^2C^2}$$

$$|Z| = \left(\frac{R^2}{1 + \omega^2R^2C^2}\right)^{1/2}$$

Figure 2. Lumped model of a capacitive microphone.

It should be noted that for any particular microphone, the noise of the network $([C_m + C_s]//R_L)$ is reduced by increasing R_L because Re(Z) (the real part of the impedance) is inversely proportional to R_L (See equation 5).

The inductive source (phono cartridges and tapeheads) is more complex to analyze because it has a much more complex model. The simplified lumped model of a phono cartridge or tape head consists of a series inductance and resistance shunted by a small capacitor. Each phono cartridge or tape head has a recommended load consisting of a specified shunt resistance and capacitance. A model for the inductive source and preamp input network is shown in FIGURE 3.

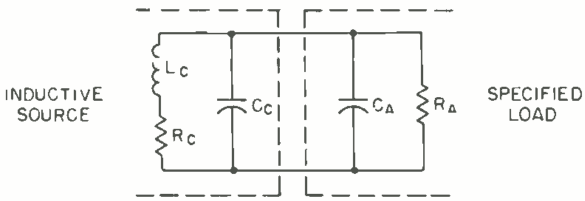
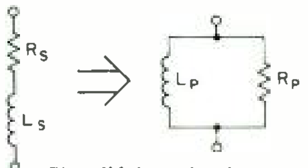


Figure 3. Phono cartridge or tape head and preamp input network.

This circuit is quite formidable to analyze and needs further simplification. Through the use of Q equations, a series L-R is transformed to a parallel L-R.

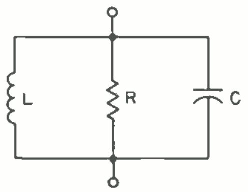


$$Q = \frac{\omega Ls}{Rs}$$

$$Rp = Rs\,(1 + Q^2) \qquad (6)$$

$$Lp = Ls\left(\frac{1 + Q^2}{Q^2}\right)$$

Simplifying the input network to:



$$R = R_A//Rp$$

$$L = Lp$$

$$C = C_s + C_c$$

Figure 4. Simplified source network.

$$Re(Z) = \frac{RX_L^2\,X_C^2}{(RX_L - RX_C)^2 + X_L^2\,X_C^2}$$

$$Z = \frac{RX_L\,X_C}{[(RX_L - RX_C)^2 + X_L^2\,X_C^2]^{1/2}} \qquad (7)$$

$$X_L = \omega L$$

$$X_C = 1/\omega C$$

The tools are now available to calculate the noise of a variety of transducers and see how this unspecified noise affects amplifier (s/n) performance.

Table 1. A summary of electret microphone calculations.

f Range (Hz)	25 50	50 100	100- 200	200–400	400 800	800- 1600	1600 3200	3200 6400	9600–12.8k	12.8k–20k
f Center (Hz)	37 5	75	150	300	600	1200	2400	4800	9600	16,400
f_{BW} (Hz)	25	50	100	200	400	800	1600	3200	6400	7,200
for R_L = 1GΩ										
Re(Z) (Ω)	74 2M	19.6M	4 98M	1 25M	0.31M	78k	19k	4 9k	1 22k	420
|Z| (Ω)	272M	140M	70 6M	35 4M	17.7M	8 8M	4.4M	2 2M	1 1M	650
e_{nz} (nV/$\sqrt{Hz}$)	1100	560	280	140	71	36	18	9	4 5	2 8
V_{nz} (μV)	5 5	3 96	2 8	1 98	1 42	1 02	0 72	0 51	0.36	0 24
V^2_{nz} (μV^2)	30 2	15.7	7 84	3.92	2 0	1 04	0 52	0 26	0 13	0 06
$(\Sigma V^2_{nz})^{1/2}$ = 7 9 μV										
R_L = 10GΩ										
Re(Z) (Ω)	8M	2M	0 5M	125k	31.3k	7.8k	2k	500	122	42
|Z| (Ω)	283M	141M	70 8M	35 4M	17 7M	8 8M	4 4M	2 2M	1.1M	650k
e_{nz} (nV/$\sqrt{Hz}$)	320	180	90	45	23	11 4	5 8	2 9	1 4	0 84
V_{nz} (μV)	1 6	1 3	0 9	0 64	0 46	0 32	0 232	0.16	0 112	0 07
V^2_{nz} (μV^2)	2 56	1 62	0.81	0 41	0 21	0.103	0 054	0 025	0 013	0 005
$(\Sigma V^2_{nz})^{1/2}$ = 2 4 μV										

Table 2. A summary of phono cartridge calculations.

f Range (Hz)	25–50	50 100	100- 200	200 400	400 800	800 1 6k	1 6k - 3 2k	3 2k 6 4k	6 4k - 12 8k	12 8k – 20k
f Center (Hz)	37 5	75	150	300	600	1200	2400	4800	9600	16 4k
f_{BW} (Hz)	25	50	100	200	400	800	1600	3200	6400	7 2k
Q = ($\omega L_s/R_s$)	0 156	0 313	0 625	1 25	2 5	5	10	20	40	68 4
Q^2	0 0244	0 098	0 391	1 56	6 25	25	100	400	1600	4678 6
$1+Q^2$	1 0244	1 098	1 391	2 56	7.25	26	101	401	1601	4679 6
$1+Q^2/Q^2$	42	11 24	3 56	1 64	1 16	1 04	1 01	1 0	1 0	1 0
R_p (Ω)	1 16k	1 24k	1 57k	2.9k	8.2k	29.4k	114k	454k	1 8M	5 29M
L_p (H)	31 5	8.43	2.67	1 23	0.87	0 78	0 76	0 75	0 75	0 75
R_p || R (Ω)	1 13k	1.21k	1 52k	2 74k	7k	18.1k	32 9k	42 6k	45 8k	46 6k
X_L (Ω)	7 42k	3.97k	2 52k	2 32k	3 28k	5 88k	11 45k	22 6k	45 2k	77 2k
X_c (Ω)	17M	8.48M	4 24M	2 12M	1.06M	0 53M	0 265M	0 133M	66.3k	38 8k
Re(Z) (Ω)	1 11k	1.11k	1 11k	1 15k	1.26k	1 73k	3.86k	12 4k	41 5k	34k
|Z| (Ω)	1 12k	1 15k	1 3k	1 77k	2.97k	5 59k	11 7k	24 4k	43.6k	40 1k
e_{nz} (nV/$\sqrt{Hz}$)	4 24	4.24	4.24	4.31	4.51	5 29	7.9	14.2	26	23.5
V_N (nV)	21 2	30	42 4	61	90.2	149 6	316	803	2080	1994
V^2_n (nV^2)	449 4	900	1798	3721	8136	22 4k	99.9k	645k	4.33M	3.98M
A^2	63 0	29.5	10 7	3.85	1.66	0 85	0 49	0 154	0.043	0.019
$A^2V^2_n$ (nV^2)	28 3k	26.6k	19 2k	13.2k	13.5k	19k	48.9k	99 3k	186k	76k

$(\Sigma V^2_n)^{1/2}$ = 3 μV unequalized noise

$(\Sigma |An|^2 V^2_n)^{1/2}$ = 0 73 μV RIAA equalized noise

EXAMPLES

Calculations of electret microphone noise with various loads and RIAA equalized phono cartridge noise is done using equations (1)-(7). Center frequencies and frequency bands must be chosen first. Values of the lumped circuit components are calculated and noise calculated for each band, then summed for the total noise. Octave bandwiths starting at 25 Hz will be adequate for approximating the noise.

In this example, the microphone capacitance is 10 pF loaded with 5 pF of amplifier and stray capacitance. Two resistive loads will be used to illustrate the effect R_L has on the microphone noise. R_{L1} = 1G Ω(10^9), R_{L2} = 10GΩ (10^{10}). It is assumed that there is no gain equalization in the amplifiers that follow. The noise calculations are summarized in TABLE 1.

The electret or condenser microphone noise [Re(Z)] is reduced when the load resistance is increased. This is one of the cases when a larger resistance means lower noise, not more noise.

The second example is the calculation of the RIAA equalized noise of an A.D.C. 27 phono cartridge loaded with C_A = 250 pF and R_A = 47k. The cartridge constants are Rs = 1.13k and Ls = 0.75H (Cc may be neglected). The noise calculations are summarized in TABLE 2 for this example.

The RIAA equalized noise of the A.D.C. 27 phono cartridge and preamp input network was 0.73μV for the audio band. Typical high quality preamps have noise voltages less than 1μV, resulting in a 3 dB or more loss in system s/n ratio when the cartridge noise is added to the preamp noise (in an rms fashion).

CONCLUSIONS

Zero noise sources and amplifiers do not exist. Specifying amplifier noise under ideal conditions will only lead to ideal specs, not a measure of actual performance. Methods of s/n ratio measurement should be used that reflect the true performance instead of hollow specs. ■

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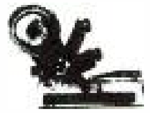

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POSITION DESIRED: Audio engineer, BA Music, 1st class F.C.C. License; 4 years' experience in music, film, spoken word recording & production. Hard working. Resume, references. **Dept. 52, db Magazine, 1120 Old Country Rd., Plainview, N.Y. 11803.**

WANTED: EXPERIENCED TECHNICAL WRITER. Experienced in professional sound reinforcement and/or recording field. Strong electronic background needed. Responsible for writing owners' manuals for electronic audio equipment. Write with resume and samples of writing to **Larry Parypa, TAPCO, 3810 148th Ave. N.E., Redmond, Washington 98052, or call (206) 883-3510.**

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db people/places/happenings

• **Harvey Zelniker** has been appointed national sales manager at **Teledyne Acoustic Research** of Norwood. Mass. He comes to Teledyne from **Pickering & Co.**

• Two new offices have been established by **Castle Associates,** manufacturer of sound level meters. A sales and service center is located at 14 Rue Cave. Levallois-Perret 92300. near Paris. A U.S. office. handing domestic distribution. is located at 515 Waverly Ct., Deerfield. Ill. **James C. Mitchell** is in charge of the Deerfield office.

• **Edmond G. Dyette, Jr.** has been elected executive vice president and director of **Super 8 Sound, Inc.** of Cambridge. Mass. Mr. Dyette. who was previously with **Scott Instrument Laboratories,** will concentrate on film sound recording.

• **Roger Faust** has formed his own international audio marketing firm. **FIMC International Marketing Consultants.** in Natick. Mass. The firm will consult with and assist U.S. and Canadian manufacturers in establishing and developing their overseas distribution. Previously. Mr. Faust was associated with the **Altec Corporation** as international marketing manager.

• Million-dollar film and recording studio. **Tulsa Studios,** located at 6314 13th St.. in Tulsa. Oklahoma. boasts the largest sound stage in the Southwest. According to president **William N. Dawson,** the facility contains over 12,000 square feet. including a 30-seat theater. editing. conforming. and projection rooms. and a computerized 15-track mixing and control room. A Cinemobile is used for location work.

• **David M. Thomas** has been named to the post of chief engineer for the **Koss Corporation.** of Milwaukee. Wis. Mr. Thomas. who comes to Koss from the **Heath Company,** will manage the engineering department. including product engineering and the coordination of technical projects.

SANTUCCI

• Marketing and sales of **Orban/Parasound** audio signal processing products and other similar lines is in the hands of newly appointed marketing coordinator **Frank Santucci.** Mr. Santucci comes from the **Ampex Corporation.** He is based in San Francisco.

• **Steve Bassett** has been named district manager of the Rocky Mountain area for **Harman International Industries,** of Northridge, California. Mr. Bassett. who will be based in Denver. will cover Colorado. New Mexico. Utah. Wyoming. and El Paso. Texas.

SIEGEL

• **Radio Shack** electronics store chain. operating out of Ft. Worth. Texas. has announced the promotion of **Hyman L. Siegel** to the post of national publicity and promotion manager. A CB expert. Mr. Siegel has been with Radio Shack since 1971.

• British **Wilmot Breeden Electronics** has opened a new showroom at 442 Bath Rd.. Slough. Equipment on display includes professional and semi-professional tape recorders. test equipment. monitor loudspeakers and marine depth sounders. as well as Wayne Kerr bridges. automatic test equipment. and electronic gauging systems.

• Several personnel changes have taken place at **RCA Broadcast Systems.** of Camden. N.J. **Jesse L. Nickels** has been appointed manager for midwestern broadcast sales. **L. A. Pinski** has been promoted to the responsibility of marketing in the northwestern U.S.. based in Seattle. A third new sales representative. operating out of San Francisco. is **D. Gerald Smith.** who will represent the firm in Nevada. Utah. southern Idaho. Oregon. and northern California.

• The exclusive distributorship for North America of disc cutting equipment from Danish **Ortofon Company** has been given to the **L. J. Scully Mfg. Corp.** of Bridgeport. Conn. Ortofon products will be stocked in the Bridgeport warehouse and a repair center specializing in cutterhead rebuilding. will be established at 138 Hurd Ave., Bridgeport.

• There have been several changes at **Design Acoustics.** of Torrance. Ca. **Steven Levine** has been appointed as national sales manager and several new rep firms have been selected. Representing the firm in Texas. Oklahoma. Arkansas. Louisiana. Mississippi and western Tennessee will be the **Texport Co. Paul Blumentritt Sales Co.** will operate in Minnesota. North Dakota. South Dakota. and western Wisconsin. **Hutmacher & Associates** are the reps for Illinois. Wisconsin. Lake County. Indiana. and Davenport. Iowa.

• The Department of Journalism and Communication at the **University of Bridgeport** and **Dondisound Studios, Inc.** of Red Hook. N.Y. will sponsor an Institute of Audio Studies. to be held at Dondisound Studios from June 6 through August 5. For information. contact **David Moulton,** Dondisound Studios. Inc.. Red Hook. N.Y. 12571.

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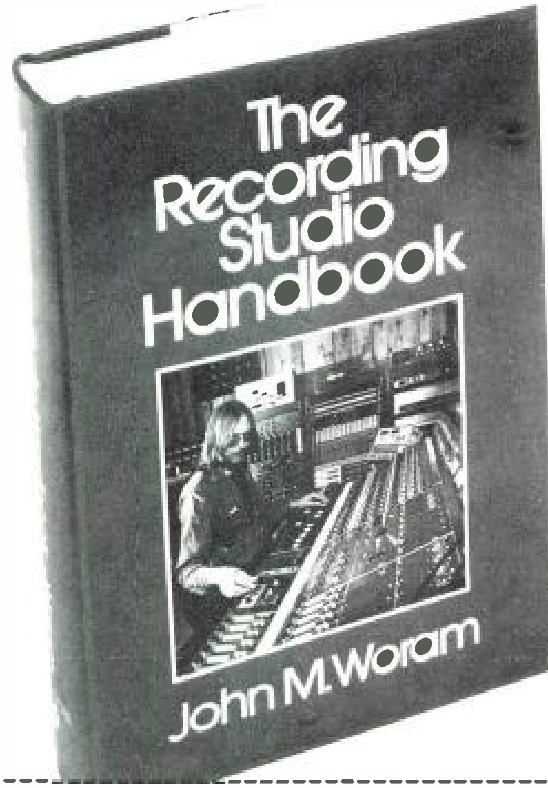

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