db

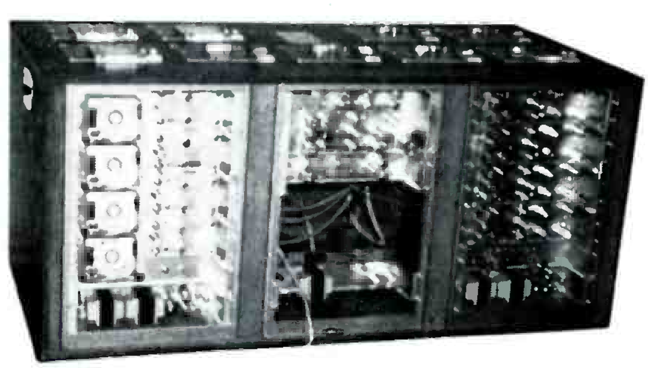
THE SOUND ENGINEERING MAGAZINE

February 1970 75c

A Primer on Noise Measurement, Part 2

Automated Radio Broadcasting

Stereo Synthesis Systems



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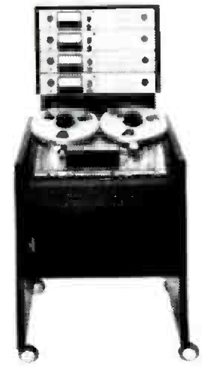

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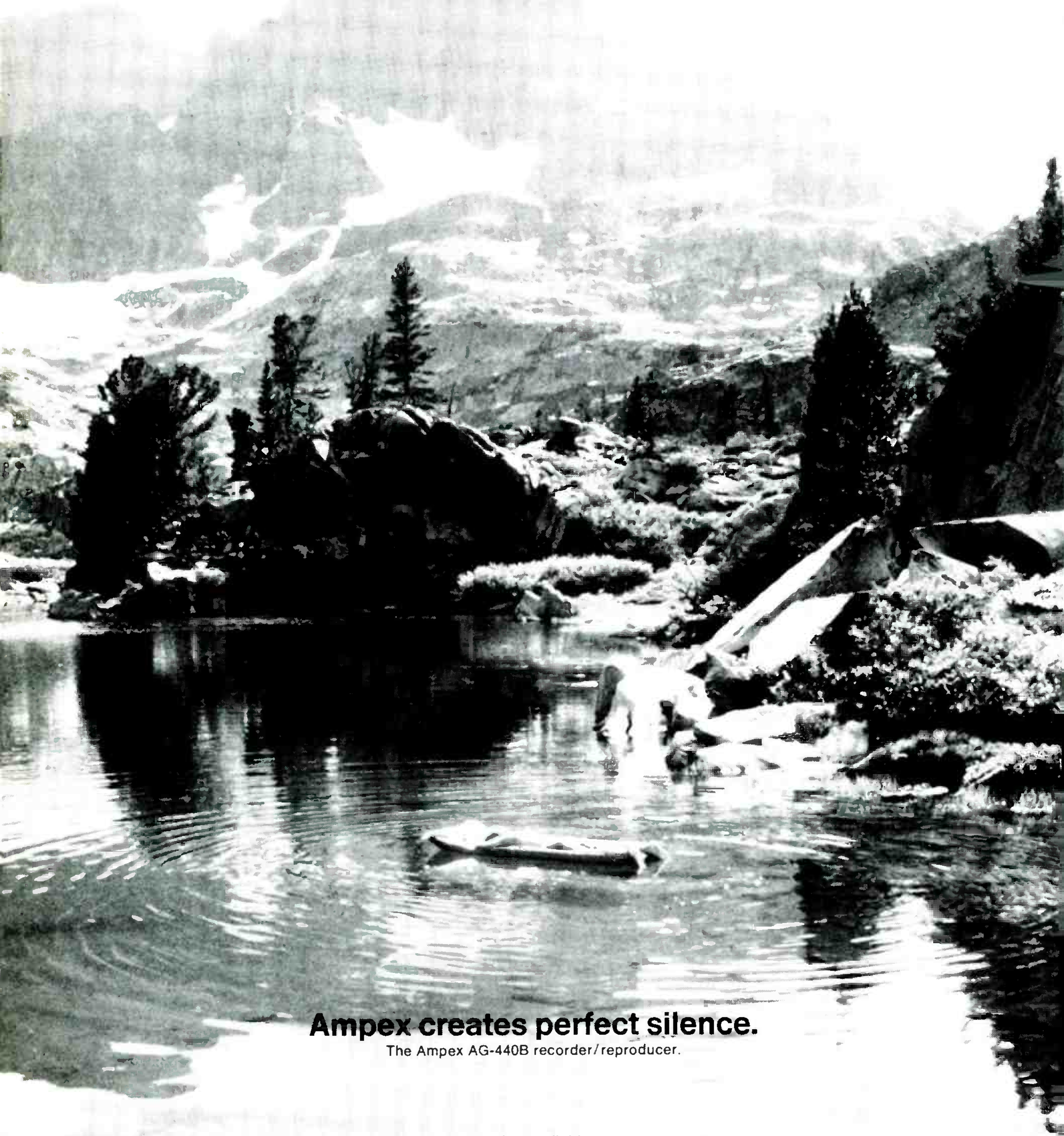

Coming Next Month

Walter Jung's newest contribution is A SWITCHING/SUMMING AMPLIFIER The circuitry described will be of particular interest to those seeking involvement with electronically generated music systems — synthesizers — of which this is a basic part. Constructors will be pleased with the product and the fact that complete parts lists are included.

THE ELIMINATION OF TRANSFORMERS IN CONSOLES is the title of Allan Smith's continuation of the theme he developed in his story that appeared in our December issue.

Elliott Full has sent in an amusing short piece titled HOW TO MIC A SPORTS BROADCAST. He describes how KXIC in Iowa City, Iowa has learned by doing just how and how not to mic a sports broadcast.

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

About the Cover

The illustrated system is an automatic radio programmer. The basic concepts of automated radio are described in Gene Hostetter's article beginning on page 18.

← Ampex Circle 10 on Reader Service Card

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THE SOUND ENGINEERING MAGAZINE

FEBRUARY 1970 • Volume 4, Number 2

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Letters

The Editor:
I read with interest the item on a projected behind-the-iron-curtain audio tour. Who is sponsoring it? What will it encompass? And what will it cost?

Robert P. Raver
Elk Grove Village, Ill.

The sponsor of the tour, which will take place April 4-25 is the Citizen Exchange Corp. This is a totally non-political organization which is attempting to create counterpart people-to-people exchanges so that this can be a force toward lessening world tension.

This specific tour is being directed by John Woram who is at once a db *columnist, a recording engineer in the employ of RCA Victor, and a director of the C.E.C. It will bring its professional audio (recording and broadcast people) to their counterparts in East and West Berlin, several cities of the Soviet Union (including Moscow), and Prague, Czechoslovakia. The total cost including air transport will be $983. Ed.*

The Editor:
Recently our company has had inquiries regarding four-channel reproduction, and the possibility of its becoming a significant factor in the marketplace. We would like to state that we do not believe that this will be the direction that the industry will take. As further proof of this, one of the three manufacturers who started to construct such a machine has now discontinued their model without a replacement being available.

An industry such as ours is always full of people who are prepared to render obsolete all existing equipment merely to focus attention on their ideas. They do great damage, and usually manage to create discussion about a subject the roots of which are too often never exposed.

Let us consider the parallel drawn between four-channel and conventional stereo, and that of stereo against mono some years ago. Firstly, all the major record companies had been producing stereo masters for many years before stereo became a significant domestic reality. This is not the case with four-channel, and only Vanguard has any library at all, and that is extremely limited. Bear in mind that until all major record producers can offer a sig-

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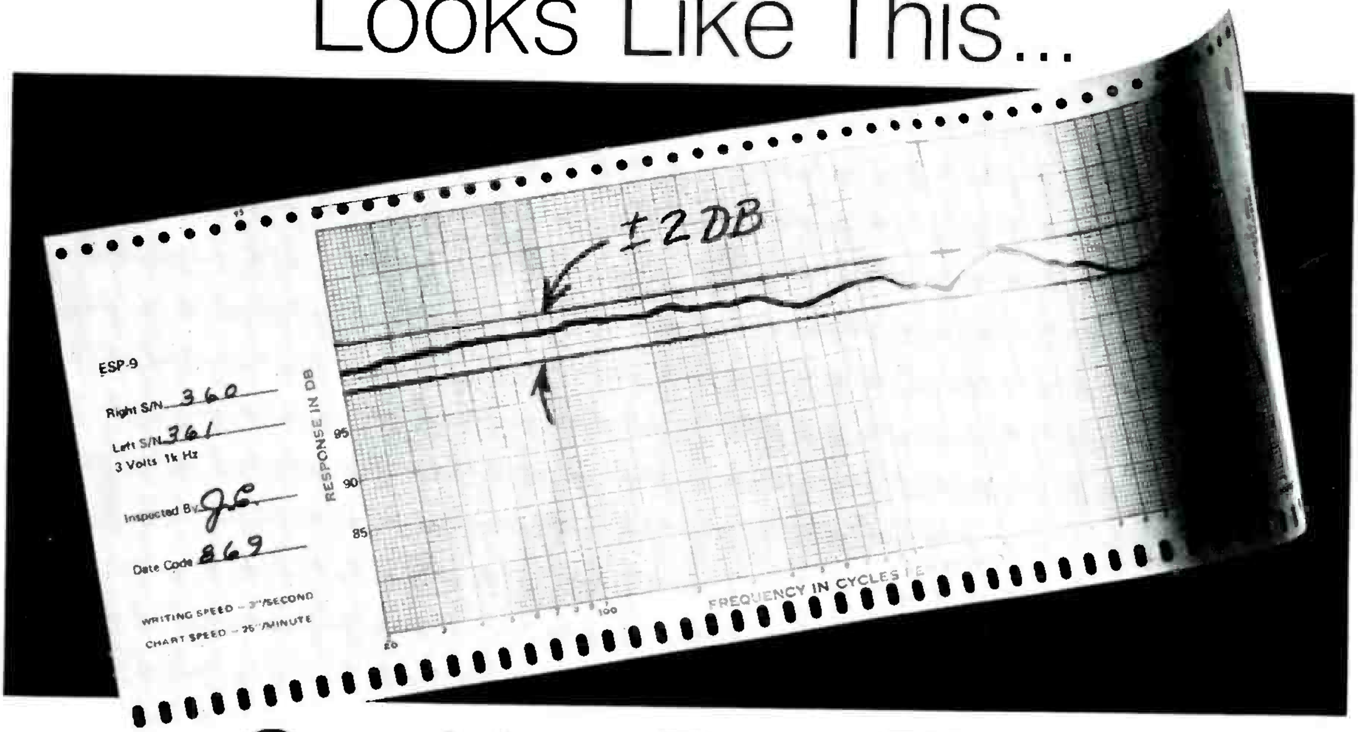
The Sound Of KOSS Looks Like This...



Can You Beat That?

To guarantee performance to specifications, this individual machine-run response curve comes with every ESP-9 Studio Monitor Headset. You get, for the first time, flat ± 2 db monitoring over the entire audible spectrum because the ESP-9 is a breakthrough electro-acoustical development achieved by exploiting electrostatic principles. Only Koss electrostatics give push-pull balanced acoustical circuitry, cancelling all second harmonic distortion to provide fatigue-free listening through long recording sessions. Now you hear what the program material *really* sounds like, uncolored by monitor room reflections. Exceeding the range and cleanliness of any speaker system, the ESP-9 gives the measure of separation and accurately positions the soloist. 40 db isolation through comfortable, fluid-filled cushions relieves the noisy distraction caused by producers, A and R men, time-killing artists, and other visitors in the control room. The ESP-9 eliminates the masking effect of blowers, breath sounds, clothes rustling and other control room ambients. So now you have a running check on low-level system noise. You monitor the sounds you only *saw* before on the VU meter, like the "whoosh" of a stage door closing, ventilator rumbles and music stand rattles — because speakers simply don't have the super-wide-range you need to hear them.

The ESP-9 has a signal handling capacity of 10 volts at 30 Hz with good wave form versus 6 volts for the integrated ESP-6 introduced last year. This is made possible by increasing the size of the coupling transformers by a factor of 4 and mounting them in the E-9 Energizer external to the cup.

The E-9 Energizer offers the option of self-energizing for the bias supply, or energizing through the ac line; choice is made with a selector switch on the front panel. When energized through the ac line, very precise level measurements can be made. Thus the unit is ideal for audiometry, and for evaluating the spectral character of very low level noise in tape mastering machines and recording consoles.

SPECIFICATIONS

Frequency Response Range, Typical: 15-15.000 Hz ± 2 db (10 octaves) 10-19.000 Hz ± 5 db. An individual, machine-run calibration curve accompanies each headset. Sensitivity: 90 db SPL at 1kHz ± 1 db referred to 0.0002 dynes/cm² with 1 volt at the input. Total Harmonic Distortion: Less than ⅕ of 1% at 110 db SPL. Isolation From External Noise: 40 db average through fluid-filled cushions provided as an integral part of the headset. Power Handling Capability: Maximum continuous program material should not exceed 10 volts (12 watts) as read by an ac VTVM; provides for transient peaks 14 db beyond the continuous level of 10 volts. Source Impedance: Designed to work from 4-16 ohm amplifier outputs. External Power Requirements: None, except when used for precise low level signal measurement, when external ac line can be selected by a front panel switch on the E-9 Energizer.

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nificant amount of material in the four-channel medium, there is little likelihood of this form of reproduction becoming universal.

Next, the cost investment between two and four channels will be greater than the changeover from mono to stereo some years ago. Remember also that f.m. stations have never produced good quality stereo and therefore we can assume that there will be proportionately less who are able to provide worthwhile four-channel sound.

Let us now consider what you get with four-channel operation. In the case of classical music, reverberation is the sole ingredient, and this can be obtained with almost the same effect using reverberation equipment externally. Such equipment has been available for some years without becoming very popular, and we see no reason why using up the full width of ¼-inch tape in one direction, and doubling tape costs merely to provide the same facility should be regarded as exciting.

In the case of light music, there has been a move to write special arrangements for the possibilities of four-channel sound, and these performances are definitely spectacular. We do not feel, however, that the limited repertoire of such material, plus the extremely high cost of going into four-channel work, can be regarded as justifiable either financially or operationally. Bear in mind that speaker placings are far more critical than with conventional stereo if any real effect is to be enjoyed.

Do not think that we regard all new developments as fads; each is equated in the light of its own reality. The Revox-Studer organization is continually researching and evaluating new techniques in the audio field. Its lead in many types of studio and broadcast equipment provides a unique position in this respect, and will enable us to be in the forefront of whatever direction the industry may take. Existing two-channel stereo, we feel, still has a long way to go before the quality enjoyed by the home user is anything as good as that available to the professional. Loud speakers and cartridges, in particular, still need considerable improvement before they can catch up on the sound of a master tape played through monitoring equipment.

We await with interest the results of the work done by Peter Scheiber. If his projected method of utilizing conventional two-channel equipment for four-channel reproduction is valid we will naturally endorse this arrangement — provided that the high standard of quality which is available with conventional two-channel operation is not degraded in any way.

Colin Hammond
ReVox Corp.
Roslyn Hts., N.Y.

The Audio Engineer's Handbook

GEORGE ALEXANDROVICH

A PRACTICAL APPROACH TO FILTER DESIGN

●We are continuing our discussion on equalization of sound reproducing and reinforcement systems in order to obtain uniform acoustical frequency response. We will now look into the design and use of simple filters, the kind that anyone with a box of assorted components can put together. With all respect to the theoretical and analytical approaches to designing filters (the kind that can be found in any text book), and which unquestionably is the most accurate method, I doubt that anyone willing to build filters for a system himself would spend days going through the calculations—even if his knowledge of math is still fresh in his memory.

My goal is to present a shortcut to figuring the needed values of components and their proper interconnection. Fortunately (or unfortunately), engineering today boils down to an ability to read and interpret tables, nomographs, formulae, and the vast amount of solutions to the arising problems being accumulated as a result of the continuous flow of information from research, technical papers, articles, etc.

Aside from the fact that very few of us have time to go through the necessary calculations, many lack a clear understanding of what has to be done and why. This discussion is directed for the benefit of those who are responsible for the design, construction, or maintenance of sound systems and who are working on a limited budget and want to improve the performance of the system.

FILTER COMPONENTS

To build filters, we need resistors, capacitors and inductors.

Resistors (composition) present equal resistance to all electrical currents regardless of frequency, polarity or phase. Resistors are linear within the range in which they are designed to perform. Their limitation is in the area of power dissipation, or the amount of heat a resistor can dissipate without changing its characteristics. A resistor by itself doesn't introduce changes in frequency response, phase, or distortion. It is a *passive* element. Wirewound resistors and some kinds of carbon and metal-deposit types have inductance which should be taken into consideration for the circuit design. Resistance is, of course, measured in ohms. Other practical units are k ohms (thousands of ohms) and m ohms (millions of ohms).

Capacitors. Capacitors or condensers are used in circuits to perform several functions: to isolating direct currents from alternating, affecting frequency response, and shifting phrase. Being a reactive component, a capacitor possesses the ability to offer infinite resistance to direct currents while passing alternating currents, offering more resistance to low frequencies and progressively less to higher frequencies. Capacitors designed to work while being biased with d.c. are called electrolytics. They are polarized and the amount of voltage they can safely handle is specified. Non-polarized types are numerous; starting with a small capacity air-dielectric type followed by paper, Mylar, glass, ceramic, and others. Some capacitors (like resistors) can be inductive because of their construction. For critical applications, this inductance should be considered or non-inductive types should be used. Capacitors can change phase and frequency response when used in conjunction with resistors or inductors. They are used in almost any circuit in existence. They change the phase so that current through the capacitor leads voltage by 90 degrees (an ideal capacitor). Capacitor values as they appear in formulae are in Farads. More practical units are μF (millionths of Farad) or $\mu\mu F$ or pF (pico Farads) millionths of a μF.

Inductance is the second reactive element—the counterpart of the capacitor. It passes direct currents while resisting alternating currents. With

Similar to a two-way speaker system, the total response range of a two-way microphone has been subdivided between a high frequency and low frequency transducer with the cross-over at 500 Hz.

The basic principle is ideal. It took the electro-acoustical competence of AKG to make it possible. It represents the most significant advancement in cardioid dynamic microphones.

The results are performance characteristics formerly unobtainable in cardioid dynamic microphones.

In practical terms this means:

- Natural, objective recordings without discoloration of sound reaching the microphone off-axis.
- More gain before feedback.
- Greater intelligibility and "reach" without deterioration of signals reaching the microphone off-axis.
- Because of elimination of proximity effect there is no rise in bass response to cause feedback or loss of clarity.

Illustrated is the D 200 E, adjacent to its components. Suggested retail net $69. Write for complete technical description of all **AKG** Two-way Microphones. AKG-29

*U.S. Patent #3,204,031

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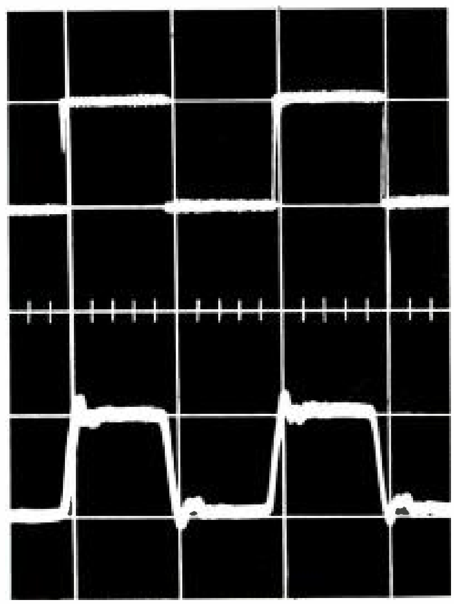

alternating currents resistance rises with frequency. Similarly to the capacitor, the inductor changes the phase of the current flowing through it, except that the current lags by 90 degrees (again an ideal indu tor). Inductance is usually a coil wound within laminated metal or powdered iron core. The core increases inductance depending on the permeability of the material. For the same number of higher turns, inductance with the same wire resistance is obtained. Inductors are characterized by their inductance, current carrying capacity, maximum voltage and Q. Current and voltage limitations are a function of the wire size and insulation, while the inductance and Q are electrical characteristics of the device. Inductance represents dynamic resistance of the coil to alternating currents (this changes somewhat with frequency, therefore it is measured and specified at the standard frequency of 400 Hz). Q of the coil characterizes the internal losses of the coil due to ohmic wire resistance and other losses. If ohmic resistance of the wire is eliminated through supercooling, current can be made to circulate in the shorted coil indefinitely without losses. Q(which is the ratio of the reactance over the resistance) becomes infinite when resistance is infinitely small.

Inductive Reactance $X_L = 2\pi fL \qquad Q = \frac{2\pi fL}{R}$

$\pi = 3.14$
f = frequency in Hz.
L = Inductance in Henries
R = Resistance in ohms

The combination of an inductor with a capacitor is called a tank circuit and is distinguished by its ability to "resonate" at one specific frequency. An inductor with low Q will produce resonance with a broad peak. Low Q inductors are generally coils wound on coil forms without the metal core. Q figures are considered low if they range below 10. Figures of tens or hundreds are harder to obtain and are considered high. The Q in filters is generally kept high (it is easy to lower it) so that when equalizing out a sharp peak, little effect is produced at neighboring frequencies.

If we are equalizing a broad peak, it would make sense to match the Q of the filter to the Q of the peak, especially knowing that higher Q inductors cost more. Inductance is rated in Henrys, mH (milliHenries) and μH (micro Henries).

Now that we have refreshed our memories on the basic qualities of resistors, capacitors, and inductors we will spend next month seeing what we can do with them in circuits for filters and equalization.

The Professionals.

Bill Bell is known as "The Ear." He's the owner of Bell Sound Studios, Hollywood. Bill does commercials, some of the best. You've heard a lot of them. He orchestrates each one, every element of sound from the soft spoken solo voice of Marvin Miller to the high dB blare of acid rock.

Bill's fussy about sound, and so are his engineers. So are the advertising agency production men, the creative people and the account executives. If you're going to take three or more hours to get the right sound in sixty seconds of commercial, you want to make sure the sound is the best possible.

So, as a starter, Bill uses Altec "The Voice of the Theatre"® speaker systems, sixteen of them. And in his custom consoles, Bill Bell also uses Altec audio controls. Again, because he thinks they're the best. After over thirty years of developing sound systems for the broadcast and motion picture industries, that's a nice reputation for Altec to have.

Specifically, each new Third Generation Voice of the Theatre speaker system features a 15" low frequency speaker that's the finest made. It has a 10½ pound magnet structure, a cast aluminum frame and a 3" edge-wound voice coil of copper ribbon. All this—plus the efficient new Symbiotik™ diaphragm—provides outstanding bass and transient response, and as much as two times more power handling capability than previous designs. The 18" massive cast-aluminum sectoral horn has a very wide sound dispersion angle at all frequencies.

The driver. It works from 500 to 22,000 Hz in the A-7-500-8. (The A7 crosses at 500 Hz.) It's so efficient, there's need for only one crossover in the system which eliminates those high frequency peaks and dips. All this means the crispest, cleanest, most undistorted sound you can get from low end through the high.

For complete specs on our sound equipment, just write to Altec Lansing, 1515 So. Manchester Ave., Anaheim, California 92803.

ALTEC LANSING

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The Feedback Loop

ARNOLD SCHWARTZ

● Phonograph cartridge specifications are not always clearly understood, and often are not properly related to cartridge performance. When I was a section head in charge of disc recording research at CBS Laboratories we had a continuing cartridge testing program. By this means we could select the best possible pickups for our research, and also stay abreast of current playback technology. These objective evaluations enabled me to compare cartridge performance with manufacturer's specs.

One of the most widely published figures-of-merit is *compliance*. Not too many years ago compliances of 1.0 to 2.0 x 10^{-6}cm/dyne were considered adequate. Today we have cartridges with advertised compliance as high as 45 x 10^{-6}cm/dyne, and values of 20.0 to 35 x 10^{-6}cm/dyne are common. What does compliance mean? Does higher compliance mean less distortion? What about tracking force and record wear? I would like to answer these questions, and to place compliance in its proper perspective in evaluating cartridge performance.

COMPLIANCE

Tracking refers to the ability of the playback stylus to maintain contact with the groove walls. Loss of contact results in distorted output. Compliance is the quantity that is usually associated with tracking and is defined as the displacement of the playback tip per unit force applied to the tip. Mathematically, compliance (C_m) is defined as follows: $C_m = \frac{x}{f}$ where x is the displacement of the tip in centimeters, f is the force in dynes applied to the tip, and compliance is measured in centimeters per dyne. In the *CGS* system of measurement a mass of 1 gram exerts a force of approximately 1,000 dynes. *Stiffness*, a quantity that may be more familiar, is the reciprocal of compliance, and is defined as the force required for a unit displacement of the tip.

We will attempt to visualize what compliance means by referring to FIGURE 1. The undulation of a stereo groove is shown here viewed 45 degrees to the record surface for ease of explanation. The stylus tip in the mean or unmodulated position (see left side of FIGURE 1), rests against the groove wall with tracking force perpendicular to the groove wall (see arrow). The relatively rigid stylus arm is held in place by a compliant mounting at one end; the diamond tip is mounted at the other end. The compliance at the tip is determined by the mounting of the stylus arm. As the groove moves past the tip the stylus arm and the tip are deflected up and down about the mean position. When the tip is deflected upward (see right side of FIGURE 1) an upward force is generated in opposition to the downward tracking force. The higher the compliance the less the upward force for this given deflection. Conversely, the less the compliance (i.e. stiffer) the greater the upward force for the same deflection. If the upward force generated by the deflected tip is greater than the downward tracking force then the stylus loses contact with the groove wall, and the out-

THROUGH THIS DOOR

PASSES THE INDUSTRY'S MOST COMPACT 16 TRACK RECORDER WITH LOTS OF ROOM TO SPARE, AND LOTS MORE TO OFFER!

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Added to this space economy, studios continue to enjoy the top performance that has made 3M's professional audio recorder a new standard in the recording industry. In the last three years, playback pre-amp and overdub noise has been improved by 10 db. Mincom's exclusive Isoloop® tape drive utilizing a differential capstan maintains constant tape tension in just 3½" of unsupported tape. This results in the lowest wow, flutter and scrape in the industry. Erasure has been improved by 10 db.

Plus: DC transport and dynamic braking, advanced electronics, unequalled no-stretch no-snap tape handling, stable and rugged 1.5-inch cast top plate—and *Remote Sync Control, 7"x7"x4", included at no extra cost.*

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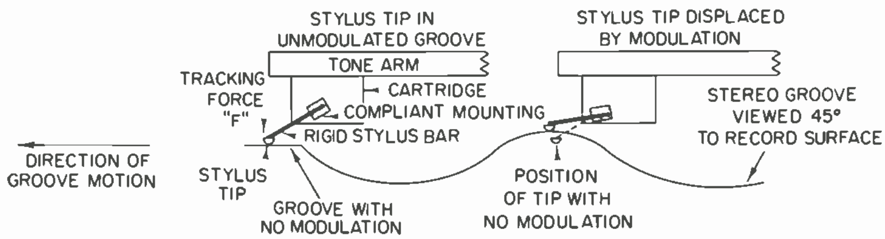

Figure 1. Deflection of the stylus tip caused by stereo modulation.

put waveform becomes distorted.

COMPLIANCE AND TRACKING

A specific example should clarify the concept of compliance. A major cartridge manufacturer claims that their best cartridge will track a *22* cm sec, 400 Hz recording. We can make some compliance calculations using our formula by taking this level as an example of what we can expect at the highest modulation levels, and converting the recorded velocity to its equivalent amplitude. Peak amplitude at this velocity is slightly less than 0.009 centimeters. If we use a 1.0 gram tracking force the cartridge would have to have a minimum compliance of 9×10^{-6} cm/dyne. If 2.0 grams were used with the same compliance we would have a safety margin for tracking even higher modulation; at 2.0 grams no increase in distortion would be expected. Increasing the tracking force to 2.0 grams would, however, reduce the minimum compliance requirement to 4.5×10^{-6} cm/dyne to track the 0.009 centimeter amplitude cited above.

Compliance and recommended tracking-force data published by cartridge manufacturers do not correlate very well. For example, one high-quality pickup is listed as having a compliance of 12×10^{-6}cm/dyne and a recommended minimum tracking force of ¾ gram. A second high quality pickup (of different manufacture) is listed as having a compliance of 45×10^{-6}cm/dyne and the recommended minimum tracking force is also ¾ gram. Here are two cartridges with the same minimum tracking force but whose compliances vary by almost four-to-one. It seems that either the manufacturer of the second cartridge is giving us too much of a good thing, or there are other unspecified criteria for low tracking force.

COMPLIANCE AND DISTORTION

Compliance by itself does not reduce distortion. A highly regarded consumer magazine, in an evaluation of phonograph cartridges, discussed the merits of a highly compliant cartridge that tracks at ¾ gram. The magazine implies that we can expect *reduced distortion* because of low tracking force. This is not correct. Compliance defines the minimum force needed to track the highest recorded amplitude expected on the record. All other things being equal, any cartridge, regardless of its compliance, can play back high amplitude grooves without mistracking if the tracking force is appropriately adjusted.

COMPLIANCE AND RECORD WEAR

Higher compliance allows the user to track his cartridge at lower force without mistracking at high recorded amplitudes. Does lower tracking force reduce record wear? Record wear can mean two things; wearing out (erasure) of the modulation, or deterioration of the groove surfaces.

Records are most vulnerable to erasure at high frequencies where groove curvature is high. At these frequencies recorded amplitudes are low, and the resulting force of the stylus against the groove walls due to stylus stiffness (that is lack of compliance) is not significant. However, effective mass at the stylus tip is an important consideration but we will have to defer discussion of this point to a later article.

Deterioration of the record grooves, which increases background noise and causes ticks and pops, is due to improper record handling, dust in the grooves, and imperfections in the polished diamond tip. This latter point is most important since a minute roughness of the tip may cause a minute amount of material to be gouged from the groove walls. Irregularities in the groove as small as 15 micro inches can cause high frequency noises only 20 dB down from standard recording level. Reduced tracking force can help reduce this type of wear, but is by no means the only factor to consider.

A more fundamental question may be asked. Is record wear a serious problem today? The quality control department of one of the large record companies has estimated that of all records returned to the customer complaint department less than 1 per cent of the complaints are for the "record wearing out" too quickly. Don't forget, this includes people that have record players tracking at 3 to 20 grams as well as those with quality players tracking at ¾ to 2 grams. This may suggest that record wear is not as serious a problem as may have been supposed.

CONCLUSION

In the seemingly accelerating race for higher compliance there is a tendency to overlook other equally important cartridge characteristics. In some of the current cartridge designs the highly compliant mounting of the stylus arm, while giving us a very high compliance, necessarily reduces the damping effect of this very same mounting. As a result the high frequency stylus-groove resonance becomes more pronounced. In these cases flat frequency response has been sacrificed for very high compliance—even though this latter quantity may not affect record life nor improve any of the cartridge performance characteristics.

Many readers do not realize that they can also be writers for db. We are always seeking good, meaningful articles of any length. The subject matter can cover almost anything of interest and value to audio professionals.

Are you doing something original or unusual in your work? Your fellow audio pros might want to know about it. (It's easy to tell your story in **db.**)

You don't have to be an experienced writer to be published. But you do need the ability to express your idea fully, with adequate detail and information. Our editors will polish the story for you. We suggest you first submit an outline so that we can work with you in the development of the article.

You also don't have to be an artist, we'll re-do all drawings. This means we do need sufficient detail in your rough drawing or schematic so that our artists will understand what you want.

It can be prestigious to be published and it can be profitable too. All articles accepted for publication are purchased. You won't retire on our scale, but it can make a nice sum for that special occasion.

Theory and Practice

NORMAN H. CROWHURST

● Nowadays enough questions come in to keep me busy writing explanations in this column, without having to fall back on odds and ends. One reader wants to know about the electrical and historical (practical) explanations for balanced mike and line techniques. He further asks, "Which way is common practice headed, towards single-ended, or balanced, or both?"

The electrical and historical origins of balanced line started with telephone usage, over long distances. By balancing the line to ground, and then transposing the positions of pairs of wires at regular intervals, in the once familiar overhead line system, crosstalk was minimized. In the newer, and almost universal underground cable, the individual pairs are continuous twisted, to effect the same result.

Mutual coupling between two circuits can occur due to one or both of two effects: inductive or capacitive. The twisting or transposing will neutralize any inductive coupling, due to signal current in each pair of wires. But if voltages conveyed are unbalanced, capacitive transfer can still occur, because one side of each circuit is live, and the other one at or close to ground potential.

By keeping the voltages and current both balanced at all points along all lines, both forms of transfer are minimized and can theoretically be completely neutralized. This matter also involves working impedance, governed by the physical dimensions of the line, whether it is open wire or a twisted pair in a cable.

To avoid electrical reflections at the end of the line, which can produce echo and ringing effects, the line should be terminated with a resistive impedance to match the natural value, which is usually about 500 ohms.

Working at this impedance, the tendencies toward inductive and capacitive transfer are usually about equal, because the natural impedance is governed by the electromagnetic properties of the line, involving its inductance and capacitance per unit length of line.

That's the story, as far as connections to long lines is concerned. The telephone company will insist on balanced, matched lines. If you happen to use lines of comparable length privately, you will find it essential, not only to avoid picking up crosstalk from other circuits, but also to minimize hum pickup from power lines or distribution systems.

In the days when tubes were the only active elements used in audio circuits, the only economical and efficient way to match a 500-ohm line was to use a well-designed audio line transformer. Nowadays, using solid-state devices, it is impossible to match a line and achieve balance without such expensive components.

In those early days, some of the

"best people," for the sake of simplicity and complete interchangeability, went to balanced line for all interconnections, short or long, including microphone inputs. While this was a costly procedure, they found that trying to use mixed impedance could prove more costly in a different way: they were apt to arrive on a job with parts that wouldn't work together! Thus came the decision to standardize.

That was fine for p.a. and other commercial installations. But for studio equipments, it was unnecessarily costly. Against this, at least for the studios with money, is the fact that having the best in the studio pays, wherever else a system may try to cut costs. But as the industry grew, there were also budget studios: people who didn't have the funding of a network or a big record company, behind them.

For these people, the natural laws of economy came into the picture again. For the many interconnections in a console, the convenience of line impedance can be met by using cathode-follower outputs, with high-impedance inputs.

The cathode follower holds the line impedance down, to avoid capacitive crosstalk or pickup, while the high-impedance input avoids loading the cathode follower to cause distortion—a point we have discussed before—with the side advantage that the quality of this arrangement can be higher than audio line transformers.

Finally, enter solid-state devices: transistors. Now line impedance, however achieved in individual circuits—and we've discussed this earlier too—is a natural; transformers are no longer necessary to economic or efficient matching.

For short interconnections, around a console or studio, there is little point in going to balanced line. That is an unnecessary refinement. It is simpler to use single-ended, with concentric (shielded) interconnections, which exclude capacitive transfer or pickup, and also inductive at the same time, provided proper grounding procedures are observed (another matter we have discussed before).

But for long line lengths, there is still no substitute for balanced lines. It would not be economically feasible to run multiple shielded lead clear across the United States, for example—just figure the cost!

Of course, microwave links are serving more and more of the uses at one time delegated to long lines so, in this sense, possibly the balanced line is on the way out, purely quantitatively. But any time a long line is needed, it will still have to be balanced, and matched.

We will continue this discussion in columns to come.

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The Sync Track

JOHN M. WORAM

ANYONE FOR TWO TRACK?

● The advantages of the big multi-track tape recorders are by now well known to almost everyone. There are, in fact, some folk who won't come near a studio that doesn't have at least one 16 tracker in the house. The idea is that everyone (almost) can have his own separate track. This is a wonderful tool—sometimes. Sometimes it's not so wonderful — such as when a producer shows up with a seventeen-man group and gets the sweats because he has to combine something on the session. And then you have to convince him that he doesn't really need seventeen tracks. He looks at you with that "You just don't care about my group" look. If you want to *really* make an impression, suggest doing the whole session on two tracks!

And here's where we very quickly discover one of the disadvantages of 16-track recording. As soon as a studio gets one of these $25,000 beauties, everyone forgets about everything *but* 16-track recording. The 8-track studio down the block is regarded with something approaching amused tolerance, and 2 track isn't even discussed anymore.

This is probably a big mistake, since there are, at times, some advantages to *not* recording multi-track. For example, 16 tracks of separation is 16 tracks of tape hiss. If you don't absolutely need the separation, you certainly don't need the hiss. And there are cases when separation actually works against you. If you record a piano on one track only, all the post session manipulating in the world isn't going to make it sound as room filling as a little leakage would. An electric guitar may be a point source of sound, but a grand piano certainly isn't, unless its recorded on just one track. Of course, if you can spare two tracks for piano only, that's just fine. A Hammond organ is a little closer to a point source of sound, yet it often sounds noticeably better when miked both close up, and at a distance—again using two tracks.

A studio that can come up with a good tight drum sound is well on its way to success, yet the multi-track studio is witnessing a trend toward two, and more, tracks for drum pickups. And then there is the sweetening session, with a large string and/or brass section. Again, two tracks sounds so much better than one.

Obviously, in an 8-track studio you can't record piano, organ, drums, strings, and brass each on two tracks. Even with 16 tracks, you're not leaving yourself much room for guitars, percussion, chorus, solos, what-nots, and afterthoughts. So, something has to give. At one extreme, you confine each instrument, or family of instruments, to its separate track, giving you a collection of point sources. At the other, you mix everything on the actual session down to two tracks (Just as in the old days). Even a split session can be done in this way. The rhythm section is recorded as a homogeneous unit onto just two tracks. Later, the sweetening is also recorded on two tracks. Of course it's not the same two tracks. But it is basically, a two track concept, and you get the advantages of both mini- and maxi-track philosophies. (You'll also get some producers who won't come near you again.)

It's always a little strange to see a big pop orchestra recording onto 16 tracks all at once, just so that later on you can go to a re-mix room and do the balancing that could have been done better in the studio.

Engineers who record the classical repertoire may by now be wondering why this column was written, since much classical recording is (more or less) a two-track pickup plus a center-fill and perhaps a fourth track for highlighting certain soloists. Balancing is done before the microphone, rather than after the session. And some excellent masters have been made that are in no way inferior to a multi-track recording. The two-track session generally suggests a more distant mic pickup than the same session recorded multi-track. The result conveys more of a feeling of spaciousness than the up-tight pickup, and is most likely inappropriate for recording a rock rhythm group of guitars, drums and bass. Today's tastes usually require these instruments to be miked very close, with maximum separation between microphones.

However, once the rhythm section has been laid down, you don't have to continue on the separation kick. The remaining instruments can be miked in a more classical way (assuming room acoustics are suitable), and perhaps recorded at once onto two tracks.

So, the point is: although multi-track recording is a must for *some* recording, it is not a must for *all* recording. Before the session, consider the advantages of using a few less tracks sometimes. Of course, it cramps the style of the *We'll fix it in the mix* set, but it does steer you into thinking in more musical terms, which after all is what the whole thing is about anyway. And at times a judicious mixture of multi-track, plus two-track, technique on the same piece of tape, may sound superior to a recording made exclusively in either mode of operation.

Automated Radio Broadcasting

GENE HOSTETTER

This article presents a brief discussion of the state of the art of this controversial subject. It should help to dispel the fear that exists at some places that automation is either too complicated or will eliminate jobs. Neither is true.

Contrary to the impressions most people have when they first view an automated broadcast station in operation, there is no magic and very little "brain" involved in even the most complex operation of today.

MUSIC SOURCES

Reel-to reel tape recorders are usually used as music sources because of their relatively large recording storage capacity. Automatically playing records directly on the air does not allow the records to be easily announced and leaves a greater possibility of mechanical error such as a warped record jamming the mechanism or the needle hanging up in a groove. Selections from record players are limited to 45 rpm records or to entire 33-1/3 rpm album sides. There are, however, some 45 rpm jukebox-type mechanisms in use in broadcast automators, particularly in areas and at times where it would not ordinarily be economically feasible to employ more complex systems. Also, some record changers are in use at stations where entire album sides are to be played, as with a classical music format or, more recently, at "underground music" stations. The owner of such a station must be willing to tolerate a light, often erratic, schedule of announcements since album sides run from about ten to twenty minutes in length.

In order to air a variety of events from several sources, it is necessary to index or *cue* each selection from each tape source. Cuing is accomplished on reel-to-reel tape machines by recording a low-frequency tone (between 20 and 35 Hertz) between each selection. The low tone is inaudible, yet capable of triggering electronic circuitry which stops the tape at the end of the selection, readying it to start later at the beginning of the next selection, and triggering the start of some other program source.

Normally the tapes used have two tracks. That is, it is necessary to use only half of the width of the tape for recording as in most home tape recorders. If the station broadcasts FM stereo, both tracks—one for each ear—are used for each selection. A reel of tape is loaded on the machine, eventually plays its so many selections and runs out. It must then be rewound and rethreaded, or replaced with a fresh tape. If the programming is not in stereo, only one of the two available tracks is played back at a time. One track may be played as the tape is travelling from head to tail end. Then, if the direction of tape motion is reversed, the second track may be played as the tape travels from tail back to head. Automatic tape direction reversal can be implemented by placing strips of electrically conducting foil or paint on the tape. The tape rolls over contact bars and the conducting foil or paint completes an electrical circuit, keying reversal of the tape motion. If conductors are placed at each end of a reel of tape, the tape can be made to play upon command continuously from one end to the other, back and forth until it is replaced. Such a capability is important when the system is left unattended for long periods of time or when the operators cannot be depended upon to rewind and reload tapes on schedule without fail.

Other cuing or reversing schemes are sometimes used. One method *silence sense*, which is most often used for jukebox and record changer music sources, depends upon a period of silence between selections or at the ends of the tape. When an electronic circuit detects a sufficiently long period of silence, the tape is stopped (or the record rejected) and a signal is given to start the next program source. Unfortunately, some musical selections contain pauses which occasionally trigger the circuitry prematurely. There is also the undesirable side effect of the period of silence on the air before the next program source can be started. Conducting strips between selections on the tape can be used for cuing, but then the recording or copying of tapes becomes quite time consuming. Instead of the conductor, the oxide coating may be removed from a segment of the tape, making it transparent. An electric eye can sense these segments and initiate cuing or reversal, but like foil cuing, photoelectric cuing between cuts is time

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consuming and thus costly.

CARTRIDGE SYSTEMS

Another programable audio source for the broadcaster is the continuous-loop tape cartridge. Most all radio stations—automated or not—use these cartridges for the airing of announcements. The operator plugs the appropriate cartridge into a playback machine and, when ready, starts the tape drive to play the announcement. As with the reel-to-reel tape machines, two tracks are available on the tape to record information. One of these tracks carries the recorded announcement and the other is left blank except for a short period just before the start of the announcement. A tone is recorded on the second track during that time to indicate when the tape should be stopped (cue'd) in order to be ready to play the announcement again. Thus, when the tape cartridge is started, it plays the announcement and then keeps on running until the cue tone on the second track is sensed, at which time the tape is automatically stopped just before the beginning of the next announcement. Of course, more than one announcement may be recorded in this manner on the cartridge and each will be ready to play in sequence after the last, so that several commercials for the same sponsor may be automatically rotated throughout the day.

The low-frequency tone method of cuing which is commonly used on the same track as the program material on reel-to-reel tapes is not practical for cartridge tapes because the cartridge is not always capable of sufficiently smooth tape motion to reproduce such low tones. When stereo announcements are desired, the ¼-inch tape used in the tape cartridge is usually divided into three rather than two tracks, two for the two stereo channels and the third for the cue tone.

Since the tape cartridges are manufactured with fixed lengths of tape in the loop, more often than not there will be silent segments of varying length between announcements. When tape cartridges are used in automated broadcasting, it is desirable to start the next program source not when the cartridge stops on cue at the start of the next announcement, but before that, when the previous announcement has just ended. Separate tones are used for the cue and *transfer* functions. (A tone of about 150 Hz indicates that the next program source is to be started, while a tone of about 1000 Hz will cause the tape to be stopped, ready for the next announcement.)

While it is possible to record in order all the announcements for a broadcast day on a single reel or cartridge tape, there are several disadvantages to this concept. First, a single error in recording or playback will throw the entire announcement sequence off. Second, a considerable amount of time is required each day to do the recording job itself. At a live rather than automated station, most commercials these days are recorded in advance and played many times before being discarded anyway, so the compiling of all the announcements for a day on a single tape is largely the busy work of putting the already-recorded commercials in the desired order. There are several options open which offer improvement over the daily recording of an entire announcement sequence. Some are essentially cartridge jukeboxes. Two such machines are the *Automatic Tape Control '55'* and the *MaCarta Carousel*. In the 55, cartridges are stacked in slots, one above the other. A single tape drive and playback mechanism is mounted on motor-driven feed screws which allow it to be positioned opposite any one of fifty-five cartridges, to play that cartridge upon command. The *Carousel* is a smaller, less expensive machine in which twenty-four cartridges are arranged in trays as the spokes on a wheel. The playback mechanism is stationary and the trays are rotated into position for playback. The basic models of these two machines are wired to play one announcement from each cartridge in sequence. When the first cartridge has played and stopped, the next available cartridge is made ready for play upon command. The already-recorded announcements need only be loaded in the desired order for playback. More complicated versions of these multiple cartridge handlers have *random selection*, the capability of selecting a particular sequence of cartridge play, eliminating the need for using several copies of the same cartridge if the same announcement is to be repeated several times before the machine can be reloaded.

Figure 1. A commercial automation system with two reel machines, a single cartridge player, and two Carousels. This unit was built by Continental Electronics and is installed at KOL-FM, Seattle, Washington.

Other systems employ reel tapes on which the announcements have been recorded. Random selection equipment allows any present sequence of play.

SIMPLE AUTOMATION SYSTEMS

Perhaps the most simple automation system is one in which a series of program sources are connected in a chain so that a transfer from one source causes the next source in the chain to be started. Such an arrangement is only workable when the timing of each of the aired recordings from each of the sources can be carefully planned so that the number of commercial announcements to be played can be controlled. Also, station-identification announcements must be made to play as near as possible to the hour and half hour as required by law. Little flexibility is possible. The same rotation of playback machines repeats again and again, with each machine playing only once per rotation through the sequence. These shortcomings can be eliminated by the addition of a *controller*. A good controller, or "brain", is capable of selecting the order

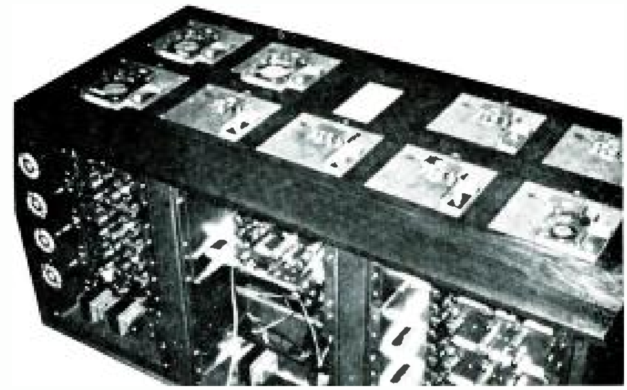

Figure 2. An automated system, designed by the author, using cartridge machines.

of machine playback, based upon a pre-selected sequence controlled by time clocks.

For example, a good music station desires to program four 12½-minute blocks of announced music per hour with an average of 2½ minutes of announcements between each block of music. Because it is difficult to make each music block exactly 12½ minutes in length and because the series of announcements between the music may not always total 2½ minutes, each music segment is ended with fill music. The last musical selection in each block, starting before 12 minutes of the recording have passed and ending beyond the 13 or 14 minute mark, is especially chosen so that it can be inconspicuously faded out at any point during the playback. A time clock is used to initiate the music fade out (although the music tape runs to cue) and start the announcement tapes at :00, :15, :30, and :45 of each hour so that timing errors do not accumulate. A second clock is used to insure that the station-identification announcements are played as close to the hour and half-hour as is possible. Additional clocks may be used to determine which of a number of possible music tapes are played at various times of the day. By classifying tapes by tempo, the tempo of the music at each time of the day may be controlled.

CONCLUSION

The ultimate objective of broadcast automation is to improve the sound and quality of a station by reducing both on-air and behind-the-scenes errors, and by freeing station personnel from tedious minute-by-minute tasks for more productive utilization. Most of the present general purpose systems are highly reliable.

Having decided to employ automation, management must choose a philosophy for its operation. How near to complete self-sufficiency should the system be and how much should the operators be bent to the machinery. If enough machines were available for the entire music library, it would not be necessary to change music tapes. Nonetheless, it is normally necessary from an economic standpoint. The question is: how much tape-changing should be done in a given operation? Hourly? Twice a day? The same sort of questions arise in a host of other areas. How much time should be devoted to announcement setup? Should newscasts be live or should they be taped for airing?

The answers to questions such as these will in turn determine the type of system best suited for a particular station's operation.

Stereo Synthesis Systems

ROBERT ORBAN

The two products described in this article have been designed to function in the gap created by the diverse needs of mono and stereo production and listening.

A NEW CLASS of audio signal-processing devices has recently been introduced to the professional audio market. The devices are the Stereo Synthesizer and the Stereo Synthesizer-Matrix. Just as a compressor controls dynamics and an equalizer controls frequency response, this new class of processor controls stereo space.

The first device to be developed was the Stereo Synthesizer. Its development was prompted by the need to develop realistic-sounding stereo from mono originals. Most techniques in use at the time of its development were unscientific, witches'-brew techniques whose results were, at best, unpredictable. The stereo synthesizer was therefore developed around the following two criteria: *first*, the electrical sum of the output channels was to be proportional to the mono input; *second*, the sum of the power spectra of the left and right channels was to be proportional to the power spectrum of the mono input.

The technique of Bauer[1], using pure phase shift between the channels, met these criteria. However, the Bauer technique had the disadvantage of being largely ineffectualized

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Robert Orban is involved in many musical and technical projects in his native California. Among them is the production of products, including those described in this article, for the professional audio trade.

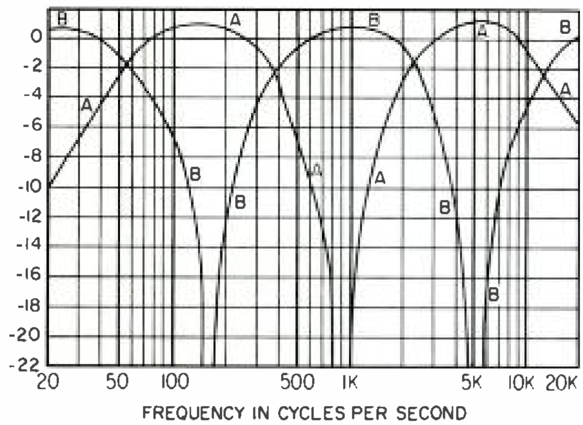

Figure 1. The frequency response of the Stereo Synthesizer at one position of the dimension control.

by reverberant listening acoustics which tended to destroy phase relationships, particularly at higher frequencies.

Nevertheless, the criteria above remain desirable. The first implies that the lateral modulation on a stereo disc, and the main channel of an f.m. stereo broadcast will both be equal to the original mono, thus producing stereo/mono compatibility. The second implies that the perceived frequency and musical balance of the stereo will be identical to the mono.

It turns out that these criteria may also be met by a system which combines frequency band-splitting with phase shifting. This is the system used by the stereo synthesizer. Frequency response of the two channels for one position of the *dimension* controls (which are frequency band pan-pots, moving bands from one output channel to the other) is shown in FIGURE 1. Listening tests revealed that the use of five bands gave highly satisfactory results for virtually all program material. The positioning of the bands is adjustable for different spectral content of different program material so that good left-right subjective channel balance may always be obtained.

The stereo synthesizer has six controls: two *dimension* controls, a *separation* control, an *input* pad, a *stereo/mono* mode switch, and a *power* on/off switch. These controls were chosen to specifically affect the stereo effect without changing any other parameters of the system. The dimension controls affect nothing but spectral balance between left and right, while the separation control adjusts the amount of stereo spread, and actually serves to adjust the amount of difference signal (L-R) appearing in the output. None of these controls in any way affect the stereo/mono compatibility, which is automatically maintained provided that the two output channels beyond the synthesizer are kept in gain balance.

Interestingly, the synthesizer has a good immunity to phase errors in the output tracks, such as errors that occur in any stereo tape machine due to the left and right head gaps not lining up precisely. Most audio people are familiar with the swishing, constricted sound of phase distortion, which occurs, for example, when a full-track mono tape is played on a stereo machine, and the stereo outputs are added together. It occurs because the spectrum cancels at the point when the tape machine's phase shift causes a given frequency to go 180 degrees (or odd multiples thereof) out-of-phase on one track compared with the other.

By splitting the sound into bands, the stereo synthesizer makes it highly unlikely that there will be equal amplitudes on left and right. Therefore, at the 180-degree points, unequal cancellation will take place, and the hole in the frequency spectrum will be much shallower. One application therefore suggests itself: the *phanto n center* may be processed through a stereo synthesizer in the course of left/center/right multi-track to two-track mixdown. In this way, phase distortion from the phantom center track (which may frequently be heard when listening to f.m. stereo stations monophonically) will be eliminated. An added bonus will be that the classic 3 dB center-channel buildup in the mono mix will be entirely absent.

Other applications of the synthesizer besides processing old mono material into stereo may be found. These include conversion of mono echo returns to stereo, selective localization and movement of instruments without pan-pots, and use of one channel only to obtain various special filter effects (including "phasing").

It is possible to realize the synthesizer with circuitry yielding harmonic distortion in the order of 0.05 per cent with an overload/noise ratio in the order of 100 dB, so that it will not cause degradation in any current application.

THE MATRIX SYSTEM

Out of the concept of the stereo synthesizer arose the stereo synthesizer-matrix. This device was designed to solve a number of problems which currently plague the pan-pot technique of creating stereo space in multi-track to two-track mixes.

Most of today's pop music has been recorded stereophonically only in name. It is a group of *mono* tracks, pan-potted across the stereo stage and smeared with echo to give some semblance of a real acoustical environment. This is a far cry from true stereo recording, where two or three microphones in the same acoustic space were used to get a true sense of acoustic environment to the listener. Yet, the advantages of the multi-track technique in terms of control of individual tracks makes it the essential technique for pop music recording today. Simultaneously, the requirement for mono compatibility causes severe problems with the classical stereo miking technique, since this technique is highly subject to acoustic phase distortion in mono mixdown.

The stereo synthesizer-matrix was designed to return to the big space of classical stereo recording, while simultaneously permitting total control and automatic stereo/mono compatibility. FIGURE 2 shows a block diagram of the matrix. It consists of four stereo synthesizers and a phase-shifter. The left and right synthesizers have had one of their output channels phantom-centered through −4.52 dB pads. When these synthesizers are fed by a left or right mix, they will spread this mix between left or right and center. The synthesizers are connected so that the low bass is phantom-centered.

Another synthesizer is connected straightforwardly to the left and right outputs. Its purpose is to spread and diffuse the echo return signal throughout the stereo space, thus giving the impression of reververation in a hall with good acoustics.

Still another synthesizer has been connected directly to the left and right. This one has been modified so that frequencies below about 250 Hz appear equally in the left and right outputs. This is to aid the mixer in keeping the bass in the center, for compatibility, and for the most efficient use of the power output capabilities of the typical consumer stereo amplifier. This synthesizer is designed to take a center feed.

Finally, a wide-band phase shift network is included. Its purpose is to handle soloists or featured instruments, when it is desired to hold these on dead-center. The center synthesizer will tend to move instruments to either right or left of center, because a large portion of its spectral output is predominantly

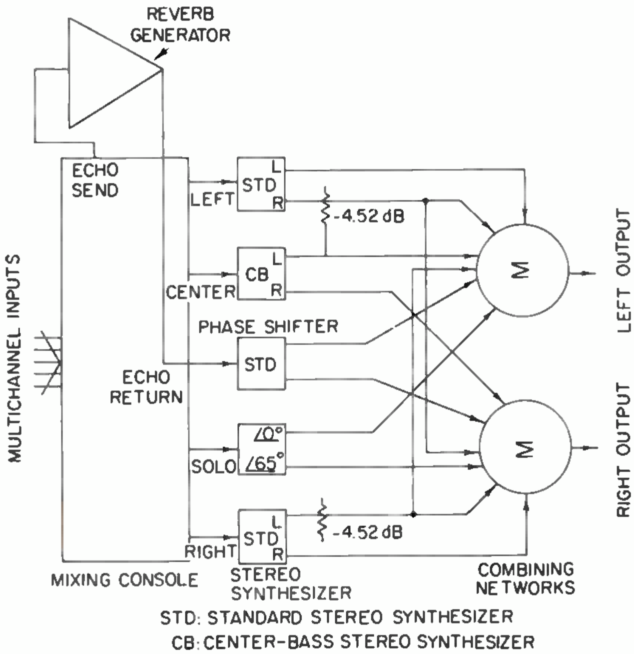

Figure 2. A block diagram of the Stereo Synthesizer-Matrix system.

left or right, as opposed to being equally split between the channels.

In the Bauer discussion on phase shifters mentioned earlier, he states that as the phase is varied from 0 degrees to 90 degrees between channels, that the apparent source of the sound moves from a point source until it fills the entire space between the loudspeakers. It turns out the stereo/mono compatibility requirements call for a phase shift of 65 degrees in the synthesizer-matrix mode of operation. Listening tests have revealed that the apparent width of a 65 degree shifted source is much narrower than a 90 degree source, so the soloist does stay well-confined to the center.

It is, nevertheless convenient and economical to provide a *center phase split* mode of operation, available by means of a mode switch. This technique for obtaining stereo/mono compatibility has been described in the literature[2]. In this mode, left and right are applied directly to the outputs, and the center input is phase-shifted 90 degrees (quadrature split), thus avoiding the 3 dB center-channel buildup in mono.

This technique has all of the spatial problems of the pan-pot technique (point source left and right), with the added problem that the apparent width of the center has been spread to the entire width of the stereo stage. The author feels that the synthesizer-matrix technique gives a far more natural spatial effect. Nevertheless, some producers may prefer the former technique, so it has been made available.

The third mode which is available is a left/right quadrature split. This is used in creating *ex post facto* compatibility from two-track mixes that have already been mixed through pan-pots. Again, it has been described in the literature[3]. Here, the phase of one track is shifted 90 degrees with respect to the other, affecting how the material common to the tracks adds up in mono. In addition to the problems encountered with the center-phase-split technique, this process also passes the left and right information through phase-shifting networks, which may alter the transient response of this material unnecessarily.

A few comments about practical operation in the synthesizer-matrix mode are in order. The multitrack master should be mixed down to four tracks: left, right, synthesized center, and phase-shifted center. All equalization, compression, etc., should be applied to the individual tracks before mixing, so that maximum control can be exercised. The four output tracks and the echo return are applied to the five inputs of the synthesizer-matrix. The dimension controls are adjusted to get the spatial effect that the producer desires. The output of the matrix is the source which should be monitored for all adjustments. In practice, it will be found that the matrix utilizes the psycho-acoustical properties of the ear far better than the pan-pot technique, so the matrixed mix will sound louder for a given peak level, and the ear will be better able to separate the various instruments. In addition, the stereo space will sound smooth, panoramic, and real. Mono compatibility will be automatically maintained for all inputs except for the reverberation input. Here, in order to get maximum field spread, it is necessary to lose 1.5 dB of reverb level in the mono mix of the stereo output. It was felt that the mono, with the constriction of its single channel, could well withstand the increase in clarity that the loss of 1.5 dB in reverb level could bring.

It turns out that material produced by the matrix technique also has distinct advantages in providing the mechanical compatibility of a stereo disc. It is well known that reducing the vertical modulation results in a disc which is easier to cut, easier to manufacture, and easier to track. However, reducing the vertical modulation has the same effect as reducing the separation in ordinary circumstances. This is not true in the case of the synthesizer-matrix. It turns out that the vertical/lateral ratio for the center inputs is a constant 0.65. For the left and right inputs, it is frequency-dependent, ranging from 0 to 1.00 and averaging (under certain assumptions) about 0.619. We then conclude that the typical vertical/lateral ratio is about 0.6 to 0.7., giving us simultaneously no loss in the width of the stereo field. The loss of different information goes instead into *filling in* the field.

In contrast, a conventional left/center/right recording gives ratios of 1.00 for the left and right, and 0.0 for the center. The center phase split technique, while solving the balance compatibility problem, makes the mechanical compatibility problem far worse: here, the left and right ratios stay at 1.00, and the center ratio has also been made 1.00, making the vertical modulation equal in magnitude to the lateral modulation under all conditions. Any attempt to solve the problem by the use of difference signal bass rolloff or elliptical equalization will only result in a different frequency balance between the stereo and the mono mix in the bass region. Since the lateral signal is usually bass-boosted to compensate for the loss of power spectral material in the vertical signal, the power spectrum of the stereo may remain constant, but the mono must suffer a bass boost.

The stereo synthesizer system, then, solves many problems of extant techniques, and, if used with taste and judgement, should further the state of the art of professional audio recording.

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Primer on Methods and Scales of Noise Measurement, Part 2

WAYNE RUDMOSE

Last month's installment dealt with the fields of measurement and the relationship of media to sound propagation

FREQUENCIES AND THEIR RELATION TO SOUND MEASURES

The range of frequencies that concern us can be classified as the sonic range (20-20,000 Hz), the ultrasonic range (20,000 Hz and above), and the infrasonic range (20 Hz and below). The problems of hearing and noise exposure are normally associated with the sonic range; jet engines, ultrasonic cleaners, etc., are common sources which generate ultrasonic as well as sonic energy; and machines with parts which rotate very slowly, as well as large rockets, are common sources of infrasonic and sonic waves.

The specification of frequencies in acoustics is usually in frequency bands, although certain sounds such as the turbine whine of our jet aircraft engines will be specified as pure tones or a one-cycle per sec (Hz) bandwidth. The reason for using the larger bandwidths is that psychophysical experiments relating psychological response to physical stimuli have shown that very narrow-band subdivision by frequency of the stimulus is not especially meaningful in relating the psychological response to the physical stimulus. Clearly there are exceptions to such a generalized statement, as evidenced by pure-tone audiometry, but for the vast majority of the cases the general statement is a reasonable description of the situation.

SOUND MEASUREMENT EQUIPMENT

The most commonly used instrument is the sound level meter. If the sound is reported as having been measured with the sound level meter, the data will most likely be reported as sound level in dB or sound level in dBA. When the sound level is reported as dBA, the frequency range should be interpreted as approximately 400-10,000 Hz. If it is reported simply as sound level in dB, the measurements were probably made on the "C" frequency scale which covers a range of approximately 25-10,000 Hz. Unfortunately these frequency ranges are not as well defined as the numbers might indicate since the response of the sound level meter does not stop abruptly either below or above the frequencies which have been listed as the frequency range. Stated another way, the sound level meter has certain simple electrical circuits which provide a degree of frequency analysis of sounds. These circuits are referred to as "weighting networks" in that they give greater "weight" or importance to sounds in certain frequency ranges than in other frequency ranges. The labels associated with these networks are the "A", "B", and "C" scales. The A scale has an upper range of about 10,000 Hz and discriminates against frequencies lower than about 400 Hz. The B scale, which is seldom used, has about the same upper range as the A scale and discriminates against frequencies lower than about 125 Hz, whereas the C scale represents little or no discrimination over the frequency range of approximately 25-10,000 Hz.

It behooves anyone interested in acoustics to study in detail the response characteristics of sound level meters so that he may gain an appreciation of the meaning and implication

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

[1]An earlier reference was 10^{-13} watt, which is now superceded. The reference is on page 24, January.

of data taken with such a device. More and more data are being reported based upon sound level meter measurements, a trend that represents a significant change in our method of measurement. Beginning with World War II a very high percentage of all acoustical data were measured with acoustical instruments much more sophisticated in frequency discrimination than the sound level meter. Although the trend of sophisticated instrumentation still continues, the sound level meter is once again considered a useful instrument for the measurement of sounds.

Let us turn to those more sophisticated instruments just referred to. They fall under the generalized heading of "analyzers." The most common type is an octave-band analyzer. An octave is any frequency range in which the value of the upper frequency of response (f_H) is twice the value of the lower frequency of response (f_L). The octave analyzers commonly used before 1960 had such frequency limits as 75-150 Hz, 150-300 Hz, 300-600 Hz, etc. This older instrumentation defined the octaves in terms of their lower and upper cutoff frequencies. Beginning almost 10 years ago the method of designating the octave bands was changed from designating the cutoff frequencies as illustrated above to designating the center frequency of the band. Unfortunately, the center frequency of an octave band is not the arithmetic mean of the low and high cutoff frequencies but is their geometric mean. The geometric mean is equal to $\sqrt{f_L f_H}$, and since $f_H = 2f_L$, this makes the center frequency numerically equal to $\sqrt{2}$ times the lower cutoff frequency ($1.414f_L$). The center frequencies of the new octave analyzers start with 1000 Hz and extend up and down in frequency. If the center frequency is stated, a quick and reasonably accurate way of determining the low frequency cutoff is to multiply the center frequency by 0.7 which is equal to $\sqrt{2}/.2$. The upper cutoff frequency of the bandwidth can then be calculated as two times the lower cutoff frequency. As an example, if the center frequency is 2000 Hz, the low frequency cutoff is $0.7 \times 2000 = 1400$ Hz; and the upper cutoff frequency is $2 \times 1400 = 2800$ Hz.

It is only natural that once the frequency domain was divided into octaves the next logical step would be to divide it into ranges of fractions of an octave. For a short period there was an "instrumentation struggle" between one-half and one-third octaves, but the battle was short-lived and the one-third octave analyzer was the victor. A one-third octave-band analyzer simply divides each octave into three one-third octaves. Thus an analyzer with 8 octave bands would now become an analyzer with 26 one-third octave bands. If one has data based on one-third octave band measurements, it is straightforward to calculate the octave-band data; however, it is not possible to calculate a one-third octave-band analysis from an octave analysis.

The development of all analyzers naturally follows the advances and capabilities of the electronics art, since the frequency bands are all obtained electronically within the analyzer. Recognizing this, one must accept the fact that in the competitive world of instrumentation other forms of analyzers will be developed according to the ingenuity of the people engaged in research. Manufacturers normally will wait until the market for new analyzers has been defined before embarking upon the development of commercially available products. This type of research and development has led to analyzers whose cutoff frequencies can be varied more or less continuously over wide ranges of frequencies. The size of the bandwidth will vary from one instrument to another, and it may or may not remain constant as a function of frequency. There are, in fact, two common types of narrow band analyzers: one called the constant bandwidth analyzer and the other a constant percentage bandwidth analyzer. A constant bandwidth analyzer has a passband which is constant and independent of the center frequency of the passband. A 10-cycle bandwidth analyzer would imply that if the center frequency of the passband is at 200 Hz the bandwidth would be 10 Hz, and if its center frequency is at 2000 Hz the bandwidth is still 10 Hz. On the other hand if the analyzer were a 10% constant percentage bandwidth analyzer and the center frequency is at 200 Hz, the bandwidth would be 20 Hz. If, however, its center frequency were at 2000 Hz, the bandwidth would be 200 Hz. It is relatively easy to explain the mathematical relationships just given, but the effect on the data obtained using these two types of analyzers is not as easy to explain.

It must be understood that the output indication of any analyzer depends upon the bandwidth of the analyzer. If one therefore measures the same sound with an octave-band analyzer, a one-third octave-band analyzer, a 10-cycle constant bandwidth analyzer, and a 10% band analyzer, the data obtained with these four analyzers would differ significantly from each other even though the noise measured is identical. If these four analyses are displayed on the same graph, the four spectral curves will be different. An untrained observer might therefore think that there were four different noises, whereas the sophisticated person would recognize these four spectral curves as representing the same noise, provided sufficient information is shown on the graph to indicate the type of instrumentation used to obtain each of the four spectra.

It is clear that as one divides the sound energy into various size frequency bands, the energy in a particular frequency band cannot equal the total energy in the sound. If a sound has equal energy in every cycle (this is called white noise), a little thought will convince you that as the bandwidth becomes more and more narrow, the output of the analyzer will decrease accordingly.

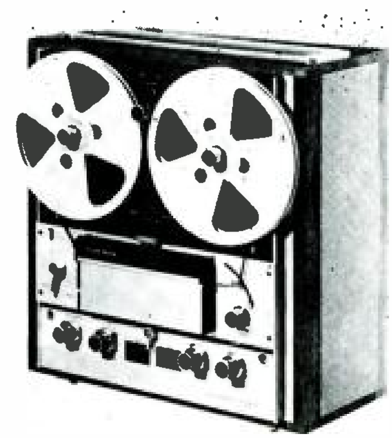
Equipment Limitations. Up to this point I have not concerned myself with the method of displaying the output of noise measuring instruments. Most of you are aware tha tthe common form of output registration is a simple meter. Data are secured by recording the deflection of the meter needle in terms of the scale on the meter, and this deflection is normally combined with the setting of a control or controls associated with the instrumentation. In most cases these controls move in fixed steps of 10-dB intervals.

The point to be stressed is that the mechanical and electrical characteristics of the indicating meter itself cannot be disregarded. If the sound is more or less steady, the characteristics of the meter have well-defined effects upon the deflection of the meter needle. If, however, the sound changes in intensity rather quickly with time, the effects of the meter's characteristics can no longer be well defined. The meter movement does not respond to alternating current (AC) signals but is responsive to direct current (DC) voltages; hence, the AC signal which corresponds to the acoustic signal must be converted into a DC signal before being applied to the output meter. The needle of the output meter can respond to changes in the DC voltage, but not if these changes take place in a time shorter than about 0.2 sec. As a consequence, the use of conventional sound level meters and analyzers must be restricted to steady types of noises or to steady noises that are not rapidly interrupted.

(Continued next month)

New Products

TAPE RECORDER



● The British-made Ferrograph tape recorders are now available in a 60-Hz, 117-volt version distributed by Elpa Marketing Industries. The Series Seven are three-speed machines with three-head facility and editing features. Full servicing and stocking facilities will be maintained by the distributor, on a national basis.
Mdr.: Elpa Marketing Industries
Price: (approx.) $650
Circle 61 on Reader Service Card

DUPLICATOR

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Mfr.: Infonics, Inc.
Circle 65 on Reader Service Card

REVERB SYSTEM

● The Reverbertron 659 features full range equalization, an improvement in s/n of 10 dB ove the previous 658A model, complete metering of all signals in the system, selection of short, medium, or long decay times, local and remote selection of three degrees of reverb, continuous mix, and plug-in p.c. electronics. A transformer isolated input and output of 600Ω or 150Ω balanced or unbalanced is provided.
Mfr.: Fairchild Recording Equipment Corp.
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Lesson Number 11

Would you pass?

Today's lesson is on responding to opening bids of one-of-a-suit.

Cover the box below with a sheet of paper so you won't see the answers to the questions until you've come up with your own answers.

You hold the hand in the photograph.

Your partner has opened the bidding with (1♠).

What do you bid?

Pass READ A — 1 NT (*Pass*) READ B — 2♥ (*3♥*) — 2 NT (*3 NT*)

What do you re-bid? (after 2♥ / 3♥)

Pass READ C — 3 NT (*Pass*) READ D — 4♥ (*Pass*) READ E

What do you re-bid? (after 2 NT / 3 NT)

Pass READ F — 4♥ (*Pass*) READ G

A) You can't pass with a hand that strong(
B) Your response shows 6-9 high-card points and is not forcing.
C) You have enough for game; don't stop now.
D) This is game, but the wrong one. You found a heart fit—that's your spot.
E) Correct bidding. Congratulations.
F) Wrong game. Your 2 NT response is to blame. You can get to no-trump later if that's right, but first you should show your good heart suit.
G) You should have bid 2♥ in the first place, because now you really took a stab in the dark by bidding 4♥. This time you were lucky — 4♥ is the right contract. But what if partner had no heart support for you?

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Sound with Images

MARTIN DICKSTEIN

LIGHT SOURCES AND THEIR COLOR

●A most important factor in the installation of the projector and its light source in any good audio-visual installation is the color response. Each available light source has a slightly different color output from any other and this is important in the fidelity of the final image seen on the screen.

Let's first take a look at the eye and its sensitivity. Most audio engineers and installers are familiar with the Fletcher-Munson curve which describes the response of the normal human ear to various frequencies and intensities. A curve is also used in describing the color sensitivity of the normal human eye. It is much simpler than the ear-curve. It looks very much like a normal probability curve with a low flat start from zero on the left, a gradual increase in rising slope to a rounded peak and a fall-off to the right resembling the rise on the left of the peak. The lower right side gradually flattens out and disappears into the zero line again. What has been graphed is the wavelength in Angstrom units on the horizontal axis with the relative response of the eye on the vertical axis.

It has been found that the upward rise starts at about 3800 Angstroms and rises to a peak of 100 at about 5550A and drops to zero again at 7600A. This plot is made under normal daylight illumination conditions, and shows that the eye is most sensitive to the green area of the spectrum. (1A equals 10^{-8} cm.) The slope of the curve is such that for the same amount of energy of 5000A light incident on the eye, the eye is only about a third as sensitive. Thus, three times as much energy at 5000A is required for the same response as for light at 5550A. It can be seen that the eye is only sensitive to a very narrow band of electromagnetic radiation ranging from violet at the shortest wavelength through blue, green, yellow, orange and red at the maximum visible wavelength.

Our principal source of light is the sun. The photosphere envelope around the sun which gives us our light is at about 6000°C. The hotter the temperature of the source in the projector, therefore, the closer will the output approach the white of sunlight.

Among the most common sources of light is, of course, the incandescent lamp. The modern lamp of this type contains a tungsten filament and is filled with a gas such as nitrogen (as opposed to an evacuated bulb and carbon filament used in the original models by Edison). The gas prevents oxidation of the filament and keeps the inside of the bulb relatively free of evaporated tungsten by preventing too rapid evaporation of the filament. Although different forms and windings of the filament are possible for special effect, the usual lamp is heated up to about 3000°C with the resulting characteristic output.

An intense source of light used a great deal in searchlights as well as in projectors is the carbon arc. The carbon-arc lamp consists of two carbon electrodes connected to a source of power through a resistor. When the carbon rods are brought into contact then separated, an arc is formed and the rods are heated by the current flowing through them. The points of separation become very hot, some of the carbon vaporizes and acts as a conducting path between the two rods. The positive terminal, white hot, is the main source of light at about 3700°C with the vapors and negative rod contributing a small amount of additional illumination. Special purpose carbon arcs are also made with the positive electrodes filled with a material other than carbon. The resulting light contains the characteristic color of the fill material superimposed on the white of the arc.

The mercury arc is another important source of light. There are two types of this lamp: low pressure and high-pressure arc lamps. In the former, the lamp contains a pool of mercury at the bottom of a glass or quartz container with two metal electrodes sealed in the ends. The pool is in contact with one of the electrodes with vapor filling the rest of the tube. By tilting the tube momentarily, the mercury makes contact with the other electrode. When the bulb is returned to the normal upright position, the vapor that has become ionized becomes the conductor for the current from the electrode at the top of the tube. The resulting emitted light comes from the positive electrode and is characteristic of the mercury source—rich in blue and violet light.

The high pressure mercury arc contains a small amount of liquid mercury which is completely vaporized by a current passing through the tube. Although the arc starts as a low-pressure type, the pressure quickly builds up to as much as 100 atmospheres within the tube. The container is always made of quartz. The temperature is about 5000°A and is a good source of ultra-violet radiation.

Although not used in projectors, it might be interesting to mention that the common fluorescent lamp is essentially a mercury-vapor lamp, made into a long glass tube with a coating on the inside. To make it easier to start the lamp, a small amount of the inert gas Argon is introduced. To light the lamp, a current is passed through the two electrodes sealed into the glass. The filaments emit electrons when hot, the electrons ionize the gas and vapor and start the arc. However, because a higher potential is needed to start the lamp than to keep it going, a special starting device is used to provide a high starting current and a lower operating potential. The principal source of light is the low-pressure mercury vapor which is in the blue-violet range. Also, a small amount of ultra-violet light is emitted. This is absorbed by the internal coating on the glass and a secondary source of light results. This secondary source can be red, orange, yellow, or almost any desired color by the use of the proper coating on the inside of the tube.

To illustrate the relative efficiency of the sources mentioned, it is only necessary to determine the output luminous flux (with a photometer) and form a ratio of the total luminous flux emitted to the power supplied. This factor is expressed in lumens-per-watt. A typical 25-watt tungsten lamp has an output of 260 lumens for an efficiency of 10.4 lumens per watt. A 100-watt tungsten lamp has an efficiency of 16.3 lumens per watt and a 1000-watt tungsten lamp with an output of 21,500 lumens has an efficiency of 21.5 lumens per watt. A 1000-watt carbon arc has an efficiency of 60 lumens per watt, a 100-watt high-pressure mercury arc has an efficiency of 35 lumens per watt, a 1000-watt lamp of the same kind has 65 lumens per watt and a 40-watt white fluorescent lamp has an efficiency of 58 lumens per watt.

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MICROPHONES — dynamic, condenser, ribbon. Precision-built in Germany. Top value, performance and appearance. Write for your catalog, distributor cost sheet, and free booklet "How to Choose and Use Microphones". **Stanford International, San Carlos, California 94070.**

THREE GOTHAM GRAMPIAN CUTTER systems — including head, amplifier and power supply. $800. each or $2300 for all three. Also available, syncronous slave system, allows four acetates to be cut simultaneously, ready to use, best offer. **Universal Recording Corp., 46 E. Walton, Chicago 60611.**

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

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

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DISTRICT SALES MANAGER for leading commercial sound line. To qualify you must have a minimum of 5 years' experience in commercial sound sales (industrial school, etc.) plus solid technical understanding of the field. Applicants must be capable of building sales volume through effective selection, training, and supervision of commercial sound distribution in multi-state territories. Send full resume and references to **Rauland-Borg Corporation, 3535 W. Addison St., Chicago, Ill. 60618, attn: C. Dorwaldt.**

YOUNG SALES ENGINEER needed by reputable manufacturing firm of professional audio equipment. Some knowledge of application engineering. Opportunity for advancement. Little travel, offices in New York City. Write **Box 1A, db Magazine, 980 Old Country Road, Plainview, N. Y. 11803.**

SERVICES

CUSTOM CARTRIDGE RELOADING, clean-up and replacements. Write for price list. **Professional Audio Services, Box 1953, Fort Worth, Texas 76101.**

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

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

People, Places, Happenings

● The **Audio Engineering Society** has made its call for papers for the 38th convention to be held at the Los Angeles Hilton the 4th through the 7th of May. If you wish to present a paper at this convention check with the AES at its New York headquarters at 60 East 42nd St., New York, N. Y. 10017. Address titles and abstracts directly to the California chairman. **Hugh S. Allen, Jr., Gotham Audio Corp.,** 1710 N. La Brea Ave., Hollywood, Calif. 90046.

● Midwest residents note the dates of the **NAB** show to be held in Chicago April 5 through the 8th. As usual there will be halls full of equipment to see. The place is the Conrad Hilton Hotel on Michigan Avenue.

● An agreement to create a new company to manufacture and distribute recorded entertainment tapes throughout the United Kingdom was announced today by **GRT Corporation,** Sunnyvale, California, and **Pye Records Limited,** London, England. The new firm, a joint venture, will be owned equally by GRT and Pye.

In making the announcement, **Alan J. Bayley,** GRT president, said that the joint venture extends GRT's international marketing program by making possible an immediate entry into the United Kingdom recorded tape market via a substantial distribution system that is already in existence. He said that the new firm will have distribution rights in the U.K. to the Pye catalog and to all of the available GRT properties on a royalty basis. Of particular significance, he said, is the ability of the joint venture to obtain additional properties via licenses from other record companies and independent producers.

Pye Records, Limited is now considered the third largest record company in the United Kingdom and distributes in the U.K. recordings of such performers as Petula Clark, Frank Sinatra, Donovan, and Herb Alpert and the Tijuana Brass. The new company will be headquartered in London and will employ GRT duplication equipment and technology in the manufacture of the tapes. They will be sold under a variety of labels and marketed through the existing Pye distribution system and through new channels as well.

● **Dr. Peter Goldmark,** president of **CBS Laboratories** was the recent recipient of the **Franklin Institute's** Elliott Cresson Medal. Dr. Goldmark was instrumental in the invention of CBS' color television system (recently used to transmit Apollo 11 space color t.v.). The medal is being presented for three specific contributions: the development of the long-play record, the color t.v. system, and the new electronic video recording system (evr). All told, Dr. Goldmark is credited with 150 inventions.

Hegeman

● The formation of **Hegeman Laboratories, Inc.,** a new corporation specializing in top-of-the-line high-fidelity kits and factory-wired units, color vtr and electronic musical instruments has been announced by **A Stewart Hegeman,** chief technical officer. Called **Stereo Age I,** Hegeman Laboratories' line of audio equipment will embody a modular approach providing units which can be purchased singly or in combination, plus the switching equipment necessary for complete flexibility. It will include products ranging from components to complete receivers—enabling the buyer to tailor his system to meet specific requirements and add to it as his budget space or sphere of interest expands. First products in the line—a preamplifier control unit, switching panel, vu meter and power amplifier—are scheduled for introduction early this year. In conjunction with the introduction of Stereo Age I products, Hegeman Laboratories plans to complete development of a consumer-priced color and black-and-white vtr, according to Mr. Hegeman.

Passin

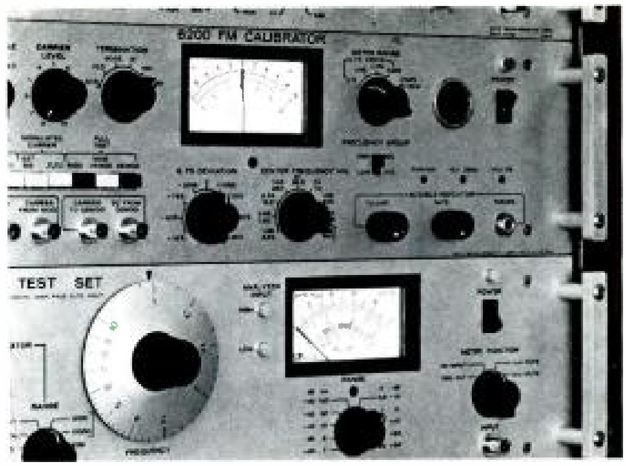
● The appointment of **Eli Passin** to the position of vice president was announced by **Stephen F. Temmer,** president of **Gotham Audio Corporation.** Mr. Passin was heretofore national sales manager for the company. **Hugh S. Allen, Jr.** continues as v.-p. of and now directs the operations of Gotham's West Coast Office.

A new affiliate, **Telden Sales Corp.,** was founded to act as both leasing and financing agents for Gotham Audio products, thereby adding a most important service for Gotham clients who wish to obtain long term leases or financing plans on both individual items and complete systems.

Tillett

● **George Tillett** has been appointed editor of **Audio Magazine,** the monthly consumer publication serving the high-fidelity market. Mr. Tillett was formerly director of engineering at **Fisher Electronics, Inc.,** and had been executive vice-president of **Audio Dynamics Corp.** Prior to these assignments he was with **Armstrong Wireless Company, Wharfedale, Heathkit,** and **Decca;** all of these companies are in his native Britain. He is also the author of many books and articles published over the years.

Should an FM Calibrator stay home in the rack...



or follow wherever you go?

Data Measurements Corporation thinks it depends upon what your requirements are. So we've developed FM calibrators to meet both needs.

Our Model 6200 is a conventional bench or rack-mounted FM calibrator, designed for checking and aligning FM modulators and demodulators, particularly those associated with instrumentation tape recorders. It consists of an accurate dc supply and crystal oscillators for all reference frequencies, providing nine-point calibration. The Model 6200 provides a stimulus to an FM modulator or demodulator under test, then analyzes the response. It compares the response to internal standards, and displays the error on a zero-counter null meter, or by an audible signal null. The null meter indicates error directly in percentage of bandwidth — with no manual operations required — on one of four possible ranges.

For field service engineers and other people on the go, or for small laboratories, universities and other facilities where simple and rapid testing is required, DMC offers the Model 6275 FM calibrator. The 6275 meets instrumentation magnetic tape recording requirements, weighs about 7 pounds, comes in its own carrying case, and costs $550.

Along with its portability, the Model 6275 can perform independent three-point calibration of modulators and demodulators as well as overall FM channel calibration. Other inexpensive FM calibrators align the demodulator first and then use the demodulator as a reference for modulator calibration, usually leading to calibration errors.

Whichever calibrator you select — the stay-at-home 6200 or on-the-go portable 6275 — DMC's professional quality will meet your FM calibration requirements.

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DMC

Data Measurements Corporation
855 Commercial Street
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