The Sound Engineering Magazine

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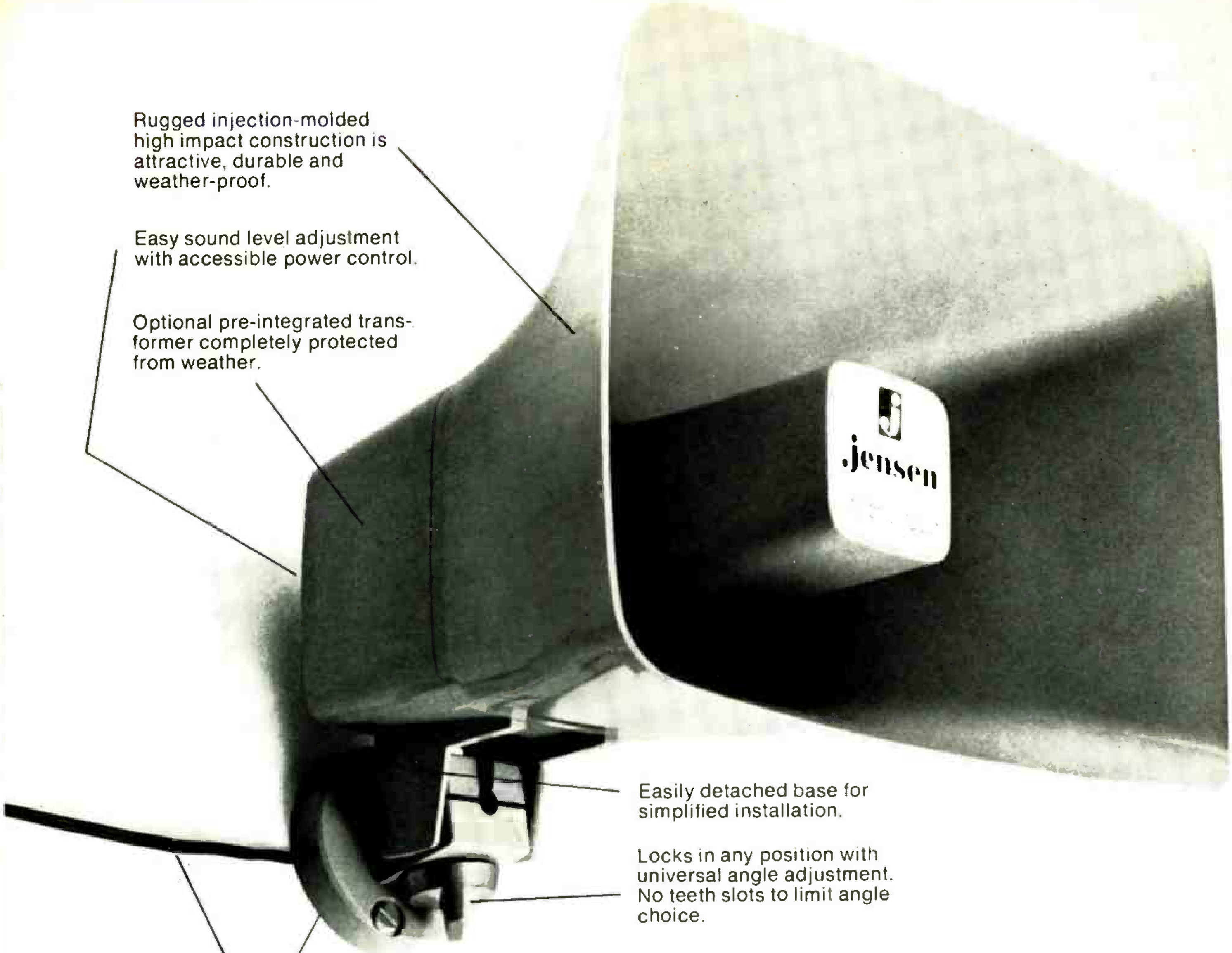
November 1967 60¢

Premiere Issue

PROFESSIONAL AUDIO:

The State of the Art



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November 1967 Vol. 1, No. 1

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Coming Next Month

The December issue of **db** will continue to explore the state of the art—with an emphasis on studio operations.

Harry F. Olson of RCA will explore the needs and expectations of users of **Monitor Loudspeakers**. Needless to say, he also will cover the limitations of this important piece of studio gear.

Richard Vorisek of Reeves Sound takes us on a tour of the highly sophisticated techniques of film and tv sound dubbing and mixing. His article is entitled **Modern Mixing Techniques Boost Efficiency**. After reading it, you will agree.

Philip Erhorn who has designed and installed many a studio console will present some meaningful opinions that may well decide you on the direction you want to go when it is time to purchase that next console.

Emil L. Torick of CBS Labs offers an up-to-the-minute report on **Automatic Audio Level Controls**. There is so much that is new in this field that you will not want to miss this article if you have any interest in studio operations.

PLUS: Monthly columns by **John A. McCulloch** and **George Alexandrovich**. And **New Products and Services** for the audio professional.

Be sure you receive your December copy of **db** and all the monthly issues to come. If you are a professional in audio you qualify for a free subscription. Fill in the postage-paid card at the back of this issue. Do it now.

Editorial

Welcome to db, the Sound Engineering Magazine.

In a field as broad as professional audio has become, the dissemination of information necessary to the improvement of performance becomes vital. However, with the exception of the highly respected professional societies, no one has yet come forth to provide such service. db's emphasis, therefore, will be to complement the theoretical orientation of existing publications with a focus on the practical applications of new concepts. We will make db a central source for reports on new techniques as well as previews of new equipment and new ideas.

Since our purpose is to establish an intra-industry dialogue with immediate relevance to everyday applications, we trust that db will become an invaluable journal for all audio engineers and technicians in broadcasting, recording, sound reinforcement, film and tv sound, video recording, the audio-visual field, and closely allied areas.

For this premiere issue of db we have asked a group of audio experts to provide us with a state-of-the-art statement about their special areas. Before we travel in new directions, it is good to establish where we are now.

Monthly issues will concentrate on a specific area of the audio field. Next month, for example, we will have several feature stories on studio operations. The following month will concentrate on sound reinforcement. Monthly columns will start in the near future. Next month John A. McCulloch will begin a series that will free-wheel through the equipment end of the audio field. It will also be a question and answer forum offering solutions to the many technical problems that come up in daily operations. Columns on Sound Reinforcement and CCTV are also planned. In addition a column devoted to certain theoretical and mathematical aspects of professional audio will soon be introduced.

Since a dialogue implies an exchange of ideas, your reaction to all aspects of this new venture will be welcome and will contribute to the growth of this journal.

We at db will endeavor to maintain a consistently high standard of technical accuracy and editorial excellence. We sincerely trust that our objectives will meet your needs and earn your encouragement.

* * * *

Subscriptions to db are free to all members of the professional audio fraternity regardless of their position. However, it is essential that you return the postage-free reply card bound into this issue if you wish to continue receiving future copies. This is necessary even if you received this copy in the mail. Be sure to fill out the card completely, sign it, and include your zip code. L.Z.

Audio Amplification

—Paul Weathers*

Amplification has come a long way since carbon amplifiers. And each era considered its state of the art as perfect. How close to an ideal product are we now?

Before Thermionic Vacuum Tubes

Most of us in the audio profession take for granted that audio amplification is a practically perfected art in the chain of recorded, transmitted and reproduced sound. Compared to the early efforts of engineers before vacuum tubes were developed, the audio amplifier of even 30 years ago seemed perfect. The first successful sound amplification was accomplished by the use of highly resonant carbon amplifiers. These were the first "solid-state" amplifiers and were current amplifiers of low impedance. They operated in the same way that carbon microphones operate by varying the pressure on carbon granules. The carbon granules provided a variable resistance inversely proportional to the applied pressure used to modulate a current flowing through the carbon granules. In the case of a carbon microphone the pressure is applied directly by the diaphram. In the amplifier application the pressure is applied by an electromagnetic transducer to produce current amplification.

Vacuum-Tube Amplification

The vacuum tube was the first development which made possible the amplification of tiny voltages at high impedance with relatively little noise and distortion. In fact a good class "A" vacuum-tube amplifier in the 1920's could boast of distortion figures of under 2% and most engineers felt that very little needed to be done since these amplifiers were so far advanced over any transducers available.

Inverse Feedback

Fortunately there are many engineers who are never satisfied with what others consider perfection. Out of the laboratories of engineers dedicated to progress came *inverse feedback*, a fundamental development which completely revolutionized the audio industry in the early 1930's and made possible a really tremendous advancement in the audio art. This development was, in fact, so fundamental that it is the basis for all present day "nearly perfect" amplification systems, whether they be vacuum tube or solid state.

How Can Amplifiers Be Further Improved?

The weakest link in the chain of an audio amplification system is the coupling the input and output of amplifiers to the transducers or modulating systems. Although transformers have been greatly improved by advanced winding techniques and magnetic materials, they have not completely solved the problem of matching transducers to the input or output of otherwise nearly perfect amplifiers. One might wonder, why bother with the most perfect link in the audio chain? Concentrate on the transducers so that their impedances do not change with frequency, or combine the amplifier with the transducer to improve the control of the transducer by intergrated feedback. In fact, much *is* being done to perfect transducers by closer integration with amplifiers, but when electrical isolation is required, coupling by transformers, (even with their limitations) is the accepted practice.

The Development Of Solid-State Devices

The development of solid-state devices is undoubtedly as great an advancement in the audio art as was inverse feedback as a perfecting technique. Solid-state amplification makes possible the matching of power amplifiers to low-impedance loads without using bulky power output transformers with their characteristic hysterisis loop and limited frequency range. The elimination of the output transformer makes possible an all-encompassing inverse feedback loop from transducer to input stage to provide less critical

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matching to output transducers and the virtual elimination of distortion within the loop. Concentrated feedback around one or two stages is often combined with over-all output to input feedback to keep phase shift within bounds.

Input matching in solid-state amplifiers is now more versatile than with vacuum tubes and, in cases where electrical isolation is not needed, without using transformers. A solid-state device can be selected which will match low-impedance transducers for minimum noise and maximum energy transfer or, when high-impedance inputs are required, to match high-impedance devices even up to many megohms, there is a solid-state circuit to effect a perfect match.

Are the arguments which we sometimes hear that vacuum-tube sound is superior to transistor·sound valid? There are also those who are just as certain that there are some subtle phenomena associated with transistor sound which makes it superior to vacuum-tube sound. These are arguments reminiscent of *triode sound* versus *pentode* sound in the 1930's. The differences are based upon easily measurable parameters and has nothing to do with whether the amplification is solid state or vacuum tube, triode or pentode. Most of the effects noticed are traceable to:

1. *Difference in damping factor on the output transducer.*
2. *Slow recovery of circuits when overloaded by short duration transients.*
3. *Regeneration or positive feedback at some part of the amplified spectrum.*
4. *Development of parasitics when excited by certain signals.*
5. *Instability of the power supply in its relationship to linearity of the amplification system as polarizing voltages vary with signal.*

Although variation in damping factor represents one of the most obvious differences between triodes and pentodes as output devices, engineers found that strategically applied inverse feedback could easily eliminate this difference. However, unless careful attention was given to phase shift, very often the amount of inverse feedback had to be increased to the point where parasitics might develop at super-audible frequencies or regeneration might occur at subaudible frequencies thus producing conditions described above in 2, 4, and sometimes 5.

It became a well-known fact that reasonably good results could be obtained by almost any engineer when triodes as output devices are used but it takes a highly experienced and talented engineer to design a superb sounding pentode or beam-power output tube power amplifier employing optimum feedback capable of feeding a resistive or reactive load with complete stability.

It has been demonstrated that solid-state power amplifiers using no output transformers can exceed the damping characteristics of the best triode power amplifiers, even when using feedback. The difference is in the elimination of the transformer. With solid-state amplifiers the damping factor can be so great that the power amplifier controls the movements of loudspeakers with extreme accuracy at low frequencies resulting in sound with less coloration; a noticeable improvement to the discriminating listener.

There are those who will argue that a certain amount of coloration is desirable in a sound system. Coloration by choice is the important objective, since the lack of coloration is the more desirable achievement. Coloration can be added in a variety of ways to any sound system if it is desired but coloration should not be the criterion in choosing between "triodes *versus* pentodes" or "vacuum tubes *versus* transistors". All of the differences between triode, pentode and transistor amplifiers can be measured or visually observed on oscilloscopes. These differences can be so minimized by careful design that listening tests comparing triodes, pentodes, beam power and transistors sound so neatly alike that a blindfold test completely confuses the most critical observers. One reason for this is that the most perfect program material and the most perfect transducers have far more distortion and coloration than that contributed by the differences in the various but highly perfected amplifiers under comparison.

Solid state provides the designer with so many new approaches to the perfect amplifier that there is little doubt in the minds of progressive engineers that solid state has taken over in audio leaving vacuum tubes as one of the great stepping stones in the search for perfect electronic amplification.

An examination of some of the advantages of solid state and the myriad of new circuits made possible to improve audio, leaves the imaginative engineer almost spellbound with the possibilities of miniaturization, elimination of heat, pinpoint control of polarizing potentials, elimination of noises due to long lengths of connecting wire, plug in modules for easy servicing, intregration of amplifiers with transducers and general reduction in costs.

Miniaturization is not just an advantage in portable equipment. The new IC's eliminate many components and reduce the stray noise pickup area making possible a simplification of shielding to attain a new perfection in noise rejection. The elimination of heat makes possible closer spacing of components along with vastly longer life. Housings can be smaller, requiring little or no ventilation, at less cost.

Zener diodes make possible pinpoint control of polarizing voltages and when combined with transistors, can provide filtering and isolation of voltage sensitive circuits literally not possible by other means.

FET's

FET's take away one of the principal advantages which tubes had over solid-state devices i.e. high-input impedance and low noise. Some vacuum tubes still have a slight edge over FET's in high gain applications for consistently low noise, but the fact that laboratory samples of FET's have approached and exceeded the quietest vacuum tubes is an indication that it is only a matter of time, by perfecting production techniques, that FET's will replace vacuum tubes for all high-impedance input applications. FET's have other characteristics of great interest to audio designers. It is possible to incorporate them in automatic compression or expansion circuits without utilizing balancing devices to eliminate control transients and the degree of compression and expansion can go much farther than was possible with more complicated circuitry using the old familiar techniques of balancing out control transients.

It is conceivable that a low-level pilot tone pitched above audibility and modulated by the compressor can be utilized in reproduction to recreate the original dynamic range with minimal time lags in the control circuits and without undesirable transients. This is an area which should be explored more thoroughly.

The characteristics of FET's and other solid-state devices lend themselves to automatic control of dynamic range, frequency range, and activation of additional transducers to produce dynamic effects never before attained.

State-of-the-art amplifiers can be manufactured so that little servicing will ever be required but the service technician will be under a tremendous handicap to correct a non functioning amplifier whose parts he cannot see and whose wiring junctions are smaller and closer together than the end of his finest probe. That is unless a plug-in modular substitution system is used.

This brings us to the final Utopian concept. Let us make all amplifiers in modular form and have the modules plug in using techniques which have been highly perfected in the computor field. Modern connectors have become so perfect that they are consistently as good as connections soldered by hand.

There are those who will dispute this statement but these connectors are used in critical application where failure could be disastrous and have proven their dependability. Besides, vacuum tubes have been plugged in, even in the most critical applications. Why not modules containing circuits?

The Ideal Amplifier

To sum up, the state of the art amplifier of today can have:

1. *A fully regulated power supply with no measurable ripple and practically zero source impedance, making class "B" amplifiers attain even greater perfection.*
2. *Capability of matching to selected output load impedances without output transformers (except for line feeding amplifiers requiring balanced lines and exact line matching).*
3. *Capability of matching input impedances with maximum transfer efficiency and low noise from a few hundred to many megohms.*
4. *Modular construction for substitution servicing.*
5. *Compact, to take up a minimum of room, with a minimum of required ventilation.*
6. *Output damping characteristics so great that the output load looks back into a virtual short circuit, so that the lack of coloration of output transducers is limited only by the transducer's series resistance.*

The resistance of the audio industry to changing technology is no longer as persistant as it was a decade ago. In fact, the avalanche of advanced technology has been applied so rapidly during the past three years that obsolescense of equipment has progressed at an unprecedented rate. Also, service technicians have been unable to keep pace with rapidly advancing technology and they are looking for some sound answers to their pyramiding maintenance problems.

It is very important that these highly advanced products be more easily assimilated and maintained at their peak of perfection without requiring an engineer in constant attendance. It is highly important that state-of-the-art products include simplification of application and maintenance to assure a continued peak of performance. The hardware and information is available in quantity and quality. It is up to the design engineer to break away from traditional concepts which have outlived their usefulness thus making maximum use of the new tools at their command.

THE STATE OF THE ART

Disc Mastering

Sidney L. Silver*

The author covers the basic elements—lathes, cutterheads, cutting styli, and instantaneous recording discs, in his discourse on the current state of the art.

In the process of transferring taped material to an acetate record for mastering purposes, the prime consideration is to make certain that none of the high quality of the master tape is lost in the process. To achieve consistent, reliable results with a minimum of service or attention, computer technology is now being utilized to provide complete automation of the entire recording system in performing the functions of switching, timing, counting and programming.

By merely depressing a button, it is possible to initiate a series of events which include dropping the cutterhead, cutting a coarse spiral lead-in groove to a fixed diameter, recording the program material, cutting coarse spirals for banding if necessary, cutting a coarse spiral lead-out groove, and finally lifting the cutterhead at a predetermined diameter. Other functions automatically performed are pitch and depth control and on-off switching of the vacuum, stylus head, and tape machine.

In this article, some of the salient features of recording equipment available in today's market will be discussed, as well as some of the more recent technical improvements.

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New York, N.Y.

Recording Lathes

Generally, the drive mechanism consists of a hysterisis-synchronous motor, utilizing belt coupling for speed selection. Belt-driven units are characterized by extremely low rumble (vertical and lateral) figures, of the order of -70dB below a 5 cm/sec. signal. To minimize belt stretching problems when switching from 33 1/3 to 45 or 78 rpm, a two-speed motor is sometimes employed instead of the usual single-speed units. By this means, the motor pulley sizes are kept dimensionally close to each other, so that changes in motor speed compensate for the otherwise large differences in pulley diameters.

Since low wow and flutter figures (typically .02% peak-to peak) must be maintained, the turntable is designed for maximum flywheel effect and is adequately isolated from the drive mechanism by a mechanical vibration filter assembly. Ideally, the turntable is fabricated of an aluminum casting to eliminate the problem of magnetic attraction between turntable and the cutterhead. Annular rings are machined at various diameters for the purpose of suction holddown, to accommodate record blanks from 7-in. to 16-in. in diameter. Provision is easily made to shut off the holddown action at the outer diameters when not required, thereby eliminating objectionable escaping air noise.

Normally, the turntable drive system is connected by mechanical linkage to the feedscrew assembly via a planetary drive and gear assembly. Some designs, however, incorporate a self-powered feedscrew for guiding the cutterhead carriage, entirely independent of the mechanical linkage from the turntable's drive system. In this arrangement, a separate motor powers the feedscrew, thus eliminating any extra load on the drive system. There are also models in which an auxiliary feedscrew is employed for the inspection microscope, which travels in with the cutter and allows a "standstill" groove effect for easy groove inspection.

Monophonic lateral recordings are commonly mastered with variable groove pitch in order to fully utilize the available recording area. This may be accomplished automatically by a servo-controlled system which derives its feed from an advance playback head reproducing the program material one or two seconds ahead of the cutterhead. The system thus provides a coarser pitch in anticipation of high-amplitude low-frequency passages during the tape-to-disc mastering operation. In some designs, pitch is continuously variable from 32 to 800 lines-per-inch with the ability to manually record over 1000 lines-per-inch if necessary.

Stereo recording, however, involves both lateral and vertical stylus excursions so that in addition to control of groove pitch, control of groove depth is also required. In stereo cutterheads which employ advance-ball suspension, variable depth control may be achieved by operating a solenoid on the advance-ball assembly. This type of suspension, however, is prone to rumble caused by undesirable vertical modulation of the cutterhead, due to any unevenness of the acetate surface. Furthermore, a condition known as "scoring" may occur when a dirt deposit adheres to the sapphire advance ball, plowing through the soft lacquer as the ball rides the acetate blank. To offset these problems, some carriages accommodate a floating- or moving-coil suspension assembly to either replace or work in combination with the advance-ball suspension.

Two typical examples of such sophisticated recording equipment are the Scully Automatic Disc Recording Lathe and the Neumann AM-66 Computer-Controlled Lathe.

Cutterheads

The need for a reliable, rugged cutting head is of major importance if we are to meet the increasing demands for quality disc recording. Particular care must be taken in cutterhead design to achieve stable operation under widely varying environments, without danger of burnout or mechanical damage.

In modern professional cutterheads, the force to actuate the cutting stylus is provided either by a moving coil attached to a fixed armature, or a fixed coil and moving armature, either in a magnetic field. The two types are classified respectively as *dynamic* or *magnetic* transducers. In the dynamic configuration, the mechanical resonant peak lies in the 800 to 1500 Hz range and is effectively removed by *motional feedback*. This arrangement consists of an amplitude-sensitive feedback coil magnetically shielded from the driving coil and placed in close proximity to the stylus chuck. The feedback signal, which is directly proportional to stylus motion, is amplified and fed back (out of phase) to the input of the driving amplifier, thus correcting changes in level and response. An advantage of this technique is that no loading effect is imposed on the driving coil, so that a wide effective range of feedback control is obtained with good linearity throughout the usable frequency range. Harmonic distortion levels with dynamic cutting systems are significantly less than 1%. Dynamic cutters in common industry use are the Westrex 2-B monophonic and the Neumann SX-45 stereo cutting systems.

In *magnetic* cutting systems, the resonant peak generally lies in the 4 to 8 KHz range and may be damped out by silicone damping material in the form of plastic, grease, or fluid, sealed into the gaps between the armature and pole pieces. Cutterhead feedback may be provided by a secondary winding placed closer to the armature than the cutter driving coil. This feedback arrangement is *non-motional* since the secondary winding senses only armature saturation. Any distortion created within the armature due to eddy currents or other magnetic circuit losses is thereby reduced.

Magnetic cutters are somewhat sensitive to the acetate loading effects which increase with decreasing linear velocity, so that the high frequencies are attenuated at the inner diameters of the disc. In some cutterhead designs, this condition is avoided by increasing the stiffness of armature motion so that the resonant frequency is raised to about 10 kHz. Unfortunately, this results in reduced sensitivity so that the power requirements of the driving amplifier are greatly increased. Although magnetic cutting systems are more rugged and considerably less costly than dynamic types, they are less sensitive and display higher harmonic distortion levels (of the order of 2 to 4%). Among the magnetic types commercially available are the Haeco SC-1 and Grampian "D" cutters.

Cutting Styli

Present-day research is being directed toward the development of the perfect recording stylus. Sapphire is still considered to be the most suitable material available for producing a very fine cutting edge. This is mainly due to its crystalline structure, which is characterized by a lack of grain and absence of cleavage planes, thus enabling the material to be ground to a sharply acute angle.

The common hot-stylus technique offers a practical solution to the problems of achieving higher frequency response, longer stylus life, and better signal-to-noise ratio. With this technique, a substantial reduction in surface noise is indeed obtained, particularly at the inner diameters of the recording disc. A further advantage is that it minimizes mechanical loading on the cutterhead, thus increasing the efficiency of the cutting system.

In a more recent development, the conventional flat-faced stylus surface is replaced by a *scooped* stylus, in which the cutting face is ground in a circular arc. The curved surface helps to eliminate some of the errors caused in the vertical radial motion of the stylus by reducing the deviation of the cutting face from the perpendicular to the disc surface.

Discs

Curiously, there has been relatively little change in the basic formulation of instantaneous recording discs over the past thirty years. So-called "acetate" discs, which actually are fabricated of plasticized nitrocellulose lacquer on an aluminum substrate, are still regarded as the best instantaneous recording medium. Major progress, however, has been achieved in recently improved coating techniques leading to flatter, cleaner surfaces. For example, improved drying equipment with rigid control of air flow and temperature, together with improved lacquer filtration methods, have greatly contributed to the fabrication of defect-free discs.

Electronic Elements

Despite the recent trend toward micro-miniaturization and transistor design, some recording systems have retained vacuum tubes and spacious layouts. In any case, the driving system for the cutterhead must provide the necessary power (typically 75 to 100 watts-per-channel) if we are to cut the entire frequency spectrum without clipping. Driving amplifiers are characterized by large open-loop gain, with heavy feedback to insure low dynamic impedance and phase-shift reduction of all stages. A useful innovation is the stylus oscilloscope which instantaneously monitors the output signal and indicates the overload level to which the system can be pushed. RIAA equalization from 30 Hz to 1 kHz and constant-velocity equalization from 1 kHz to 15 kHz is provided by passive networks (usually mounted on a plug-in card) for precise correction of individual cutting systems.

The development of disc recording to its present state-of-the-art has required relatively little electronic sophistication compared to the difficult mechanical and electro-mechanical problems that have had to be solved. Even with present-day mastering techniques, disc recording technology is continually progressing in a continuing search for superior performance.

Sound Reinforcement

—Martin Dickstein*

The elements that make up a sound reinforcement system have come a long way since the days of the cupped hand. But the present state of the art is merely a stopping point. Where do we go from here?

Back in the very, very old days, it was necessary to be within "earshot" of a sound's originating point. Earshot depended for the most part on the frequency characteristics, loudness, wind direction, surrounding noises, terrain or location of source and listener, just as it does today, whether the receiving device is an ear or mechanism. In those days, though, earshot distance actually meant to an ear. There was no way to preserve the sound for later listening, nor was there any way to extend the distance. If anyone wanted to hear the sound they had to be there, fairly close to the source. Attempts to increase the "throw" distance through the use of cupped hands or other horn-like devices helped the situation a little. In time, it was learned that sounds could be directed somewhat toward the listeners if the source was located in front of something such as a hill or structure. Carefully shaped and strategically placed vases in the proximity of performers, prevailing winds, and higher but closer seating areas were used in further attempts to permit more listeners to hear the sounds originating from the source.

*Television Utilities Corp
Long Island City, N.Y.*

Through the many years that followed, more attempts were made to enlarge the listening area, but meeting halls, auditoria, concert halls, churches and ampitheaters had to be built to take advantage of natural acoustics to "spread the word." It was not until the availability of electronic amplification that something was done to substantially enlarge the listening area.

Recordings could now be made using a microphone which changed sound impulses to electrical impulses. These recordings could be played through speakers which distributed sound to listeners in a fairly good size area. When movies started to talk, though, a real need for "big sound" became apparent. Theaters could seat pretty big crowds to see the screens, but now it became necessary that all these people also be able to hear the voices, sounds, and music. Larger speakers, higher-output microphones, and more powerful amplifiers had to be developed. And they were.

This solved some problems but, as seems to happen in almost every field, created new ones.

As microphones came into use for addressing live audiences, methods had to be found to make the mikes smaller in order to avoid hiding the speaker or performer. Lapel microphones, lavaliers to hang around the neck, and button-hold mikes were designed to permit the speaker or singer free use of the hands during a performance. As sensitivity of microphones improved it became necessary to manufacture units with different pickup patterns so that some had very narrow cones of sensitivity, some could pick up through greater angles and some were made to be non-restrictive, or omnidirectional.

Many types of microphones are now made for music, offering better frequency response in the bass range or high-frequency range or with smooth response through a very wide band of frequencies. Some distort sound less than others during loud, impact sounds. Others have pre-emphasis in the voice range built in. Still others are made to pick up through an extremely narrow angle but at a greater distance. Each microphone manufacturer today tries to cover a particular market or use with a variety of types and shapes especially designed and engineered for each application.

For locations where microphones are used near loudspeakers or in highly reverberant areas, the choice is a combination of narrow pickup angle, smooth frequency response and high front-to-rear rejection ratio to diminish the possibility of acoustic feedback. Today, microphones are made with these singular characteristics, combining those qualities that make them particularly suited for individual requirements, in almost all sizes and shapes and at almost any price.

Amplification

Many changes in the construction of microphones were caused by, and resulted in, changes made in the amplifier units developed to raise the minute power output of the microphone to the level necessary to drive a loudspeaker strongly enough to enlarge the listening area by a substantial amount. Early amplifier designs for power applications were made to match existing microphones and speakers. The desired power output was achieved first with existing tubes and components through the use of some known circuits, then with modified circuits and newly developed tubes, new ideas on impedance matching, energy transfer, interstage coupling and feedback.

Inherent tube and circuitry noise was slowly eliminated with the development of new materials for use in filaments, grids, resistors, and the resulting modifications in power supplies and filtering methods. Frequency response was broadened to provide smoother lows and more natural highs. Components were designed to withstand the heat developed by high-power tubes.

Mixing circuits were devised to offer greater possibilities for multi-source originations. Special filters were designed to compensate for recording characteristics. Mixer units were separated from the power chassis to permit the more delicate low-voltage circuitry to operate to its own best advantage without interference from tube heat and power transformers. Bridging circuitry and matching networks were calculated and engineered to permit flexibility of inputs. Power amplifiers could not be located at a distance from the mixing unit whenever desired without any appreciable loss of quality or energy.

Amplifiers were modified to reduce distortion, increase frequency response, provide higher power and take up less space. Preamplifier mixers were also modified to operate with a greater frequency range and lower noise and dis tortion level.

As power ratings of amplifiers were increased to provide greater audience coverage, constant-voltage output systems were devised to allow more and more speakers to be added without loss of level.

Separate mixer/amplifier units and packaged, or combined, mixer-amplifier equipments are now available from many manufacturers in various price ranges depending on the application.

With the arrival of stereophonic reproduction, mixers and amplifiers have been again redesigned to permit two and three channels to be available on units built to the same dimensions as previous mono equipment, and with a minimum of cross-channel interference.

The disassociation of mixer from amplifier has permitted the development of equipment specifically to solve some of the problems incurred in multi-input and high-level speaker output systems. Limiters and compressors have come into fairly extensive use to prevent sharp bursts of sound from distorting the quality of reproduction or damaging the equipment. In the same way, expanders to raise the level of the source sound automatically if it should fall below a predetermined point, are now common.

Most recently, equalizers of various types have been engineered to permit a greater opportunity to emphasize desired frequencies and eliminate undesired ones. In reverberant auditoria greater intelligibility can be realized by raising the relative importance of certain frequencies and diminishing the level of others, thus limiting the discrete ones causing the disturbing reverberation and, as a result, permitting higher levels of output in the desired ranges. Less sophisticated equalizers are available that operate on calculated curves at fewer, but carefully selected, frequencies from those units that are specifically designed to permit exceptionally sharp operation at a great number of frequencies.

In recent studies of acoustics and reverberation, it has been found that if output frequencies can be made slightly different from the original ones, by an amount insufficient to be heard, then feedback can be reduced. This will allow higher output levels. The result is the development of a frequency shifter which raises each input frequency

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Solid State

In most recent years, preamplifiers, amplifiers, compressors and even power supplies have undergone a most revolutionary change with development of the transistor. In its infancy, the transistor introduced its own sound characteristic when used in audio equipment. It also could not withstand the punishment that a tube could when something happened in the internal circuitry of the equipment or in the speaker lines. With further development, this device is now being used in almost all equipment. New circuitry has been devised to keep the transistor from breaking down in the event the equipment becomes defective, or if trouble develops in the output lines, or even if there is a sudden power surge. Transistors of various designs and internal structures have been developed for special applications and while some are still being made that cannot be handled without special precautions, for the most part, these tiny units have made such inroads that they have almost taken over completely.

Audio equipment can now be made smaller, lighter, with very good frequency and noise characteristics, but not yet quite at the same price as the tube equipment.

Since the heat developed by a transistor is much less than that given off by a tube of equivalent operation, more equipment can be mounted in smaller housings. Nevertheless, all transistor equipment should still be protected from being overheated by any tube units which may be used in the same system.

Speakers

As other units of a sound distribution system have changed, so did the speaker. Extremely bulky, inefficient loudspeakers with coil-generated fields changed to smaller, lighter, more efficient units with permanent magnets. Cone material changed along with the type of suspension of open ends. Frequency response improved, rattle was eliminated in better speakers during peaks, shapes of magnets changed as new materials were found to create stronger fields in a smaller space, and efficiency improved as less power was required to move the cone. Speakers, as with microphones, have been developed for different applications, with unique specifications of frequency response and smoothness, power ratings, angle of dispersion, efficiency and size. Smoothness of the response curve is desired to reduce feedback when used in the vicinity of open microphones. Concepts of speaker housings and their internal structure, front openings and sizes, dimensions and material have undergone study and modification to improve particular characteristics deemed necessary by the variety of applications.

A study was also made of the interaction of sound waves when speakers were located close together. It was found that a mounting of several speakers in a particular arrangement provides characteristics unique to that arrangement. A linear array, with speaker centers a definite distance apart, creates a "spotlight" effect with the vertical angle less than that of a single speaker alone. That angle decreases as the number of speakers increases.

Sound columns are now used in areas where reverberation time is high to direct the sound towards the audience and to decrease the wasted sound energy which would otherwise hit the ceiling or floor and create reflected, disturbing sound patterns. Straight line and curved speaker arrays, multiple speaker arrangements, filter networks and other concepts have sprung from the original theory in an effort to produce units that will suit an application range or fit a price category. Still, specially designed units are required when the circumstances are sufficiently unique to warrant this engineering, as for example tremendous audiences in large outdoor areas or music halls where highest fidelity is required at each seat.

As more and more equipment has been developed and produced in each phase of sound distribution systems, it has become obvious that standardization is required. Microphone output impedance must match preamplifier inputs, amplifier outputs must match speakers, preamplifier outputs and amplifier inputs, as well as those of intermediate equipment, must also match for proper operation of a complete system. Characteristics of equipment, depending on several variables, have to be defined at particular points of measurement.

Prognostications?

Now that all these concepts, systems, equipment and many of the standards have become accepted, tried and developed to meet a great variety of uses and conditions, have we gone as far as we can in sound reinforcement and public address? Can anything more be done to improve what is now available to compare with the strides in development up to now? Further miniaturization seems to be part of the answer.

Will mikes be made with mini-transmitters to eliminate the use of connecting cables altogether? Will receiver-amplifiers be further diminshed in size through the use of multi-function modules so that they can be located at each speaker instead of in central racks? Will speakers become smaller to permit them to be hidden entirely? Will new studies of acoustics, reverberation, sound transmission and dispersion create havoc with present standards? Or will future studies of the ear and its operation, the response of the brain and greater use of psychedelic sound provide the new incentives for the developments yet to come? It might prove interesting to live long enough to hear for ourselves.

33rd CONVENTION AUDIO ENGINEERING SOCIETY

Technical Sessions

THEATRE — Barbizon Plaza Hotel

MONDAY, OCTOBER 16

9:00 A.M. — Annual Business Meeting
9:30 A.M. — Audio Applications
1:30 P.M. — Transducers
7:30 P.M. — Speech and Music

TUESDAY, OCTOBER 17

9:30 A.M. — Amplifiers
1:30 P.M. — Control Systems
7:30 P.M. — Sound Reinforcement

WEDNESDAY, OCTOBER 18

9:30 A.M. — Broadcasting
1:30 P.M. — Standards
6:00 P.M. — Social Hour
7:00 P.M. — Banquet

THURSDAY, OCTOBER 19

9:30 A.M. — Disc Recording and Reproduction
1:30 P.M. — Tape Recording and Reproduction
7:30 P.M. — Film Recording and Reproduction

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The Recording Studio

—George Alexandrovich*

This state-of-the-art picture of the recording studio is the preamble to a series under preparation. In this installment the author ranges wide to cover much of what is likely to be encountered in the studio. Future installments will tend to concentrate on specific areas of interest.

The past decade has seen advances in the field of electronics no one has expected. New discoveries in the field of semi-conductors have led to new technologies and changed most of the design concepts overnight. Gradually new technologies are making their way into the audio field, awakening audio engineers and specialists to the advantages of the new concepts over the older kind, a kind inferior in performance reliability and convenience.

The advent of semiconductors into audio closely coincided with the development of the stereo disc. This led to further developments of multi-track tape recording. Such tape machines in turn led to the redesign and rework of the mixing consoles and control rooms. As a result, the problems of the recording engineer and maintenance man suddenly increased two fold. Several channels of audio now had to be recorded simultaneously and monitored at the same time, each channel individually as well as the total mono mix, to be sure of proper phase relationships. Instead of caring for one channel only, maintenance had to be pulled on several, keeping proper balance, frequency response, phase and separation.

Disc cutting rooms had to be fitted with new equipment capable of stereo. Recording and cutting personnel had to be trained to use stereo equipment as well as to maintain it. Enormous new problems arose when semiconductor circuits were applied to the design of audio equipment, because of the inability of a majority of maintenance men to grasp immediately the basic operating principles of these new circuits. For quite some time a negative attitude existed among audio engineers and management toward the transistorized equipment. True, the first transistorized circuits were not designed with the same degree of sophistication as they are now, at times leaving more to be desired from their performance. At this time, the advantages of transistors over tubes have still not been recognized by many professionals. Many not yet set to accept semiconductors are those that were probing for faults in the equipment available at an earlier time and, as a result, continue criticizing it. Some of these criticisms are utterly ridiculous; for example, the claim that transistorized equipment does not produce "air around the sound."

I can prove that this is not so and that transistorized equipment can outperform tubes in distortion, frequency response, noise, in stable operation and reliability. As far as the air around the sound is concerned I need only say that transistorized circuits with their extended performance range reproduce information more faithfully than tubes without creating any side effects. True, they reveal more faults in the recording than do tubes, leaving the impression that tube circuits are cleaner sounding because of poorer transient response and restricted frequency range.

These innovations and changes in operating as well as in maintenance procedures have met strong opposition for another reason. Up to this time, recording was more often an art, with skill in it acquired more through experimentation and cut and try methods than through a proper scientific approach and knowledge of what conditions have to be met in order to achieve good recording. Today, new technologies require more science prerequisites for the audio man; he must have more theoretical knowledge and skill.

Advances and improvements have been made in every branch of audio recording. The most significant are in the design of the *equipment*. This inevitably has affected *operating procedures* and *techniques*. With new equipment and new technology *maintenance* gets the lion's share of the changes in procedures and requirements. It is my intention to use these three topics for future discussions, developing each one into an individual review of the tasks and problems facing the audio recording engineer today.

And those problems are numerous. As systems become more complex, operating procedures have to be more precisely controlled, maintenance of unfamiliar equipment and circuits becomes more painful. It is my goal to guide you men in the studio, behind the mixing consoles, and in front of the editing machines or disc-cutting lathes, to the correct approach used in solving the myriads of individual problems. I would like to shed light on the facts about the new equipment and operating methods showing how much their flexibility offers chances for a successful session. I want to cover the methods of preparation of the studio and control room for a session; how to conduct the session; proper storage of recorded information; mixing and editing; preparation of the master tape and master disc. Some topics will be discussed in greater detail than others; I hope to be able to share with the reader a few "trade secrets." These

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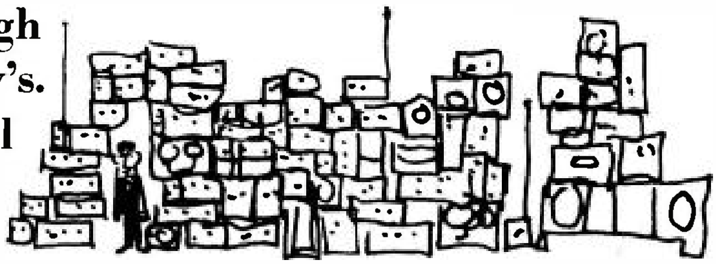

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secrets are nothing more than short cuts to the solution of problems or ways in saving money that may turn out to be important to smaller studios with limited budgets. I will direct my efforts to be as down to earth in these discussions as possible so that every recording man can understand all that is being talked about. And there is a lot to talk about.

Today you can hardly find a studio not already in possession of a multi-track recording system (or at least contemplating acquiring one). A few studios are still operating on two tracks, more are set for four tracks and some for eight tracks. Since eight-track machines are possible because of miniaturization with transistorized equipment quite a number of professionals are now thinking of eight and even fifteen or sixteen tracks on one- or even half-inch tape.

The advantages of an entry into multi-track narrow tape recording should be obvious to every engineer as well as his management. The ability to record virtually every microphone on a separate track offers an easy re-mixing job with better chances for correction if during the take balance between the microphones was other than acceptable. Equalization, reverberation, and other effects can be added at will to an individual instrument or groups.

With the help of selective recording, better known as *selsync*, recording on the multi-track machines can be economical. One or more tracks can be recorded independently of each other. When the remaining tracks are to be recorded, the previously recorded ones are played back through the record head and fed into a separate circuit. This output is fed into the headphones placed on the heads of performers so that perfect synchronism is achieved between all tracks. In this way a few performers proficient on several different instruments can be used, eliminating the need for large group, yet achieving the same results. If the take is unsuccessful one track or any number of selected tracks can be erased and re-recorded again without affecting other tracks.

Almost all of the newer consoles incorporate separate equalization on each microphone channel, with separate echo or reverb feed, sometimes compression, as well as many other features. But it is important to know how and when to use these features. For instance, compression during the original take should be used *only* as an overload protection rather than for altering the dynamics. It means that the threshold of compression should be set above normal operating levels in the console. Equalization should also be used with discretion. In the original recording it should be used only to improve crosstalk by restricting the frequency range or as a means to better noise figures.

With multi-track recording there is a strong trend to record and store audio information with the least amount of deviation from the original sound. This way original performance is always at hand and special effects can easily be added during the remixing session.

The storage medium for original recording or tape has also seen numerous improvements. Electrical as well as physical properties have been affected. Tapes are manufactured today from better materials (Mylar) and coated with oxides capable of carrying higher magnetization forces and producing lower electrical noise when fully demagnetized. They also offer less friction with the recording heads. All of this produces a wider dynamic recording range.

Work is being done on high frequency bias to lower the hiss level normally generated by the bias currents. Networks, producing predistortion into the recording, compensate and cancel 3rd harmonic distortion caused by the tape itself when it is being recorded with levels approaching saturation of the oxide. Naturally this predistortion would vary with the type of tape used and should be adjusted for each individual brand. This technique can improve the dynamic range of the recorder up to 6 dB with attendant low distortion.

To ease the tasks of phase control, equipment has been developed to insure proper mike placement in the studio. Phase detection monitors are used in some installations to avoid phase cancellations at low frequencies.

All these improvements allow the recording engineer to work with wider margins of safety for better recordings.

A great deal has been accomplished in the past decade in the field of tape recording. Nevertheless, disc recording and disc pressing are still with us and are sure to remain for a long time to come. With the advent of the stereo disc, a multitude of new problems arose. Precision control of groove geometry and position as well as stylus alignment for best channel separation and minimum distortion calls for an increased ingenuity by the cutting engineer or technician. Earlier I talked about the conversion of an art into a science in general, but this part of sound recording belongs in a separate category, since there is still much "know-how" and ingenuity as well as experience required of a cutting man. Every step in setting up for cutting stereo is a critical one. In order to achieve optimum results in both the sound and the appearance of a record one must be thoroughly familiar with all facets of this skill. This part of recording contains the most trade secrets and shall be treated as such in a future separate section on disc recording. Many recording engineers can benefit from the information that has been assembled from many cutting rooms and as many ingenious operators and technicians.

I have reviewed hastily the basic problems of sound recording and some of many improvements that have been made in this field in the past few years. But no equipment is immune to mishandling or misuse. One of the biggest handicaps of studio setups today is the lack of maintenance and quality control. This might be through incompetent personnel or simply because of an absence of trained technicians.

It is quite common to find multimillion dollar installations without a good 'scope or signal generator; never mind looking for a distortion analyzer. Commonly, there is an absence of any maintenance records or studio block diagrams.

On numerous occasions it has been found that because of the lack of maintenance or negligence the best equipment was operating as poorly as the worst kind. Hum, noise, susceptibility to clicks, crosstalk, and intermodulation distortion have been found as results of wrong terminations and a lack of proper grounding. Correct phasing, proper wire identification, good soldering, and wire dressing as well as equipment location, are major items which shouldn't be forgotten. Maintenance may well make or break the studio, so I will continually place special emphasis on proper maintenance.

Let this short review be an introduction into the coming series of talks about the wide field of practical professional audio engineering.

Measurement Techniques and Standards

— Norman H. Crowhurst

The author explores the essentials of audio measurements from a professional point of view. Special emphasis is placed on proper termination and its importance in gain measurements. The various standards of measurement are also covered.

AUDIO MEASUREMENTS are usually regarded as comprising three basic items: gain, frequency response and distortion. A fourth dimension in audio is dynamic range, limited at one end by the maximum relatively distortion-free level and at the other by noise.

Thus, one way or another, distortion is inevitably linked with level. Usually the percentage of distortion rises as level rises, up to a point where distortion fairly suddenly becomes quite considerable—unbearable to listen to. This is the absolute upper limit of dynamic range. Preferably it should be beyond the upper working limit, so it is never reached during operation.

Below this upper level limit, most high quality equipment shows lower distortion the lower the level, but in some equipment this does not happen. Suppose, for example, that a circuit uses class B amplification, producing crossover distortion: below a certain level the magnitude of the distortion component may not diminish any further as the magnitude of the wanted signal does, so that the percentage distortion actually rises as level is reduced. Such a system sounds far from clean and would be unacceptable professionally.

Gain and frequency response are also interlinked with level and distortion, as well as being dependent on an important factor we didn't name above: impedance. All these factors tend to complicate measurements, and make the establishment of meaningful standards and techniques difficult. Let's take the various standards and look over some pitfalls in their use. This will help avoid some of the common mistakes, and also lead to the use of better methods.

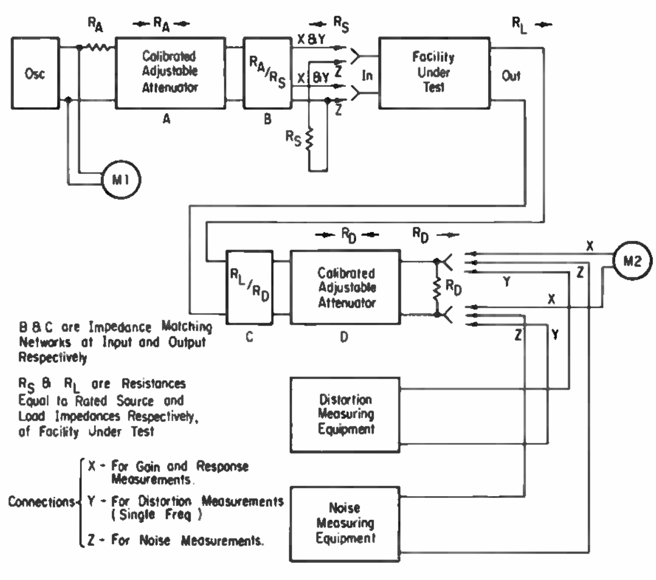


Fig. 1. Typical Measuring System for Audio Facilities (From EIA Standard: Audio Facilities for Radio Broadcasting Systems).

Gain.

Gain, frequency response and distortion tests are all made with a similar measurement set-up, according to audio standards. This consists of an audio oscillator with a metered output, fed through an attenuator and input matching network to the item of equipment being tested, which is terminated with a further matching load and output metering (fig. 1).

Most professional equipment operates at line impedance, commonly 500 ohms. So the input resistor will be 500 ohms, the attenuator will work at 500-ohms and probably, unless it is a monitor amplifier, also an output impedance of 500 ohms.

Insertion gain is defined as the change in level brought about by inserting the amplifier or other item of equipment between input and output, as compared with a direct connection (fig. 2). This is a somewhat academic definition.

If input and output impedance are the same, a direct connection will result in a 6 dB loss: voltage at the input will be double that at the "output." If the equipment is now inserted and attenuation added till the same double-value input voltage produces the same output voltage, the attentuation inserted will be equal to the insertion gain.

That's fine in theory, and it's the best we can do toward a general or universally true statement, based on standard practice. And it will be correct *if* the input and output impedances are themselves 500 ohms, as well as requiring to be matched to these impedances. But this is not always true.

For example, an amplifier may be designed to work from a 500-ohm source, but it does not provide precisely 500-ohm loading for the input at the same time. It will work satisfactorily from a 500-ohm line, because that is what it is designed to do. But its actual input impedance may be a few thousand ohms.

So using this amplifier's input as termination for a 500-ohm line, or for a 500-ohm attenuator, will result in not as much loss as the 6 dB theoretical—maybe only 1 or 2 dB. If this termination is applied to a real line, needing 500 ohms to avoid reflections, it is customary to apply a 500-ohm termination directly across the line as well. The equipment will now look out of 250 ohms, instead of the ideal 500 ohms. This addition can invalidate either gain or frequency response, or both.

At the output end, an amplifier can be designed to feed a 500-ohm (or any other desired impedance) load, without necessarily providing a source impedance of the same value. Most often, unless special measures are taken in the amplifier design to equalize these impedances, the source impedance will differ from the impedance of the load it is designed to feed into.

These two possibilities for deviation mean that insertion gain may not hold to its academic definition. Whenever source and load impedance, whether it's at the input or output end, are not equal, putting the equipment into a circuit changes the basic, theoretical 6 dB loss effective without the equipment in circuit.

The error this causes is particularly noticeable when equipment is cascaded. For example, a system may be built up of mixers, equalizers, compensators, etc., with line amplifiers or preamplifiers interspersed between these various groups of controls to equalize level losses. When the whole system is measured for over-all performance, the total gain may be considerably less or more than the sum of the parts, because of these discrepancies.

The discrepancies can best be overcome by figuring things out in a little more detail, although the standards do not call for this procedure. Know the input and output impedances, both internal values as well as the nominal, or matching values. This may require a little more measuring, but it can save some disappointments.

Frequency Response.

The same variations in impedances can invalidate frequency-response measurements. The measurement procedure calls for use of correct input and output termination. Actual systems may connect together inputs and outputs with the same rated, or nominal impedance, but in which neither has the *actual* value named. Each is merely intended to *work with* that value—say 500 ohms.

Thus, for example, an output designed to feed into 500 ohms may have a source impedance of 20 ohms, while an input designed to accept from a 500-ohm circuit may present a terminating load impedance of 2,000 ohms. Connecting these items together seems correct: two 500-ohm circuits, and input and an output, are connected together—matching is correct; but in fact, a 20 ohm source is connected to a 2,000 ohm load.

Gain as well as frequency response of both units is predicated on coupling to a true 500-ohm circuit. So, as well as changing the apparent insertion gain of both units, the frequency response of each may be different from its measured or specified performance. An input circuit designed to work from 500 ohms may work quite differently when connected to 20 ohms, as may the output connected to 2,000 ohms.

Putting in attenuator pads that reduce level by a known amount, and bring the impedance nearer to its nominal value, can minimize both these deviations (fig. 3). If you know your situation, you can correct accordingly. For example, a 500 ohm 6 dB pad would make the 20-ohm source look like about 300 ohms and the 2,000 input look like about 650 to 700 ohms—an improvement. A 10 dB pad will do much better. The loss of level must be available—amplification sufficient to compensate it.

The only way to really know is to run gain and frequency response checks under all possible operating conditions, particularly extreme settings of any controls, and never to assume that the over-all performance of a system will be the sum of its parts!

Distortion.

Although errors can creep into the measurements of gain and frequency response as just discussed, unless carefully applied, these errors can be corrected fairly readily. With distortion measurements we encounter more difficult problems. While the differences already mentioned as affecting gain and frequency response can also affect distortion, there are some things even more basically deficient about distortion-measuring standards. And extra distortion due to unintentional mismatch may not be so easy to compensate out, as deviations in gain or frequency response. The only way is to carefully see that matching is correct, both ways (forward and backward).

The more basic deficiencies about distortion measurements have shown up to some extent in the variety of tests that have already been developed. The earliest, still

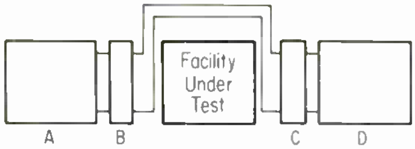
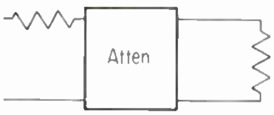


Fig. 2. The concept of insertion gain: the gain with the facility under test inserted is compared with the condition when a simple electrical by-pass is made, as shown here.



dB Atten	Mismatch Ratios 1.5	2	3	5	7	10	20	100
3	1.2	1.4	1.7	2	2.2	2.4	2.7	3
6	1.1	1.2	1.3	1.4	1.5	1.5	1.6	1.7
10			1.1	1.1	1.2	1.2	1.2	1.2
13	Less Than 10% Error Reflected							1.1

Fig. 3. Inserting a two-way attenuation pad to minimize mismatch due to internal impedance not being the same as the nominal or rated impedance. Such an attenuator should be of the two-way matching type (T, pi, H, box or lattice configuration). The table shows the reduction in mismatch for various attenuation values.

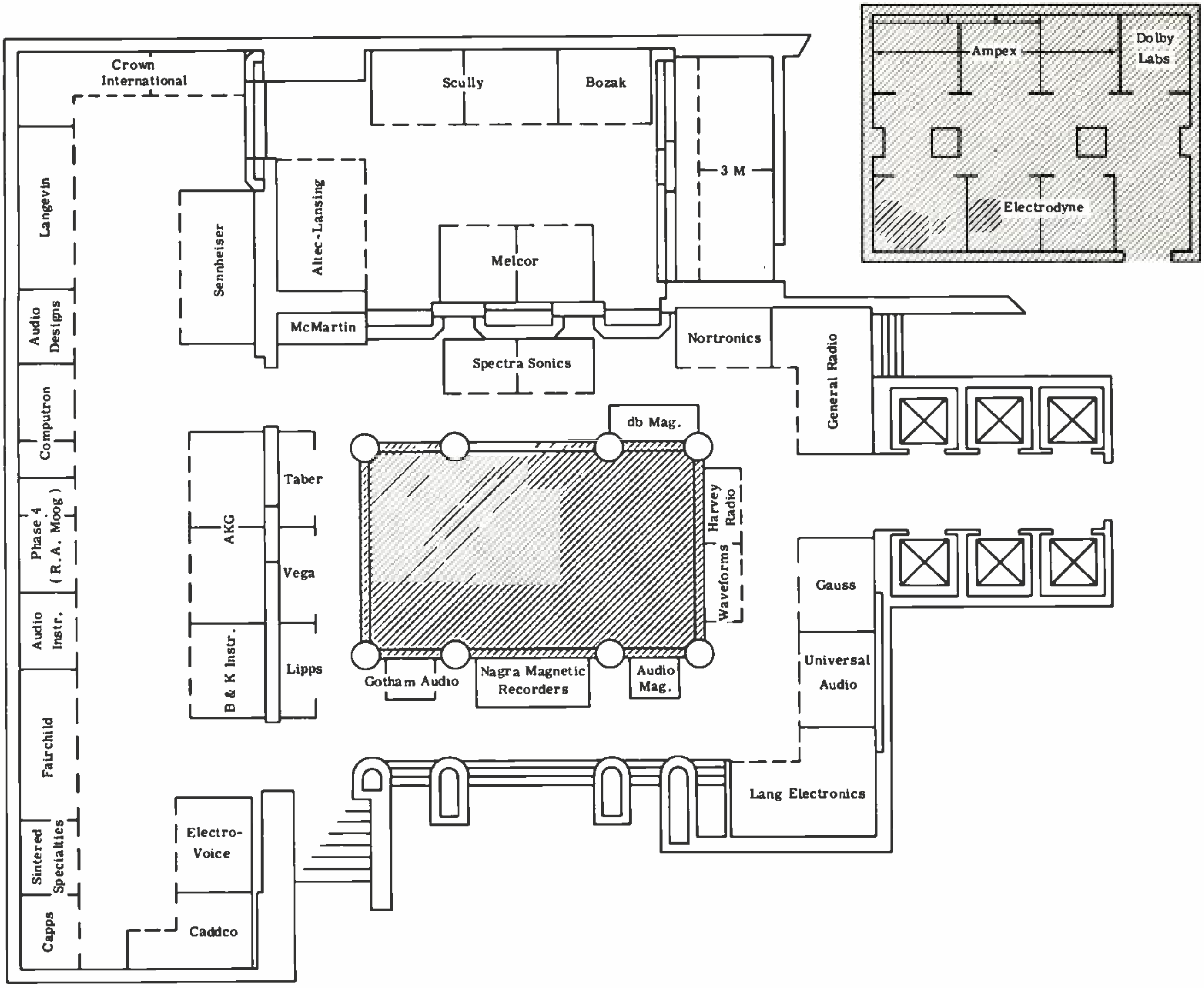
accepted as the basic standard, is the test with a single input frequency, to determine the generation of spurious frequencies (harmonics) in the output.

Later, recognizing some of the limitations of this test, came various intermodulation tests, using more than one frequency.

Two methods are available for the single-frequency test. Each must use the purest possible input signal tone—at least an order of magnitude better than the expected distortion content of the equipment tested: if the distortion is expected to be 0.1%, the input must have less than 0.01% harmonic present.

These two methods differ in how the output is analyzed. The classic method uses a wave analyzer, measures harmonics, usually up to the 5th, but higher if appreciable amounts are suspected present, from which the resultant distortion is calculated by taking the square root of the sum of the squares.

(Continued on page 24)



33rd AES Convention

Registration

LOBBY – Barbizon Plaza Hotel

Monday, Tuesday and Thursday,
October 16, 17 and 19
9:00 A.M. to 8:00 P.M.

Wednesday, October 18
9:00 A.M. to 6:00 P.M.

Banquet

BARBIZON ROOM – Barbizon Plaza Hotel

Wednesday, October 18 Social Hour – **6:00 P.M.**
Banquet – **7:00 P.M.**

Exhibit Hours

Monday, October 16	**4:00 to 9:00 P.M.**
Tuesday, October 17	**Noon to 9:00 P.M.**
Wednesday, October 18	**Noon to 5:00 P.M.**
Thursday, October 19	**Noon to 7:30 P.M.**

Technical Sessions

9:00 A.M. – MONDAY, OCTOBER 16, 1967

ANNUAL BUSINESS MEETING

9:30 A.M. – MONDAY, OCTOBER 16, 1967

AUDIO APPLICATIONS

Chairman: Frank A. Comerci, CBS Laboratories, Stamford, Connecticut

A REVIEW OF SOME OF THE APPLICATIONS OF ACOUSTIC WAVES TO OIL FIELD PROBLEMS – Gerald J. Crawford, Schlumberger Technology Corporation, Ridgefield, Connecticut

MORTAR LOCATION WITH ORTHOGONAL GRADIENT MICROPHONES – Edward J. Foster, L. T. Fiore, and B. B. Bauer, CBS Laboratories, Stamford, Connecticut

ACOUSTICAL MEASUREMENTS BY TIME DELAY SPECTROMETRY – Richard C. Heyser, Cal Tech Jet Propulsion Laboratory, Tujunga, California

R-C ACTIVE FILTERS FOR AUDIO FREQUENCY APPLICATIONS – David Friend, RCA Laboratories, Princeton, New Jersey

DESIGN OF AN INDUCTIVE COMMUNICATION SYSTEM FOR SHIPBOARD APPLICATION – Philip Marino, Michel Copel, US Naval Applied Science Laboratory, Brooklyn, New York

MAGNETIC TAPE CONSIDERATIONS FOR CONTINUOUS LOOP OPERATION – Ralph Cousino, Orrtronics, Inc., Toledo, Ohio

1:30 P.M. – MONDAY, OCTOBER 16, 1967

TRANSDUCERS

Chairman: Philip B. Williams, Jensen Mfg. Div., The Muter Company, Chicago, Illinois

DISPERSION AS A FACTOR IN LOUDSPEAKER PERFORMANCE – Charles L. McShane, Acoustic Research, Inc., Cambridge, Massachusetts

NONLINEAR DISTORTION IN DYNAMIC DIRECT RADIATOR LOUDSPEAKERS – Harry F. Olson, RCA Laboratories, Princeton, New Jersey

TRANSIENT RESPONSE OF LOUDSPEAKER CROSSOVER NETWORKS – Oliver McDaniel, Pennsylvania State University, State College, Pennsylvania

AN IMPLANTABLE APPLANATION TRANSDUCER – Philip Kantrowitz, Sonotone Corp., Elmsford, New York; Paul Freed, Maimonides Medical Center, Brooklyn, New York; Dr. T. Okura, University of Tokyo, Japan

APPLICATION OF A MINIATURE NOISE SUPPRESSING MICROPHONE TO AN OXYGEN MASK – George R. Sebesta, Arthur J. Mellen, Alan Hofer and Richard W. Carlisle, Dyna Magnetic Devices, Inc., Hicksville, New York

BASIC CIRCUIT CONSIDERATION IN FET-EQUIPPED CONDENSER MICROPHONES – Stephen F. Temmer, Gotham Audio Corporation, New York, New York

SOME PROBLEMS IN HEAD-WORN AID RESPONSE MEASUREMENTS – Hugh S. Knowles, Knowles Electronics, Inc., Franklin Park, Illinois

THE "HUMANIZED" HEADPHONES – Robert Amplatz, AKG, Vienna, Austria (To be read by Andrew A. Brakhan)

7:30 P.M. – MONDAY, OCTOBER 16, 1967

SPEECH AND MUSIC

Chairman: Robert A. Moog, R. A. Moog Company, Trumansburg, New York

INFORMATION CONTENT OF A SOUND SPECTROGRAM – Tiong Suy Yu, Wayne State University, Detroit, Michigan

LIMITED-VOCABULARY ADAPTIVE SPEECH-RECOGNITION SYSTEM – Paul W. Ross, RCA Laboratories, Princeton, New Jersey

APPARATUS FOR GENERATING SERIAL SOUND STRUCTURES – Hugh Le Caine, National Research Council of Canada, Ottawa, Canada

ELECTRONIC MUSIC IN COMMUNICATIONS – Eric Siday, Identitones Incorporated, New York, New York

COMPUTER CONTROL OF SOUND APPARATUS FOR ELECTRONIC MUSIC – James Gabura and Gustave Ciamaga, Department of Computer Science, University of Toronto, Toronto, Ontario, Canada

CREATIVE ASPECTS OF LIVE-PERFORMANCE ELECTRONIC MUSIC TECHNOLOGY – Gordon Mumma, Cunningham Dance Foundation, New York, New York

MUSIC SIGNAL PROCESSING FOR FUN AND PROFIT – Robert A. Moog, R. A. Moog Co., Trumansburg, New York

INTEGRATED CIRCUITS IN ORGANS – Harold O. Schwartz, The Wurlitzer Company, North Tonawanda, New York

9:30 A.M. – TUESDAY, OCTOBER 17, 1967

AMPLIFIERS

Chairman: A. W. Linder, H. H. Scott, Inc., Maynard, Massachusetts

THE CHARGE AMPLIFIER – David Bell, Electro-Voice, Inc., Buchanan, Michigan

SOURCES OF NOISE IN MAGNETIC TAPE REPRODUCTION AND THEIR MINIMIZATION – Norman P. Doyle, Fairchild Semiconductor, Mountain View, California

DESIGN FACTORS AND CONSIDERATIONS IN FULL COMPLEMENTARY SYMMETRY AUDIO POWER AMPLIFIERS – George C. Haas, Motorola Semiconductor Products Inc., Phoenix, Arizona

PROTECTION OF AMPLIFIERS – Daniel R. von Recklinghausen, H. H. Scott, Inc., Maynard, Massachusetts

LATEST ADVANCES IN LINEAR INTEGRATED CIRCUITS FOR PROFESSIONAL AUDIO AMPLIFIERS – Basil T. Barber, Sperry Gyroscope Corp., Great Neck, New York

POWER OUTPUT AND POWER DISSIPATION IN CLASS B TRANSISTOR AMPLIFIERS – R. S. Hartz, RCA, Somerville, New Jersey

AUTOMATED PRODUCTION TESTING OF AUDIO AMPLIFIERS – Derek Wheeler, McCurdy Radio Industries, Toronto, Ontario, Canada

1:30 P.M. – TUESDAY, OCTOBER 17, 1967

CONTROL SYSTEMS

Chairman: Philip C. Erhorn, Philip C. Erhorn and Associates, Inc., Stony Brook, New York

ACTIVE ISOLATION "TRANSFORMERS" IN STUDIO CONSOLE DESIGN – William G. Dilley, Spectra Sonics, Ogden, Utah

A MODULAR AUDIO FACILITIES MIXING SYSTEM – John P. Jarvis, Langevin, Santa Ana, California

A VERSATILE TABLE TOP AUDIO CONTROL CONSOLE – Donald N. McLaughlin, Electrodyne, North Hollywood, California

REMOTE CONTROL OF MODULAR EQUIPMENT IN AUDIO CONSOLES – George Alexandrovich, Fairchild Recording Equipment Corp., Long Island City, New York

A NEW CUSTOM MODULAR CONSOLE CONSTRUCTED WITH STANDARD STOCK COMPONENTS – Arthur C. Davis, Controls Division, Altec Lansing, Anaheim, California

ABC'S NEW STANDARD AUDIO PRODUCTION CONSOLE FOR TELEVISION – James R. Baker, American Broadcasting Company, New York, New York

RERECORDING CONSOLE FOR THE PREPARATION OF DUPLICATING MASTER TAPES – Mitchell G. Heller, Tape Transfer Techniques, Chicago, Illinois, Alfred Antlitz, Radio Station WFMT, Chicago, Illinois

A HIGH PERFORMANCE CONTROL CONSOLE WITH FLEXIBILITY – Allan P. Smith, Naval Training Device Center, Orlando, Florida

A COMPLETELY SOLID STATE AUDIO FOLLOW VIDEO SWITCHING SYSTEM – Michael H. Stoll, Sarkes Tarzian, Inc., Bloomington, Indiana

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7:30 P.M. – TUESDAY, OCTOBER 17, 1967

SOUND REINFORCEMENT

Chairman: Wayne Rudmose, Tracor, Inc., Austin, Texas

AUDIO-VISUAL ENGINEERING – A NEW DISCIPLINE – William Szabo, Will Szabo and Associates Ltd., New Rochelle, New York

CHURCH SOUND REINFORCEMENT – Daniel W. Martin, D. H. Baldwin Co., Cincinnati, Ohio

A DIFFICULT CHURCH SOUND SYSTEM DESIGN – G. R. Thurmond, Tracor, Inc., Austin, Texas

CHARACTERISTICS INFLUENCING THE GAIN OF SPEECH REINFORCEMENT SYSTEMS – Daniel Queen, Perma-Power Co., Chicago, Illinois

A 1/3 OCTAVE BAND VARIABLE NOTCH FILTER SET FOR PROVIDING BROADBAND EQUALIZATION OF SOUND SYSTEMS – Donald B. Davis, Altec-Lansing, Anaheim, California

SOUND AT EXPO '67 – N. E. Rudback, N. J. Pappas and Associates, Montreal, Canada

SPECIAL REPRODUCE SYSTEM FOR THE INDEPENDENCE HALL AT "KNOTT'S BERRY FARM" – Akio Hosoda, Ampex Corporation, Redwood City, California

INTEGRATION OF HIGH LEVEL SOUND SYSTEMS AND ACOUSTIC ENVIRONMENT IN THE HOUSTON MUSIC THEATRE Charles F. Webber, Audio Acoustic Engineering, Santa Ana, California

A SOLID-STATE SCHOOL PAGING AND MUSIC SYSTEM – Philip C. Erhorn, Philip C. Erhorn and Associates, Inc., Stony Brook, New York

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9:30 A.M. – WEDNESDAY, OCTOBER 18, 1967

BROADCASTING

Chairman: Harold L. Kassens, FCC Broadcast Bureau, Washington, D. C.

THE MEASUREMENT OF LOUDNESS LEVEL – Benjamin B. Bauer, Emil L. Torick and Allen J. Rosenheck, CBS Laboratories, Stamford, Connecticut

AUTOMATIC CONTROL OF LOUDNESS LEVEL – Emil L. Torick, Richard G. Allen, and Benjamin B. Bauer, CBS Laboratories, Stamford, Connecticut

CONTROL OF BROADCAST PROGRAM LOUDNESS – Arnold Schwartz, American Broadcasting Company, New York, New York

LOUDNESS CONTROL AT NBC – J. L. Hathaway, National Broadcasting Company, New York, New York

EDUCASTING – Lewis D. Wetzel, Triangle Publications, Inc., Philadelphia, Pennsylvania

AUTOMATIC SELF-MONITORING FOR FM TRANSMITTERS – J. L. Smith, Collins Radio Company, Richardson, Texas

ADVANCEMENTS IN SCA RECEIVER DESIGN – Leonard Hedlund, McMartin Industries, Inc., Omaha, Nebraska

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1:30 P.M. – WEDNESDAY, OCTOBER 18, 1967

STANDARDS

Chairman: Floyd K. Harvey, Bell Telephone Laboratories, Inc., Murray Hill, New Jersey

RIAA ENGINEERING COMMITTEE – H. E. Roys, Indianapolis, Indiana

IEEE'S WORK ON STANDARDS IN THE FIELD OF AUDIO – Robert H. Rose, Newark College of Engineering, Newark, New Jersey

A BRIEF REVIEW OF EIA STANDARDS IN THE AUDIO FIELD – J. A. Caffiaux, Electronic Industries Association, Washington, D. C.

THE STANDARDS ACTIVITIES OF THE INSTITUTE OF HIGH FIDELITY – Daniel R. von Recklinghausen, H. H. Scott, Inc., Maynard, Massachusetts

THE ROLE OF THE SOCIETY OF MOTION PICTURE AND TELEVISION ENGINEERS IN AUDIO STANDARDIZATION – H. W. Knop, Jr., E. I. du Pont de Nemours and Co., Inc., Rochester, N. Y.

NAB AUDIO STANDARDS – Warren L. Braun, Consulting Engineer, Harrisonburg, Virginia

RULES AND REGULATIONS OF THE FEDERAL COMMUNICATIONS COMMISSION PERTAINING TO

THE AUDIO FIDELITY CHARACTERISTICS OF FM BROADCAST TRANSMITTING EQUIPMENT – John T. Robinson, Federal Communications Commission, Washington, D. C.

USA STANDARDS FOR ACOUSTICAL MEASUREMENTS – Richard K. Cook, Environmental Science Serv. Admin., Rockville, Maryland

INTERNATIONAL STANDARDS FOR AUDIO – William W. Lang, International Business Machines Corp., Poughkeepsie, New York

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WEDNESDAY EVENING, OCTOBER 18, 1967

6:00 P.M. SOCIAL HOUR

7:00 P.M. THE JOHN H. POTTS

MEMORIAL AWARDS BANQUET

Honor Guests – Past Recipients of the John H. Potts Memorial Award.

Presentation of Awards
Guest Speaker – JEAN SHEPHERD

9:30 A.M. – THURSDAY, OCTOBER 19, 1967

DISC RECORDING AND REPRODUCTION

Chairman: Welton H. Jetton, Pepper Sound Studios, Inc., Memphis, Tennessee

TRACKING ABILITY SPECIFICATIONS FOR PHONOGRAPH CARTRIDGES – J. H. Kogen, Shure Brothers, Inc., Evanston, Illinois

AN AUDIO NOISE REDUCTION SYSTEM – Ray M. Dolby, Dolby Laboratories, London, England

FACTORS AFFECTING THE NEEDLE/GROOVE RELATIONSHIP IN PHONOGRAPH PLAYBACK SYSTEMS – C. R. Bastiaans, Westinghouse Research Laboratories, Pittsburgh, Pennsylvania

PHONOVID, A SYSTEM FOR RECORDING TELEVISION PICTURES ON PHONOGRAPH RECORDS – Kenneth E. Farr, Westinghouse Electric Corporation, Pittsburgh, Pennsylvania

THE SONY SERVOMATIC TURNTABLE – Sam Mabuchi, Sony Corporation of America, Long Island City, New York

THE GENERAL ELECTRIC LETTERWRITER – Jack L. Kelly, Audio Products Department, General Electric Company, Decatur, Illinois

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1:30 P.M. – THURSDAY, OCTOBER 19, 1967

TAPE RECORDING AND REPRODUCTION

Chairman: Arthur E. Gruber, Consultant, East Rockaway, New York

THE GRANDY TDS-1 TAPE DUPLICATING SYSTEM – Donald E. Killen, Grandy, Inc., West Caldwell, New Jersey

NEW ELECTRONICS FOR THE 3M MASTER TAPE RECORDER – C. Dale Manquen, 3M Revere Mincom Division, Camarillo, California

DESIGN FEATURE OF A FLEXIBLE MULTI-TRACK RECORDER-REPRODUCER – John Curtis, Scully Recording Instruments Corp., Bridgeport, Connecticut

PITCH AND TIMING ERROR IN TAPE RECORDING AND REPRODUCING – John McKnight, Ampex Consumer and Education Products Div., Los Gatos, Calif.

ALIGNMENT OF STEREOPHONIC TAPE MACHINES – Henry Korkes, CBS Radio Division, New York, New York

MAGNETIC HEAD LAPPING FOR PROFESSIONAL TAPE RECORDERS – William Taber, Taber Manufacturing and Engineering Co., Alameda, Calif.

DESIGN CONSIDERATIONS IN THE BOSTON UNIVERSITY LIBRARY HEADPHONE LISTENING SYSTEM – William R. Lewis, Audio Lab Inc., Cambridge, Massachusetts

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7:30 P.M. – THURSDAY, OCTOBER 19, 1967

FILM RECORDING AND REPRODUCTION

Chairman: John Arvonio, Photo-Magnetics, Inc., New York, New York

OPTICAL SOUND RECORDING WITH A SILICON CARBIDE ELECTROLUMINESCENT DIODE – Allan S. Miller, Paul L. Vitkus, National Research Corp., Cambridge, Massachusetts

SYNCHRONOUS SOUND FOR MOTION PICTURES – Loren L. Ryder, Ryder Sound Services, Inc., Hollywood, California

A MOBILE ELECTRO-ACOUSTIC LABORATORY – Allan P. Smith, Naval Training Device Center, Orlando, Florida

THE FILM PROJECTOR – MAGNETIC REPRODUCER DOUBLE SOUND SYSTEM AT CBS TELEVISION, NEW YORK – Arthur C. Helmer, CBS Television Network, New York, New York

AUDIO FACILITIES FOR THE LABYRINTH PAVILION, EXPO '67 – Peter Mundie, N. J. Pappas and Associates, Montreal, Canada

THE SURVEY OF CURRENT PROBLEMS IN TV TRANSMISSION OF 16 mm SOUND-ON-FILM – James Townsend, Townsend Production Service, New York, New York

Crowhurst

(Continued from page 19)

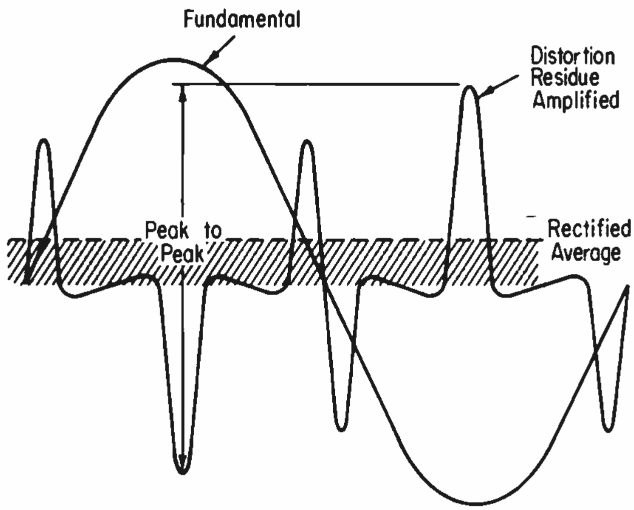


Fig. 4. Why peak reading of residual composite harmonic is more meaningful than mean reading: in this typical waveform, for a low-distortion signal, the peak value has about five times the mean value (peak-to-peak is about 10 times).

For example, if the readings are: 2nd, 0.04%; 3rd, 0.06%; 4th, 0.01%; 5th, 0.007%; the resultant is

$$0.04 \quad + 0.06 \quad + 0.01 \quad + 0.007$$
$$= 0.0016 + 0.0036 + 0.0001 + 0.000049$$
$$= 0.005349 = 0.073\%$$

The alternative may find a less accurate "rms" distortion (for whatever that is worth), with a simpler measurement method: it uses a frequency-selective bridge to null out the fundamental or input frequency, leaving the total harmonic waveform to be measured by either a mean-reading or peak-reading meter.

The harmonic residue, with the fundamental removed, will have a peak value that has been shown to be somewhat more meaningful than the average value, because the peak value represents the departure from the true waveform, which is nearer to the relative effect of the spurious sound heard (fig. 4).

So with the simplified method there is some merit to using a peak indication. But, using a wave analyzer to read individual harmonics, peak/average always has the same ratio (1.414) so there's no point to the distinction with this method. Only by reading the composite residue, with fundamental removed, can this improvement be utilized.

But there is a way to make the whole measurement easier, that many equipment manufacturers have adopted for their production test facilities, although no professionally available measuring set as yet employs it. This consists of a bridge that balances out *input* against output (fig. 5) rather than nulling out the *frequency*.

This method has several advantages: most obvious to the user, the null is far less sensitive to work with. Using frequency nulling, if frequency changes at all (even a hertz or so) balance is completely lost. Using input/output nulling, change in test frequency has comparatively little effect on balance. It is much more comfortable to use.

Another advantage is that harmonic content in the test signal does not have to be a whole order of magnitude lower than the quantity to be measured. This method allows the whole output waveform to be nulled by a component representing the whole input, leaving only spurious components due to the equipment being tested.

While nulling of harmonic components may not be perfect when the fundamental is nulled, it will usually be well within 10%, which gives the order of magnitude margin, without having to be that much "cleaner." Thus, to measure a harmonic content of 0.1%, an input signal of 0.1% harmonic is satisfactory. The harmonic in the input will null its part of the output to within 0.01% or better.

IM Distortion.

Multi-frequency distortion testing uses one of two signal forms. Most popular in this country is the *SMPTE* test, using a lower frequency with a higher one "riding" it, usually in 4:1 ratio. When intermodulation occurs, the higher frequency is modulated by the lower one, causing spurious components, usually shifted from the higher frequency by difference frequencies that are multiples of the lower frequency.

The usual method here employs filters that first remove the lower frequency from the output, then rectify the higher frequency and "smooth" it, to measure any fluctuations in its amplitude occurring at the lower frequency (fig. 6). For the kinds of distortion in this category that were common in earlier amplifiers (before feedback) this method gave a fairly good indication.

But with modern feedback amplifiers, the lower frequency can frequency modulate the upper one, as well as amplitude modulating it. The traditional method detects only amplitude modulation. Audibly, both forms are equally annoying. Searching for the spurious components with a wave analyzer and then calculating the rms resultant intermodulation would overcome this deficiency.

Alternatively, an adaptation of the input/output nulling method used for harmonic measurement would also detect both forms of intermodulation more simply.

The second kind of intermodulation test, adopted as the European or International standard, uses two higher frequencies, usually spaced by a controlled difference frequency, such as 100 Hz. Thus the input frequencies might be 5,000 Hz and 5,100 Hz. The output is then tested for content at 100 Hz, which is a distortion product.

This first-order product is due to asymmetrical distortion. And one problem of this form of test is that the product magnitude is not necessarily related to the annoyance value of the distortion it represents. And a complete test must explore for multiples of the difference frequency.

Each kind of test has some validity. There is no fixed relationship between the results of the three tests (the single-frequency harmonic and the two forms of IM test) and each is subject to interpretation, according to the method of measurement and the form of distortion present in a particular case.

For example, the difference-frequency test can be quite invalidated if there is a bass cut in the system, because the difference frequency will be seriously attenuated in the measurement. But the form of distortion could still be quite objectionable on music.

Noise.

These measurements are relatively simple, so we'll not take space to describe them in detail here. The input must be terminated with its nominal impedance, suitably shielded, to prevent spurious pickup (hum) and the output measured for noise, with or without weighting.

Whether or not weighting is used is important in comparing results obtained with other equipment. Weighting is intended to make the reading indicate more closely the audible effect of the noise. Without it, noise concentrated in the lower range may indicate much higher than it sounds, while noise concentrated in the upper frequency band may sound much higher than the indication would indicate.

Transient Performance.

The big problem in audio measurements is that the tests just described don't account for everything that happens when handling program, especially music. Most of the troubles experienced in modern equipment are related to transient handling ability. Many amplifiers that perform impeccably on steady tone test signals seem to "break up" in some way on certain parts of musical program.

One form of transient test uses tone bursts, where the signal level is switched at a lower frequency, to give opportunity to observe or measure what happens when a loud signal suddenly hits an amplifier and again how it recovers after the loud signal is removed.

An important part of this, which can be detected by tone-burst testing, but for which we may yet find better methods, is the overload recovery effect in an amplifier or system. Most tests, both with tone bursts and steady tones, keep the signals within the rated maximum level—or at least they don't exceed it.

Program often produces isolated "spikes" that do overshoot the maximum level for quite a brief moment. Some amplifiers or equipment find this really crippling—blocking for a fraction of a second after it. Others handle these spikes without trouble. While tests have been devised to demonstrate the difference, there is room for test standardization in this important area.

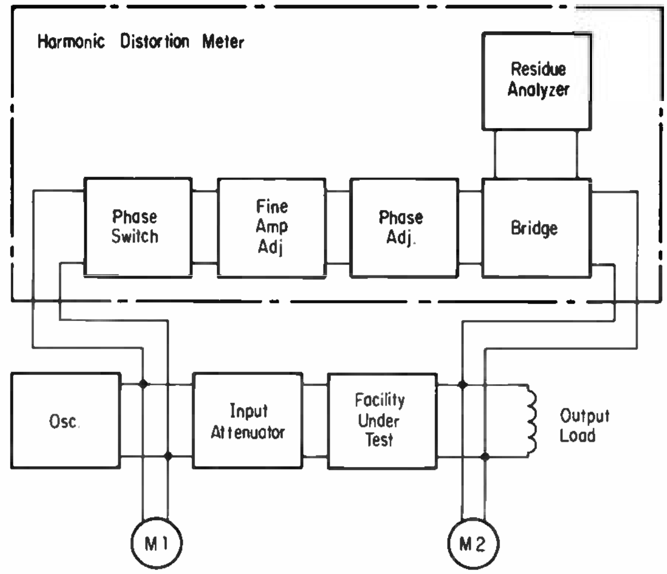
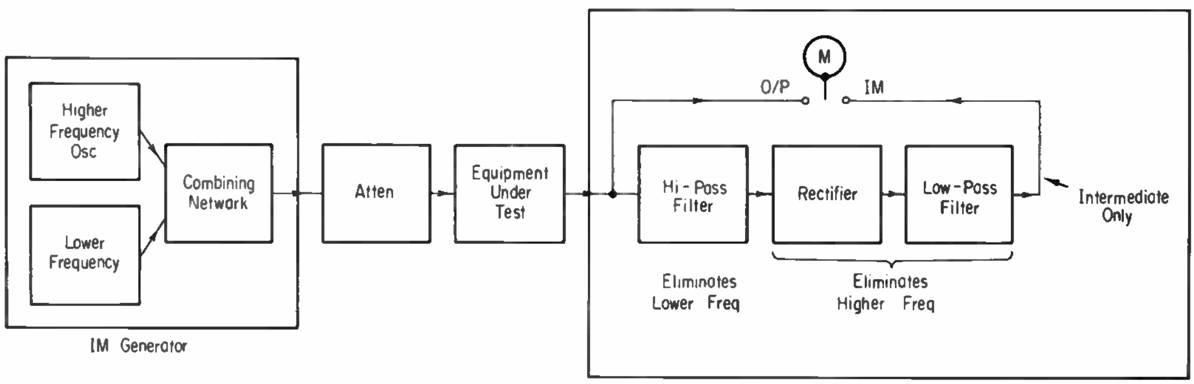
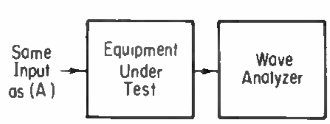


Fig. 5. An alternative measurement technique for harmonic distortion that is both easier to use, and gives greater accuracy within limitations of oscillator purity.

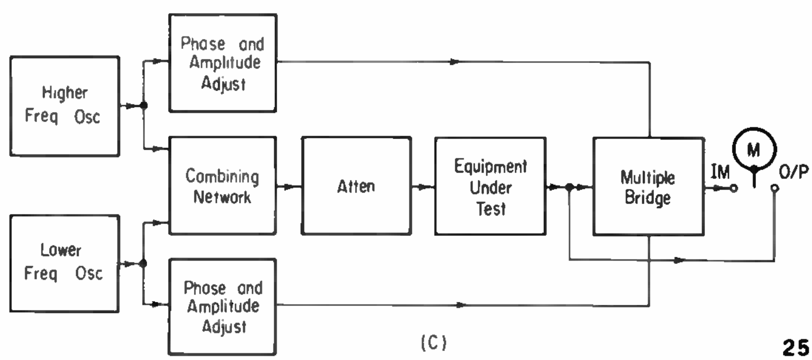
Fig. 6. Three methods of measuring intermodulation distortion of the SMPTE type: (a) the classic SMPTE method, which detects only amplitude intermodulation effects, fails to detect frequency modulation effects; (b) using a wave analyzer and calculating the rms value from all the spurious-frequency readings obtained, will catch what the classic method misses; (c) a simpler form of test, using bridge nulling, that catches all forms of intermodulation occurring with this form of test signal.



(A)



(B)



(C)

Tape Duplicating

Thomas Everett*

New consumer tape formats have placed formidable demands on commercially recorded tape duplicators. Here are Ampex's answers to the demands made by the Philips cassette system, and the constant clamor for higher quality in all formats.

The proliferation of stereo tape playback configurations now being used by today's consumers—open reel, four and eight-track cartridge, and cassette—has brought about tremendous demands on tape duplicating engineers for faster and more sophisticated mastering and duplicating systems. These engineers are faced with such sticky problems as multiple channels on narrow-width tape and playback speeds as slow as 1-7/8 ips. Even the familiar four-track open-reel playback system is the object of further improvement in reproduction quality.

This all adds up to the fact that we are in a period of intense competition to develop further and perfect the art of tape mastering and duplicating so as to offer the consumer, no matter what his choice of tape system, a better product than ever before.

The newest medium on the scene requiring the attention of duplicating engineers is the mastering and duplicating of cassette stereo tapes. At Ampex the duplication of four tracks on 150-mil width tape posed some formidable problems for engineers whose goal was to produce a high-quality sound from a tape that played back at 1-7/8 ips.

The new cassette duplicating system utilizes two Ampex 3300 master units at the head of each duplicating line, rather than the usual single unit. This permits continuous duplicating by eliminating the need to stop the slaves while the master tape is rewound. This dual mastering system lends itself especially to cassette duplicating because it gives a much higher production rate and eliminates the possibility of tape stretch during start and stop.

The reels of duplicated tape are 3,600 feet in length, hold up to 23 to 24 complete albums and can be duplicated on the tape on a continuous basis. Future use of larger reels of tape could produce as many as 50 albums on one reel.

*Manufacturing Engineering Manager
Ampex Stereo Tapes
Elk Grove Village, Illinois*

All four channels are mastered and duplicated simultaneously. Unlike open reel tapes, cassette tape channels are adjacent; channels 1 and 2, left to right, and 3 and 4 right to left. Stereo track widths are 0.024 inches wide.

A special tape duplicator was designed by the Ampex special products groups in Redwood City, California for use in cassette duplicating. This new slave, uniquely qualified to produce high-quality tapes in this narrow-width format, is a precision transport about the size of a portable closed-circuit video recorder.

Unlike any other slaves, these duplicators feature a tape tension control mechanism that prevents the tape from stretching when a change in mode is made. Dual capstan drive on these slaves virtually eliminates flutter and wow. Both of these features are prime requisites for quality tape reproduction at 1-7/8 ips. Also, a highly sophisticated guide system eliminates any tracking problems—a critical factor when recording four tracks on 150-mil width tape.

Another uniquely designed companion machine used for quality control checkout is similar to the duplicators but

Fig. 1. These two Ampex 3300 mastering units, using 1/2 inch tape, feed a line of ten cassette duplicators. Continuous tape duplicating is possible with the two masters operating alternately. All illustrations have been made at the Ampex facility in Elk Grove Village, Illinois.

Fig. 2. This line of ten cassette duplicators can produce as many as 10,000 albums a day if it is operated on an around-the-clock basis.

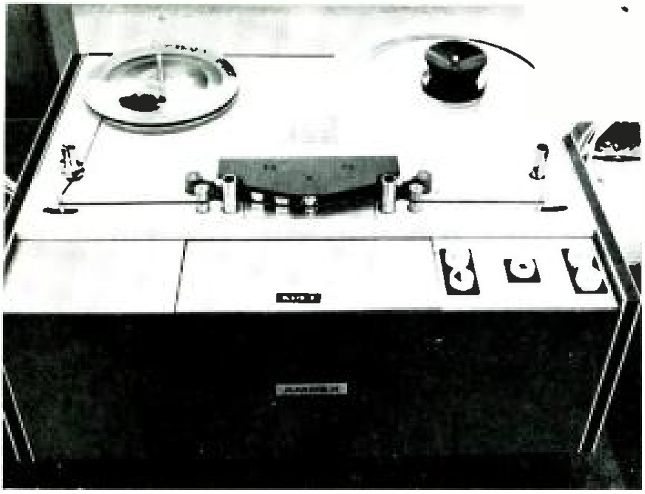


Fig. 3. This closeup of an individual cassette duplicator slave shows this specially designed product. Careful attention to tape tension control prevents tape stretch and a precise guide system eliminates tracking problems.

has two sets of heads, plus an automatic reversing mechanism. This allows complete auditing of cassette tape samples without the need for removing the reel and turning it over. Also featured is a motion sensor that preserves tape quality by completely stopping tape travel before reversing direction.

Production

The new system masters at 120 ips on 1/2-inch tape while the duplicators operate at 30 ips. Because of its ability to duplicate a large number of albums at relatively high speeds, and a highly efficient system of tailoring the duplicates onto single-album hubs and placing them into their cassette cases, this system makes possible production of cassette stereo tapes at a rate of seven to eight times faster than ordinary open reel tapes. One line of 10 slaves with two masters can produce 10,000 cassette albums a day on a three-shift basis.

The stereo tape duplicating in Elk Grove Village, Illinois produces cassettes that offer a flat frequency response up to 10,000 cycles, which, while below the 15,000 cps attainable on 7-1/2 ips open reel tapes, does produce a high-quality sound.

The new cassette duplicating technique will be of special interest to sound engineers and commercial dubbing operators because the cassette concept itself is one that will be coming into widespread use in the very near future. Thought should be given to the use of proper equipment. In addition to large-scale music duplicating, there will be much small-lot duplicating for business firms to be used as sales tools and for schools as playback devices. There is already considerable interest in the concept on the part of the people who sell audio/visual equipment to schools.

Lower Noise Duplication

In addition to developing systems that produce stereo tape duplicates in new configurations, Ampex sound engineers have come up with a new stereo tape mastering and duplicating process that substantially reduces background noise during tape playback that will be of interest to professional audio people.

Utilizing a new equalization curve the new system, which is being called EX+, permits up to a 100 per cent increase in the volume of the sound recorded on the master tape over standard recording techniques. The greater volume, which is passed on to the tape duplicate, reduces extraneous noise as much as 50 per cent, vastly improving the signal-to-noise ratio.

The EX+ recording technique is made possible by the elimination of the "saftey zone" used in ordinary recording. The new equalization curve is monitored on a new highly sensitive meter coupled with new record and reproduce amplifiers, thus permitting maximum recording level. The safety zone heretofore had been maintained at approximately 6 dB below the distortion level which reached its maximum at an rms flex of 600 pico webers per millimeter of track width.

The result is a substantially louder signal recorded both onto the master and duplicate tapes while the background noise remains the same. All Ampex EX+ albums are recorded on polyester tape, virtually eliminating the possibility of breakage and prolonging head life.

The Ampex EX+ process is in many ways complimentary to another recent development that represents the state of the art in tape duplicating. The new Dolby system is a highly effective means of reducing noise level from 10 to 15 dB at the point of original recording. This is accomplished by boosting or stretching low-level signals in four

Fig. 4. These tailoring machines take reels of duplicated albums and wind them onto individual hubs that will fit the cassettes. A sensing device activates a cutting mechanism that separates the individual album hubs of tape.

separate frequency ranges during recording and reducing or unstretching them on usable masters. Ampex has processed several Dolbyized masters for duplicating. And when combined with the EX+ process Dolbyized tapes produce a sound of great purity.

The technology of tape duplicating and mastering is moving fast. Small, easy-to-use stereo cassettes operating at extremely slow speeds are producing sound quality that would never have been considered technically possible as recently as a few months ago. At the same time new techniques for open reel mastering and duplicating are making the purest form of music playback even better.

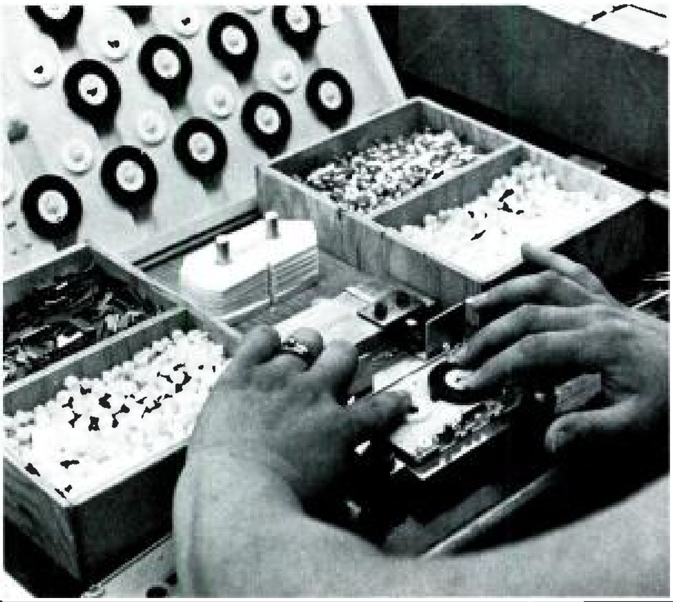


Fig. 5. The hubs of tape which have already been secured to a takeup hub by the tailoring machine, are inserted in the cassette casings prior to final sealing.

CLASSIFIED

Classified advertising is an excellent and low-cost way to place your products and services before the audio professional. If you are a prospective employer seeking skilled help or an employee seeking a change you will find that the classified pages of **db** reach the people you want. Special low rates apply for this service.

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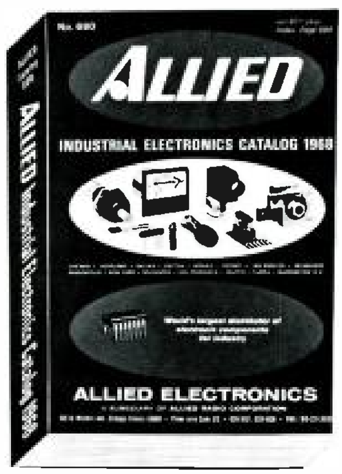
3 times — 10%
6 times — 20%
12 times — 33 1/3%

Agency discounts will not be allowed in any case.

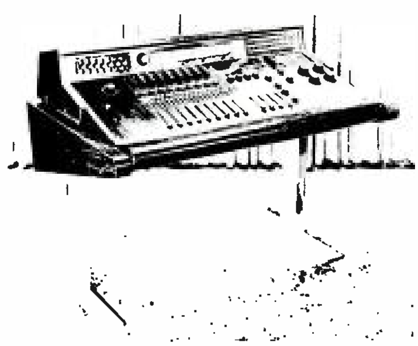
Closing date for any issue is the **fifteenth** of the **second** month preceding the date of issue.

New Products and Services

Audio Control Center
The model AC-155 offers a combination of compactness, mobility, appearance, and strength, plus technical features of professional quality. The four-foot-wide unit has 14 inputs and accommodates six low-level and eight high-level audio sources. An all-channel cue system, muting, monitor amplifier and speaker are included. Also part of the console are two turntable/arm/cartridge systems. Accessories include a matching bench/lid which gives protection to the console in transport and becomes a seat for the operator on location. Tape cartridge systems and a cardioid microphone are also available.
Mfg: Sparta Electronic Corp. Price: On request
Circle 50 on Reader Service Card



Industrial Catalog
This wopper of a catalog has over 600 pages with over 50,000 stock items listed. More than 500 manufacturers are listed. The listings show prices for purchase in various quantities of every type component, including IC's, semiconductors, vacuum tubes, relays, times, transformers, resistors, capacitors, connectors, soils, chokes, sockets, plugs, jacks, switches, fuses, batteries, clips, lamps, wire, and cable. Other sections show test instruments, two-way radios, recording equipment, sound equipment, power supplies, hardware, etc. There is a full index.
Mfg: Allied Radio Corp. Price: No charge
Circle 51 on Reader Service Card

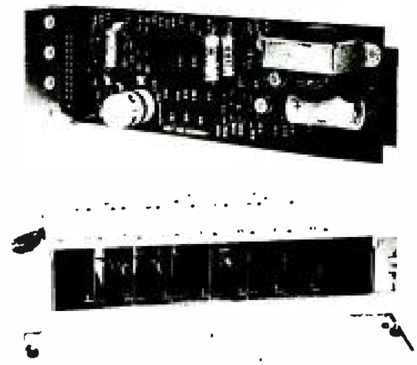


Modular Console, Model 9200
This is a basic, unitized enclosure for modular assembly. A full line of components may be selected to meet the specific requirements of a particular installation. As many as 27 strip modules may be installed, along with up to 4 VU meters. Each module is hinged forward for easy access. Each input panel may contain a rotary mixer, straight-line mixer, equalizer, echo send attenuator, and a multi-switch assembly. Pan pot panels, talk-back panels, playback and monitor panels, jack strips, graphic equalizers, and the meter assembly are all variable to individual requirements.
Mfg: Altec, Div of Ling Altec, Inc. Price: List available on request
Circle 52 on Reader Service Card

Theater Sound Systems

The recent entry of Ampex into the theater sound system supply field has been announced by the California-based company. A building-block concept is being used. Custom systems can be assembled to meet the needs for sound distribution or reinforcement in arenas, gyms, stadiums and and auditoria. Modular products include power amplifiers, preamplifiers, and power supplies. Preamplifiers are available with inputs for microphone; others accept magnetic tape heads from recorders and motion picture projectors. Availability is scheduled for November.

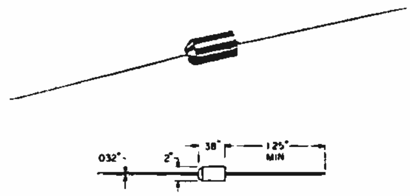
Mfg: Ampex Corporation Price: On request
Circle 53 on Reader Service Card



Two Amp Silicon Rectifier

The pictured rectifiers are miniature, 0.2-in. diameters and 0.38-in length, with transfer molded, void-free bodies. Surge ratings are 200 amperes. PIV ratings range from 50 to 1200 PIV. The EDI miniature 3 ampere axial lead rectifiers replace 7/16-in. stud units and eliminate heat sinks.

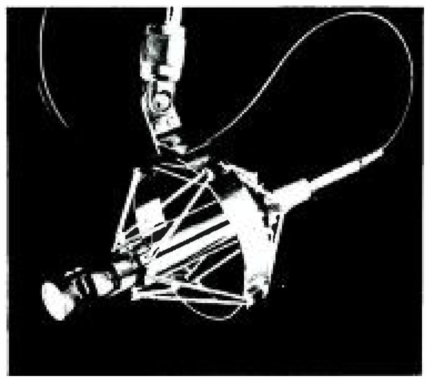
Mfg: Electronic Devices, Inc. Price: (200 PIV) $0.64 each in hundred lots
Circle 54 on Reader Service Card



Self-contained Condenser Microphone

The model S-10 Syncron Condenser is again available now that Vega has acquired the assets of Syncron. The S-10 is completely self-contained and battery-operated. Frequency response is flat from 40-20,000 Hz with cardioid or omnidirectional versions available. Other accessories include an a.c. power supply, windscreen, elastic suspension, and desk stand.

Mfg: Vega Corporation Price: $260
Circle 55 on Reader Service Card



Equalizer/Amp Module using IC's

This is an audio input channel with 71 dB of gain for use in control consoles. The module incorporates a low-noise mike preamp, straight line or rotary attenuator, separate low and high-frequency equalization, isolated echo, cue, and program outputs. An input switch selects line or mike inputs. Two additional switches reduce mike gain by respectively 10 and 20 dB. A coaxially mounted switch selects echo send output from before the attenuator or after the program amplifier. All solid-state construction utilizes the latest IC differential amplifier design techniques.

Mfg: Electrodyne Corp. Price: On request
Circle 56 on Reader Service Card

FET Amplifier, Model 1725

High input impedance (1010 ohms) and low input offset currents (50 pico-amps) characterize this compact unit. The unit is similar to Melcor's existing 1619 amplifier with the exception of output current and open loop gain. These two elements are approximately half those found in the 1619. Other features include operating temperature range of -55-deg. C, to +85-deg. C; output voltage minimum ±10 volts; 10 mHz unity gain crossover and 200 kHz full power bandwidth. Applications are in control instrumentation, computation, test equipment, and data logging.

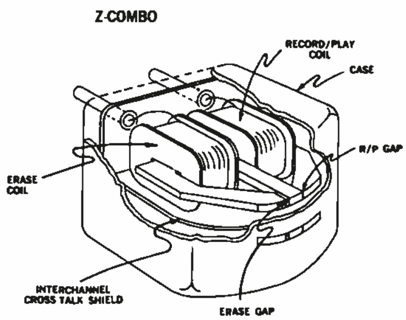
Mfg: Melcor Electronics, Corp. Price: $35.00 each
Circle 57 on Reader Service Card

New Products and Services

New Catalog

The 1968 Lafayette catalog is off the press, showing the latest in mail-order availability from this Long Island based distributor/retailer. The latest in stereo hi-fi, CB radio, tape recorders, test equipment, ham gear, and other home entertainment products is listed. In addition there are extensive lists of current component availabilities for industrial and experimental use.

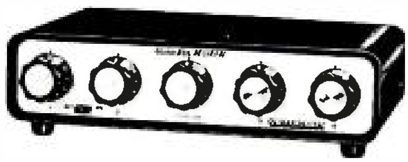
Mfg: Lafayette Radio, Inc. Price: Free
Circle 58 on Reader Service Card



Eight-track R/P-E Head for 1/4-in. Tape

This new head makes available a combined record, play, erase assembly in a single head structure designed to operate with the standard eight-track configuration. The new head is of the 3-leg or "Z-Combo" type with the record/play and erase sections sharing a common center leg. Each gap thus produced is 50 thousands of an inch. This close gap results in greatly improved track location accuracy, vital for the eight-track stereo system. Pressure pads in existing tape cartridges will easily cover both gaps of this new head.

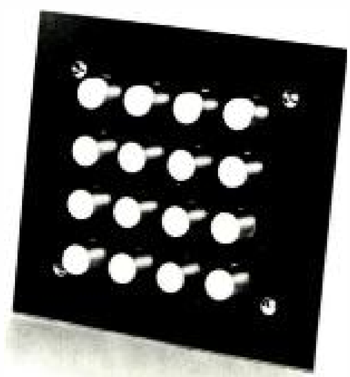
Mfg: Nortronics Company, Inc. Price: On request
Circle 59 on Reader Service Card



Battery Operated Studio Mixer

This small unit is designed specifically for studio, remote, and hobby high-fidelity use. It functions in stereo or mono modes as a mixer/amplifier. Power is from self-contained "D" batteries. Output is better than 2 volts into a high-impedance load. Distortion is 1 per cent maximum (0.5 per cent typical) at 1.5 volts output. Frequency response is 20 to 20,000 Hz. S/n is 60 dB minimum referred to 1 mV input. One to four mono inputs or half the stereo inputs may be used for any type of input including magnetic phono cartridges (RIAA equalization is provided). Each program input is level controlled individually. There is a master over-all gain control. Standard phone and phono-type jacks are used throughout.

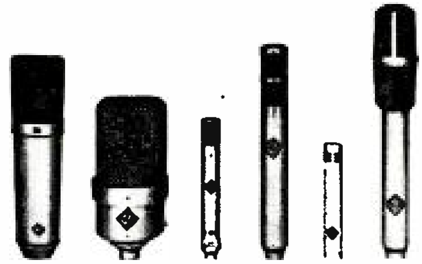
Mfg: Switchcraft, Inc. Price: On request
Circle 60 on Reader Service Card



Pushbutton Multiswitch

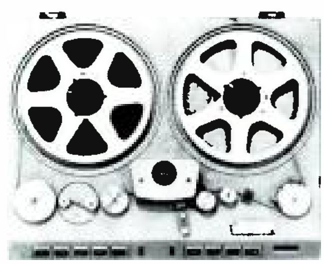
These new boards have been designed to fit in with the Sealectoboard© modular approach to matrix-board programming. They are supplied with any number of single-pole, momentary contacts. Ratings are 500V d.c. at 5 amperes, with contact resistance less than 10 milliohms. Front panels are fabricated with engraving stock and can be furnished marked to suit specific requirements.

Mfg: Sealectro Corp. Price: On request
Circle 61 on Reader Service Card



What do Neumann microphones have in common

with this...



and this...

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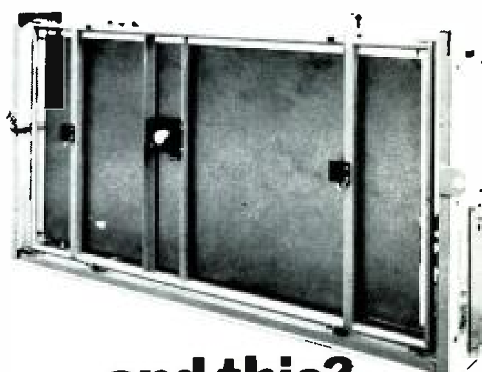
and this...

and this...



and this?

Gotham Audio

Surprised? So are a lot of other sound engineers when we tell them. That's why it's time we set the record straight.

Gotham Audio is the sole U.S. importer of Neumann — the world's finest microphones. And we're proud of it.

But Gotham Audio is also proud of the more than 300 other quality sound products and services we offer. Including tape recording equipment. Disk recording lathes. Disk cutting systems. Disk reproducing equipment. Studio and control room equipment. Test equipment. And other products.

These products are made by Europe's best-known companies. Beyer. EMT. Studer. And Neumann. Companies whose products have consistently made significant contributions to the technical excellence of American audio.

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Telephone (212) CO 5-4111
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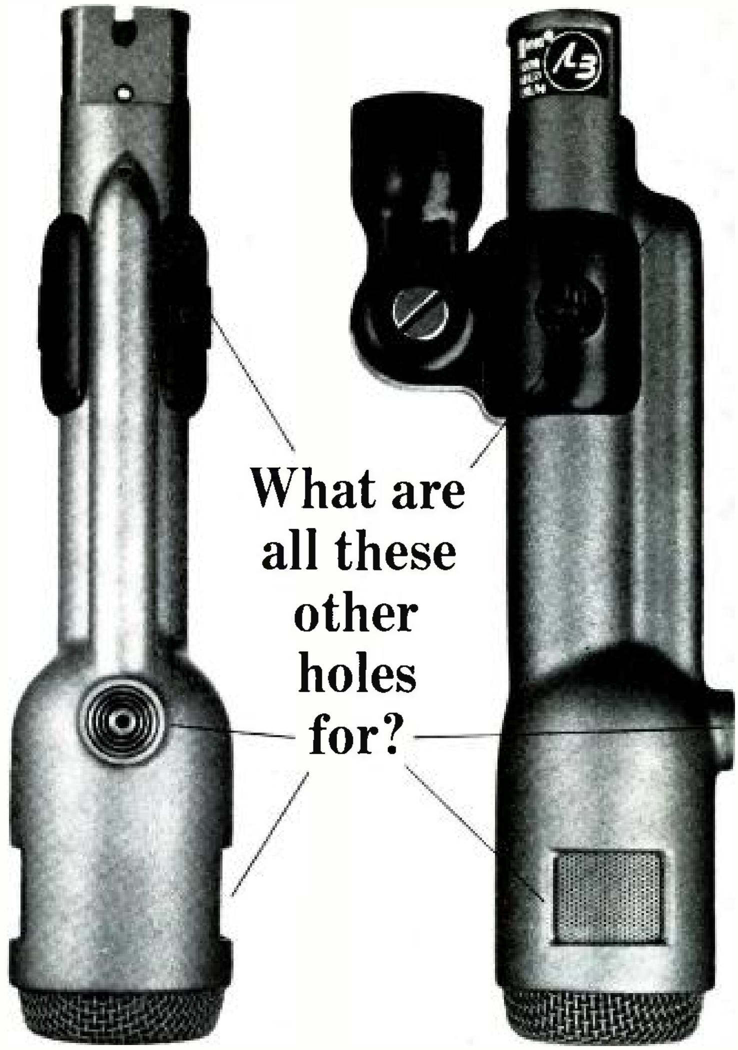
Circle 2 on Reader Service Card

If the Electro-Voice Model 666 picks up sound here...



EV. The holes in the top, sides and rear of the Electro-Voice Model 666 make it the finest dynamic cardioid microphone you can buy. These holes reduce sound pickup at the sides, and practically cancel sound arriving from the rear. Only an Electro-Voice Variable-D® microphone has them.

Behind the slots on each side is a tiny acoustic "window" that leads directly to the back of the 666 Acoustalloy® diaphragm. The route is short, small, and designed to let only highs get through. The path is so arranged that when highs from the back of the 666 arrive, they are cut in loudness by almost 20 db. Highs arriving from the front aren't affected. Why two "windows"? So that sound rejection is uniform and symmetrical regardless of microphone placement.

The hole on top is for the mid-range. It works the same, but with a longer path and added filters to affect only the mid-frequencies. And near the rear is another hole for the lows, with an even longer path and more filtering that delays only the bass sounds, again providing almost 20 db of cancellation of sounds arriving from the rear. This "three-way" system of ports insures that the cancellation of sound from the back is just as uniform as the pickup of sound from the front—without any loss of sensitivity. The result is uniform cardioid effectiveness at every frequency for outstanding noise and feedback control.

Most other cardioid-type microphones have a single cancellation port for all frequencies. At best, this is a compromise, and indeed, many of these "single-hole" cardioids are actually omnidirectional at one frequency or another!

In addition to high sensitivity to shock and wind noises, single-port cardioid microphones also suffer from proximity effect. As you get ultra-close, bass response rises. There's nothing you can do about this varying bass response — except use a Variable-D microphone with multi-port design* that eliminates this problem completely.

Circle 3 on Reader Service Card

Because it works better, the E-V 666 Dynamic Cardioid is one of the most popular directional microphones on the market. Internal taps offer 50, 150, or 250 ohm impedance output. Frequency range is peak-free from 30 to 16,000 Hz (cps). Output is—58db.

To learn more about Variable-D microphones, write for our free booklet, "The Directional Microphone Story." Then see and try the E-V 666 at your nearby Electro-Voice professional microphone headquarters. Just $255.00 in non-reflecting gray, complete with clamp-on stand mount. Or try the similar Model 665. Response from 50 to 14,000 Hz (cps), $150.00 (list prices less normal trade discounts).

*Pat. No. 3,115,207

ELECTRO-VOICE, INC., Dept. 1071 BD; 686 Cecil Street, Buchanan, Michigan 49107

Electro-Voice®
SETTING NEW STANDARDS IN SOUND