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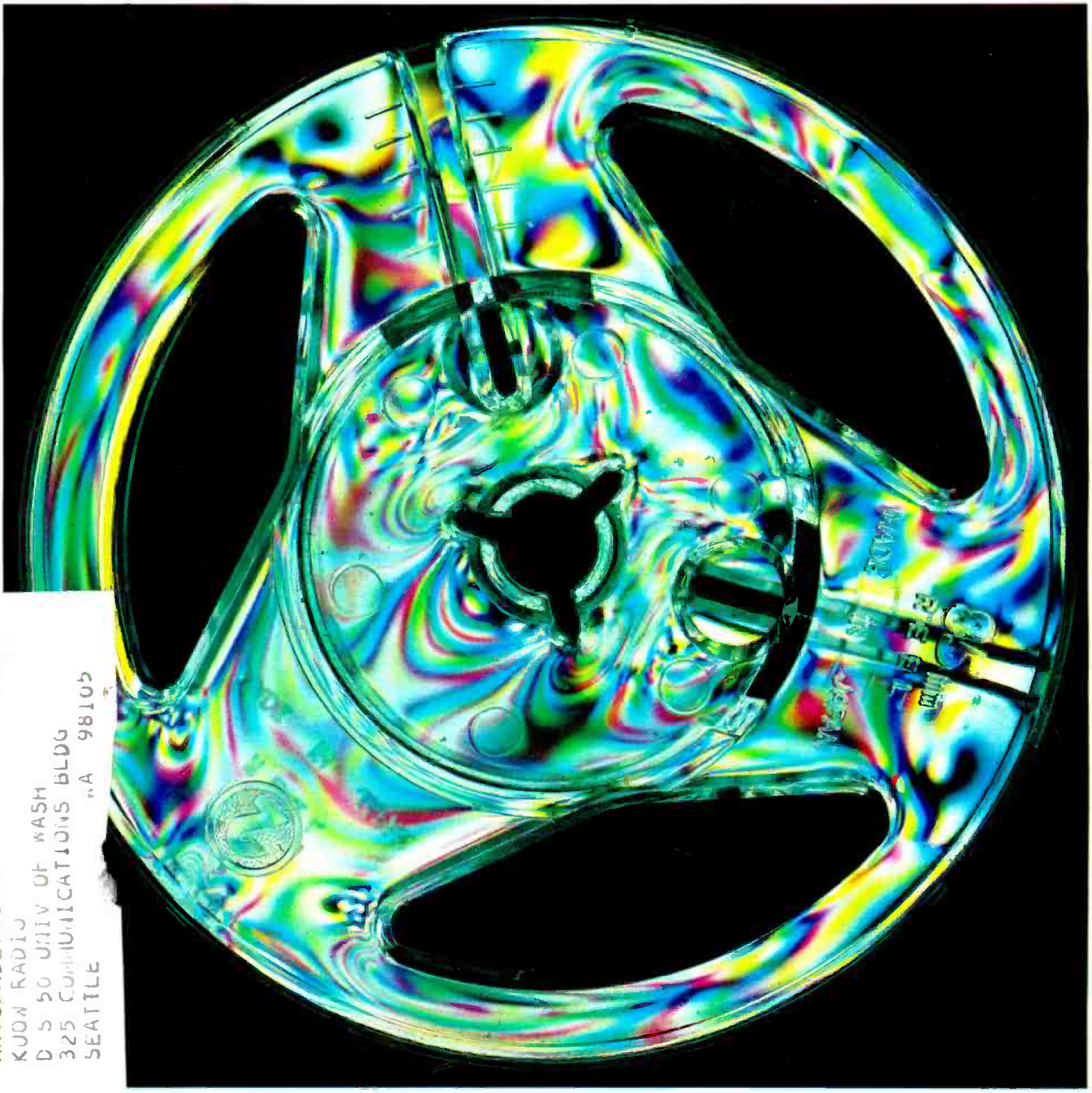

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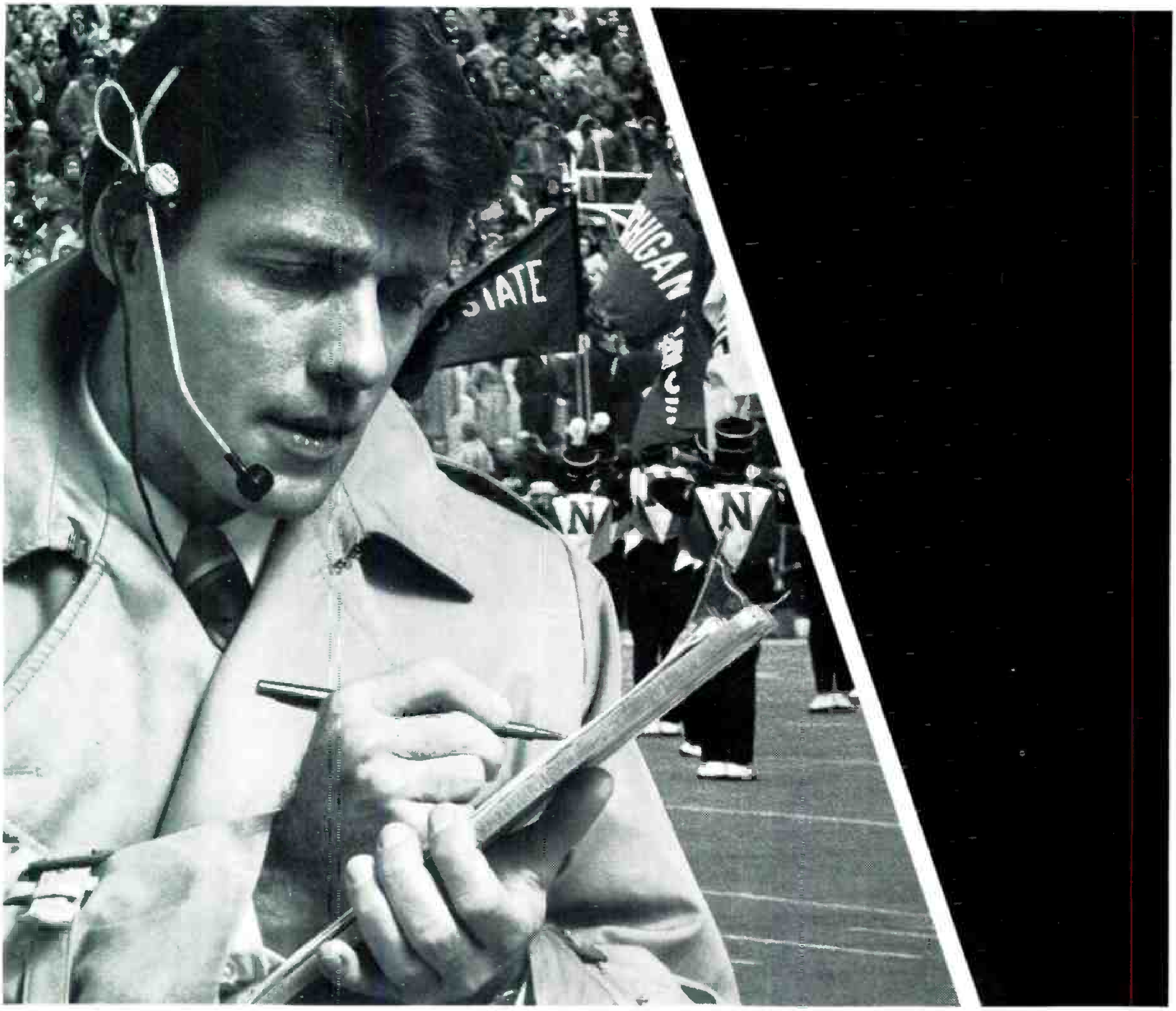
STATE
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coming next month

• Walter Jung will be back with us with Part II of his SIGNAL PATH, this time discussing sine wave oscillators, particularly the Wien Bridge oscillator.

• **db** was represented in full force at the May AES show in Los Angeles, and a complete report of the doings will appear in our July issue.

about the cover

• This is a psychedelic interpretation of a tape recorder reel.

db

THE SOUND ENGINEERING MAGAZINE

JUNE 1976, VOLUME 10, NUMBER 6

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266 pages lists thousands of electronic components and equipment. This catalog sells for $5.00, is offered free to **db** readers. Specify Dept. DB. Mfr: Allied Electronics.

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MAGNETIC SHIELDING

Physical data, application notes, and fabrication methods are contained in catalog MG-3, covering Co-Netic AA alloy. Mfr: Perfection Mica Co., Magnetic Shield Div.

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STATE-VARIABLE FILTER CONFIGURATIONS

Detailed instructions, including numerous mathematical formulae, are aimed at the design and specification of "biquad" active filters. 51 pages. Mfr: Kinetic Technology.

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SPLICING TIPS

Along with a listing of their splicing equipment, the manufacturer offers tips on splicing in this leaflet. Mfr: Nagy Research Products.

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32 pages detail power supplies, including open frame, miniature encapsulated, dual output, modular, high power, and special power applications, as well as a listing of overvoltage protection devices and a.c. line conditioners and regulators. Mfr: Tele-Dynamics.

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OVER/UNDERVOLTAGE PROTECTORS

More over/undervoltage protectors are covered in a four-page brochure. Mfr: Heinemann Electric Co.

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ROTARY SWITCH

Mechanical and electrical characteristics of the Monodex one-inch diameter rotary switch are covered in this brochure. Mfr: Oak Industries, Inc.

Circle No. 92 on R.S. Card.

CUSTOM FILMSTRIPS

This brochure, including a small sample of their work, describes a service creating filmstrips to order. Mfr: De Luxe Laboratories.

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660 clamp-mounted . . . $75.00; 661 stud-mounted . . . $78.00; 671 . . . $72.00 (suggested resale net—slightly higher in western states).

Electro-Voice, Inc.
Dept. 662BD, 686 Cecil Street
Buchanan, Michigan 49107

db letters

THE EDITOR:

I am prompted to comment on the coupling circuit shown in the article by Norman Crowhurst in Figure 7, page 40 of the March issue. This coupling circuit should provide no problems of its own, providing maximum signal levels have been determined. The circuit does suffer from poor stability, which Mr. Crowhurst has failed to mention. Bias stability may be simply stated as: $SF = \frac{\Delta Ic}{\Delta Ico}$.

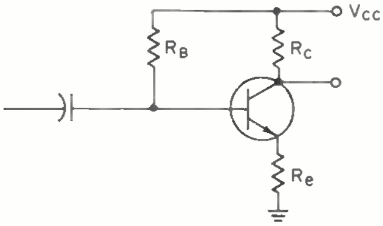
In the circuit shown, the no-signal collector current may be expressed as:

$$Ic = B_{IB} + (1 + B)\ Ico$$

and $I_B \cong \frac{Vcc}{RB}$ Therefore,

$$Ic = B\frac{Vcc}{RB} + (1 + B)\ Ico.$$

Since $B\frac{Vcc}{RB}$ does not change with Ico, this term diminishes and the stability factor ≅ 1 + B, and since B is usually large. SF is poor and totally dependent on the beta of the transistor. To realize a desirable SF of 10 or less, the addition of an emitter resistor, Re, can provide significant improvement.



$$SF = \frac{1 + \frac{Re}{RB}}{1 - \alpha + \frac{Re}{RB}}$$

A stability factor of 10 or less is easily realized, making the circuit almost entirely immune to variations in beta of the transistor used, which is consistent with good engineering design and practice.

CHARLES BLAISDELL
Engineer, IBM Corp.

Mr. Crowhurst's Reply:

I am glad that Mr. Blaisdell raised this question, because it illustrates very well what I meant, although I did not take it that far. First, all his formulas are correct. He is regarding β as the only quantity that is variable, but he is not noting all the factors that make it change.

It will change from transistor to transistor. It will also change with the temperature of the same transistor, thus leading to the instability factor he discusses. But that is not all. β also changes with operating point. Quite specifically, at cut-off β becomes zero. Now *where* cut-off is, and thus where β becomes zero depends on the stability factor he discusses, because this has the effect of moving the operating point.

Once the transistor has gone beyond the cut-off point, the base-emitter junction acts as a rectifier, changing the operating point even more, biasing it further into cut-off, with a return time determined by R_B and the coupling capacitor, which can be quite long, relative to the lowest frequency the stage is designed to handle, thus causing blocking.

Now, his suggested remedy, the using of R_e: To apply his formula, R_e/R_B needs to be substantially larger than unity to have any effect in improving the stability factor. Consider the relationship R_B/R_c. To provide an operating margin, this should be of the order of 2β. So R_e needs to be more than 2β times R_c.

Presumably he plans to use a bypass capacitor across R_e; otherwise the stage is extremely degenerative. But even then, consider the d.c. voltage distribution. R_e must drop at least 2β times the voltage across R_c. This severely limits the handling capability of the stage, even more than the stability factor it seeks to remedy.

Perhaps it is time to write another series, dealing more fully with the varieties of solid state circuit, and discussing in more detail the effect that circuit values have on operation, both linear (as given by the kind of formulas Mr. Blaisdell quoted) and non-linear (what happens when the so-called linear range of operation is exceeded).

On that last point, note that the limit does not necessarily mean too large a signal has been presented. It can also occur because the operating point is incorrect, caused in turn by a poor stability factor. In short, the poor stability factor is only the beginning of a poorly designed circuit's troubles.

NORMAN H. CROWHURST
Dallas, Oregon

THE EDITOR:

Regarding the letter from R. Dennis Alexander of Radex Productions concerning warped reels and the Revox A77: WPLZ is a beautiful music-automated f.m. station which uses three A77s, and the problem of warped reels has been a heartache to us as well.

My advice is to remove the screws from the warped reel and take off the two side plates. Obtain a suitable board or other surface with an absolutely smooth, flat surface (a Formica-covered sink cutout from a newly constructed house is excellent). Using the flat surface as a gauge, gently bend the side as required until it will lay perfectly flat. Repeat for the other side plate and then reassemble the reel. This process can be time-consuming and boring, but it is the only solution I know of that doesn't cost anything. Those reels can sure hammer up the surface of a Revox—not to mention what it can do to the sound.

PAUL H. BOCK, JR.
Chief Engineer, WPLZ
Petersburg, Va.

THE EDITOR:

Mr. Alexander's problem with reels scraping the front panel of his Revox (March, 1976) is quite a common one among broadcasters. During the course of a broadcast day, we encounter many defective reels, inasmuch as a good deal of the syndicated material we air is supplied on pancake plastic reels.

To solve the problem simply cut a 3 x 5 index card in half, cut a hole large enough for the spindle to pass through, and place the card under the reel, between the reel and the spindle. Not only will this solve the front panel problem, but it will also prevent the tape from scraping the sides of the reels as it winds on and off.

JOSEPH A. MARTIN, JR.
Station Manager, WSCI
Charleston, S. Carolina

Editor's note:

While this method and those described in earlier issues (careful straightening of the metal reels) will cure the scraping problems, the simplest suggestion is to secure a pair of vinyl reel shims such as are included with tape decks supplied by Otari and Teac. These inexpensive (and durable) shims come in several thicknesses and should be readily available at least from the aforementioned two suppliers.

db the sync track

JOHN M. WORAM

• Recently I received the following letter.

I was interested to read John Woram's comments on magnetic tape (**db**, January, 1976 Sync Track), and I am sure that his factual article will help many technicians. However, when it come to the *confusion factor* of zero vu, high output tapes, and the dreaded "operating level," we Europeans never cease to be amazed at American practice.

Firstly, just what is zero vu? Okay, it's the line level when a vu meter reads zero which, with a proper standard vu meter, happens to be +4 dBm. Unfortunately, most so-called vu meters have little relation to the genuine standard vu meter, which should have well-defined ballistics and use a full wave average rectifier characteristic.

Even when the genuine instrument is used, the indicated level—in relation to the peak level of the waveform—is a very variable factor which depends enormously on the program material, it being generally accepted that zero vu should lie somewhere between 8 dB and 12 dB below the 3 per cent distortion point of tape. It is here that the confusion really starts!

"Operating level" was originally designed so that one put an operating level calibration tape on the recorder and set the replay chain for zero vu output. That was just fine, because tapes available at the time all had their 3 per cent distortion points about 8 dB above operating level, and if the meter indicated somewhere around zero on peak program (with a genuine vu meter), all was well.

Fortunately, from a point of view of quality, we now have so-called high output tapes and Dolby. Also, we have the peak program meter, which in many ways is to be preferred to the vu meter. And, with the sole exception of Dolby, there is every reason to bury "operating level;" if the best performance is to be had from modern tapes, a quick death would save much head scratching!

All we need to align levels is a standard reference level and a knowledge of the tape's performance in terms of that reference level. It happens that the standard reference level, or operating level, is around 185 nWb/m tape flux (in the USA) and that in Europe we have standard levels of 320 and 510 nWb/m. These levels are all of equal use, as we can easily calculate the difference in decibels from the simple formula:

$$\text{Difference (dB)} = 20 \log \frac{\text{flux A}}{\text{flux B}}$$

In practice, we only need to remember that there is about a 4 dB difference between 185 nWb/m and 320 nWb/m, or between 320 nWb/m and 510 nWb/m.

If we have the tape manufactturer's specifications for distortion and noise in terms of a reference level, we know exactly where we stand, and we can set our headroom between zero vu and 3 per cent distortion as we like.

Another important factor is that if recorder manufacturers would forget the magic signal-to-noise specification and specify reference level-to-noise, we would then be in a position to determine the available dynamic range for a given type of tape.

Yours sincerely,
HUGH FORD

Well, I ought to be able to milk this letter for at least two columns of confusion factoring. Hugh points out that zero vu corresponds to +4 dBm. Sure enough, if you follow an engineer around for a while, sooner or later you'll hear something like, "Um, let's see now, uh, zero equals plus four, so therefore this meter is really reading a plus seven, or is it mumble mumble, etc. etc,?"

0 = +4? What sort of nonsense is that? Does it mean that dBm readings are simply vu readings, plus four? And if so, why? And if not, why not? (Long live confusion factors!)

To get to the bottom of this, let's back up and find out what a vu is, and then compare it with a decibel.

VU MEANS VOLUME UNIT

Vu stands for volume unit, and it is *not* a symbol for dB (decibel). Now and then, we read that a certain sound has a sound pressure level of so many decibels. For instance, the noise level in a factory may be about 80 dB The background noise in an office building may be about 50 dB. Both sounds are reasonably constant in noise level, and we would certainly agree that the factory is noisier than the office. But occasionally, someone slams a door in the office, and there is a momentary peak far above 50 dB. The peak is over and done with almost instantly, and we hardly take it into account.

And so it is with music. In a concert, there may be a peak here and there that is indeed quite high in sound pressure level. But the peaks are not sustained, and our ear tends to average the level over a period of time, so that we don't think of the moderate level program with infrequent peaks as being "loud." Note that "loud" and "soft" are subjective terms, and may have little to do with the levels measured on instrumentation devices. That's why highly compressed top-40 radio stations strike us as being "loud." The peaks have all been squashed, and then the overall level is brought up. The average level is up, and we think, LOUD. On the classical station (assuming they haven't gone compressor-crazy), much of the music is low level. There may be an occasional peak, but the low average level makes us think of the total program as being soft in loudness level.

INDICATOR FOLLOWS EAR'S CHARACTER

Well, the volume indicator more or less follows our ears' characteristics; it averages out the program level, and sudden transients do not really register at their actual dB sound level. This makes the volume indicator a valuable studio tool, since its visual display pretty much conforms to our subjective impression of loudness. But, we must remember that those infrequent peaks may indeed be—as Hugh suggests—some 8 to 12 dB higher than the volume indicator reads. Or, to paraphrase his letter, when "operating level" is some 8 dB below saturation, those now-and-then peaks that seem to be safe (about zero on the meter) may actually exceed operating level by 8 dB, and yet not saturate the tape.

To make sure we don't mistake the volume indicator display for the actual dB level, we read it in volume units, not dB. (Or at least we should). To calibrate the meter face, we feed in steady state decibel levels, and mark off the meter in 1 dB increments. No, this is not a contradiction of what was just said. As long as the signal is a steady level, the volume unit corresponds to the decibel. But, when we're reading music (which is most of the time), the volume indicator follows our ear, while a real dB meter would be reading much higher on those transient peaks.

So, why not just say 0 vu = 0 dBm, and understand that this equality only applies to steady-state test tones?

Well, it turns out that this wonderful volume indicator, that conforms so nicely to our subjective impression of loudness, also manages to load the circuit across which it is placed. In other words, every time we switch the volume indicator into the circuit, the level drops a dB or so! And to think this turkey was designed by an engineer!

We get around the problem by putting a resistor in series with the meter movement. A 3.9k ohms value seems to eliminate the loading effect nicely. But now, the meter reads 4 dB too low, and an actual 0 dB is read as —4 dB. To put it the other way around, a meter reading of zero indicates an actual level of +4 dB. So, since we're going to read the meter in vu anyway, we simply say that 0 vu = +4 dBm. What could be simpler? (I can think of a lot of things.)

But now, all we have to remember is that although a sustained zero vu reading represents a +4 dBm level, if the program contains a lot of transient peaks (A snare drum track, for instance), that same safe-looking zero may conceal a lot of instantaneous peaks that are some 8 to 12 dB higher. We don't hear them as being that loud, but the tape does, so watch out!

Many engineers are finally re-discovering the peak reading meter, which more closely follows the actual program peaks, and allows us to keep a closer look out for those distortion-producing transients that escape notice on the vu meter

(To be continued, unfortunately) ■

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Circle 22 on Reader Service Card

CALENDAR

JUNE

7-11 **B & K Seminar: Industrial Noise Control.** Contact: B&K Instruments, 5111 W. 164th St., Cleveland, Ohio 44142. (216) 267-4800.

8-10 **Synergetic Audio Concepts Training Session.** Columbus, Ohio. Contact: Don Davis, P.O. Box 1134, Tustin, Ca. 92680. (714) 838-2288.

8-11 **Communications '76,** Brighton, England. Contact: British Information Services, 845 Third Ave., New York, N.Y. 10022. (212) 752-8400.

8-11 **Information Retrieval Exhibition, London, England.** Contact: British Information Services.

8-11 **International Audio-Visual Aids, London, England.** Contact: British Information Services.

21-July 9 **Audio Recording Technology Workshop.** Brigham Young University. Contact: Special Courses & Conferences, Brigham Young University, Audio Recording Technology Course, 242 HRCB, Provo, Utah 84602.

• Nine technical institutes will be offered this summer by **Union College,** Schenectady, N.Y.: Noise Control, July 12-16; Fracture Mechanics, July 12-16; Advanced Fracture Mechanics with Industrial Applications, July 19-22; Instrumentation, Measurements Engineering and Application, July 19-23; Computer Workshop in Finite Element Methods of Analysis for Stress and other Field Problems, July 26-30; Computer Workshop in Earthquake and Structural Dynamics, July 26-30; Micro-Processors and their Applications, Aug. 2-6; Energy Constraints on Transportation Systems, Aug. 2-6; Fatigue and its Relation to Engineering Problems, Sep. 13-17. Contact: Office of Graduate Studies and Continuing Education, Wells House, 1 Union Ave., Union College Grounds, Schenectady, N.Y. 12308.

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db sound with images

MARTIN DICKSTEIN

• In the recent past, I have been involved in presentations which were given in strange surroundings for the presenters. A technician was on hand from the site who was assigned to assist with the operation during the presentation in any way possible. Although some coordination did take place before the time of the meetings, some of the circumstances that existed were cause for thought and worthy of mention to those of you who might run into some of the same conditions.

One of the facilities was a rear screen projection system with controls on the lectern to operate remotely the audio visual equipment. The illuminated buttons indicated what they controlled. Labels *front*, or *rear* showed they either opened the little trap door in the ceiling to bring down the recessed front projection screen or opened the panelled doors in front of the rear screen projection wall. A button said *drapes*, another *curt* (for curtains), and four light buttons each had a different indication. Other buttons said *full*, *half*, *meet* (meeting, we learned later which meant about two-thirds) and *aux* (about one-third with side and sconce lights). There were controls for two slide projectors with forward, reverse, and focus and for two overhead projectors. There was also a control to run and stop a 16mm film projector and a volume control to adjust the level of sound in the room. A rack-mounted tape recorder could be started and stopped, and the microphone on the lectern or the lavalier could be turned on and off. It was a model of engineering ingenuity and amazed as well as perplexed the visiting presenters, who were not accustomed to such complex controls. They finally learned which buttons to push for their own sections of the presentation and all seemed to go well before the meeting started during a brief rehearsal-like rundown.

The house technician stood by behind the screen "just-in-case" . . . but it was found out that he was not yet familiar with the system because it was new. To make things worse, the techniques of getting the stereo tape recorder to feed into the room and adjusting the film level. etc. were not yet completely worked out. In fact the system was so new it had not yet been turned over to the new owners. and there was no manual or operating instruction book with the system! Fortunately, there was enough time to figure things out.

Obviously, the system had been designed according to the specifications of the architectural and audio visual consultants who in turn did what was necessary according to the requirements of the future owners. The a/v contractor did his job according to his contract. However, it was slightly unnerving, to say the least, for the presenters not to be sure that the audio-visual aids to be used in the talks could be made to work without a hitch. No one is really to blame, yet it seems that something could have been done to avoid just this kind of circumstance. Perhaps the technician could have been briefed on the system's intricacies, or an instruction booklet left with the equipment when it was brought in for installation.

NO LIGHT CONTROL

On another occasion, there was a similar installation, but much simpler. The front screen also came down out of the ceiling, operated by a switch on the side wall. The switch was on a multi-gang panel on which there were other switches. One of these worked the rear light, one turned the front lights on and off, and one turned all the lights in the room to half light, off, or full on. The panel on the wall also had a switch to operate a single slide projector in the rear screen setup, and a buzzer behind the screen for the operator to start a 16mm film projector or the tape recorder in the rack.

There was no provision, however, for the technician to monitor what happened in the conference room. There were also two other problems; one was the omission of a control to dim the lights in the room during a meeting with front screen projection. Either the room was at half-light or in darkness. With only fluorescents in the ceiling there was no dimmer, so the blinds on the windows in the rear of the room had to be left open slightly if a dim setting was required. The other problem was that there was another meeting room on the other side of the common rear screen projection room. A glass partition faced the rear screen in the larger meeting room; when the lights were full on in the little room, the rear screen in the other room lost some of its sharpness and brightness.

Perhaps some more thought could have been given to the design of the room's a/v systems and installation. The client said that he had requested a dimmer in the large room and could not figure out why none was installed. He also realized that a visiting speaker would have a problem with the panel inconveniently positioned on the wall, but that's the way it was and he had gotten used to it. He did say also that he was disappointed that his technician was not familiar with the equipment rack, and that he worried every time there was a meeting using the equipment. He said the audio-visual contractor was lax in giving instructions, and he had reported this to the people in his company who had worked with the design consultants. Sloppy planning like this can only lead to possible bad word-of-mouth next time.

Then there was an impressive panelled facility with a huge rear screen in a large board room. A very long conference table stood in the center of the room. The area was about 50 feet by 40 feet, and a lectern stood at the front toward one side. There were entry doors at each corner, since the room was built in the center of the floor space. The rear screen was about 9 feet wide by 6 feet high. A wide front screen up in the ceiling was controlled by a switch on the wall near the light switches. The rear projection room was spacious and occupied only by a slide projector on a specially built stand and a very large front surface mirror. The intent of the setup was, obviously, to fill the screen with the slide image. This meant that the projector was in a corner near one side of the screen and the mirror was at the proper angle to avoid keystoning.

With the mirror set up as it was, there was no room to move around in the projection room except near the walls. There was also no place to put a 16mm movie projector in the event both slides and film were to be used. The slide projector was dead center of the mirror and the screen, and any image the film unit threw would either be cut off or keystoned.

We were told that no one used the facility that way, that is, with both slides and film intermixed. The suggestion was made to run the slides on the rear screen while the film was put on the front screen from the rear of the room.

That was not feasible in this case as the presentation *did* consist of slides and film intermixed. The only solution that seemed possible in the short time for setting up was to move the mirror to another angle, move the slide projector so that it was parallel to the screen and put the film projector with its own right angle lens between the screen and the slide unit. The images were both slightly off center and smaller than full screen, but it worked. There was no keystoning and the images were still large enough to be seen clearly at the rear of the room.

NO MIC!

However, that was not the only trouble. There was no microphone in the room for voice reinforcement and no way for the technician behind the screen to hear what was going on in the room, other than craning a neck around the small window at the side wall of the projection booth which led into the conference room. The slide control "pickle" and its cable, operated by the presenters, had to be run through this same window to the lectern location, awkward for the presenters. Aside from mystifying the monitoring technician, the lack of a mic was a serious drawback. Without any microphone, some speakers can't be heard clearly at five feet, let alone at forty. Trying to speak up, for some people, can be a problem and the presentation loses some of its impact.

Each of these situations can be multiplied a hundredfold in the number of installations in which there are problems in the final system. Sometimes the culprit is limited funds, determining how many features can be built into the facility. Other times there is lack of forethought or wisdom on the part of the client. In a few instances, its lack of knowledge on the part of the people who design the audio visual setup. If you get in on the early stages of an a/v project, you might try to show some examples like the few described to point out to the future owners, and users, possible problems and inconveniences. Sometimes talking won't help, but a letter report might do some good. At least you can't be faulted for not trying. ■

• Some months back, I made a casual comment about why certain forms of music sound louder than others and I have not ceased to hear about it since! The main topic under discussion was the difference between loudspeakers intended for hi-fi and those for commercial sound of various kinds.

In the discussion, I mentioned the effect of the environment on what the listener expects of the sound, as well as the character of the program. It was suggested that rock music sounds loud because of what, in other forms of music, would be called distortion. But in rock, it is deliberate, built-in.

One reader wrote to point out the preference that many guitar users have for the old tube amplifiers, having never accepted solid state. As he pointed out, the reason is that tubes possess something that engineers of the era had sought to design out of them: microphonism. Early amplifiers were limited in the gain that could be used in the same room as the loudspeakers before early stage tubes would pick up the sound and produce a howl, just like acoustic feedback.

Guitar users liked to operate just below that point so that the tubes would act as rather poor microphones and produce a sort of extra reverberation effect. At that level, the distortion limited the output for the early part of the note—the pluck sound and the first part of the decay—and the *pongy-ness* of the tubes would cause the later part of the decay to hang on longer.

This was a characteristic sound that certain guitarists loved, and that solid state could not duplicate, because it is too "clean." Yes, I remember that situation well. I was consultant to one of the guitar amp manufacturers at the time. The manufacturer naturally wanted distortion-free reproduction—something more like hi-fi products—but when he demonstrated them to guitarists who were prospective customers, they said the amplifiers were "gutless!" No distortion—or not enough!

MUSICAL DISTORTION

So why not use a 1 watt amplifier, and push it to what would be 50 watts, if distortion did not hold it down to about 2 watts? Wouldn't that be the ultimate? It seems there is an optimum in distortion, for each guitarist's ear. Some like more, some like less. Now let me hasten to add that, in calling it distortion, I am thinking as an engineer, not a musician, because I am well aware that it is really a musical form, and thus the musicians who like it would hate to hear it called "distortion!"

Another letter takes off in a different direction. The writer inquires whether any recent work has been done on the effect of distortion on the listener, psychologically. He refers back to the early work, done to determine how much—or rather how little—distortion is audible, and wonders whether any of these findings have been updated, scientifically. He is aware that much has been said about it, with not too much scientific support.

Reading his letter reminds me of an experience many years ago, in the early days of electronic reproduction. A veteran musician preferred an old acoustic phonograph for listening critically to a musical performance, rather than one of the modern high fidelity systems. To my ears, the acoustic phonograph completely obscured the music with scratching. Whether it produced distortion, I could hardly tell for the scratch.

Yet the old musician complained that the electronic reproduction prevented him from hearing some nuances of the music that he could hear in the acoustic phonograph reproduction. Inasmuch as my ear was not trained as well musically as the old musician's, I was not about to argue with him, but it did make me think. Rather than telling you my conclusions, I'll let you think about it too.

One thing, however, was obvious. The old man could ignore all that scratch, and any distortion that might be with it, and "listen through," to whatever it was he was hearing. And in spite of the much greater apparent clarity of the electronic reproduction, to my ears, there was something about it that obscured whatever it was the musician was hearing in the acoustic reproduction.

Although I found this view difficult to believe, I had to accept it, on the authority of the old man's experience and capability as a musician. Too often, we think we know something, and thus reject something else that appears to disagree. This can happen in an engineering context, in a musical context, in an educational context, or in a political context. If we are to learn, we must be prepared to accept other people's experience.

SUBJECTIVE EVALUATIONS

The old Fletcher-Munson curves have been verified by much more recent work. Work on masking has also been performed, and verified, over the years. The work, though it involved subjective measurement—what subjects observed—had an objective basis. The observers were told to listen for audibility, or for audible equality. Though these were subjective evaluations, they have a definitely objective property to the observer.

The effect of distortion on our perception is much less objective, more subjective. Perhaps we can illustrate this with a somewhat parallel phenomenon, sometimes called the "cocktail hour effect." You are in a room where a great many separate conversations are going on at the same time. If you completely relax, the sound is a hubbub of voices, in which no conversation predominates. You may hear a word emphasized here, and another one there, but you are not following any particular conversation. If you make a recording of it, your capacity for unscrambling the sounds are even more reduced.

But perhaps something said captures your attention because it interests you. By some means, inside your interpretive faculty, you begin to isolate that piece of the conversation. It is probably nowhere near the loudest sound, yet you manage to hear it above all the others. There may be people closer to you who are also talking, and at your ears, their voices are much louder than the ones you are listening to. But you can ignore what the people nearby are saying, and listen to the intriguing conversation taking place in another part of the room.

Has anybody measured that capability? Is it even feasible to measure it with any precision?

FUNDAMENTAL FREQUENCIES

Here is another piece of the puzzle, that has been documented. If a series of harmonics of a certain fundamental frequency are reproduced together, the tone you *identify* as hearing, even if that frequency is totally absent, is the fundamental. In terms of distortion, I suppose you could say you were listening to infinite distortion, and hearing something that was not there at all. But is that how you would describe the sound of bells?

Reverting to the question of how much distortion is audible, which would obviously be a first step toward determining its psychological effect, the early measurements suggested that any amount of distortion less than 5 per cent was completely inaudible. Today, that sounds a little ridiculous. Here we are, struggling to get distortion down to point zero-zero-something per cent. If you cannot hear anything more than 5 per cent, why all that fuss? Obviously we can hear the lower levels. Then why did not those early measurements show it?

This really illustrates the folly of taking a statement of alleged fact without checking the context of the original statement. Earlier, we suggested that this kind of mistake occurs in education and politics, too, with perhaps even more disastrous effects. The same occurred later, with measurements on the stereo illusion.

CHECK THE CONTEXT

It is always important to check the context in which measurements of any kind are made, to be sure *exactly* to what they relate. If you take a pure fundamental, sine wave, and add controlled amounts of second harmonic, reproducing the result through a system that has 5 per cent distortion, largely 3rd harmonic and higher, then you may have difficulty detecting the addition of second harmonic until it gets over 5 per cent.

But adding second harmonic is not the only effect of distortion. In the early days, when single-ended tube amplifiers were the vogue, and feedback had never been heard of, second harmonic was probably the most common form. When push-pull amplifiers were first introduced, we remember claims made for them that they "eliminated distortion." So how come we are still struggling with it?

Obviously what they eliminated was what had been regarded as *the* distortion, up to that point in time: second harmonic. A correctly adjusted push-pull amplifier does eliminate second harmonic distortion. But there are lots of other harmonics that it does not eliminate. And there are many other kinds of distortion.

INTERMODULATION DISTORTION

Next into the audio vocabulary came i.m.—intermodulation distortion. Again, do not forget the context. We have gone through these differences in earlier columns. But we still encounter plenty of people who talk glibly about i.m.d. without bothering to specify which kind they mean, or how they measure it.

If they do specify the method of measure, such as SMPTE, they then assume that their meters tell them the gospel truth about that form of distortion. For example, did you ever think that Doppler distortion is a form of i.m.d., virtually indistinguishable from the kind that the SMPTE test measures when you listen to it, but that it will get by the SMPTE test completely?

This is because the lower frequency frequency-modulates the higher one, rather than amplitude-modulating it. The SMPTE test detects the amplitude modulation, but not the frequency modulation. Once the signal has been let loose in an acoustic environment, and you listen to it, you cannot tell the difference.

Or do you specify i.m.d. according to the CCIF test? That is where you use two high frequencies as input, and measure for presence of the difference frequency. For example, you may use 5,000 hertz and 5,100 hertz as test frequencies, and check for presence (or absence) of 100 hertz in the output. What does that test tell you?

Assuming the test is clean, that is, of satisfactory low reading, it means that you have no second harmonic distortion at 5,000 or 5,100 hertz. You could have bags of 3rd, or any other harmonic, and the test would never tell you.

As an engineer, one tends to think in terms of such purist distortion. But the kind of which you are more often aware never conforms to these precise definitions. They produce spurious effects in the program content, having various degrees of audibility, connected not only with the level at which the spurious components appear, but also with the perceptive capability of the individual listener.

We still know very little about all this. ■

PATRICK S. FINNEGAN

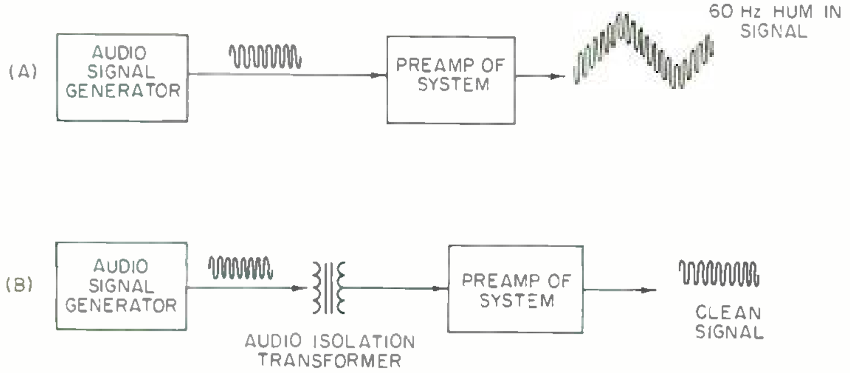

Figure 1. Hum can be introduced if there are grounding or other differences between the signal generator and the system input. An audio-isolating transformer will correct the problem.

Preparing for Proof

Before undertaking Proof of Performance, the station engineer should spot check his system

• The FCC rules require each broadcast station to make a complete set of measurements on its system once each year. The particular measurements to be made are specified and so are the tolerances. This is called the "Annual Audio Proof of Performance." Proof of Performance means just that—a demonstration by actual measurements that the station operation is in compliance with the technical Standards of the Rules.

The bulk of the measurements center around the audio response, distortion and noise in the overall system—from main microphone input terminals to antenna output. Since there are so many measurements to make, and since the results must demonstrate compliance, the station engineer would be wise to invest time in making spot checks of the system before making the actual measurements. These spot checks will point out problem areas that exist and must be corrected before making the Proof measurements. In this article, we will discuss some of the pertinent areas where problems can arise, and also some methods for spot checking.

NOISE

Noise in the system will affect *almost all other* measurements. It is immaterial what form the noise takes; so long as it is in the audio bandpass it will be a part of the measurements. During programming, for example, noise may be masked. But even though the actual level may be inaudible, it can still be high enough to cause other measurements to go out of tolerance.

Distortion measurements are particularly susceptible to erroneous indications because of noise. This is due to the type of distortion analyzer in use. The audio tone to be measured is fed to the input of the analyzer, which internally nulls or cancels out the fundamental frequency. The measuring section then indicates everything remaining as distortion. The remaining components are harmonics of the fundamental tone and system noise. If the noise in the system is too high, the apparent harmonic distortion will indicate high and can well be out of tolerance.

Most distortion analyzers provide an output for oscilloscope viewing of what is being measured as distortion. During the spot checks, if the distortion appears unduly high, use the oscilloscope to observe what is being measured. When noise is a significant part of that distortion reading, the 'scope will show this very plainly. Try to identify the type of noise, as this will be very helpful in pointing out the possible source of the problem.

NOISE SOURCES

Noise may be introduced into the system at many places. It is not at all unusual to find that the particular test equipment setup and connections to the system are major sources. The signal generator can be the offending unit, because it connects to the low level microphone input of the system. But noise entry at this point can also be misleading, since it will appear in the distortion measurements but not in actual noise measurements. This apparent paradox is due to the fact that the input signal is removed and the input of the system terminated during a noise measurement. This simple action also removes the noise source! If there are problems in this regard, insert an audio isolation transformer between the signal generator and the system. In most cases, the hum will disappear immediately.

Try to identify the character of the noise when observing it on the oscilloscope. If this is 60 Hz, then look for a.c. primary circuit problems. There may be 120 V a.c. power circuits close to low level microphone circuits, or the magnetic field around a power transformer might be coupling to a transistor stage or other nearby circuit. But if this is 120 Hz hum, it almost always comes from power supplies. Look for poor rectifier filters or decoupling capacitors in the audio units.

The problem can also be ripple on the d.c. bias voltage to the vvc diodes in the f.m. transmitter modulator. The major power supplies in the transmitter will be single or 3-phase. These have their own peculiar appearance on the 'scope and may be accompanied by switching transients (spikes) in the rectifier. Any noise in the audio system or input to the transmitter will appear as noise in the recovered f.m. audio. Generally, noise problems caused by the transmitter power supplies, or cathode-filament leakage, or shorts in the rf stages will produce amplitude modulation of the f.m. carrier. (This is another of the special f.m. measurements that must be made.)

The first spot check of the system should be a noise measurement. If this is higher than it should be, then a simple isolating technique can help run the noise down to the offending unit quickly. Once the measurement is set up and indicating, terminate the input to the transmitter with a resistor termination. If the noise indication drops to a very low value, then the noise is in the audio system, but if it does not, then the noise is in the rf system. If the noise is in the audio system, then move the termination to the input of each unit in the chain, observing the noise reading each time. As soon as the offending unit is passed, the noise will rise to the original value. Some maintenance at that point is called for.

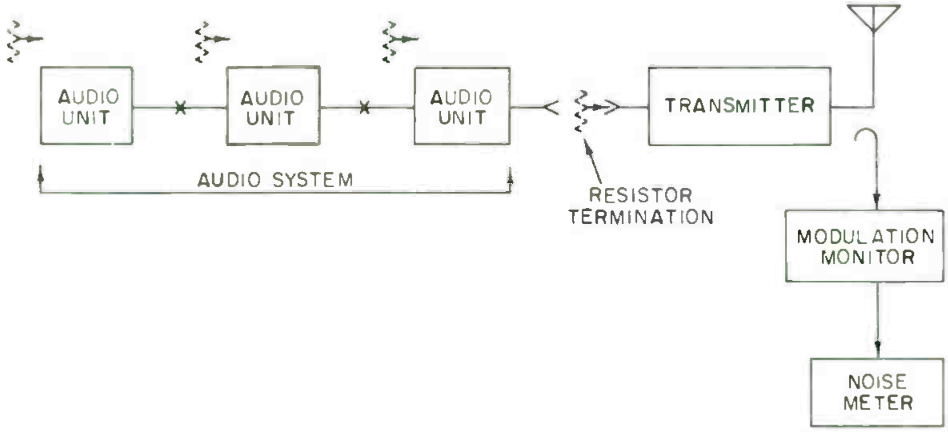

Figure 2. As an isolation technique, terminate input to each unit in the audio system, starting at the transmitter input.

When the offending unit is passed, the noise reading will rise to the original value.

DISTORTION

The Proof requires that total harmonic distortion (thd) measurements be made at many audio frequencies. So, make three or four spot checks for distortion, spread out across the audio bandpass. If these measurements are high (assuming you have eliminated noise as a factor), either use the oscilloscope to observe the waveform out of the signal generator or use the distortion analyzer and measure the generator itself. Some component or other fault may have occurred in the generator so that its output is actually distorted. The system cannot measure better than the test equipment, so if there is a fault, correct it before going on.

Improper signal levels throughout the system are a very common source of distortion. Stages may be overloaded so that the signal is clipped or may be driven into non-linear operation, and both these conditions will produce distortion. During regular programming, an emergency situation may have occurred that caused signal levels to be misadjusted in the system. When the emergency is over, the level may be forgotten. Should distortion be high, check the proper setting of levels throughout the system. This is one of your preliminaries to the measurements anyway, so it can be done now.

If the levels are adjusted correctly and the distortion persists, you need to isolate the offending unit. Begin by moving the distortion analyzer to the *output* of the audio system, and measure distortion here. This will indicate whether it is an audio problem or an rf problem. If the distortion is in the audio, use the same techniques as employed for isolating noise, but in this case, work with signal. Patch the signal generator to the input of each unit in the chain as you work towards the analyzer. Remember to match correctly each time and of course, use the correct signal level.

Now if the distortion proved to be in the rf section of the system, then some different techniques are required. First, measure the audio signal level at the input of the transmitter. Compare this with previous maintenance measurements or a previous Proof. If the required input is now high, for example +13 dB instead of the normal +10 dB, it can be that the monitor is out of calibration and the transmitter is seriously overmodulated. Check the actual deviation of the f.m. carrier by use of a communications receiver and listen for the carrier nulls. This will determine if the monitor is in or out of calibration. If this is a stereo situation, then connect the composite output of the stereo generator directly to the monitor and check it that way. Use the oscilloscope and check the stereo generator itself, which may have problems or be out of adjustment. If the monitor and stereo generator appear to be normal, then check out the bias on the vvc diodes in the modulator or the diodes themselves.

FREQUENCY RESPONSE

The other major area of measurements concerns the system's audio

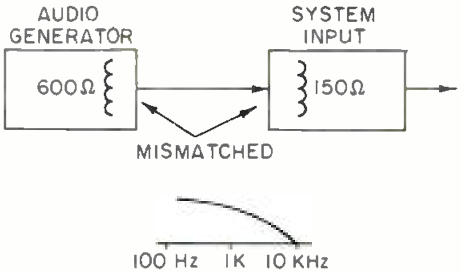

Figure 3. The first place to look for an impedance mismatch is at the generator interface to the system.

bandpass. When making the spot checks for distortion, response measurements should be made at the same time. The f.m. transmitter is required to have a 75 μsec. pre-emphasis that enters into this response curve. But these are usually passive elements which seldom fail. Aside from component failures in some unit, poor high frequency response is very often caused by improper impedance matching somewhere in the lineup.

The first place to look for mismatch of impedances is at the signal generator connection to the system. The generator may be set for a 600 ohm output, but the microphone input will be in the neighborhood of 150 ohms. And if during the noise tests it was

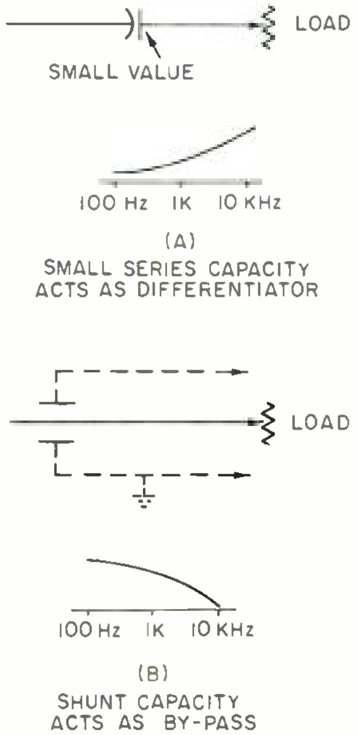

Figure 4. Different effects of capacity on the response curve.

necessay to insert an isolation transformer, this transformer may have the wrong impedance taps or it may not have a good enough bandpass of its own. A few spot checks at the output of the transformer will soon prove this out.

Some amplifiers use a "T" input or output control but the system is balanced. When system levels are such that this type of control must operate at one of its extreme ends, expect problems with the response of the upper audio band. Reset such controls to mid-range, and if necessary, add fixed pads or readjust system levels. These pads are intended for unbalanced circuits, but broadcast circuits are balanced.

A potential problem source can be in modifications that have been added to reduce or eliminate rf interference. The one who made the modifications may have been so elated to be rid of the interference, he forgot to check what effect it may have had on the audio. The capacitors may also be bypassing the high audio frequencies.

In general, when there are problems with the low end of the audio band, look for coupling capacitors that may have opened or dried up and now present only a small series capacity to the audio. This will act as a differentiator circuit. A dirty jack can behave in this manner. For problems in the upper audio band, look for capacity across the circuit or to ground. This will by-pass the upper audio frequencies. This does not have

Rock and roll is in it's third decade and there are mountains of blown diaphragms and discarded speaker systems as evidence of the difficulties loudspeaker manufacturers have had in meeting the challenge. The SP1 was designed and tooled by a new loudspeaker company dedicated to solving the basic difficulties of high level sound reinforcement in order to meet that challenge.

Our two-way is a compact, powerful, reliable, high fidelity loudspeaker with dispersion and power response so uniform that the "sound" of the system is stable in different environments. The SP1's multi-flare radial horn is the most significant advance in the control of high frequency dispersion since the invention of the radial horn half a century ago. Undoubtedly the design will become the industry standard.

The real marvel of the SP1 system, however, is the Model 22 Compression Driver. We have been producing it since August 1975 demonstrating that it is possible to combine adequate high end response (13 KHz), efficiency (30% midband), high power handling (40 watts pink noise 8 hours continous), reliability (6 forms of on-line analysis plus listening), and good sound in a compression driver. Until "22" a high performance 2-way like the SP1 was not possible.

The low-end of the system is provided by a 15" horn loaded cone speaker covering the range of 60 - 500 Hz. The extensive Q.C. system devised for the driver has been expanded to allow the same scrutiny of the SP1.

Some strong statements have been made here, but we know we can deliver. The demand is so great for a system with the SP1's performance that our dealers have ordered over 1,200 units (as of Dec. 1,'75) based on word of mouth from the few people who have heard samples from pilot production.

The SP1.

AN ALTERNATIVE TO THE ESTABLISHED WAY

$499.50* Soon at your Peavey Electronics Dealer.

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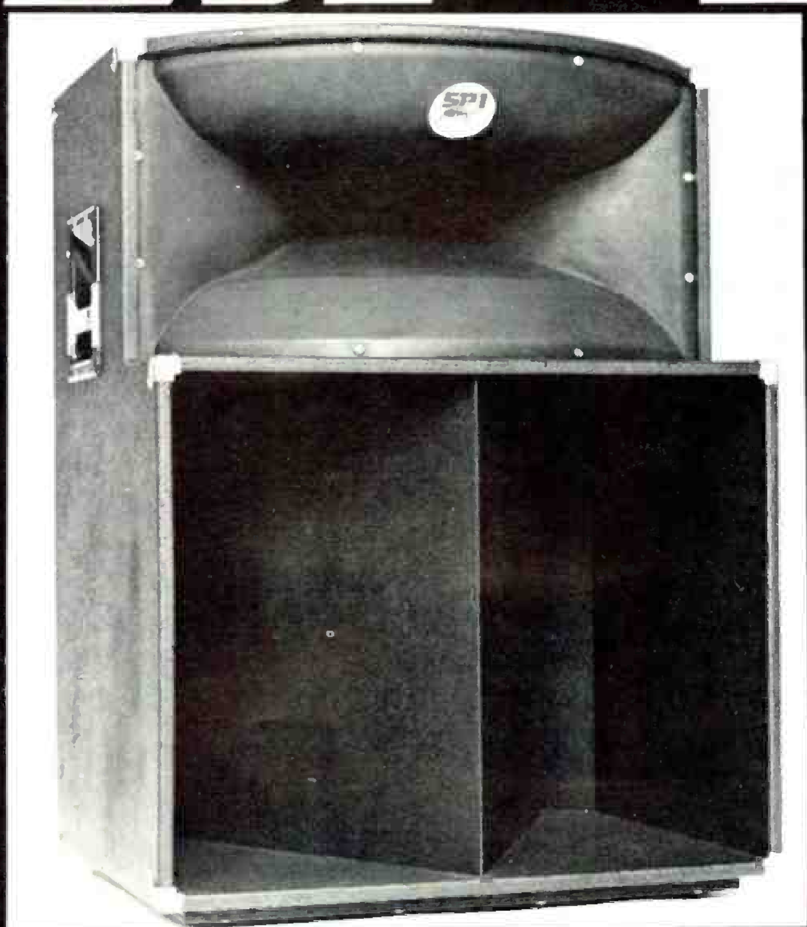

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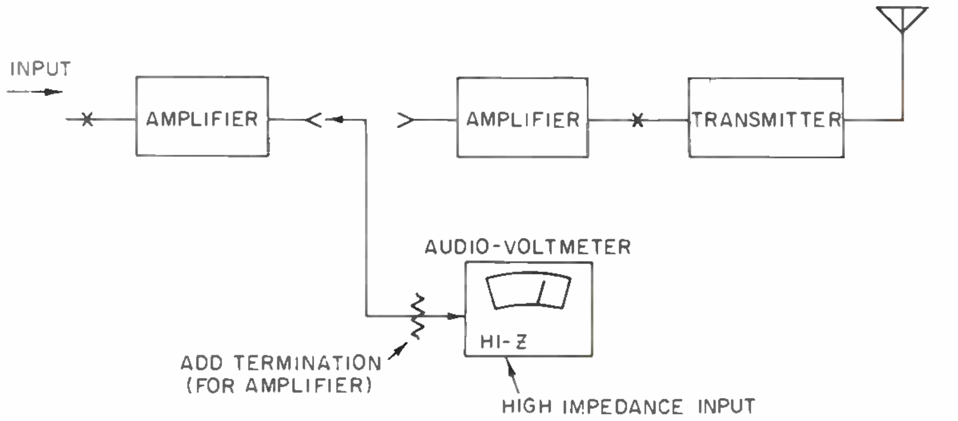

Figure 5. The audio voltmeter has a high impedance input. When measuring levels in the audio chain to isolate a response problem, add a termination to the voltmeter input so as to load the amplifier properly.

to be a discrete capacitor across the circuit, but it can be *anything* which acts as a capacity.

Response problems can be isolated with the same techniques used for the other problems, but be sure to use correct impedances. You may work "up" to the voltmeter, or the voltmeter "down" to the generator. When moving the voltmeter, remember this has a high impedance input and the amplifier must be terminated. So, add a terminating resistor across the voltmeter input.

SUMMARY

A station cannot "fail" a Proof. If the system can't meet specs, then the problems must be corrected. A Proof attests to the fact the station *does* meet specs. Before making the many required measurements, the engineer should make a few spot checks of the system. This will determine if the measurements will fall into place or that maintenance is required before making the measurements. A few simple isolating techniques will help in finding the offending unit. ■

DIGITAL DELAY LINE

• Random access memories instead of shift registers are used in model 1745M digital delay line. Two independent outputs are offered, each providing up to 320 milliseconds of delay in 20 microsecond steps. Delay is read out in milliseconds on solid-state digital readouts, one for each output. The front panel includes switches for instant zeroing of all delays, plus switches for audio recirculation and delay doubling (up to 640 milliseconds in 40 microsecond steps at one-half frequency response). An input level control and optimum level indicator are included. Up to three additional outputs are available.
Mfr: Eventide Clockworks, Inc.
Circle 50 on Reader Service Card

CIRCUM-AURAL HEADPHONE

• Circum-aural headphone K-240, weighing only 298 grams (10.5 oz.) contains a combination of a main driving transducer, and six passive slave diaphragms in each ear piece. 45 degrees apart, three slaves are situated forward of the mid-line of the transducer with three slaves to the rear on the main coupling wall. Acoustical resistance presented by the slaves provides an effect of "open listening" at frequencies above 200 Hz. Below 200 Hz, the compliance of the passive diaphragms acts as a closed wall. The interaction between the passive diaphram membranes and air volume also produces diffraction effects for a simulation of spatial hearing. The unit is fitted with a low-pressure auto-adjust dual headband and soft circum-aural ear pads.
Mfr: Philips Audio Video Systems
Price: $69.50.
Circle 51 on Reader Service Card

CONDENSER MICROPHONES

• Phantom-powered MKH series condenser microphones contain rf-biased transducer elements. They operate with common 48 V phantom powering systems where the positive supply voltage with reference to ground (pin 1) is supplied through both audio lines (pins 2 and 3 of the mic's XLR connectors). Models in the series include: MKH 416, frequency response 40 to 20,000 Hz, narrow supercardioid/club-shaped directional pattern. MKH 406, cardioid, overload level up to 132 dB SPL for close miking; MKH 816 distance shotgun mic with confined club-shaped pickup pattern, frequency response, 50 to 20,000 Hz, can be used hand-held or on a boom. Central phantom power adaptation is available with separate dual power supply, Model MZN, 16 P 48 U.
Mrs: Sennheiser Electronic Corp.
Prices: MKH 406: $495. MKH 816: $629. MZN: $176.
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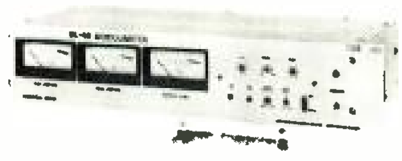
A.M. BROADCAST LIMITER



• Tailoring a modulation envelope to any program format or transmitter characteristics is possible with automatic Modulimiter BL-40. Independent adjustments are provided for rms and peak limiting and for variable positive overmodulation up to 125 per cent. Unwanted overmodulation without peak clipping is claimed for the fast f.e.t. peak limiting section. A phase optimizer maintains most favorable signal polarity, automatically and silently reversing polarity to maintain maximum upward modulation. The unit features integrated circuitry. Adjustments are located behind a removable security panel.
Mfr: U.R.E.I.
Circle 53 on Reader Service Card

AUDIO INTERFERENCE FILTER

• Self-contained LINX-60L removes power line hum and buzz from audio systems through automatic interference tracking in the tape. The digital comb notch filter removes objectionable fundamental and harmonic frequencies of the a.c. line through the action of −55 dB equally-spaced narrow bandwith (1 Hz) notches combing the input. The attenuation is locked to the interfering signal fundamental, automatically holding any drifting interference at the center of the notch. The LINX will track any interference from 40 to 80 Hz and provide up to 250 separate distinct harmonic notches.
Mfr: Xetron Corp.
Circle 54 on Reader Service Card

TELECOMMUNICATIONS INDICATOR

● Bright red led indications with a 180 degree viewing arc emanate from a solid-state lamp in the Tel-Led indicator, series LT-70. The indicators fit all standard telephone lamp jacks which accommodate T-2 slide base lamps. Features include solid-state construction, low power consumption, low heat generation, and shock protection. The units operate on 48 Vdc, are available color-coded in red, black, green, white, or yellow bezel colors.
Mfr: Switchcraft, Inc.
Price: $1.44 (1,000 lot quantities)
Circle 55 on Reader Service Card

FULL RANGE SPEAKER

● Full frequency range speaker system with five drivers, model PRO 100, is designed for floor placement. A 15 in. high compliance acoustic suspension type woofer with aluminum voice coil and crossover at 700 Hz provides bass drive in a 3.4 cu. ft. sealed enclosure. Two 4½ in. cone midranges and two 1 in. dome tweeters cover the upper frequencies with crossover at 3,500 Hz. One midrange/tweeter pair faces forward and the other upward. Frequency response is ± 4 dB from y0 Hz to 20 kHz. A controlled impedance feature maintains the load impedance presented to the amplifier at not less than 4 ohms at any audio frequency. Power handling capability is 125 watts program material. A three-position control sets the balance between the forward facing and upward facing drivers to adjust to room variables.
Mfr: H. H. Scott, Inc.
Price: $350.
Circle 56 on Reader Service Card

DYNAMIC HEADPHONES

● Two new headphones provide facilities for special usage. DT100.1 leaves one ear open for monitoring ambient noise. DT108 is similar, but has a boom microphone, highly directional, and featuring exceptionally wide frequency response. Both models are completely modular.
Mfr: Beyer Dynamic (Revox Corp.)
Price: DT100.1: $45.
DT108: $63.50.
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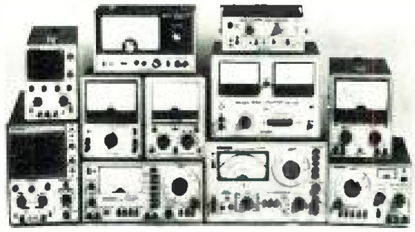
12-INSTRUMENT TEST CENTER



● Instruments needed for servicing audio equipment are grouped conveniently in the Audio Test Center. The complement contains two oscilloscopes, model LBO-552, a 5 in. solid state dual channel/dual trace oscilloscope/vectorscope which provides simultaneous left/right waveform display and a 20mVp-p/cm sensitivity, and 3 in. model LBO-310A, offering a 4MHz bandwidth and a.c./d.c. coupled vertical and horizontal inputs. In addition, the package contains a 3-in-1 wideband audio analyzer, the LAV-190, which acts as a generator, attenuator and a.c. millivoltmeter. A companion instrument is the LDM-170 distortion meter which measures a signal-to-noise ratio and signal levels in all circuits, measuring total harmonic distortion as low as 0.01 per cent. Then there are four generator-type instruments, including LSG-231 multiplex/f.m. stereo generator and LAG-125 low distortion generator which offers sharp square waves. The third generator, LAG-120, is a sine/square wave unit. LAG-26 generator tests transient response.
Mfr: Leader Instruments Corp.
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There's never been a wider range studio monitor. Of <u>any</u> size.

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Ampex ATR-100. World's

Ampex introduces an audio recorder/reproducer that's so much better than anything else on the market, you'll find it impossible to make a comparison.

Performance specifications for the ATR-100 are significantly better than you have ever seen for a commercially available tape recorder.

Used with today's top-of-the-line mastering tapes, the ATR-100 will give you better recording performance and better playback results than you thought possible. Used with tomorrow's higher coercivity tapes, the ATR-100 will achieve new levels in the capture of transients and the recording of high intensity sound passages against a background of normal-level material.

There's never been an audio recorder like the ATR-100. It handles tape better than any previous machine, and it processes audio signals more faithfully than anybody else's best recorder.

Transport and Heads: All New Tape Handling

The ATR-100 has a triple-servo transport. Both reel motors and the capstan are servo-controlled in a tightly coupled loop. This servo arrangement does away with pinch rollers forever, and moves tape positively in both directions.

The transport is heavy cast aluminum, and the reel motors are strong enough to bring the tape up to the flutter spec in 0.5 second at 30 in/s. And because all elements are servoed, cueing is accomplished by twirling a knurled knob on top of the capstan. Movement is sure, positioning is positive.

Best Audio Recorder.

The ferrite heads are new, too, and will last a lot longer than any we've ever used before. Flux-gate head design gives unbelievable response; at 15 in/s, the ATR-100 will deliver ± 3/4 dB from 100 Hz to 15 kHz. You'll see "flatter" curves than ever before.

Precision machining of the heads means that you can change or replace them without going through a mechanical alignment routine. A limited azimuth adjustment is all that's needed. A single screw is also the key to changing heads—loosen it, unplug the head assembly and that's all there is to it. You'll take care of your ATR-100 with a minimum of simple tools.

Another tape handling improvement is provided by ceramic tape guides. These edge guides resist wear by fast-moving tape and team up with the hard ferrite heads for more hours of use with minimum maintenance.

Electronics: New Ideas Throughout

The basic ATR-100 contains only audio and transport electronics. There's an unbalanced input and output circuit, and that's all. No meters or operator adjustments.

Normal balanced input and output circuitry is contained in the optional channel input/output units. You use one, two, or four I/O channels (located in an overhead bay) with an ATR-100, giving you your choice of full track, stereo or four-track recording.

Each I/O module has its own metering, line drivers, balancing transformers, amplifiers and controls. We've provided the ability to switch meter circuits, giving you a choice of ASA ballistics with VU indication or the European EBU ballistics with Peak indication.

Controls and Special Features: A New Concept

Transport controls have international symbols, and no restrictions on sequencing. You can go from any transport mode to any other mode without going through stop, and without waiting for anything to slow down or speed up. Furthermore, the ATR-100 has dynamic braking that takes over in the event of a power failure. This machine always stops in a programmed manner, even without power.

You won't stretch tape with ATR-100, because it always checks to make sure the tape is fully tensioned before actuating the servos.

The power supply is heavy duty and universal. No matter where you take the machine, you'll be able to plug it in the wall.

All record, playback and transport controls are located on a compact, keyboard control panel with LED indicators. It looks somewhat like a small calculator and stows on either side of the machine. Pick up record capability (PURC) is standard equipment.

Speeds: Select Any Two

Every ATR-100 is capable of the four standard recording speeds—3-3/4, 7-1/2, 15 and 30 inches per second. You can select any pair, adjacent or not, and change them later if you wish.

And no matter what speeds you select, you'll get a rewind and fast forward capability of 2,400 feet in just 60 seconds. That's really moving tape.

Specifications: The Ultimate Measure

Complete, detailed performance specifications are contained in our new ATR-100 brochure. Call any Ampex sales office or your nearest Ampex Distributor. You'll see why we say that this is the finest audio recorder/reproducer ever offered for sale.

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Circle 36 on Reader Service Card

JUERGEN WERMUTH

Compander Increases Dynamic Range

A new device, using four independent control systems, eliminates the need for alignment, controls precisely, and increases the dynamic range of equipment.

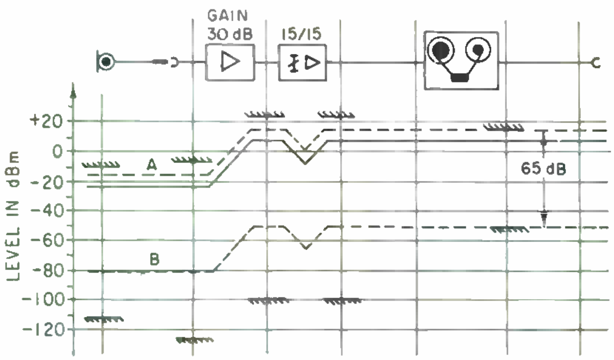


Figure 1. Level diagram of a single channel.

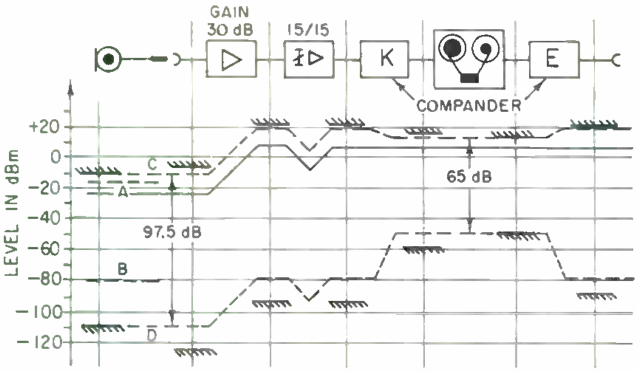


Figure 2. Level diagram with use of the compander developed by Telefunken.

Juergen Wermuth is an engineer with the German Telefunken firm. Stephen Temmer, president of Gotham Audio Corp., N.Y.C. translated the article from the German, and contributed additional material.

THE IDEA OF a compander system to suppress noise goes back some 35 years. Few still remember the Fairchild expander, which was interposed between power amp and loudspeaker and which operated passively on the resistance characteristic of the filament of a pilot lamp. The EMT *Noisex* system of the early sixties was also based on this idea, albeit somewhat refined. But it wasn't until Dolby developed the 301 system almost ten years ago that companding reached the stage where it could stand reasonable objective and listening tests. Much of the credit should go to English Decca, who encouraged Ray Dolby to abandon his quest for a video signal compander and to shift his attention to audio. The rest is history—over 25,000 channels of his professional system are in use today around the world—both western and eastern.

In recent years, another compander has appeared, the dbx system, and in spite of predictions that "the world has made its mind up to go Dolby," it appears to attract an ever increasing number of admirers. If all of this proves anything, it indicates that there is no system or device so perfect that it will last forever. Technology marches ever onward, and the elusive crest of the hill of achievement appears unattainable.

Mr. Wermuth is convinced that the improved system here described could well have been developed by Dolby, if in fact he did not actually have such a design in his desk drawer. But could Dolby ever market it? The answer is likely *no*, simply because all of those thousands of Dolby equipment owners would have done him in had he, by issuing such a new system, instantaneously obsolesced all the equipment he had sold to them.

It is therefore not surprising that it fell to someone else in the field to pick up the torch and to carry it forward. Telefunken has had a long tradition of torch carrying. I rather suspect their design has genuine merit. S. F. T.

In recent years, there has been an increase in the demand for systems designed to improve the dynamic range in professional studios. The signal-to-noise ratio of critical equipment, such as tape recorders, is significantly worse than that of the rest of the system. Obviously, the weakest link determines the quality of the entire chain, leaving out the acoustic weakness of the loudspeakers in this analysis.

Let us examine solely the dynamic range for the mo-

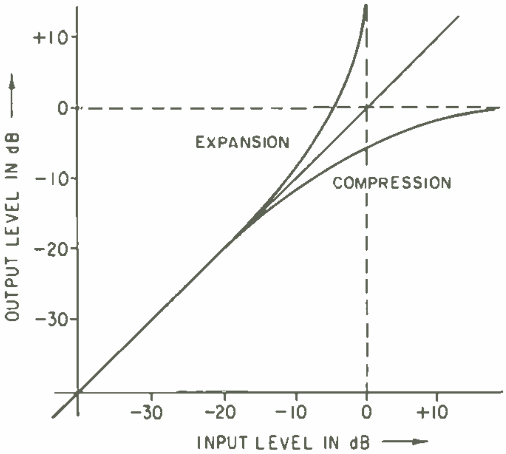


Figure 3. Transfer curves of an older compander system.

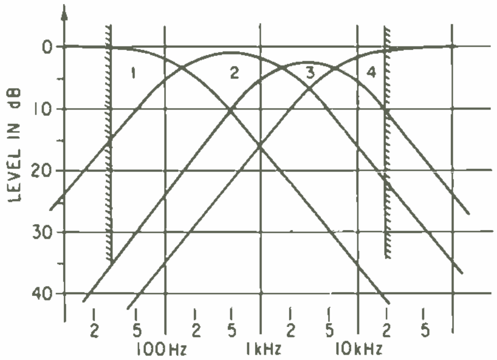


Figure 5. Frequency band splitting used in the Telefunken compander.

ment. FIGURE 1 shows the level diagram of a studio channel. The narrowing of the dynamic range resulting from the tape machine is readily apparent. Present day condenser microphones have a usable dynamic range of about 100 dB (25 dB(A) to 125 dB(A). This value can be managed by modern mixing consoles. Studio tape recorders allow a maximum dynamic range of about 65 dB, and we, therefore, need an improvement of about 35 dB to remove this restriction on the dynamic range.

For physical-technical reasons, only minor improvements can be expected in the devices themselves. Therefore, engineers have been making use of supplemental compander systems which provide only 10-15 dB of dynamic range improvement and therefore do not fully restore these narrowed dynamics. The system developed by Telefunken permits transmission of virtually the entire dynamic range without any annoying side effects (FIGURE 2).

TRADITIONAL COMPANDERS

To permit an evaluation of the reasoning behind this system, it is useful to discuss the functioning of traditional companders. Companders operate, in contrast to noise filters, in a complementary fashion: the information to be

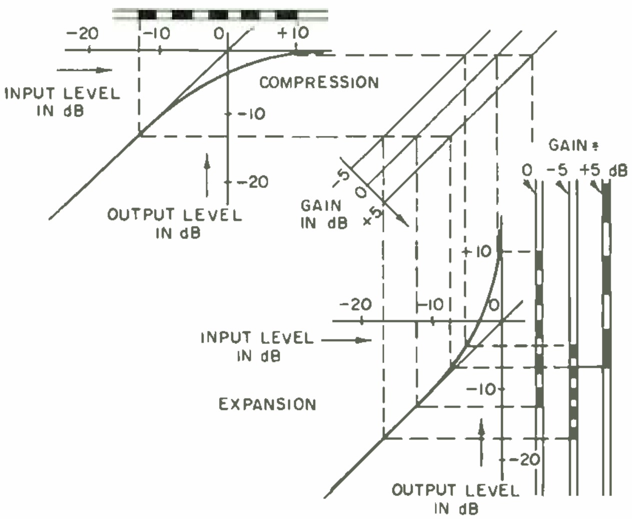


Figure 4. Presentation of the dynamic relations of an older system when levels are misaligned.

transmitted undergoes a process at the beginning of the system chain (for example, prior to recording), which is reversed by a contrary process at the end of the system chain (for example after playback).

Interference signals such as tape noise, which enter the chain are only exposed to the reversal process and are therefore "disadvantaged" *vis a vis* the modulation signal, in other words, attenuated. A familiar example of such a complementary system is the pre-emphasis/de-emphasis of high frequencies, as used in disc recording and f.m. broadcasting. This sort of device is very limited for use in magnetic recording.

Traditional companders generally operate by using a level-dependent output control system, which is automated with the help of control stages. The control transfer curves are designed to produce a compression at the beginning of

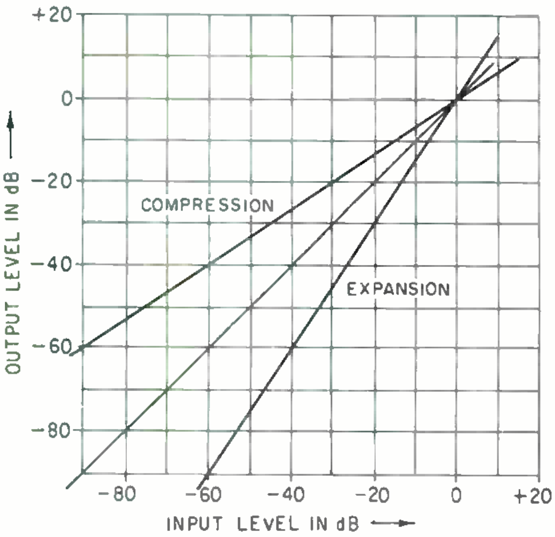


Figure 6. Transfer characteristics of the compander.

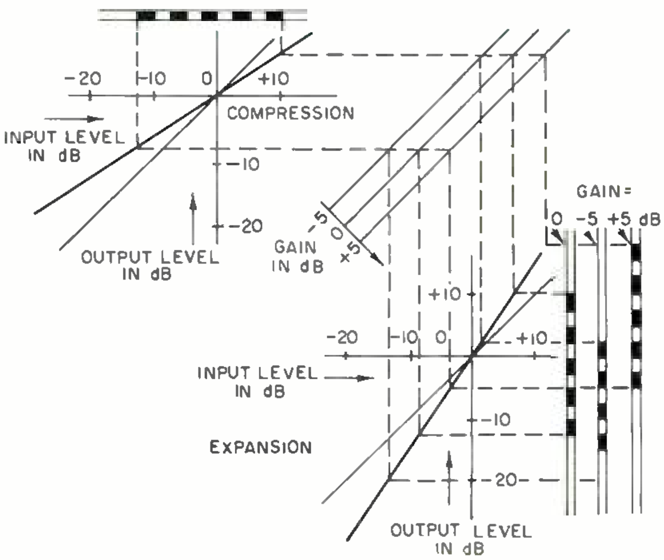


Figure 7. Presentation of the dynamic relationships of the Telefunken compander.

the system (corresponding to a flattening of the transfer curve, which is compensated for by an expansion at its end (corresponding to a steepening of the transfer curve). (See FIGURE 3). The s/n gain depends on the magnitude of the compression and expansion ratio.

Practice has shown that this kind of system performs rather imperfectly. One of the reasons for this is the fact that the compressor only processes program peaks after they have occurred. Since the control circuit is saddled with a time constant, the ensuing device will be overloaded for an instant because of the compression ratio factor. This time constant, aside from technical considerations, cannot be arbitrarily decreased due to the fact that the compander must react "lethargically" to the lowest frequencies to be processed.

There is another negative factor attached to existing companders. The noise "breathes" in synchronization with the output level because it is not treated in a complementary fashion. This effect becomes especially noticeable if the frequency spectra of the noise and modulation are highly different. For example, very peaky high-frequency tape noise will breathe noticeably if the compander is fed a dull, low frequency modulation signal.

Most compander functions are sensitive to level misalignment within the system and serious dynamic distortion can result, due to their curved transfer characteristics, if the system is not accurately aligned both during recording and playback because the compression and expansion characteristics are not exactly complementary. (FIGURE 4.)

COMMON ERRORS OF COMPANDERS

To sum up, these are the errors from which older companders (1-5) and new ones (1-3) suffer.

1. Faulty complementary behavior and maximum control function at peak modulation audibly falsify the program dynamics when they occur simultaneously.
2. Excessively long attack time and short release time result in higher distortion percentages.
3. Dynamic distortion is unavoidable in a misaligned system.
4. Application of only a single control system for the entire band width results in the "breathing" of noise.
5. The modulation noise of the tape itself is not suppressed.

A NEW TYPE OF COMPANDER

The system here described uses new methods to eliminate the disadvantages listed above. The elimination of the need for alignment, which requires test time, was placed high on the list of priorities. The Telefunken compander uses four independent control systems (FIGURE 5.) Each of these is associated with a particular frequency band. This results in the suppression of the tape noise's breathing, as well as the tape's own modulation noise.

The control circuit time constant problem is best stated as follows: the attack time has to be so short that it prevents overload of the following device for the highest step function level increase, even at the highest frequencies.

This means, in other words, that the attack time may not exceed one quarter wave of the upper boundary frequency.

Restated in numbers, this means that it must be shorter than 12.5, assuming 20 kHz. The release time is determined by the dynamic distortion factor which results from the logarithmic amplitude change of a sine wave. Assuming a 0.2 per cent distortion factor and a low boundary frequency of 30 Hz, the gain may not increase by more than about 6 dB per second. For a 30 dB control, we therefore obtain a release time of about five seconds!

An examination of the very great ratio of attack-to-release time of 500,000:1 shows that older systems must, of necessity, utilize compromise solutions to this problem. The separation into four frequency bands, each of which processes only a part of the spectrum, brings with it a reduction in the time ratio by a factor 100 to about 5,000:1. This is an order of magnitude which is readily realized in practice.

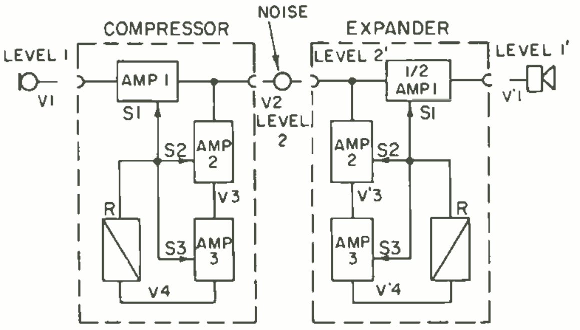


Figure 8. Block schematic of the compressor and expander system.

The static transfer curves of all four systems have a constant slope over their full operating range (FIGURE 6). The slope is 33 per cent in the logarithmic scale and independent of level. An electro-acoustic signal with a dynamic range of 90 dB is reduced by the compressor by 33 per cent to 60 dB and is, as a result, 30 dB less sensitive to noise. The expander restores the 90 dB range at the end of the transmission chain.

The compander is designed for a dynamic range of 110 dB apportioned by the alignment level into 94 dB above self-noise and 16 dB overload reserve. At the alignment level of +4 dBm, the gain of both the compressor and expander is unity at all frequencies. Magnetic tape recorders have a working range of about −57 dB to +8 dB, relative to alignment level. This compander system appears to shift this to −85.5 dB to +12 dB, relative to alignment level. In other words, the dynamic range has increased from 65 dB to 97.5 dB!

The apparently unused area between the lower boundary value (−85.5 dB) and the self-noise level (−94 dB) has the following reasoning; it serves to retrieve the dynamic range loss which would result from a gain displacement within the transmission chain. Aside from that, it assures that the theoretically attainable value of −85.5 dB, referred to the tape recorder alignment level, will not be deteriorated by the self noise-level of the compander.

OBTAINING A COMPRESSION CURVE

The following basic method is used to obtain the compression curve with its 33 per cent slope over such a wide range. Three identical control amplifiers (V_1 to V_3 in FIGURE 8) are connected in series with their control inputs

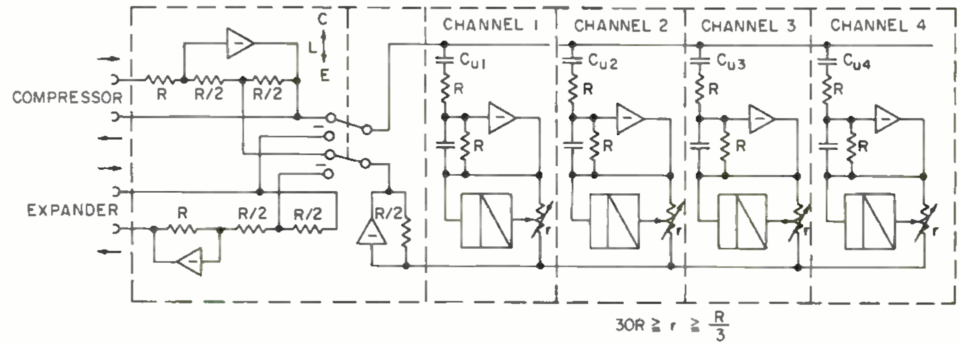


Figure 10. Basic schematic of the new Telefunken compander.

(S_1 to S_3) connected in a parallel. A control system assures that the output of the last amplifier (V_3) remains constant regardless of the input signal (E_1) of the first amplifiers. The following equations will show how this circuit produces 33 per cent logarithmic compression.

Proper dimensioning of the circuit components produces

$$V_1 = V_2 = V_3 = v$$

$$\frac{E_2}{E_1} = \frac{E_3}{E_2} = \frac{E_4}{E_3} = v; \quad \frac{E_4}{E_1} = v^3; \quad \frac{E_4}{E_2} = v^2$$

$$\Delta P_1 = 20 \log \frac{E_4}{E_1} = 20 \log v^2 = 40 \log v$$

$$\frac{\Delta P_2}{\Delta P_1} = \frac{40 \log v}{60 \log v} = \frac{2}{3}; \quad \Delta P_2 = \frac{2}{3}\,\Delta P_1$$

where
v = gain
P = level

The dynamic range of the level after the first amplifier (V_1) is compressed to 2/3 of the range at its input, or by 33 percent in the logarithmic scale.

This method has the advantage of operating very precisely over its entire range due to the high degree of feedback in the control system. Furthermore, the control range of each of the three amplifiers is reduced to one-third of the input dynamic range, simplifying the design of such circuitry. The retrieval of the original dynamic range in the expander is obtained in a complementary way.

The level (E), which is held constant by the control system, passes through a weighting network and is processed as a peak value. The weighting network (FIGURE 9) has an 18 dB/octave filter steepness. This assures that, for example, a bass note will not cause breathing tape noise.

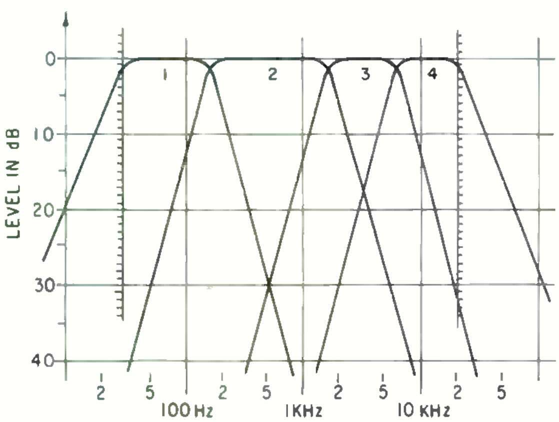


Figure 9. Weighting filter characteristics.

For the control function, the Telefunken system uses electronically stabilized control parameter f.e.t.s. This results in a complementary dynamic behavior which, even with fluctuations in alignment level, remains a constant for the expander function. A possible version of this expander is shown simplified in FIGURE 10.

APPLICATION OF THE COMPANDER

The Telefunken system is applicable to all systems where sound transmission demands are subject to dynamic range restriction. A partial list of these might include magnetic recorders, echo or delay units, whether of the electronic or magnetic type, transmission lines with hum or click interference, optical film track, and not least, multi-track recording and mixdown in the production of phonograph records. It is also applicable in multiple dubbing of magnetic recordings, as well as in broadcast links looking toward improved dynamic range.

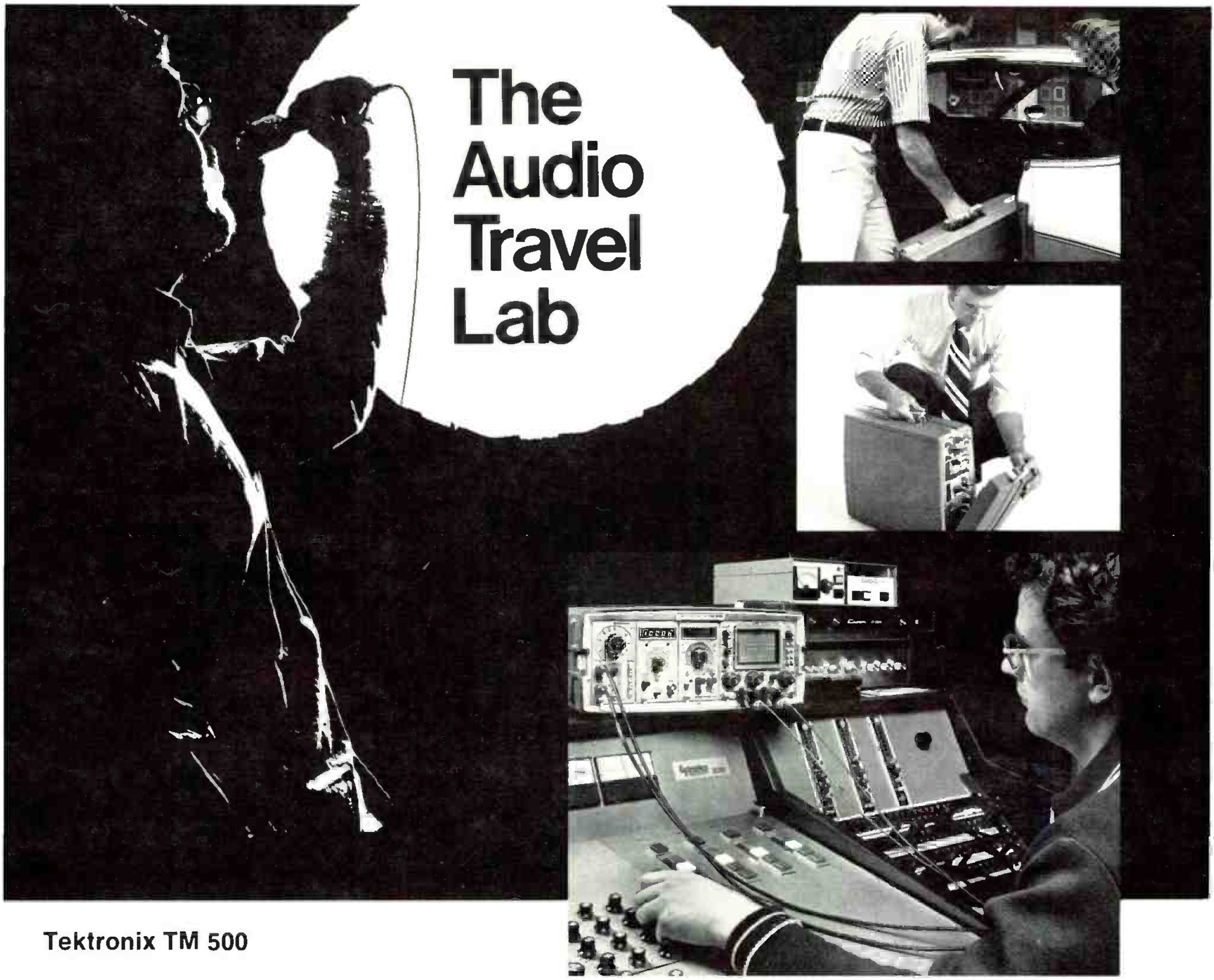
The following example illustrates the effectiveness of this compander. It is a well known fact that a signal which is passed through a device, the self-noise of which equals the s/n ratio of the signal, is degraded by 3 dB. Passing the signal through nine such devices results in a degradation of 10 dB. Using a compander as described here would theoretically permit such a signal to pass through *one thousand* such devices without any degradation of the original dynamic range.

TABLE OF PROVISIONAL TECHNICAL DATA

Frequency range	30 Hz to 20 kHz
Control systems	4 (log characteristics)
Frequency bands	4 (30-160/160-1600-5000/5000-20,000 Hz)
Compression slope	33 per cent (log)
Nominal level	+4 dB (vu meter based)
Input level range	−88 dBm to +22 dBm
Dynamic gain	30 dB (for professional recorders
Overload reserve	≧18 dB
S/n ratio (wtd & unwtd)	≦0.2 per cent
Tot. harm. distortion	≧94 dB

More realistic is the use of a 16-track recording as an example. In a mixdown of the 16-tracks, the dynamic degradation is somewhat greater than 12 dB, compared to a single track. Application of the compander improves the individual track dynamics by 30 dB. Therefore, the deterioration of 12 dB resulting from a mixdown is not only eliminated, but an additional 18 dB is obtained!

From these observations, it is easy to deduce that the application of this compander system in the multi-track studio results in a significant easing of the mixer's work load. Level control during recording no longer requires the traditional accuracy, aside from the additional gain in dynamic range. ■



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THE INVERSE SQUARE LAW only determines the intensity of sounds that travel in a direct line from a source to the point of observation. If any reflecting or absorbing surfaces are present, reflected sound combines with the direct sound to produce a total intensity dependent on other factors.

When a complex wave strikes a surface, a percentage of its energy will be absorbed and some will be reflected. The percentage of absorption is known as the *absorption coefficient*. This coefficient will vary with the frequency, the nature of the material, and the angle at which the sound wave strikes the surface of the material. Absorption coefficient tables for materials as shown in TABLE 1 are normally listed, indicating the coefficients of a material at the following frequencies: 125, 250, 500, 1,000, 2,000, and 4,000 hertz.

You will note in TABLE 1 that the coefficients are labeled *average* coefficients. This represents an average of the coefficients for all angles at which a sound wave may strike the surface. The coefficients given are for one square foot of material.

Referring to TABLE 1 again, we see that concrete block, coarse, has an absorption coefficient of 0.36 at 125 Hz. If we had a concrete block wall 50 feet long and 20 feet high, the absorption coefficient for the wall would be 50 feet times 20 feet times the coefficient 0.36 or 360. Therefore, our absorption coefficient for this wall at 125 Hz would be 360.

Had we calculated the absorption coefficient of this wall at 4,000 Hz, we would have 50 feet times 20 feet times 0.25 or 250. It is therefore apparent that this particular wall would absorb low frequencies more effectively than high frequencies.

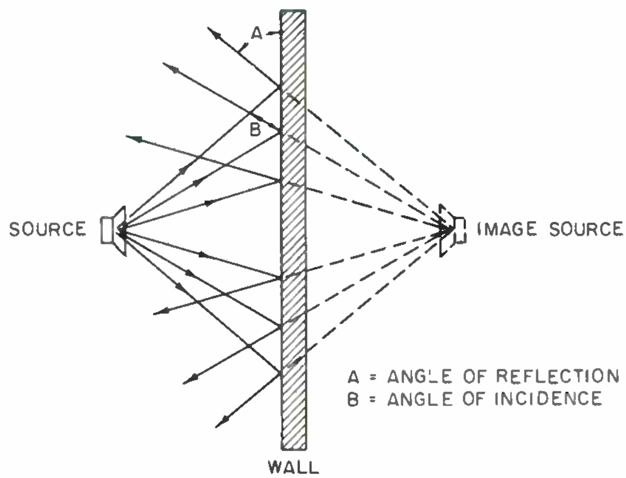

Figure 1. Sound reflections from a plane surface.

DIFFERENCES IN ABSORPTION CHARACTERISTICS

The efficiency of almost all sound absorbing materials depends on the porous structure that they maintain. The air inside the pores is set into motion by the sound waves striking it. This motion against the walls of the pores creates frictional heat energy, which is absorbed by the material. As this absorption only represents a percentage of the total striking force, the remaining sound energy is returned to the area as a reflected sound wave.

The absorption capability depends on material thickness, pore size, ratio of pore volume to overall volume, and frequency of the sound wave. To qualify this statement, a porous material must be capable of generating friction heat. The thickness plays a larger role in absorbing lower frequencies and the degree of porosity plays

Ronald Braho is associated with the Dukane Corp., of St. Charles. Ill. This article is a chapter of a book written by Mr. Braho.

TABLE 1. SOUND ABSORPTION COEFFICIENTS OF GENERAL BUILDING MATERIALS & FURNISHINGS

Materials	*Coefficients* 125 Hz	250 Hz	500 Hz	1000 Hz	2000 Hz	4000 Hz
Brick, unglazed	0.03	0.03	0.03	0.04	0.05	0.07
Brick, unglazed, painted	0.01	0.01	0.02	0.02	0.02	0.03
Carpet, heavy, on concrete	0.02	0.06	0.14	0.37	0.60	0.65
Same, on 40 oz. hairfelt or foam rubber	0.08	0.24	0.57	0.69	0.71	0.73
Same, with impermeable latex backing on 40 oz. hairfelt or foam rubber	0.08	0.27	0.39	0.34	0.48	0.63
Concrete block, coarse	0.36	0.44	0.31	0.29	0.39	0.25
Concrete block, painted	0.10	0.05	0.06	0.07	0.09	0.08
Fabrics						
Light velour, 10 oz. per sq. yd., hung straight, in contact with wall	0.03	0.04	0.11	0.17	0.24	0.35
Medium velour, 10 oz. per sq. yd., draped to half area	0.07	0.31	0.49	0.75	0.70	0.60
Heavy velour, 18 oz. per sq. yd., draped to half area	0.14	0.35	0.55	0.72	0.70	0.65
Floors						
Concrete or terrazzo	0.01	0.01	0.015	0.02	0.02	0.02
Linoleum, asphalt, rubber or cork tile on concrete	0.02	0.03	0.03	0.03	0.03	0.02
Wood	0.15	0.11	0.10	0.07	0.06	0.07
Wood parquet in asphalt on concrete	0.04	0.04	0.07	0.06	0.06	0.07
Glass						
Large panes of heavy plate glass	0.18	0.06	0.04	0.03	0.02	0.02
Ordinary window glass	0.35	0.25	0.18	0.12	0.07	0.04
Gypsum board, ½" nailed to 2 x 4's 16" o.c.	0.29	0.10	0.05	0.04	0.07	0.09
Marble or glazed tile	0.01	0.01	0.01	0.01	0.02	0.02
Openings						
Stage, depending on furnishings			0.25	0.75		
Deep balcony, upholstered seats			0.50	1.00		
Grills, ventilating			0.15	0.50		
Plaster, gypsum or lime, smooth finish on tile or brick	0.013	0.015	0.02	0.03	0.04	0.05
Plaster, gypsum or lime, rough finish on lath	0.02	0.03	0.04	0.05	0.04	0.03
Same with smooth finish	0.02	0.02	0.03	0.04	0.04	0.03
Plywood paneling, ⅜" thick	0.28	0.22	0.17	0.09	0.10	0,11
Water surface, as in a swimming pool	0.008	0.008	0.013	0.015	0.020	0.025
Air, sabins per 1000 cubic feet					2.3	7.2

Absorption of Seats and Audience

Values given are in sabins per person or unit of seating

	125 Hz	250 Hz	500 Hz	1000 Hz	2000 Hz	4000 Hz
Audience, seated, depending on spacing and upholstery of seats	2.5-4.0	3.5-5.0	4.0-5.5	4.5-6.5	5.0-7.0	4.5-7.0
Seats, heavily upholstered with fabric	1.5-3.5	3.5-4.5	4.0-5.0	4.0-5.5	3.5-5.5	3.5-4.5
Seats, heavily upholstered with leather, plastic, etc.	2.5-3.5	3.0-4.5	3.0-4.0	2.0-4.0	1.5-3.5	1.0-3.0
Seats, lightly upholstered with leather, plastic, etc.			1.5-2.0			
Seats, wood veneer, no upholstery	0.15	0.20	0.25	0.30	0.50	0.50
Wood pews, no cushions, per 18" length			0.40			
Wood pews, cushioned, per 18" length			1.8-2.3			

NOTE: For other material coefficients refer to *Absorption of Sound in Air Versus Humidity and Temperature*, Cyril M. Harris, Columbia University, New York, N.Y., January, 1967.

a big part. The more porous the material, the more area there is for the frictional effect to take place up to a certain point. Porosity is normally defined as density: the lower the density the more porous and *vice versa*.

Mounting materials of a board or panel type on furring strips with an air space in the rear increases absorption capability at lower frequencies. This type of mounting allows the surface to vibrate under the bombardment of the sound waves, which sets up internal friction in the material, creating heat energy.

It can be seen that under this condition the sound waves are being absorbed by vibration rather than the porosity effect previously discussed. The amount of absorption in this case depends largely upon the vibrational capability of the material which would indicate that the material be of a light and flexible construction. The air space behind the material has the effect of increasing the material thickness and thus increases the absorption capability.

Another method of increasing the absorption capability

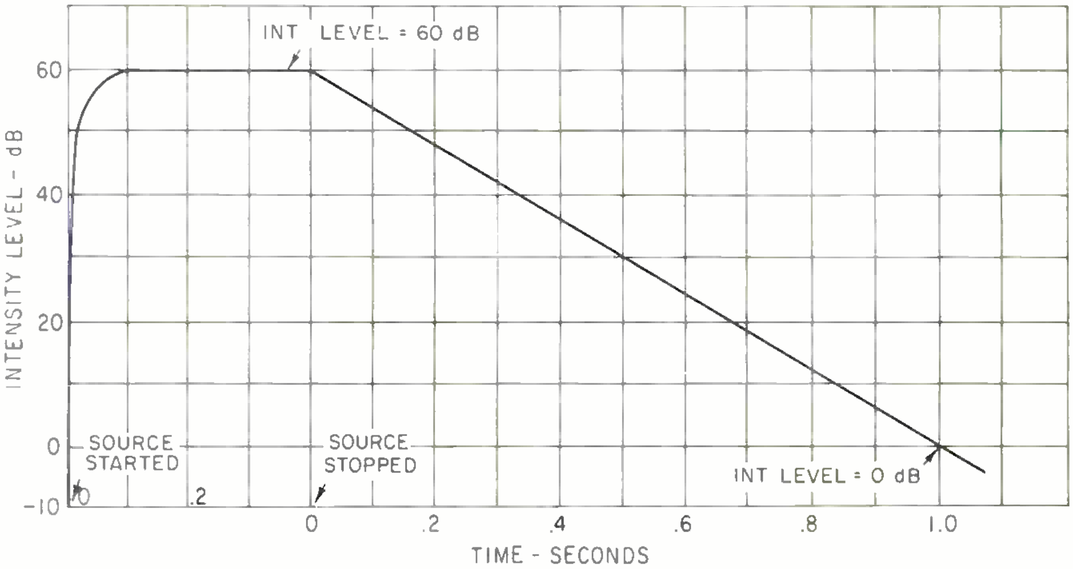

Figure 2. The buildup and decay of sound intensity in an area having one-second reverberation time.

of a substance is by drilling holes into the surface, which, in effect, exposes more surface area to the sound wave. We have now determined that thin porous materials will absorb high frequencies and thin vibrating material will absorb low frequencies. To decrease the frequency range of absorption in porous materials, we make them thicker and to decrease the frequency range of vibrating materials we increase the air space behind them. If we were to paint the surface of a porous material and seal up the pores, this also would decrease the absorption capability of the material.

ACOUSTICS AND LISTENING

As was pointed out previously, when sound waves strike an object, a percentage of the sound energy is absorbed by the object and the remainder of the energy is reflected back into the area. FIGURE 1 illustrates this reflection and what happens to the direction of travel of the original striking source.

In FIGURE 1, we see that the reflected wave always makes the same angle with the surface as that of the original wave. Another way of stating this is that the angle of reflection is the same as the angle of incidence. Also in FIGURE 1 we see that reflected waves travel in the same manner as they would had they been originated by source number two (S_2). This theory is known as the *mirror image theory*, where the image source is the real source located in front of the surface. Knowing this theory allows us to plot out the path of any sound wave in a given area. If you were to take a source of a known radiation characteristic and plot it for a given area, it would become apparent that every part of the area is filled with reflected sound waves. It is these reflections which must be kept in mind when designing a sound system for an enclosed area.

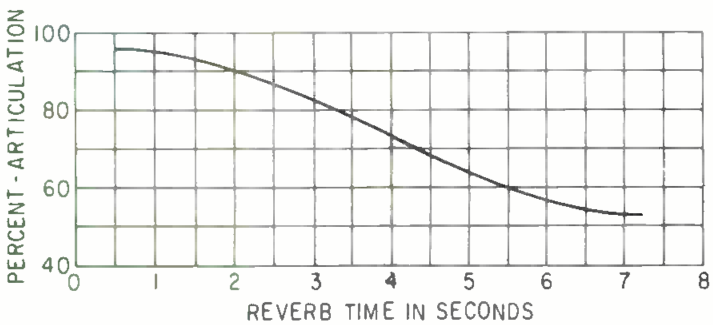

Figure 3. The effect of reverberation time in seconds on articulation.

THE EFFECT OF REFLECTED WAVES

The first effect that should be apparent by this time is the increase in intensity when compared to inverse square loss that will occur because of the reflected energy.

An example of this can be shown by the following: If you stand in an open field and talk at a normal voice level, your ears will not interpret the sound as being as loud as if you were standing in an average closet and talking at the same level. A listener in an enclosed area not only hears the main sound source, but also hears the reflected sound waves which add to the main source and thus make a louder sound than the main source alone would give.

The amount of difference between the main source level and the main source plus the reflected sources level is dependent on the absorption coefficient of the area. If the absorption coefficients of an area are low, the resulting level will be much higher than they would be for an area with a high absorption coefficient at all frequencies.

Sometimes this can be helpful and sometimes it can be a hindrance. If the reflected sound waves reach the listener's ears in rapid succession, the mind will interpret this as an elongation of the original sound (reverberation). However, if these reflected waves do not reach the listeners' ears in rapid succession, but rather in elongated time intervals of 40 milliseconds or more, the result is a confusing echo. Reducing the intensity of reflected waves by absorption has the effect of lowering the loudness buildup rate and decreasing the time required to make reverberations die out.

In areas where excessive reverberation exists, the voice of a person trying to speak to an audience reaches the audience's ears as a garbled mess rather than as a clear, distinct sound. The reason for this garbling is because at the moment the speaker is uttering one syllable, syllables that were previously spoken are still bouncing around the area and the listener hears these as loudly as the syllable that was just spoken. This resultant overlapping of syllables can only result in confusion to the listener.

DESIGNING THE ACOUSTICS

There are four general rules that must be met when designing the acoustics for an area if good listening conditions are to exist.

1. Excessive reverberation at all frequencies must be non-existent.
2. The sound must be fairly equally loud in all areas of the room.
3. Echo interference must be non-existent.
4. Extraneous noise must be non-disturbing.

If these four conditions are met, you can be almost assured of having a desirable listening area. With a sound source operating, the steady-state intensity varies from point to point in an area, due to interference patterns. Even though this variation is present, an average value exists which is fairly uniform throughout the reverberant field of the area.

Average intensity depends on the acoustic power output of the source and the total sound absorbing power of the area. The total sound absorbing power is determined by the absorption coefficients of all surfaces in the area. If the source had been operating at a steady state, and was suddenly shut off, the reverberant field would immediately start to decay in level. This decay will plot as an exponential curve as the intensity decreases by equal percentages in equal time intervals. Example: if the sound decays 40 per cent in the first second, it will decay 40 per cent of what remains in the third second, etc.

It is apparent that the decay is rapid at first and slows down gradually as the intensity nears zero. It is this decay rate that determines the reverb-time of an area.

DETERMINING REVERBERATION TIME

Reverberation time is normally defined as the time required for an average sound intensity to decay 60 dB in level. Reverberation time is given in seconds and is measured from the instant the source is shut off.

FIGURE 2 illustrates the building up and decay of a sound intensity in an area having a reverberation time of one second. Reverberation time of one to one and one-half seconds seems to be ideal for good listening conditions. Reverberation time is sometimes defined as the time required for a sound to decay to inaudibility. This definition is misleading, since there are many variables involved, such as the power of the source, the hearing acuity of the listener, the ambient noise in the area, and the acoustical properties of the area.

Because reverberant sound waves travel at a fixed speed and lose energy by the absorption of the room surface, it is apparent that the longer the average path length between reflections, the longer the sound energy requires to die out. Therefore, the larger the average absorption coefficient and the room surfaces are the quicker the sound will die out.

These factors can be put into the formula $T = \dfrac{0.049V}{a}$,

which is known as the *sabin formula* where T = reverberation time in seconds, V = volume of the area in cubic feet, 0.049 is a constant, and a equals the number of sabins of absorption, more frequently called *sound absorbing units*.

One sound absorbing unit is defined as one square foot of surface that has an absorption coefficient of 100 per cent. In other words, the surface absorbs 100 per cent of the sound that strikes it. Two square feet of perfect ab-

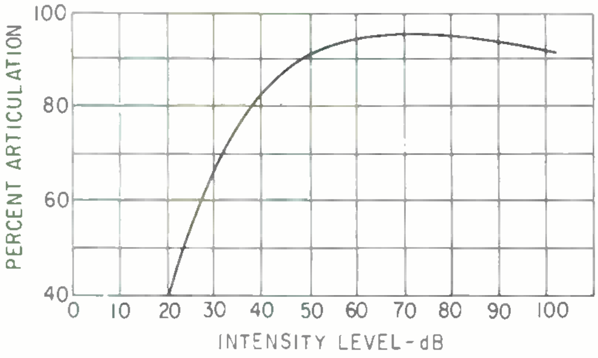

Figure 4. Relation of percentage of articulation to intensity level.

sorption would be two sound absorbing units, etc.

On the other hand, if we substitute a surface with only 25 per cent absorption coefficient for the 100 per cent unit previously mentioned, we find that it requires four times the surface area in square feet to equal 100 per cent absorption. It is, therefore, clear that the number of sound absorbing units furnished by any surface is equal to its surface area in square feet times the absorption coefficient of its material.

The sound absorbing units utilized in the sabin formula for reverberation would be equal to the sum of the number of units furnished by each surface in the area. It should also be apparent that by transposing the formula,

$T = \frac{0.049V}{a}$ to $a = \frac{0.049V}{T}$, we can determine what the

total units of absorption are if the reverberation time of an area is known.

RELATING REVERB TIME TO SPEECH

We have determined that reverberation can cause difficulties in understanding speech. A method of relating reverberation and speech intelligibility has been developed experimentally by V. O. Knudsen[1] by measuring the percentage of articulation in rooms of various reverberation times. A list of words was read to a group of listeners sitting around the area, each of whom wrote down each word as he heard it.

The average percentage of the current words written down was the percentage of articulation for that area. Knudsen's tests showed that 96 per cent articulation is the best that can be attained under ideal conditions. Other conditions tested indicated—85 per cent understandable with no effort, 75 per cent understandable satisfactorily with some effort required, 65 per cent understandable only by closest concentration, quite fatiguing. FIGURE 3 illustrates the effect of reverberation time on the percentage of articulation.

LOUDNESS AND INTELLIGIBILITY

What effect does loudness have on intelligibility and what factors govern this loudness?

The effect of loudness on intelligibility was determined by engineers of Bell Telephone Labs.[2] A high quality telephone system was utilized to conduct the tests where the intensity level in dB was set at various levels and measurements of percentage articulation were recorded. The results of these tests are shown in FIGURE 4.

Based on Knudsen's findings, it is apparent that intelligibility is satisfactory for any intensity level between 40 and 90 dB. As the level is reduced below 40 dB, the articulation drops off rapidly, and intelligibility becomes almost impossible to understand after 30 dB. It should be noted that these percentages, *versus* levels, only hold true if all other listening conditions are perfect. Loudness cannot overcome the effects that reverberation or interfering noise have on intelligibility.

Normally, the larger the area, the more absorption will be present, indicating that the source will have to be proportionately more powerful to cover the area effectively. Knudsen[1] gave an interesting comparison of speech loudness in two rooms, one of 27,000 cubic feet and the other of 240,000 cubic feet. During lectures, the average intensity level of a number of speakers was measured near the center of each area. The average intensity level varied from 44.7 to 56.2 dB in the small room. Using six speakers, this averaged out to 50.7 dB. The average intensity level for *eight* speakers was only 45.7 dB in the larger room. The speakers in the large room produced nearly twice as much sound as the ones in the small room, but this increase was not enough to overcome the added absorption of the larger room.

Knudsen[1] also prepared FIGURE 5 which relates percentage articulation to the reverberation time in rooms of various sizes. The curves were based on an average individual with normal voice capacity, adjusting his voice power to the size of the room. From these curves, we can see that the percentage of articulation decreases continuously for a given reverberation time as the room size increases. For a given room size, the percentage of articulation increases as reverberation time decreases to a certain point; beyond this point, the absorption factor begins to be a deterrent. ■

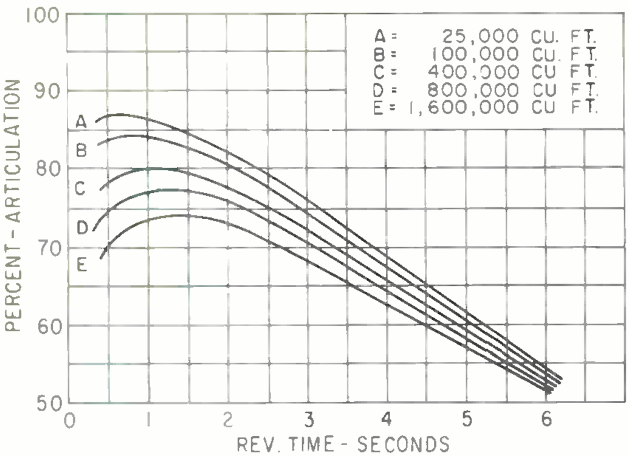

Figure 5. Relationship of the percentage of articulation to reverberation time in five auditoria of various sizes.

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1. Knudsen, V. O. *Journal Acoustical Society of America*, Vol. I, 1929, p. 56. Also V. O. Knudsen. *Architectural Acoustics*, N.Y., John Wiley & Son.

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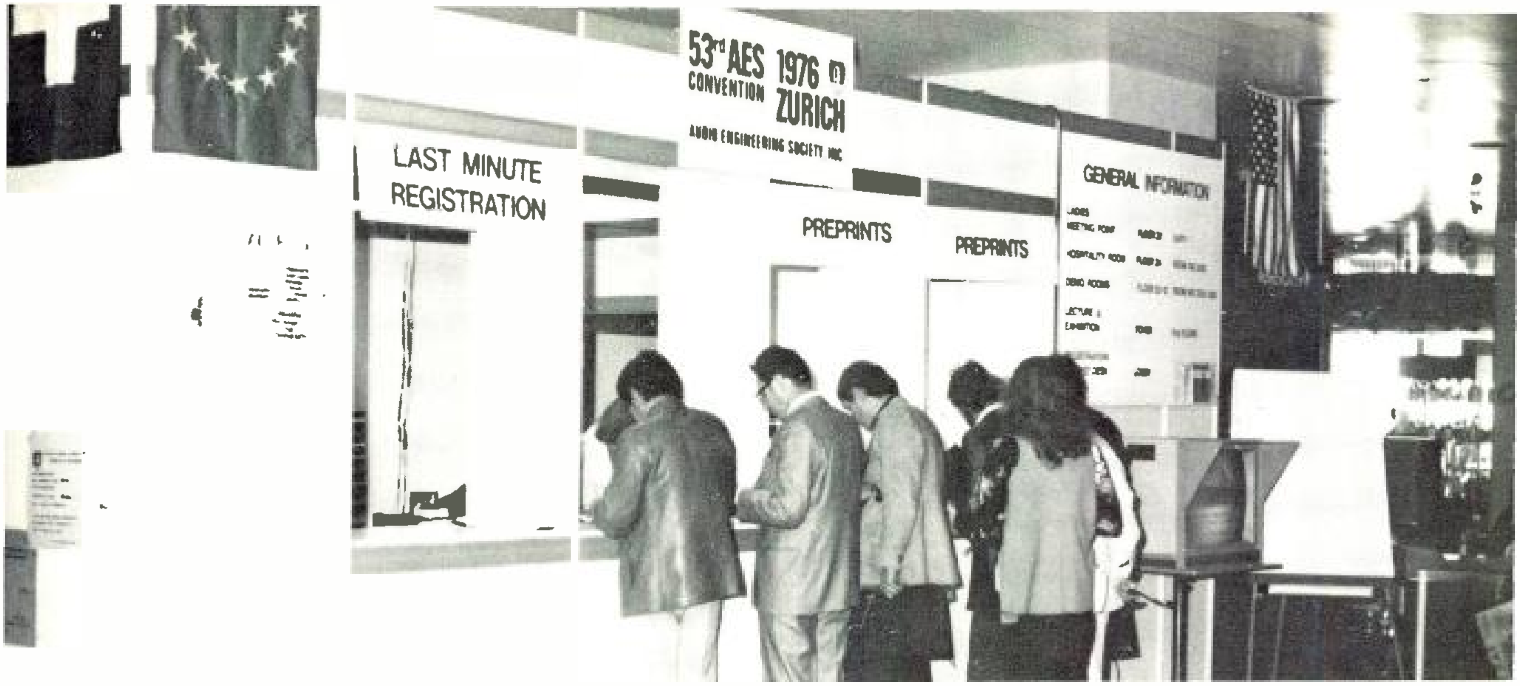


LARRY ZIDE

Trips to Conventions

Herein are reports on two recent conventions. First, the European AES and then a return to our offices, and off again to Chicago for the NAB.

IT MUST BE ADMITTED that the idea of a different city for the annual European AES Convention is a great idea. Last year, I reported from London, this year the Convention was held in Zurich, Switzerland. And just as in years past, the weather was perfect, particularly considering that these shows are held in early March.

The place was the modern Hotel International in Oerlikon (Zurich). About fifty international exhibitors gathered to show their wares, and a total of eight papers sessions were provided. These gave over 50 papers, most in English, but some in a variety of European languages.

I wasn't able to get to too many of the papers, what with the exhibitions and other planned activities. Surely the most interesting one I did see was presented by the Japan Victor staff on the subject of Loudspeaker Wave Propagation Observation. The narrator was John Eargle, who presented a color motion picture film that graphically, and dramatically, demonstrated the sound wave exit from various kinds of loudspeakers. Interference, ripples, etc. were more clearly shown than I have ever seen before.

THE EXHIBITS

For purposes of this report, I will concentrate on those exhibitors who are not usually seen in the U.S. AES Conventions.

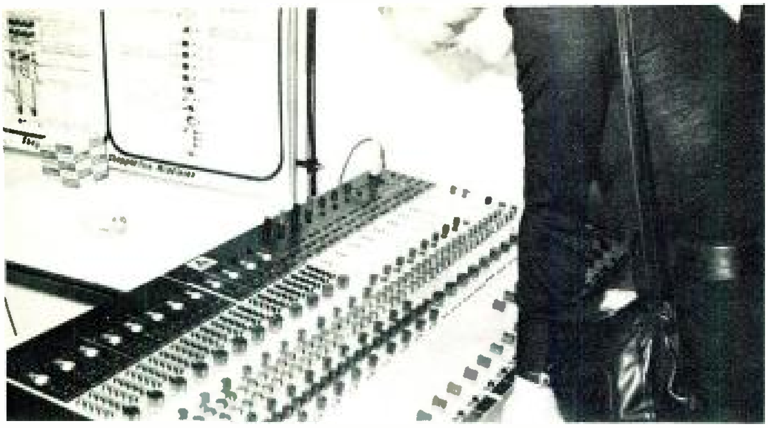
The illustrations accompanying this article show many of the most interesting ones, but I want to single out a few.

By far, the highlight of the show was the Telefunken Telcom, a noise reduction system offering something of a cross between Dolby A and dbx. That may be a misleading statement. Briefly, the Telcom is a four-band compander device that will automatically track the decoding of the encoded tape. Noise reduction is on the order of 30 dB or more. (There is a detailed story on this system elsewhere in this issue.)

I listened to the unit via headphones. In truth, this demonstration on the floor was too limited to permit an assessment of the system. Certainly, it works, but is it undetectable? I don't know yet.

Several shows ago, we reported on the BASF Unisette, an oversized cassette system. Perhaps *system* is the wrong word, because this is the first exhibition in which *equipment* has been shown that is capable of using these cassette/cartridges. The Unisette system was demonstrated on a prototype Studer machine.

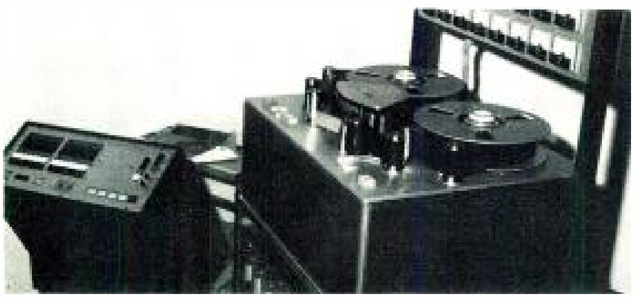
(continued)



The Trident portable console can be ganged for multi-input or output operation.

If you want to record binaurally, AKG can provide you with a mean looking artificial head.

The Telefunken Telcom noise reduction system was on working demonstration.



The sixteen track Lyrec is a beautifully assembled machine that may come to the U.S.

Ferrograph, the tape deck people, showed an audio response analyzer, model ARA1.

This is a broadcast system designed to replace with a reel-to-reel cartridge (the Unisette) the re-entrant cartridge system now used in broadcasting. Certainly, the broadcasters need something better than the present cartridge system, but will it be the Unisette?

Two alternatives exist, neither shown in Zurich: there is the Japanese *Elcaset*, their spelling for a large sized cassette using quarter-inch tape at 3¾ in./sec. speed. We should be seeing machines and tapes from several sources in Japan well before this year is out.

Then there is the new Otari professional cassette deck, a fully professional unit that uses standard cassettes but operates them at 3¾ in. sec. and lays down two tracks. More on this later.

CONSOLES

Americans know Neve, of course but Europeans know API, Audio Designs, Spectra-Sonics, and MCI, all of whom showed in Zurich. API has the widest European success of the three thus far. They have fully automated consoles operating in European studios and their European distributor (who also distributes 3M on the Continent) is doing a bang up job for both accounts.

Audio Designs recently signed a distribution agreement with Ampex (only outside the U.S.) and thus was on exhibit for the first time in Europe at the Ampex booth.

MCI was there in force with their own booth. They were showing both tape decks and consoles to visitors.

One British console manufacturer deserves to be better known here if the products he has shown (and I have seen in European studios) are any example. This is Triad, whose latest Fleximix System is a well designed portable modular console. This system can be expanded at any time without rewiring to anything from mono to 24-track. With fifteen input modules, the unit measures only 27 inches wide.

OTHER INTERESTING EXHIBITS

Several exhibitors were not on the main exhibition floor, but in rooms of the hotel. CBS was there extolling their SQ four channel matrix system, and from what I could see, they were getting a lot of interest. dbx was there, and they too were getting a lot of interest in their noise reduction system. I wouldn't be surprised to see an increase in dbx usage by European studios.

Keith Monks is both a person, and a company name who makes a lot of specialty items for record playing. He makes a powered record cleaner that is a Rube Goldberg contraption, but really does clean discs well. There are also tonearms, cable drums, mic stands, etc., and Monks has recently aligned himself with a British microphone manufacturer, H-H—also a name we may know better in the U.S. in years to come.

SUMMARY

The 53rd AES Convention had an excellent attendance. Zurich proved to be a fine host city, the equal of those in years past. I have now seen the European AES grow from almost nothing into a major third show. Thus the AES is truly an international organization operating these three annual affairs. Next year's European Convention will be held in Paris. Where else can you find such a combination of education and tax-deductable travel pleasure?

THE NAB CONVENTION

Two weeks after getting off the plane from Zurich, I was back on a plane (much smaller this time) heading for Chicago and the 54th annual Convention of the National Association of Broadcasters.

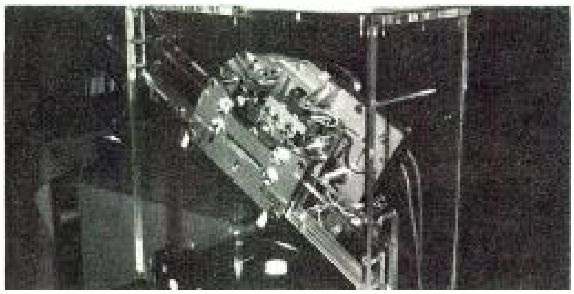
I've avoided this show in the past several years because it had become so thoroughly video broadcast oriented that audio was given short shrift. I'm happy to report that things have turned around and audio is back in force. Perhaps the video people have discovered that audio can exist without a picture, but not the other way 'round.

The Convention was held in the mammoth McCormick Place exhibition hall. This is a great barn of a place that is good for boat shows, car shows and flower shows. It's so big that the National Flower Show was occupying the main floor at the same time as the NAB was relegated to the basement lower level.

I think this was a mistake on the part of the NAB. First, the planning didn't take into account the flower show, which attracts literally hundreds of thousands of visitors. These two shows at the same time overtaxed the ability of Chicago taxis (needed to get to and from McCormick Place) as well as the facilities of the hall itself.

Further, the exhibitors of the NAB are of two types. There are the large ones such as Philips and RCA that require ballroom-type space, and the smaller ones that are best served in the kind of space represented by hotel rooms. The open barn of McCormick Place is wrong, I think.

The BASF Unisette. It looks like a standard cassette, but it's much larger.



This is the Studer Unisette machine, mounted in an acrylic case to show its operation.

ABOUT THE SHOW

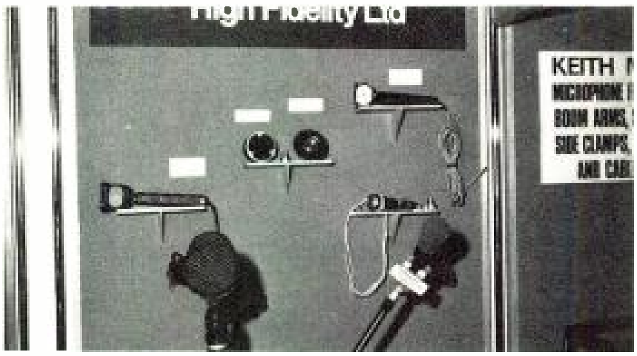
I've collected an impressive amount of literature on audio products shown this year. It was particularly interesting to see friends from the recording field who are expanding into the broadcast market, as they should. Among these companies were API, Auditronics, and Neve (with broadcast consoles), and Ampex, Scully, Sony Corporation, Electro-Voice, 3M, Cetec, and others. Some companies such as Robins/Fairchild (they are phasing out the Fairchild name gradually) once primarily in recording studio product manufacture are now primarily in broadcast work.

Which category do you put an exhibitor such as Stanton Magnetics? They provide phono cartridges that are of recording studio calibration standards, and others that can be cheerfully (and safely) backcued by broadcasters, and indeed many Stanton cartridges are found in both kinds of places.

Familiar names such as Audimax, once made by CBS Labs, are now made by the company that bought the division, Thomson-CSF Laboratories, Inc. They showed these and newer designed products.

Optimod-FM is a product made by Orban Broadcast (same Orban). This is a single product to replace conventional limiters, compressors, and stereo generators. It does all they do, and in a sonically most subtle but effective way.

Finally, because I'm running out of space, I must mention Otari. They were there showing off their already established tape recorder and duplicator line, but what may have well have stolen the show was a prototype broadcast cassette deck in its first appearance. This is a professional (balanced in and out lines) unit that takes standard cassette tapes but operates at double speed of 3¾ in./sec. Instead of the standard four cassette tracks, only two wider tracks are used. Applying theory only, the doubling of speed and widening of tracks will offer 6 dB more s/n, but this machine probably does even better. We'll have more information on it in an early issue. ■



These H-H mics were shown on Keith Monks stands at the Monks booth.

The EMT 250 electronics effect reverb generator is demonstrated by Gotham Audio's Steve Temmer (left).

MICHAEL RETTINGER

Low Frequency Slot Absorbers

Resonance control can be varied with the flexible use of slot-absorber configurations.

SINCE THE APPEARANCE of my article, "Low-Frequency Sound Absorbers," in the April, 1970 issue of **db** I have received a number of inquiries about the subject. Bringing us up-to-date on new acoustic design details for such units, it appears desirable to re-examine this type of sound absorber.

FIGURE 1 shows the principal dimensions of a slot absorber placed at an angle θ to a vertical wall.

The frequency of resonance of a slot absorber with identical slots and slats is, for a first approximation, given by

$$f_0 = \frac{c}{2\pi}\sqrt{\frac{A}{dV}}$$

	A	V	d
$= 54.3\sqrt{\frac{A}{dV}}$	sq. meters	cu. meters	meters
$= 5480\sqrt{\frac{A}{dV}}$	sq. cms.	cms.	cu. cms.
$= 2160\sqrt{\frac{A}{dV}}$	sq. ins.	cu. ins.	inches

where d = thickness of slat
A = area of slat
V = volume of space between wall and slot absorber.

To avoid having to introduce the dimensions of the absorber, which depend on the dimensions of the room, the per cent open area, *e*, provides a convenient measure, along with slot areas and slat-to-wall volumes for slots of unit length. Thus

Michael Rettinger, a frequent contributor to db, is an acoustical engineer from Encino, California.

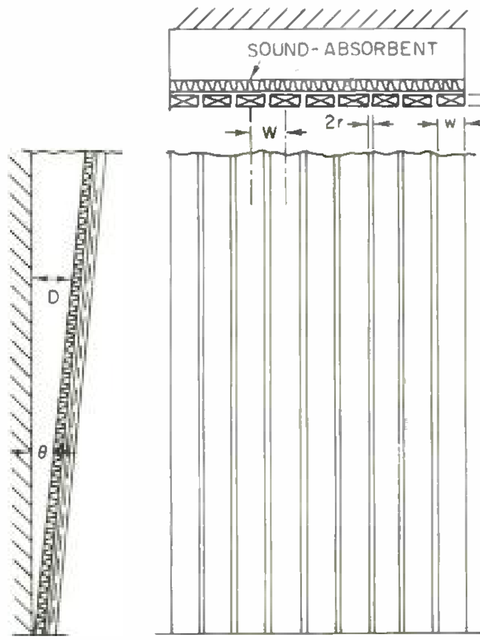


Figure 1. Principal dimensions of a slot-absorber.

$$e = \frac{2r}{w + 2r}$$

where 2r = width of slot
w = width of slat

As long as the same type of units are used for the quantities in *e*, the per cent open area will come out correctly, being dimensionless.

If we concern ourselves only with an absorber of width *w* which contains one slot and the two halves of adjoining

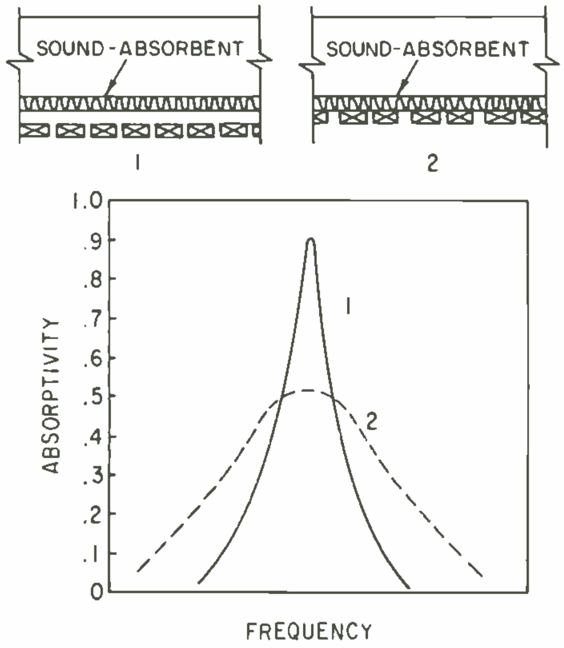


Figure 2. Sound absorption characteristics of a slot-absorber: (1) sound-absorbent material spaced at a distance from the slats; (2) sound-absorbent material in contact with the slats.

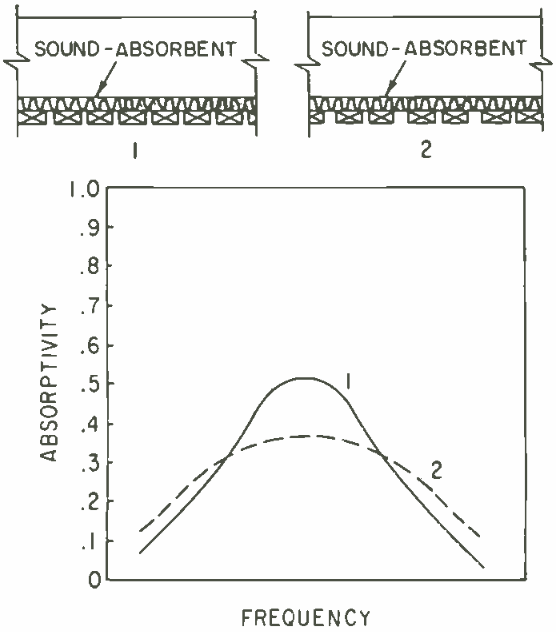


Figure 3. Sound absorption characteristics of a slot-absorber: (1) slats spaced at a constant distance from each other; (2) slats spaced irregularly.

slats, we may write

$$\frac{A'}{V'} = \frac{e}{D}$$

where D = depth of air-space
A' = area of slot for unit length of slot
V' = volume of air-space behind slats for unit length of slot.

Using English units of measurement, we get

$$f_o = 2160 \sqrt{\frac{e}{dD}} = 2160 \sqrt{\frac{2r}{dD\ (w + 2r)}}$$

For a more accurate determination of the resonance frequency, the thickness or depth of the slot *d* (essentially the thickness of the slat) should be replaced by the effective depth *d'*. This term, however, is a function of frequency, the cross-sectional shape and area of the slot, and the acoustic properties of the sound-absorbent behind the slots. Hence, the exact effective length must be determined experimentally. For estimating purposes, *d'* may be made 1.2d.

In the references provided at the end of this article, more accurate formulae are given for slot absorbers, but only for those which have no acoustic material behind the slats. In my experience, the use of 1.2d for *d'* has resulted in resonance frequencies which agreed well with tests when the slot absorber contained 1 in. thick fiberglass board behind the slats.

FIGURE 2 shows the effect of placing the acoustic material at a distance from the slats, instead of in contact with them. This results in a sharper absorption characteristic for the absorber.

FIGURE 3 illustrates the effect of irregular slots, while maintaining contact between acoustic material and the slats. This results in a broader absorption characteristic.

By slanting the absorber against the wall, a similarly broader absorption characteristic results, compared to that obtained with a parallelepiped airspace between wall and slats.

Below are the requirements for a sharp and a broad characteristic:

Sharp Characteristic	Broad Characteristic
Identical slots	Slots of various width
Identical slats	Slats of various width
Space between slats and acoustic material	Acoustic material in contact with slats
Parallelepiped airspace	Prismatic airspace

VARIATIONS

Variability of slot absorbers can also be achieved by non-parallel slots (achievable by the use of non-parallel slats), slats of tapering thickness, and other non-conformities or irregularities. In addition, a slot absorber can be designed with two sets of slats, one over the other, and so fastened that the exterior slats can slide to cover some or all of the slots of the unit. Changes in the absorption characteristic of a slot absorber can also be achieved by the use of a thick plastic curtain drawn partially over the unit.

Helmholtz resonators are used to advantage when a room has widely spaced, low-frequency modes of vibration, brought about either by a small enclosure volume or by an integral or near-integral ratio of dimensions. These so-called *eigentones* are of three types: (1) axial, involving reflections from any of the three pairs of parallel walls in a rectangular enclosure; (2) tangential, with reflections occurring in parallel along two principal orthogonal directions; and (3) oblique waves, concerning reflections from all six surfaces. Of these modes, the axial ones are frequently the most important. The equations for these vibrations are

$$f_o = \frac{565n}{L} \quad f_o = \frac{565n}{W} \quad f_o = \frac{565n}{H}$$

where *L*, *W*, and *H* are the length, width, and height of the room expressed in feet, and *n* is any integer.

FIGURE 4 shows a studio 15.5 feet high, 22.1 feet wide,

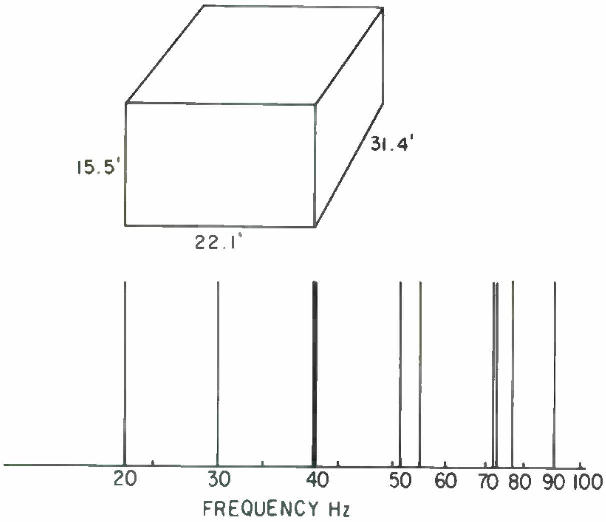


Figure 4. Number and frequency distribution of axial modes of vibration for the room shown on top of the figure.

and 31.4 feet long. It has the following axial modes along

Length	Width	Height
18.0		
	25.6	
36.1		36.5
	51.2	
54.1		
72.1		72.1
	76.7	
90.2	102.2	
108.2		108.8
126.3		
	127.8	

FIGURE 4 also shows the distribution of the axial modes along a logarithmic frequency scale. It is seen that the widest spacing is between the frequency of 54.1 and 72.1 hertz (18 hertz) and that there are double modes near 36 and 72 hertz. This condition can cause coloration (spectrum- or pitch-change), particularly during reverberation.

Assume that it is intended to install a Helmholtz absorber in the form of a slot unit employing 2 x 4 in. slats with an airspace D = 10 inches behind the slats, and that the unit should be most absorptive at 72 hertz. What should the width of the slots be?

$$f_0 = 2160\sqrt{\frac{2r}{dD\ (w + 2r)}}$$

$$72 = 2160\sqrt{\frac{2r}{2.4 \times 10 \times (4 + 2r)}}$$

By solving the above equation for 2r we obtain 0.1 in. for the width of the slots.

In the design of a slot absorber, precautions should be taken to avoid a depth dimension *D* which is equal to a half wavelength of the desired frequency of resonance. The reason for this is that the wave reflected from the hard wall in back of the slats would interfere with the high particle velocity at the interior surface of the slot, since it would be 180 degrees out of phase. Similarly, when *D* is equal to a quarter wavelength of the resonance frequency, the air stream at the mouth of the resonator would be additionally higher because the particle velocity of the reflected sinusoidal wave is highest at a quarter wavelength from the hard wall. The former cancellation effect for the 72 hertz resonance frequency would have occurred if the depth *D* of the (parallelepiped) airspace had been made 0.94 in. and the slot width *2r* had been made equal to 1.1 in. for the 2 x 4 in. slats. Similarly, additional absorption at 72 hertz would have been gained if the depth *D* had been made 47 in. and the slot width 0.46 in.

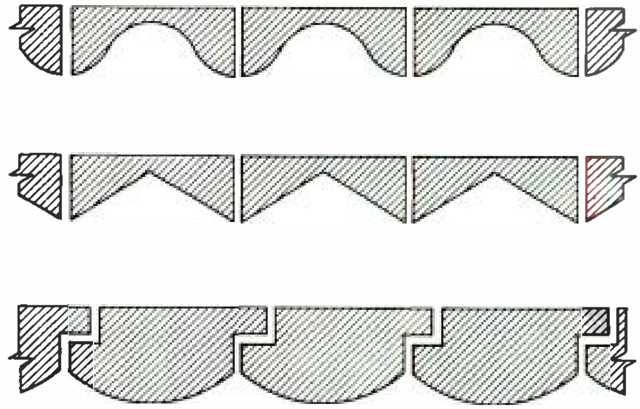


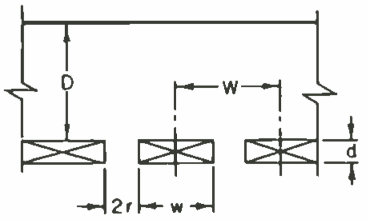
Figure 5. Slats for a slot-absorber may have any shape.

RESONATORS

When the air space behind the slats is subdivided by partitions, so that a number of resonators with smaller volumes are formed, the resonance frequency of each resonator so created will be higher than that of the total unit (with equal and uniform slots). This is evident from the fundamental equation of a Helmholtz resonator whose frequency is inversely proportional to the square root of the volume behind the slats.

The use of vases in a studio for sound-absorptive purposes must be carefully considered, because such resonators may actually prolong the reverberation in a room. It was this type of vessel, generally made of bronze, which the Greeks used in their open-air amphitheatres so that "the sound is strengthened and beautified" (Vitruvius). Such prolonged re-radiation of the signal from metal jars resembles that of a bell, which when struck may continue to vibrate for a period of time ranging from a few milliseconds to possibly a second and more.

The slats for a slot-absorber do not have to be a parallelepiped, that is, have a rectangular cross-section. FIGURE 5 shows various cross-sectional shapes for the slats. The slats shown on the bottom of the figure are similar to those used in the Herrenhaus-Sitzungssaal of the Parliament in Vienna, Austria. Indeed, the rectangular shape of the slats is not the best, particularly when a large wall area is covered with them, since such a surface can give rise to high-frequency echoes in the manner of perforated hardboard or transite. Parenthetically, the common perforated ⅛ in thick Masonite with a 10 per cent open area, and backed by 1-inch absorbent material is not a low-frequency, but more nearly a mid-frequency, and even high-frequency, absorber, depending on the spacing between acoustic material and Masonite, the size of the holes, and type of absorbent used.



$$e = \frac{2r}{W} = \frac{2r}{W+2r}$$

$$= 4 \times 10^{-8} dDf_o^2$$

$$f_o = 5000\sqrt{\frac{e}{dD}}$$

$$dD = \frac{2.5 \times 10^7 e}{f_o^2}$$

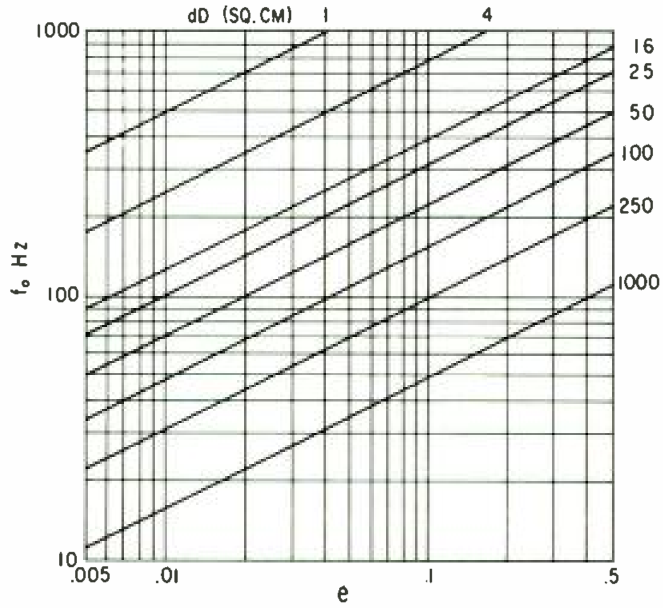


Figure 6. Chart for determining the resonance frequency of a slot absorber.

APPENDIX

For practical design purposes, the resonance frequency of a slot absorber may be written as

$$f_o = 5480\sqrt{\frac{e}{1.2dD}}$$

$$= 5000\sqrt{\frac{e}{dD}}$$

where d = slot depth, cm.
D = depth of space between slats and hard wall, cm.
e = per cent open area

$$= \frac{2r}{w + 2r}$$

2r = width of slot, cm.
w = width of slat, cm.
V = unit air volume behind slats
= D (w + 2r) x 1, cm^3.

From a knowledge of the desired resonance frequency, we may solve the above equation for any of its factors. Thus

$$e = 4 \times 10^{-8}\, dDf_o^2$$

$$D = \frac{2.5 \times 10^7\, e}{df_o^2}$$

$$d = \frac{2.5 \times 10^7 e}{Df_o^2}$$

$$2r = \frac{4wdDf_o^2 \times 10^{-8}}{1 - 4f_o^2 dD \times 10^{-8}}$$

$$V = WD \times 1$$

$$= (w + 2r)D$$

$$= \frac{2.5 \times 10^7}{f_o^2 \dfrac{d}{2r}}$$

The last expression is a simplified form of the one given in the April, 1970 issue of **db**. It should be noted that in that equation, a typographical error occurred in that a square sign was omitted in one of the factors in the denominator. The correct equation is

$$V = \frac{3 \times 10^7}{f_o^2 \left(\dfrac{d}{2r} + 6.62 - 1.462 \log 2rf_o\right)}$$

■

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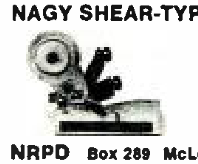

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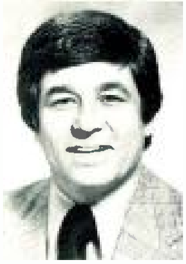

HOLLON

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BROOKS

• Two new staff changes have been announced at **JVC America,** of Maspeth, N.Y. **George E. Meyer** has been named as national merchandising manager and **Ken L. Acker** as national credit manager. Mr. Meyer was formerly director of product management for **Fisher Radio.** Mr. Acker has been midwest credit manager for JVC.

• **James M. Hollon** has been appointed sales supervisor, professional recording and broadcast markets of the Magnetic Audio/Video Products Division of the **3M Company** in St. Paul, Minn. Mr. Hollon has been with 3M since 1967.

• A fresh look at management and new business opportunities is in the wind at **Goldmark Communications Corp.,** Stamford, Conn. with the recent appointment of **Nat C. Myers, Jr.** as vice president. Mr. Myers has considerable expertise in the innovative approach to audio-visual communications marketing. Mr. Meyers comes to Goldmark from the AnCom Company and has for the past year been serving as a corporate consultant for Goldmark's Transcan division.

• A Fine Arts Convocation at **Lycoming College,** Williamsport, Pa. was the scene of the granting of an honorary degree to **Dr. Robert A. Moog** on April 5. Dr. Moog founded the firm which bears his name in 1965, for the purpose of developing and manufacturing the Moog synthesizer, now an accepted source of modern music.

• **Warner Cable Corporation,** New York City cable t.v. operators, has announced the appointment of **John S. Auld** as vice president. Mr. Auld is well known people in the broadcast product field, coming to Warner from **Philips Broadcast Equipment Corp.**

• **Component Marketers, Inc.,** of Montclair, N.J., has increased its partnership roster. **Morley Kahn,** coming from **Dolby Laboratories,** has joined the company as executive vice-president. **Robert Pett,** who has been with the firm for a number of years, has also joined the partnership. The other partners in the firm are **Jack Fields,** president, and **Jack Simon,** vice-president.

• **Carl N. Brooks** has been added to the staff of the **Turner Division** of **Conrac Corporation,** of Cedar Rapids, Iowa, as manager of engineering. Mr. Brooks comes to Turner from **Motorola.**

• Appointments of sales managers for the midwestern and western areas were announced recently by **RCA Broadcast Systems. Paul Bergquist,** manager of midwestern area sales, will be based in Camden, N.J. **Ray Harding,** based in Hollywood, Ca., is in charge of the western area, replacing recently retired **Edwin C. Tracy.**

• **William R. Brock** has been appointed manager of dealer sales by the **Auditronics Company** of Memphis, Tennessee. Mr. Brock was previously with **Scully/Metrotech.**

• Engineering design and manufacturing services for broadcasting, recording, sound reinforcement, and film applications are offered by a new firm of consultants, **Professional Audio Systems Engineering, Inc.,** 7330 Laurel Canyon Blvd., N. Hollywood, Ca. 91605. Pricipals in the new firm are **George Gaal, Richard Guy,** and **Donald Petty.**

• Touted as the most powerful computer-based equipment ever developed for broadcast, valued at over $151,-000, System 770, developed by the **IGM** division of **Northwestern Technology, Inc.** of Bellingham, Wash. has been purchased by **Storer Broadcasting's KGBS,** Los Angeles, a **CBS** f.m. station. The sale was focalized by **Chester P. Coleman,** the System 700 sales specialist for IGM.

• Distinguished educator and composer **Emerson Meyers,** founder and director of the Electronics Music Laboratory at **Catholic University** in Washington, D.C. has retired from teaching and is now available for consultation and workshops in electronic music. He can be reached at Catholic University.

• A new disc-mastering system, introduced in March, is being produced in a joint venture between **Rupert Neve and Company** of Melbourn, England and MSR Electronics, of North Coventry, England. The system, called the Neve/MSR 2000 Series Disc Mastering System, uses innovative techniques, including computer technology.

• Staff changes at **Spectra Sonics,** Hollywood, California include the promotion of **Bruce Hall** to the post of division sales manager. **Brian Morze** and **Steve Cannon** have been added to the engineering sales staff. Mr. Cannon, who has an extensive background in outdoor entertainment events, will specialize in sound reinforcement. **Julie Wahnsiedler** has joined the office staff.

• Capping a career of artistic achievement, **Richard J. Vorisek** was nominated by the British Academy Award for outstanding achievement during 1975 in the Best Sound Track category for his work as re-recording supervisor of the film "Dog Day Afternoon." Other impressive credits have been earned through Mr. Vorisek's contributions to "Lies My Father Told Me," "The Fortune," "Lenny," "Hester Street," "The Stepford Wives," and "Serpico." Mr. Vorisek is a vice-president of **Trans/Audio, Inc.,** of New York City.

• The **Society of Broadcast Engineers** is accepting papers to be presented at the annual convention November 7, 8 at the Holiday Inn in Hempstead, N.Y. Brief abstracts, 100-300 words, should be submitted to **Mark Schubin,** Society of Broadcast Engineers, P.O. Box 607, Radio City Station, New York, N.Y. 10019 by August 1.

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It's not something an amateur can do. It's an art. And that's why Yamaha has designed 3 superb mixing consoles with the qualities and range of controls that the professional sound reinforcement artist needs.

For instance, our exclusive 4x4 matrix with level controls gives you more exacting mastery over your sound than the conventional method of driving speaker amps directly from the bus outputs.

Features like that are years away except on the most expensive mixers. On the Yamahas, it's standard equipment. And so are transformer isolated inputs and outputs, dual echo send busses, an input level attenuator that takes +4 dB line level to −60 dB mike level in 11 steps, and 5-frequency equalization.

Whether you choose the PM-1000-16, the PM-1000-24 or the PM-1000-32, Yamaha gives you the flexibility you need to turn your job into an art. And because they're designed from the ground up to perform on the road, more and more professional sound men around the United States and the world are depending on Yamaha, night after night, gig after gig.

If you've never thought of your mixing console as a musical instrument, we'd like to invite you to stop by your Yamaha dealer. Once you've checked out the operation manual and tested for yourself what the PM Series can do, we think you'll come away a believer.

YAMAHA

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