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Professional Group on Audio

IRE PROFESSIONAL GROUP ON AUDIO

The Professional Group on Audio is an organization, within the framework of the IRE, of members with principal professional interest in Audio Technology. All members of the IRE are eligible for membership in the Group and will receive all Group publications upon payment of an annual assessment of $2.00.

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PGA News

ELECTION RESULTS

The following election results are reported by B. B. Bauer:

For Chairman—Harry Olson.

For Vice-Chairman and Member of the Committee for the years 1957–1960—Frank Slaymaker.

For Administrative Committee 1957–1960—Walter T. Selsted, Philip Williams.

Holdover Members—Andrew B. Jacobsen, 3/31/58
Winston E. Kock, 3/31/58
Murlan S. Corrington, 3/31/58
Alexander Bereskin, 3/31/59
Semi J. Begun, 3/31/59.

FINANCIAL

IRE Professional Group on Audio

For the period from January 1 to December 31, 1956

Balance from January 1, 1956		$ 8,163.87
Receipts during period:		
IRE matched funds	$3,217.00	
Assessments	6,911.30	
Advertising	1,400.00	
Sale of publications	972.75	
Surplus from meetings	—	
Other sources	—	
Total receipts		12,501.05
Total balance and receipts		20,664.92
Expenses during period:		
Publications	$7,893.88	
Membership service charges	215.80	
Others	1,395.97	
Total expenses		9,505.65
Balance as of December 31, 1956		$11,159.27

MEMBERSHIP

Paid	3,259
Paid students	550
Unpaid	21
Total	3,830

IRE-PGA AWARDS

The IRE-PGA Awards Committee at their meeting in New York on December 26, 1956, selected the candidates for the IRE-PGA Awards as follows:

IRE-PGA Achievement Award

H. E. Roys—For outstanding contributions to Audio Technology published in IRE publications over a long period of years.

IRE-PGA Senior Award—Paper Award

J. Ross Macdonald—For his paper entitled, "A Multi-Loop, Self-Balancing Power Amplifier," IRE Transactions on Audio, Vol. AU-3, pp. 92–107; July–August, 1955.

IRE-PGA Award—Paper Award

H. H. Kajihara—For his paper entitled, "Miniaturized Audio Transformer Design for Transistor Applications," IRE Transactions on Audio, Vol. AU-4, pp. 10–18; January–February, 1956.

W. R. G. BAKER AWARD

The Board of Directors of the Institute of Radio Engineers has announced the establishment of a new IRE award, to be known as the W. R. G. Baker Award, to be given annually to the author of the best paper published in the IRE Transactions of the Professional Groups.

This will consist of a certificate together with a cash award comprising the income from a fund donated to the IRE by Dr. Baker, Vice-President of the General Electric Company and Chairman of the IRE Professional Groups Committee.

The 1957 W. R. G. Baker Award will go jointly to R. J. Kircher of the Hughes Aircraft Company, and R. L. Trent and D. R. Fewer of the Bell Telephone Laboratories for their series of three papers which follow:

R. J. Kircher, "Properties of Junction Transistors," IRE Transactions on Audio, Vol. AU-3, pp. 107–124; July–August, 1955.

R. L. Trent, "Design Principles of Junction Transistor Audio Amplifiers," IRE Transactions on Audio, Vol. AU-3, pp. 143–161; September–October, 1955.

D. R. Fewer, "Design Principles for Junction Transistor Audio Power Amplifiers," IRE Transactions on Audio, Vol. AU-3, pp. 183–201; November–December, 1955.

The Board of Directors also announced that D. A. Buck, instructor in the Electrical Engineering Department of M.I.T., would receive the 1957 Browder J. Thompson Memorial Prize for his paper, "The Cryotron—A Superconductive Computer Component," which appeared in the April, 1956, issue of the Proceedings of the IRE. This award is given annually to an author under thirty years of age for a paper recently published by the IRE which constitutes the best combination of technical contribution and presentation of the subject.

THE IRE "AFFILIATE" PLAN

On January 4, 1957, the IRE Board of Directors arrived at a decision which may in time prove to be one of the most far-reaching in its 45-year history. On that date, the Board adopted a plan which will enable non-IRE members whose main professional interests lie outside the sphere of IRE activities to become affiliated with certain of the IRE Professional Groups *without* first having to join the IRE itself.

This plan is aimed at those specialists in other fields of science and technology whose work touches upon our own electronics and communications field only in specialized areas. In effect, the IRE is extending the specialized services of its Professional Groups to every field of science and engineering.

An outstanding example of where these services are needed may be found in the case of the medical and biological sciences. At the present time, some 1400 IRE members enjoy the privileges of membership in the Professional Group on Medical Electronics. And yet there are hundreds, perhaps thousands, of medical doctors, biologists, and others to whom the activities of this Group would be of interest and value. Both they and the Group would benefit from their participation. To require these persons, who have no interest in radio engineering, to join the IRE in order to join the Group is unreasonable, and probably futile as well. In fact, it was largely to provide an answer to this problem that the "Affiliate" Plan was first conceived, although it pertains to other fields as well, such as Computers, etc.

The "Affiliate" Plan is admittedly an experiment. So far as is known, no other society has ever tried a similar scheme. The Board of Directors feels strongly that the benefits afforded by the plan justify the risk that some persons who should join the IRE will instead become Affiliates. To minimize this risk, the plan has been carefully worked out along the following lines:

1) Participation in the Plan is at the option of each Professional Group. It is not expected that all Groups will adopt it; only those which feel it serves a need in their particular field.

2) Each Group interested in initiating the "Affiliate" Plan must submit to the Chairman of the Professional Groups Committee a list of accredited organizations which has been selected and approved by its Administrative Committee, for official approval by the IRE Executive Committee.

3) To be an Affiliate of a Professional Group, a person must belong to an accredited organization approved by that Group and the IRE Executive Committee. Moreover, he shall not have been an IRE member during the five years prior to his application. He may affiliate with more than one Group, provided the accredited organization to which he belongs is recognized by the Groups concerned.

4) The fee for Affiliates shall be the assessment fee of the Group, plus $4.50. The latter covers IRE subsidies to the Group, Professional Group overhead expenses borne by IRE Headquarters, and 50 cents which is to be rebated to IRE Sections for mailing and meeting costs.

5) An Affiliate will be entitled to receive TRANSACTIONS of his Group and that part of the IRE NATIONAL CONVENTION RECORD pertaining to his Group. He will be eligible for a Group award, and may attend local or national meetings of the Group by payment of charges assessed Group members.

6) An Affiliate cannot serve in an elective office in the Group or Group Chapter, nor vote for candidates for these offices.

7) An Affiliate may hold an appointive office in the Group or Group Chapter.

8) An Affiliate may not receive any IRE benefits that are derived through IRE membership.

The "Affiliate" Plan is a bold and farsighted venture; one that recognizes and provides for the rapidly spreading influence of electronics in every walk of scientific and technological life, and one that enables the IRE to further its aims as a professional engineering society—the advancement of radio engineering and related fields of engineering and science.

W. R. G. BAKER
Chairman, Professional Groups Committee

WITH OTHER ACOUSTICAL AND AUDIO SOCIETIES

The November, 1956, issue of the *Journal of the Acoustical Society of America* contained numerous papers of interest to the audio technologist. Many of the papers contained in this issue are from the Second International Congress on Acoustics sponsored by the International Commission on Acoustics and the Acoustical Society of America, with cosponsorship of various organizations including the IRE Professional Group on Audio.

The issue contains 17 papers from the International Congress and 26 papers among the regular contributions.

Among the ICA papers, the following will be of interest to the members of the IRE Professional Group on Audio:

Cornelius M. Van der Burgt of Philips Research Laboratories, Eindhoven, Netherlands, wrote an article on "Performance of Ceramic Ferrite Resonators as Transducers and Filter Elements." In recent experiments, the composition and methods of preparation of various ferrites were varied with the aim of achieving optimum mechanical and piezomagnetic performance. Essential improvements were obtained by small cobalt substitutions in the chemical composition and by suitable modifications of the mechanical and thermal treatments.

Dieter Goetze of Raytheon Manufacturing Company deals with "Effect of Vibration Amplitude, Frequency, and Composition of the Abrasive Slurry on the Rate of Ultrasonic Machining in Ketos Tool Steel." Results of experiments conducted to determine the dependence of the ultrasonic machining rate V in Ketos tool steel on the peak-to-peak amplitude 2ξ on the frequency of vibration f on the abrasive particle diameter d and on the ratio of the mass of abrasive to the mass of water used in compounding the slurry are given. It is shown that within the range under investigation the quotient $V/2\xi fd$ is a constant for a critical ratio of the mass of abrasive to that of water. It is also shown that events occurring within the volume of a cube having sides equal to d are of fundamental importance in the ultrasonic machining process. A phenomenological equation is derived from which the most probable machining rate can be calculated for range under investigation.

Philip L. Alger and Edward Erdelyi of General Electric Company have written an article entitled, "Calculation of the Magnetic Noise of Polyphase Induction Motors." The origin of the magnetic driving forces which are responsible for magnetic noise and a method of calculating the mechanical response of the motor structure are reviewed in this paper. A general method is developed to calculate the sound pressure levels of magnetic noise at a specified point near the motor. The magnetic noise for a particular experimental 4-pole, 30-hp, polyphase induction motor under five different operating conditions; calculations agree with measurements within 3 db.

Harry F. Olson and Herbert Belar, RCA Laboratories, have contributed an interesting article entitled, "Phonetic Typewriter." The important factors involved in the development of a phonetic typewriter are as follows: the particular form in which the words are typed, the means for analyzing the sounds of speech, the identification of the analyzed sounds, the encoding, coding, and decoding of the sounds for the operation of the actuating mechanism, and the design of the mechanism for actuating the typewriter. A study has been made of these problems. As a result of this study a simplified model of a phonetic typewriter has been developed incorporating all of these aspects. This model serves to illustrate the principles involved and provides means for further study towards a complete system.

Aram Glorig, Robert Quiggle, D. E. Wheeler, and William Grings of Research Center, Subcommittee on Noise in Industry have contributed an interesting paper on "Determination of the Normal Hearing Reference Zero." The present American normal hearing reference zero is considered by many investigators to be too high. The data from two recent studies on normal hearing provide a possible explanation. One of these studies, which was a survey type study, confirmed the present reference zero. The second study, a laboratory type, confirmed the results of other laboratory studies that had led to the opinion that the present zero is too high.

In our opinion, the difference between the results of these two studies and various other studies of the auditory threshold are due to the inherent difference in experimental technique in laboratory and survey type studies. In establishing a normal hearing reference zero, it is no longer a question of resolving differences in various studies but a question of whether survey or laboratory data will be used.

Roger E. Kirk of Ohio State University has an article on "Learning, a Major Factor Influencing Preferences for High-Fidelity Reproducing Systems." Frequency range preferences of 210 college students for reproduced music and speech were determined by an A-B-A preference test. Two groups of subjects then listened to music reproduced over a restricted frequency range and a relatively unrestricted frequency range, respectively, for six and one-half weeks. The results of a postfrequency range preference test indicate that: 1) Learning plays an important role in determining preferences for sound reproducing systems; 2) continued contact with a particular system produces shifts in preference for this system; 3) the average college student prefers music and speech reproduced over a restricted frequency range rather than an unrestricted frequency range; and 4) the frequency range preferences of college students are in part a function of the type of music to which they are listening.

From among the regular contributions, the following should not be missed:

"Twixt Earth and Sky with Rod and Tube; the Mobility and Classical Impedance Analogies," is a monumental article by the Editor of *JASA*, Floyd A. Firestone, giving a basis for analogies for mechanical, acoustical, and electrical networks by the classical (impedance) and the mobility (admittance) methods.

Warren P. Mason of Bell Telephone Laboratories deals with "Physical Acoustics and the Properties of Solids." The techniques of physical acoustics have been applied in determining the elastic properties and internal friction of polycrystal and single crystal metals, glasses, nonmetallic crystals, high polymer materials, and ceramics. They have been used as tools in investigating such solid-state phenomena as grain and domain boundary effects in metals and ferromagnetic and ferroelectric materials, in the diffusion of atoms, molecules, and vacancies through a solid, in the motion of imperfections such as dislocations; they have even detected an interaction between the lattice sound vibrations and free electrons in metals at low temperatures.

"Speech Communications in Noise: Some Equipment Problems," is a very interesting article by Mones E. Hawley of Radio Corporation of America. The design of a speech communication system begins with an operation analysis of the communication problem. When speech has been chosen as the means and when the needed linkages have been determined, the designer chooses the best compromises among the frequently conflicting factors of intelligibility, safety, comfort, quality, reliability, and economy. It is particularly important to provide good quality as well as adequate intelligibility. The latter may be predicted with reasonable accuracy if the noise and signal levels and the transfer characteristics throughout the system are known. Pressure gradient microphones, especially with noise shields, noise attenuating earcaps, and earplugs, are the primary acoustical devices that can be used to obtain high intelligibility through improvement of signal-to-noise ratios. If the listeners are in intense noise, headsets presently pose the major systems limitation. Automatic volume control and peak clipping are the audio techniques most frequently used to complement the transducers. Typical values and limitations for these kinds of processing are given. Optimum results can be obtained only if the whole system is designed together.

Keith K. Neely of the Sonics Group, Defence Research Medical Laboratories, has written an article entitled, "Effect of Visual Factors on the Intelligibility of Speech." Research has indicated the important role that vision plays in face-to-face voice communication. This study was an attempt to quantify further the visual contribution to speech intelligibility in a high intensity noise environment in terms of the angle and distance from the listener to the speaker.

The issue contains its usual excellent "References to Contemporary Papers on Acoustics," by Robert N. Thurston and "Review of Acoustical Patents," by Robert W. Young. It also contains the complete index to Volume 28 of the *Journal*.

Information about the Acoustical Society may be obtained from the American Institute of Physics, 57 East 55th Street, New York 22, N. Y.

BENJAMIN B. BAUER

Audio Applications in the Home*

RONALD D. STEWART†

> The first three papers in this issue are reprinted from the May, 1956, issue of the *Proceedings of the IRE, Australia.* The papers by Ronald D. Stewart and Arthur McLean were first presented at a general meeting of the Sydney Division of the Institution of Radio Engineers Australia on April 29, 1954. The paper by Fritz Langford-Smith was delivered on September 29, 1954. These papers are reprinted here as a general service for the purpose of acquainting the readers of the IRE TRANSACTIONS ON AUDIO with thoughts on audio subjects prevalent in other areas of the world.
>
> —*The Editor*

Summary—**The quality of home reproduction of sound is strongly dependent upon psychological considerations and the characteristics of the room. To evaluate improved methods of home reproduction it is necessary to check system performance by carefully controlled listener tests. The results obtained from such tests indicate that wide-range reproduction is only acceptable when distortion is kept to an extremely low level.**

INTRODUCTION

IN SEEKING to improve the reproduction of sound in the home, it would appear desirable to establish an ultimate goal toward which development might be directed. As with most matters associated with audio work, however, the problem is not at all clear cut and there is as wide a divergence of opinion as to what are the aims of sound reproduction, as there is to the means of attaining them. This paper attempts to analyze and discuss some of the problems involved.

Quality of Reproduction

What is "perfect reproduction?" The answer to this might be that the listener should hear exactly what he would have heard, had he been present at the point at which the sound originated. In trying to reduce this to practicable means, difficulties are encountered immediately. It would be necessary to set up pressure-type microphones at the "ears" of a dummy head. Because sound incident on the ears is modified by the shape of the head, the dummy head should be an exact replica of the listener's, any other shape of head or ears would modify the sound in a different manner and therefore depart from perfection. The signals picked up in those microphones must be kept separate through all stages of amplification, recording and playback, to be reproduced by headphones directly into the ears of the listener.

Even this perfect system can be subject to strange effects caused by the movements, involuntary or otherwise, of the listener's head. This causes the sensation of the location of the sound source wheeling round with the listener's movements. The ear's sense of direction is relatively poor and it is only by movement of the head, more or less automatically, that the directions of many sounds are established. It follows therefore that for perfect reproduction the dummy head on which the microphones are mounted would have to move exactly in accordance with the listener's movements and if the program was recorded, at every playback the listener's head would have to follow again exactly the same series of movements.

Having followed the matter this far it seems to be rather pointless to proceed further. It is not possible to imprison the listening public in headphones. Similarly it is not possible to gang all their heads together like a group of dummy clowns at a carnival side show. It is obvious that consideration of this form of "perfect reproduction" is misleading and gives no guide to improvement in existing systems.

Why should it be necessary to establish the location from which the sound has come? It is true that the dramatic value of stereophonic sound in music and voice has been demonstrated clearly in the field of cinema entertainment but in the home, conditions are very different. As an example, the reproduction of a symphony orchestra might be considered. When listening to an orchestra in a hall with an appreciable reverberation time, as is usually the case, probably about 80 per cent of the sound energy incident upon the ears has been subject to at least one reflection. In other words, the sound is not coming from the individual instruments but from the walls, the ceiling, the back of the hall, etc. In fact, everything in the hall contributes its share of reflection.

It might be suggested that at symphony concerts the origin of the sound is identified with the eyes just as when watching a motion picture show and that the enjoyment of the music has little to do with "audio perspective." It seems likely that the spatial distribution of an orchestra has only a very small bearing on the musical content of the program and that, if the sound could be all produced from the one spot in the hall simultaneously, it probably would not sound materially different from music coming from the spread out orchestra.

* Reprinted from the May, 1956, issue of the *Proc. IRE, Australia.*
† Amalgamated Wireless (A/sia) Ltd., Sydney, N.S.W., Australia.

It cannot be disputed that the sound would be different, but much greater differences occur already between positions in the hall. It is quite possible that, if the orchestra in a hall could be replaced by a single concentrated source producing the appropriate sounds, appreciation of the music would not be impaired to any noticeable extent and could conceivably be an improvement over the sound from an orchestra occupying a great deal of space. It must be admitted that seeing the orchestra has a tremendous psychological effect and even the atmosphere and attitude of the audience has considerable bearing on the feelings and reactions of the listener, but the opinions expressed are based purely on the musical content of the sound.

In bringing this music into the home similar arguments apply. In any event, the sound is reflected all round the room; therefore, why should it sound musically better to be bounced round from two or three speakers in a so-called stereophonic system than it does from one speaker? Furthermore, how does one simulate the presence of a seventy-five piece orchestra in a lounge room 18 feet by 12 feet?

In any system involving the use of loudspeakers in the home, the best that can be done is to try to create an illusion of reality under circumstances entirely different from the original. The extent to which success is achieved cannot be measured satisfactorily. It is purely a psychological reaction on the part of the listener and might depend on such factors as his physical comfort, his visual surroundings and company, as well as the quality of the music and interest in the program material.

Quality of Sound in a Small Room

Any music reproduced by a loudspeaker must be colored by the characteristics of the room. A reverberation time of about one second used to be considered quite good for music, but lately, many recordings have been made in studios which are quite "live" (artificially or otherwise). These may sound better if reproduced in fairly "dead" rooms. For good results it is necessary also as far as possible to avoid standing waves and to diffuse the sound by avoiding or breaking up large parallel wall areas. This can be done by introducing irregularly disposed areas of furniture and other objects to scatter the sound throughout the room. For these reasons and also because of speaker directional and loading effects, it is desirable to mount the speaker either in the corner of a room or, if this is not possible, on one of the short walls in a rectangular room.

Listener Preference Tests

Test Conditions

The ear has a very short memory and only direct comparison tests have any worthwhile value. Even with tests of this nature, the directional effects of cone radiator type speakers have much more influence than is generally realized and in order to get a true comparison between two speakers the listener must place the two units quite symmetrically with respect to his own position and the shape of the room. The positions of the speakers during the test should be reversed from time to time to ensure that difference in hearing between one ear and another is not a controlling factor in the test. The listener should also vary his distance from the speakers so that comparison can be obtained with varying room effects.

In listening tests there is a great tendency for the listener to come to false conclusions, quite unconsciously, by hearing what he wants to hear. In order to guard against this, it is good practice to have the speakers which are to be tested, set up by an assistant, so that they are judged by the sound and not by a prior inspection of response curves. Tests under these conditions are always carried out more carefully and unsuspected effects are sometimes noticed which might have been overlooked in a more perfunctory test. It is considered that measurements should be designed to analyze what the ear has heard, rather than to conduct a listening test to confirm some preconceived notion. In this type of work there are many factors which cannot be measured satisfactorily, but which may affect the test results. There is a very strong tendency indeed, in listening tests, to hear that which it is thought should be heard.

In carrying out comparative listening tests, it is necessary that the output from both speakers be adjusted as closely as possible to the same level. If this is not done, the louder speaker will have its bass response apparently enhanced and the reproduction made clearer and cleaner by comparison with the other one.

Having set up the listening test the sound which is heard is then judged, but what is the question that is asked? Is it "Which of these two speakers is preferred?" or "Which of these two speakers gives the better simulation of the original?" Only in very rare cases would the listener have heard the original so perhaps the second question should be "Which of these two speakers gives the better simulation of what the original is imagined to be?" Usually the listener has only a very vague idea as to how the original might have sounded and so in most cases he will choose the one he prefers.

Broadly speaking, listeners can be divided into two groups: the nontechnical listener who listens to the program and the "technical" listener who listens to the reproduction. Comparatively the first type is quite intolerant of extraneous noises such as needle scratch and rumble, and prefers restricting the frequency range if this is necessary to obtain more pleasant reproduction. The second type prefers listening to odd noises such as steamer whistles that can be felt rather than heard, and the chirps and clicks of insects that are practically supersonic. Usually such a listener is most distressed by turntable rumble but will put up with quite a lot of scratch noise in order to extend the high frequency response. Furthermore, he is usually not so critical of distortion or peaks in the high frequency region.

Discussion of Test Results

In reviewing listener preference tests it may be of interest at this stage to examine some tests carried out overseas and the conclusions drawn from them. The most controversial point is "Are the high frequencies required or not?" Working for CBS in the United States, Chinn and Eisenberg[1] attempted an analysis that was most commendable. A large number of listeners, selected at random, were given three frequency ranges from which to choose. The responses chosen by the majority favored a narrow or medium band with a response 70–7000 cps. It is obvious that no expense was spared in the conduct of these tests so far as the quality of the equipment was concerned. The recordings played were unused originals and a direct line was used to bring in a live artist program originating in a nearby studio, but the "highs" were still disliked. Similar tests were carried out in England with very similar results; wide range was rejected and the majority vote split between the narrow and medium responses.

There are two points of view in examining these results. The first is that traces of distortion, no matter how small, are heard to such a degree in extended range reproduction that the resultant sound is inferior to that involving a more restricted range. If this is the case, the line of progress is quite clear; that is, gradual reduction of distortion from one cause or another will permit a gradual extension of the range.

The second viewpoint is that most people just do not like high frequencies anyway. This idea is supported by some work done in England by Moir[2] who examined the frequency responses of Stradivarius and other accepted "master" violins and found that the richness of tone as compared with other less famous instruments was due to the reduction or absence of high order harmonics. This idea seems to be supported by players of the Spanish or the steel guitar who, quite apart from considerations of power output obtainable, prefer the mellowing effect their instruments assume when played through amplifiers and speakers of limited range. It is also known that players of the clarinet often prefer to play through limited range sound reinforcement systems because a subtone can be used to give a more pleasing tone with reduced even harmonic content while in the upper register the note sounds clearer due to fewer high-order harmonics, absence of wind whistles, and reduction of key clicks.

Following this line of thought to a logical conclusion, if a radio manufacturer can produce a cheaper set which, by restricting the range, can make a relatively cheap violin sound like a Stradivarius, he would be foolish to do otherwise. How can the listener be expected to pay more for the privilege of hearing wind whistles and key clicks? The answer to this of course is that frequency range is not the only consideration and that the cheap violin still sounds like a cheap violin but more muddy and less squeaky than the original. The sibilants have quite important parts to play in speech articulation but they are wind whistles and to cut them out with the other wind whistles is to reduce the effectiveness of the reproduction. We cannot expect sound equipment to be aesthetically discriminatory but often wrong conclusions are arrived at in listening tests because insufficient variety of program material is used and insufficient time spent on the test.

C. J. LeBel[3] has suggested that quick listening tests can be misleading and that poor reproduction is sometimes shown by the increased fatigue of the listener. Fatigue may be an indication of faults in the reproduction difficult to measure with existing techniques, but it cannot be assumed that it would not be tiring listening to the original. It might be less tiring listening to restricted range reproduction, than listening to the original.

It is believed that the only person who has investigated this question of frequency range with a purely acoustical series of tests is Olson[4] of RCA. In this test an orchestra was set up in the corner of a room and a framework was arranged between the orchestra and the audience whereby a 5000 cps low pass acoustical filter could be interposed and removed by operating a lever. In this test 70 per cent preferred the full range sound. The sound through the filter was described as muffled, muddy, and so on.

The results of such a test appear to be clear-cut and conclusive, most listeners have experienced the effect of a curtain dropped in front of the orchestra in a theatre and, while on occasions this may have given genuine relief to true lovers of good music, it must be agreed generally that under these conditions, much of the live tone color of the orchestra is lost and the resultant sound is lacking in character. The idea that this type of sound should be preferred does not appear acceptable.

Permissible Distortion

The general preference for wide range in live listening and restricted range in reproduced listening appears to be explainable almost entirely by the presence of distor-

[1] H. A. Chinn, and P. Eisenberg, "Influence of reproducing system on tonal range preferences," Proc. IRE, vol. 36, pp. 572–580; May, 1948.

[2] J. Moir, "Perfect versus pleasing reproduction," *Electronic Eng.*, vol. 19, pp. 23–27; January, 1947.

[3] C. J. LeBel, "Psycho-acoustical aspects of listener preference tests," *Audio Eng.*, vol. 31, pp. 9–12; August, 1947.

[4] H. F. Olson, "Frequency range preference for speech and music," *Electronics*, vol. 20, pp. 80–81; August, 1947.

tion and other aberrations. In many listening tests it is claimed that "distortion was low." How low? It would be a very good system indeed, particularly if disc records were involved, that could claim a sum total of distortion and noise products, taken over-all from sound input to sound output, better than 50 db below program level.

Consider then a symphony orchestra being reproduced at an instantaneous sound pressure level of 90 db above threshold. This is quite loud but not unduly so, and under this condition the distortion and noise products are reproduced at a level of 40 db. Is this negligible? It is a hundred times greater sound pressure level than the threshold of hearing. This appears to the crux of the whole problem and the frequency range can be increased only when distortions of all kinds over the entire system are reduced. Since the speaker is the very last instrument in the chain, its function is more or less of a selective gate allowing the more acceptable parts of the electrical signal through and the gate may be opened wider only if warranted by the quality of the signal.

Much has been written on the subject of distortion but knowledge so far is very limited. Even in considerations of harmonic distortion, recently published articles[5] suggest figures approaching 12 per cent harmonic distortion on the inner grooves of disc records while typical performance of output tubes gives a power rating corresponding to 5 or 10 per cent distortion. These values are many times greater than the 50 db down mentioned previously. Dr. Olson[6] says "tests show that the tolerable nonlinear distortion in a sound reproduction system with a 5000 cps high-frequency cutoff is of the order of 6 to 10 per cent while the tolerable nonlinear distortion in a sound reproducing system with a 15,000 cps high frequency cut-off is of the order of 1–2 per cent."

[5] H. E. Roys, "Analysis by the two frequency intermodulation method of tracing distortion encountered in phonograph reproduction," *RCA Rev.*, vol. 10, pp. 254–269; June, 1949.

[6] H. F. Olson, "Musical Engineering," McGraw-Hill Book Co., Inc., New York, N. Y.; 1952.

Such a generalization is far too loose to be accepted without question since the degree of distortion tolerable to the ear depends as much on the nature of the distortion as on the quantity. A simple experiment carried out some time ago by the author and his colleagues provided a good illustration of this statement. A 12-inch speaker with a top roll-off at about 7000 cps was driven with an input of 3 watts at 200 cps. When analyzed, the sound picked up by a velocity microphone showed 3.1 per cent distortion. A piece of metal was moved in from the front of the speaker until it just touched the vibrating cone. The buzz, similar to that which is well known as "speaker rattle," was quite audible, but the measured distortion had increased from 3.1 per cent to only 3.5 per cent with small detectable components up to the 20th harmonic.

Peak-clipping in an amplifier is similarly distressing although the measured harmonic distortion may be quite low. Weighted distortion factors in which the percentage of each harmonic is muliplied by its order number or even by the square of the order number have been suggested but up to date it appears that no attempt has been made to relate a weighted distortion factor precisely to listening tests.

Conclusion

In case some of the remarks made in this paper appear altogether unhopeful and defeatist, it should be reiterated that, at the moment, there appears no means of affecting suddenly a vast and wonderful improvement in domestic audio reproduction. However, it should be the aim of everyone working in this field to improve as far as possible, his own particular section, whether it be the gramophone pickup, amplifier, or loudspeaker assembly. Furthermore, it seems quite clear that properly conducted listening tests are the only means of finally judging quality and that measurements that are to be of any value must relate to factors significant to the ear.

Audio Amplifiers*

FRITZ LANGFORD-SMITH†

Summary—**In this paper the design and testing of audio amplifiers is discussed. Attention is also given to the question of comparing the performance of different amplifiers and to the lines along which future investigation should proceed.**

Introduction

IT HAS BEEN assumed frequently that amplifier design is now quite conventional and that the real problems in audio frequency reproduction are associated with loudspeakers and acoustics. However, recent investigation into amplifier design has revealed a series of problems with no direct indication as to how their solution might be obtained. This paper describes a number of these problems and suggests an approach to the solution of some of them.

This paper is limited to a discussion of "main" amplifiers as distinct from preamplifiers since, in general, the most serious design problems in a complete audio frequency reproducing system occur in the main amplifier. Preamplifiers may assume many different forms according to the requirements of each particular application but, by good design, the nonlinear distortion in the preamplifier can be, and should be, less than that of the main amplifier.

It is generally agreed that a good "main" ampiifier should possess the following characteristics:

1) Low nonlinear distortion and hence low harmonic distortion and low intermodulation distortion. Anything in the nature of a sharp kink in the linearity characteristic (input voltage vs output voltage) has a particularly distressing effect on the listener—far in excess of that indicated by the usual measurements of percentage distortion—and is therefore to be avoided where good fidelity is desired.
2) Substantially uniform frequency response over the whole audio range.
3) Sufficient maximum power output to handle peak power requirements under anticipated operating conditions over the whole audio range without noticeable distortion.
4) A good "overload" characteristic—that is, the distortion at outputs above the rated maximum power output should not increase at an excessive rate.
5) Sensitivity sufficient to provide the rated maximum power output with an input of not more than 1 or 2 volts.
6) Ability to reproduce any likely forms of transients without serious change in waveform.
7) The amplifier should not add to the information with which it is presented, for example, by overshoot, damped oscillations, etc.
8) Low output resistance—not greater than 20 per cent of the nominal load impedance.
9) Very low hum level.
10) Very low noise level. In most cases the noise contributed by the main amplifier is negligibly small compared with that from the preamplifier.

Amplifier Design Considerations

Output Valves

Effects of Loudspeaker Load: All amplifiers of the types being considered are intended to drive a loudspeaker. A very few are designed for operating one and only one loudspeaker, but the vast majority are intended for use with any one of a wide range of loudspeakers. A loudspeaker presents to the amplifier an impedance which may vary normally by a ratio of 10 to 1 while one world-famous high-fidelity loudspeaker varies by a ratio of 20 to 1. In addition, the loudspeaker impedance is reactive over most of the frequency range, presenting an elliptical loadline to the output tubes.

The effects of these variations, both in impedance and phase angle, are minimized by the use of push-pull Class A triodes, but are serious with pentodes and beam power tetrodes. The effects of a 10 to 1 variation in load impedance for purely resistive loads are shown in Figs. 1 to 3 inclusive. Fig. 1 shows the composite characteristics for a typical pair of push-pull Class A triodes. Fig. 2 shows the same characteristics for a pair of beam power amplifiers. A generally similar effect also occurs with pentodes. It can be seen that the triode characteristics are much more linear than those of the beam power tube. The linearity of the latter's characteristics may be improved if the screens are connected to taps on the plate winding of the output transformer. This arrangement is known as the "ultralinear" amplifier and typical characteristics for a pair of pentodes are shown in Fig. 3.

Pentodes vs Triodes: Pentodes have the following advantages over triodes:

1) They are more sensitive, requiring less input voltage for the same power output.
2) They have greater plate circuit efficiency, even when the screen power input is included.
3) They have good "cushioning" effect when approaching the overload point, and the rise in distortion is gradual.

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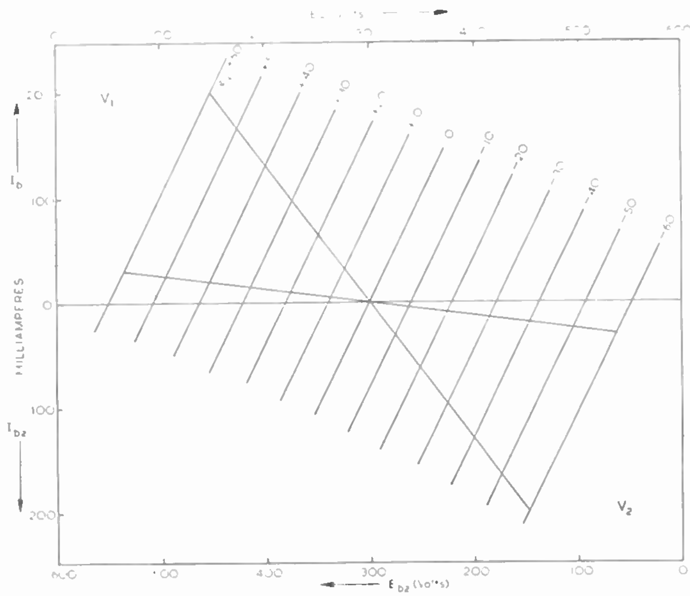

Fig. 1—Composite characteristic curves for two triodes connected as a Class A push-pull amplifier. Load lines for resistance loads of 3000 ohms plate to plate and 30,000 ohms plate to plate are shown.

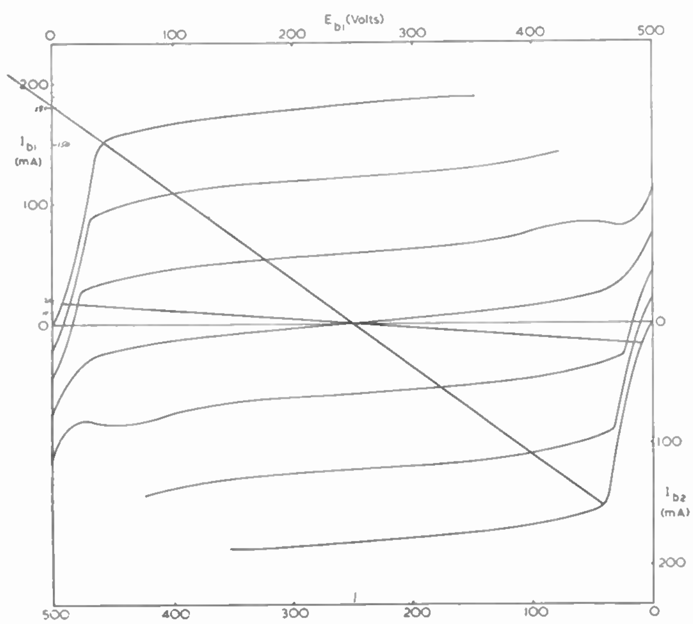

Fig. 2—Composite characteristic curves for two beam tetrodes connected as a Class A push-pull amplifier. Load lines for resistance loads of 5500 ohms plate to plate and 55,000 ohms plate to plate are shown.

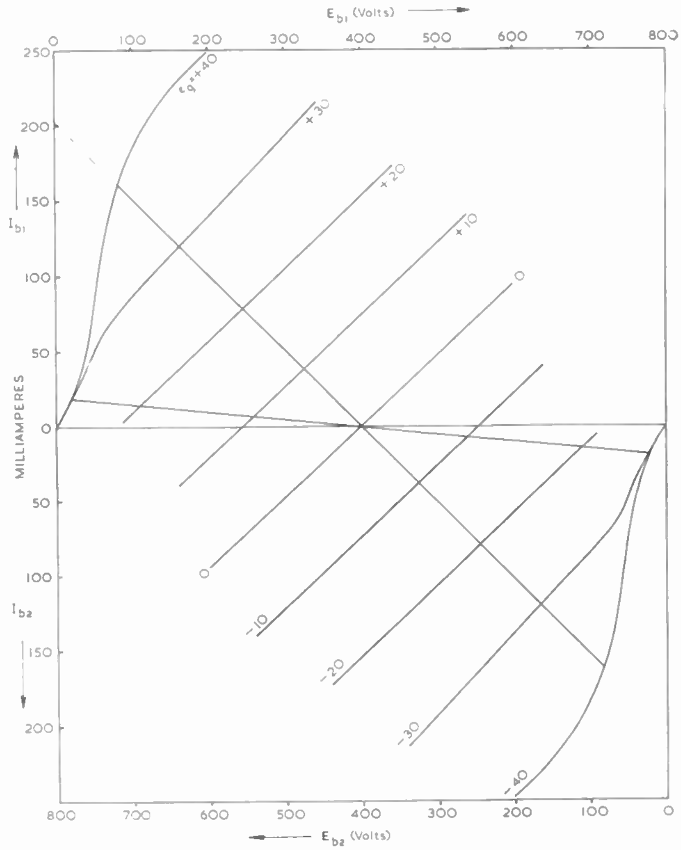

Fig. 3—Composite characteristic curves for two pentodes operated as an "Ultralinear" amplifier. Load lines for resistance loads of 8000 ohms plate to plate and 80,000 ohms plate to plate are shown.

As a result pentodes are very widely used in public address systems and medium fidelity amplifiers.

In high-fidelity amplifiers pentodes have the following disadvantages compared with triodes:

1) They are very sensitive to loudspeaker impedance variations; above and below 400 cps, power output must be reduced to limit distortion to the same as that at 400 cps. Consequently a much larger moninal power output is required with pentodes for comparable performance on a loudspeaker load.

2) In amplifiers of the 15-watt class, it is possible to use pentodes or beam power tetrodes which give the required power output and operate with equal plate and screen voltages. For higher power outputs the usual arrangement is to use a screen voltage lower than the plate voltage, and the regulation of the screen circuit then becomes a problem. Various ways of obtaining good regulation of the screen circuit have been developed but the best methods add appreciably to the total cost.

3) With pentodes the linearity characteristic always shows a gradual curvature over a considerable portion of its length and the slope of the characteristic at maximum rated power output is appreciably less than that at low levels. Consequently the reduction in distortion due to negative feedback at maximum output is considerably less than that at low levels. A greater degree of feedback is therefore required than would be anticipated from the simple theory of feedback. This effect does not occur to any appreciable extent with triodes such as in the Williamson amplifier, which has a substantially straight linearity characteristic even without feedback.

With triodes it is possible to drive quite a distance into the grid current region, with almost perfect linearity so far as the plate characteristics are concerned, provided a sufficiently low impedance driver stage is used. As a result the overloading characteristics are very good. One way of achieving the low impedance driver is by using a push-pull cathode follower although this method is of very little value with pentodes.

It is the author's opinion that, other factors being equal, push-pull triodes are superior to pentodes or beam power tetrodes for the reasons stated. However this optimum performance is obtained at a cost—at least the additional cost of a larger power supply.

Much ingenuity has been shown in the design of amplifiers using pentodes or beam power tetrodes in an attempt to produce results approaching those obtained with triodes. In this field there is strong competition between pentodes operating as pentodes and those operating as so-called "ultralinear" amplifiers. At this stage it is difficult to quote a quantitative comparison, but the "ultralinear" circuit is being adopted widely in both England and the USA for high-fidelity amplifiers. It has the advantage of extreme flexibility of performance merely by moving the tapping point on the output transformer.

Matched Tubes: Certain amplifiers give their specified distortion values only if matched tubes are used but this condition is highly undesirable. The Williamson amplifier requires tubes matched within 5 ma plate current, but the specified distortion of 0.1 per cent is obtained only when the two plate currents are closely matched by using the potentiometer provided. Unfortunately, the tubes tend to drift apart, resulting in increased distortion, and it is unusual for such an amplifier to be checked periodically.

The Leak amplifier Model TL/12 does not require matched tubes nor does it rely on balancing controls, and yet its distortion on a resistive load is less than 0.05 per cent.

Certain other makes of amplifiers also claim to operate within specified distortion limits without requiring matched tubes, but in practice all of them are found to have considerably higher distortion. It seems that the two main features providing for satisfactory operation with unmatched tubes are the design of the output transformer and the use of separate cathode resistors, each separately by-passed.

Feedback

Point from Which the Feedback is Taken: Feedback may be taken from either the primary, the secondary, or a tertiary winding of the output transformer. The primary winding is the simplest and safest to use, since it does not introduce serious problems with instability. Its defects are that it does not reduce distortion caused by the transformer core, and that it results in a high hum level, hence requiring more elaborate filtering. The high hum level is caused largely by feeding back to some earlier stage the full hum voltage on the plates of the output tubes.

The best method for the reduction of distortion is to take the feedback from the secondary winding, but it introduces serious problems in stability usually accompanied by high peaks at low and high frequencies, unless only a small degree of feedback is used. These peaks are due to the large phase shifts which occur in output transformers at very low and very high frequencies, resulting in positive feedback at those frequencies.

The use of a tertiary winding on the output transformer is only slightly inferior to using the secondary of the transformer but it has the advantage that a greater margin of stability is obtained. Its great disadvantage is that a special transformer is required.

Multiple Loops: Most present-day designs have only one main feedback loop—usually from the output to the cathode of an earlier stage. Actually there is a small but useful, subsidiary loop caused by the unby-passed cathode resistor, which not only adds feedback but also helps to improve the stability. It seems that modern design will tend towards the use of multiple loops. One good arrangement comprises two partially-overlapping loops with a third subsidiary loop. This gives the required reduction in distortion with improved stability. The Leak TL/12 amplifier has three loops, both first and second stages having local loops which are enclosed within the main loop over the complete amplifier.

Combined Positive and Negative Feedback: In the usual form, positive feedback is used to increase the gain of the first stage, and over-all negative feedback is used to reduce the distortion. This method seems to have quite a useful future, not only for economical amplifiers, but also in the fairly good fidelity class. It is possible to control the attenuation of the stage employing positive feedback at very low and very high frequencies to improve the stability margin of the whole amplifier.

Transients

In musical reproduction the amplifier is forced to handle a succession of transients of many waveforms, which should be reproduced by the amplifier without distortion of form. In order to achieve this objective the following characteristics are necessary: a rapid rise, small overshoot, very small phase shift, and a slow rate of fall of the "flat top."

The most common method used in the laboratory for determining response to transients is the square wave. All amplifiers give better square-wave response from an electronically regulated plate supply, since an ordinary filter condenser is not capable of maintaining full voltage over the half cycle, especially at low frequencies.

An amplifier required to give good square wave response from 50 to 15,000 cps must possess substantially uniform gain, without peaks, over a very wide frequency range. The precise bandwidth required for a specified performance cannot be stated on present knowledge but all known amplifiers in this class appear to have a substantially uniform response from below 5 cps to about

200 kc. The testing of amplifiers with square wave input voltage is described in the next section.

A sawtooth waveform is approximately equivalent to a square wave of half the frequency, and therefore provides a much more severe test. The Williamson amplifier distorts a 50 cps sawtooth wave, although the reproduction is quite good when the time constant of the intertube coupling is increased four times. Good reproduction of some particular waveforms (*e.g.*, square wave or sawtooth) implies very low phase angle shift, since, for example, the phase angle of the twentieth harmonic is 20 times that of the fundamental. For example, one degree phase shift of the fundamental is equivalent to twenty degrees phase shift of the twentieth harmonic, and the twentieth harmonic of such waveform is quite appreciable.

Stability Margin and Peaks in Gain[1]

It is desirable to have a large stability margin under any possible condition of operation with any value of resistive, inductive, or capacitive source or load impedance—including open and short-circuited conditions. Stability margin may be defined as the increase in feedback, expressed in decibels, which may be applied before sustained oscillations are set up immediately following a large transient input signal. Most feedback amplifiers have their lowest stability margin when their normal load resistance is shunted by a particular value of capacitance and this condition is closely linked with the height of peaks in gain.

The peaks in gain due to feedback which occur at very low and very high frequencies should not, in the author's opinion, rise appreciably above the gain level at 1000 cps. However, some designers (including Williamson) allow one of the peaks to rise up to 6 db above the 1000 cps level on a resistive load, which means that, on a loudspeaker load, one or both peaks may rise considerably higher. When the high-frequency peak is somewhat above the frequency at which the particular loudspeaker changes over from inductive to capacitive impedance, it is likely that the high-frequency peak will be accentuated. Considerable differences are to be expected in the height of the high-frequency peak with various loudspeakers, and it seems preferable for amplifier design purposes to use a normal resistive load shunted by capacitance, the value of the capacitance being determined by trial of successive increments (of the order of ± 20 per cent) to give the maximum peak height. In this way the worst possible condition may be obtained. The effect of any appreciable high-frequency peak, even $+2$ db, on a square wave input is to produce overshoot, followed by damped oscillation commonly known as "ringing" at the frequency of the peak.

In many cases the low-frequency peak is higher than the high-frequency peak, and its effects on performance are more serious. The presence of a low-frequency peak with a level of about $+3$ db or more is usually associated with a fluctuation in level when a high-frequency tone is instantaneously decreased from maximum to say half output voltage. The simplest means of testing for this condition appears to be the application of a 10,000 cps tone switched at 50 cps from maximum to half voltage.

Loudspeaker Damping

A low amplifier output resistance assists considerably in increasing loudspeaker damping in the vicinity of the bass resonance frequency but has no beneficial effect at much higher frequencies. It is possible, by the use of combined positive and negative feedback, to reduce the amplifier output resistance to zero, but this provides very little improvement in the damping compared with a value of the order of 10 per cent of the load impedance. With loudspeakers having high efficiency and high flux density it is possible to reach or even exceed critical damping. With a low efficiency and low flux density loudspeaker there is no hope of reaching critical damping, although even a partial degree of damping is beneficial.

Distortion

It may be asked why it is desirable to have an amplifier with harmonic distortion of the order of 0.1 per cent and whether it is possible to detect aurally such a low level of distortion. Tests by Dr. Olson[2] have shown that approximately 1 per cent total harmonic distortion is the lowest amount perceptible, under the conditions of his tests. These tests were conducted on single triodes and single pentodes and, although no details are given, it is obvious that the test amplifiers did not include any sharp kinks in the linearity characteristics (which only occur in an amplifier when a tube is completely cutoff during part of the cycle), and that there was no incipient instability. Consequently this figure of about 1 per cent is not of general application, and some amplifiers with less than 1 per cent total harmonic distortion are likely to have perceptible distortion.

The total distortion heard by the listener is the sum of the distortion levels in the source, amplifier, and loudspeaker. Therefore any reduction of the distortion in the amplifier is generally desirable and beneficial, although a very small reduction would probably not be noticed by the listener. Another reason for the desirability of a low distortion level is that it gives a comfortable margin for deterioration during the life of the amplifier, particularly in the output stage.

The Testing of Amplifiers

Electrical Testing with Steady Sine Wave Input

This includes all the usual tests for nonlinear distortion, power output, frequency response, output resistance, hum and noise etc. Unfortunately, these tests do not give a true representation of the performance of the

[1] Further experience since the date of submission has shown the importance of avoiding positive feedback at any frequency. This is indicated by a higher response level with feedback than without, at the frequency in question.

[2] H. F. Olson, "Elements of Acoustical Engineering," D. Van Nostrand Co., Inc., New York, N. Y., 2nd ed.; 1947.

amplifier under conditions existing in the reproduction of music. In particular, three features are criticized strongly:

1) The tests for distortion (both total harmonic distortion and intermodulation) do not give values which are truly indicative of the subjective effect on the listener.
2) They are carried out normally with a constant resistive load, and are therefore quite misleading when used to compare different types of output tubes, such as triodes and pentodes.
3) They do not indicate the performance with transient input voltages, this being the usual condition when reproducing music or speech. In the case of Class AB operation, the results obtained with transients depend on the immediate past history of the amplifier, that is, the type of input signal, which has been applied during the preceding moments and its effect on the static grid bias and plate currents.

In addition to these routine tests, it is highly desirable that the gain of any feedback amplifier be measured and plotted, both with and without feedback, over the whole effective frequency band of the amplifier, including both low- and high-frequency peaks. In the case of high-fidelity amplifiers in the Williamson class it is necessary to cover from about 1 cps to 500 kc, and this involves serious problems both with oscillators and measuring equipment.

Electrical Testing with Pulse Type Input Voltages

The most practical laboratory methods for simulating the types of transients existing in speech and music utilize either repetitive pulses or a white noise input. The former includes square wave, sawtooth and other forms and also a high-frequency tone (10,000 to 20,000 cps) which may be pulsed either on/off or pulsed from maximum to reduced-amplitude. A square wave has both a steep rise, which is a transient, and a flat top which is not a transient. Significant features for measurement are the rise time in microseconds (from 10 to 90 per cent of the flat top height, conveniently measured at 10 or 20 kc), the percentage overshoot, the recovery time after the overshoot, and the percentage fall of the flat top measured at some convenient low frequency such as 50 cps. The oscilloscope should give good performance up to 500 kc. It is unfortunate that there is no single figure which can be quoted to indicate the distortion of a square wave.

Although white noise input has been used for the transient testing of both loudspeakers and amplifiers, its peculiar qualities make it a very difficult tool for the amplifier designer to handle, both in use and in interpretation of the results. Consequently no comment is offered at this stage.

Subjective Tests

Subjective tests with a critical listener are vitally important in evaluating the performance of an amplifier particularly in view of our present limited state of knowledge regarding objective electrical tests. It is well known that some amplifiers with less than 1 per cent measured total harmonic distortion at 400 cps give very poor results on a listening test. Some of the many possible causes of this effect include parasitic oscillations over a small portion of the cycle, a sharp kink in the linearity characteristic, and damped oscillations following a sharp transient.

Reliable conclusions from listening tests are obtained only when care is taken with the equipment and the methods used for the test. For comparing two or more main amplifiers with essentially flat frequency response, all the equipment other than the main amplifier (*i.e.*, record player, preamplifier, and loudspeaker) should be common to all tests, and a suitable volume control should be placed before the input terminals of all, or all but the most insensitive, main amplifiers. These volume controls should be adjusted by ear to give identical loudness for all amplifiers under test.

The best available loudspeaker should be used for all the tests; it should have wide frequency range and low distortion, otherwise distortion in the loudspeaker will prevent aural detection of distortion in most amplifiers. The listener should be seated on or near the axis of the loudspeaker, at a distance of 4 to 6 feet. At a greater distance the room characteristics tend to dominate the direct sound. The listener should have in his hands both the volume control and changeover switch. Comparisons should be made at all levels, including overloading, and various types of program material should be used, both speech and music.

It is difficult to arrange a "fair" test when comparing amplifiers having widely different maximum output powers, since in most well-designed amplifiers the distortion decreases as the level is decreased. If the power ratio is two to one, it might be possible to connect two loudspeakers in parallel to the larger amplifier, one being placed in another room. The alternative of using a partially resistive load to absorb the excess power of the larger amplifier is not satisfactory except perhaps with Class A triodes.

Conclusion

No amplifier is perfect, or is claimed to be perfect. As with all other products, amplifier design is a compromise. Exceptionally fine performance may be obtained from a "laboratory" amplifier requiring matched tubes and periodical adjustment, but this is of limited interest.

Unfortunately at the present time there is difficulty in defining suitable tests for amplifiers and so it is extremely difficult to make fair comparisons between different types of amplifiers. Within very limited categories it is possible to make at least an approach towards a fair comparison, for example all pure Class A push-pull triodes may be compared on the basis of total harmonic distortion, performance on square waves, and on pulsed wave-train input voltages. But it is not possible, with our present limited state of knowledge, to compare performance figures of Class A triodes with Class AB triodes, or to compare any type of triodes with pentodes.

This serious lack of knowledge, particularly about the transient performance of amplifiers, is seriously hindering both the design and the comparison of amplifiers. As a result, a long-term laboratory program is now being undertaken on the investigation of amplifier problems, among which are:

1) The operation of resistance-coupled triodes and pentodes following high-impedance stages; also more comprehensive and convenient forms for giving the distortion under a wide range of resistance-coupled operating conditions.
2) Methods of minimizing the distortion rising from the use of output tubes which are not specially matched.
3) A method of measuring nonlinear distortion in such a way as to give a true indication of the effect on a listener.
4) The development of a "dummy" loudspeaker load to be used for the measurement of distortion, power output, and gain vs frequency, which gives results closely approaching those obtained on a loudspeaker load.
5) The investigation of the relationship between the heights of the high- and low-frequency peaks and overshoot and other defects in the reproduction of square wave and pulsed wave-train input voltages, and to put these into the form of numerical values for comparison.

In conclusion, it is the author's hope that the contents of this paper will stimulate discussion on this subject which is of great interest to many people.

Loudspeaker Design and Application*

ARTHUR McLEAN†

Summary—**This paper is concerned with the problems involved in the design of loudspeakers and in the assessment of their performance. The incorporation of design data into speaker applications is also discussed.**

Introduction

Mr. STEWART concluded his paper with the theme that the ear is the final arbiter. A wide difference exists between the performance as expressed by present-day instrumentation and that perceived by the ear, because in most cases too many unjustified assumptions and omissions exist in the former. As an example offered to explain this statement a modern amplifier has been analyzed below.

It is known that it handles single-frequncy sine waves down to 5 cps uniformly, yet where Fig. 1(a), opposite, shows a saw-toothed wave of 50 cps applied to the input, the output depicted in Fig. 1(b) is considerably altered. While the amplifier is known to handle single-frequency sine waves uniformly up to 100 kc, its output as depicted by Fig. 2(b) is different from its input shown in Fig. 2(a) when handling white noise of uniform energy per cycle from 40 to 16,000 cps. The conditions under which the tests were conducted are as follows:

1) The oscillograms are photographs of the trace on a high-quality cathode-ray oscillograph.
2) While two beams and their associated amplifiers are available on the equipment, one system only is used in case the two systems do not provide identical performance on complex waves.
3) The amplifier is terminated in a fixed resistor equal to the nominal value of its output impedance.
4) The peak-to-peak values of voltage applied are well below the overload point of the system.
5) The gain of the cro, not the amplifier under test, is adjusted to give a suitable amplitude in each case.

The selection of a saw-toothed wave form as a basis for testing is not due simply to chance. It is the fundamental wave form of speech and animal sounds as well as that of all bowed musical instruments. When a blockage is caused by the vocal cords being closed, air pressure is built up until the cords open and the pressure decreases. To simulate these vocal sounds a steady flow of direct current from a battery is modified in a relaxation oscillator, where the voltages developed depend upon the charging and discharging of a capacitor by a critically-controlled trigger circuit. Thus Fig. 3(a) shows the wave form, using a probe microphone at the vocal cords and Fig. 3(b) depicts the wave form at the lips after the basic saw-toothed wave form is modified by a series of more than the twenty resonant chambers between the larynx and the lips. Thus a suitable synthetic wave form of speech comprises a sawtooth with super-

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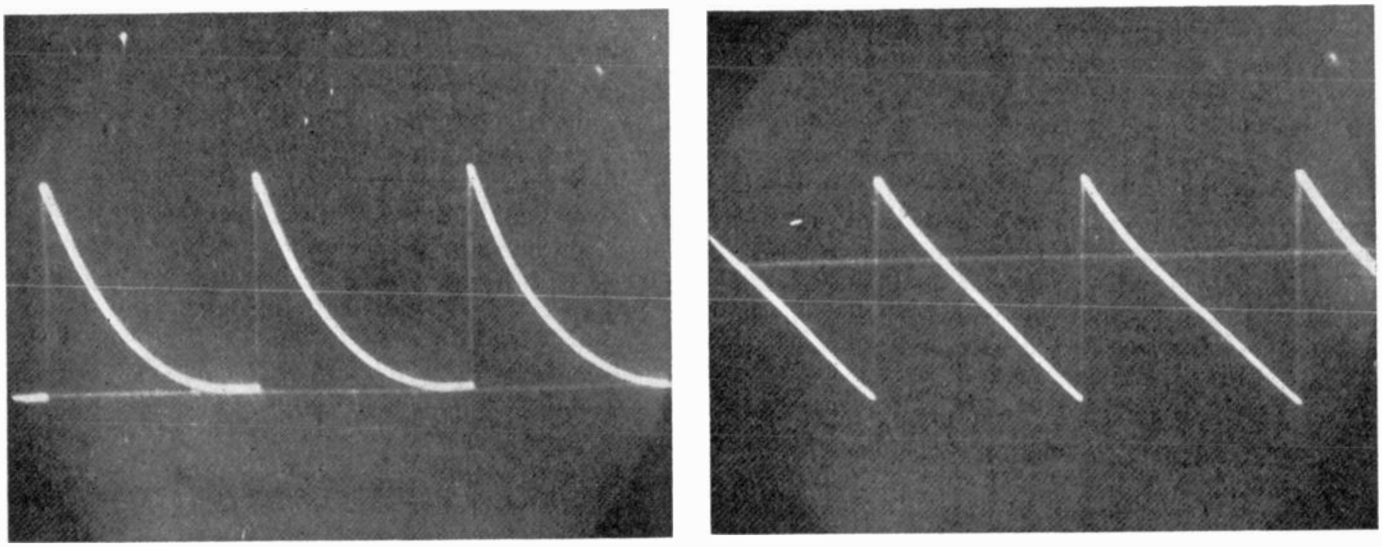
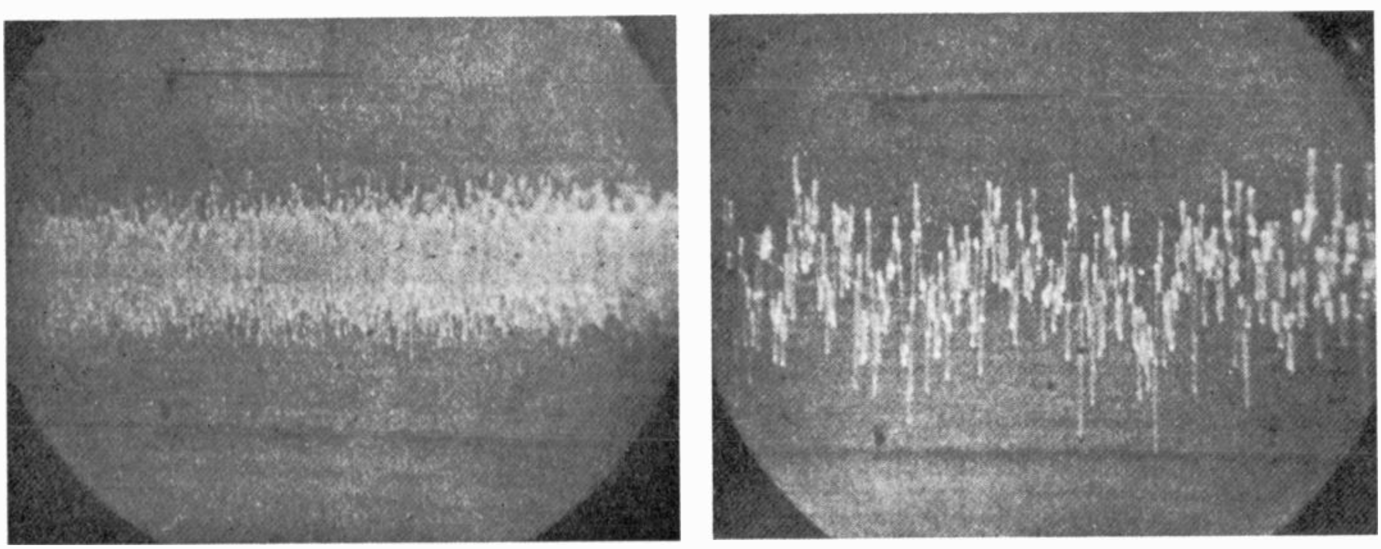
† Amalgamated Wireless (A/sia) Ltd., Sydney, N.S.W., Australia.



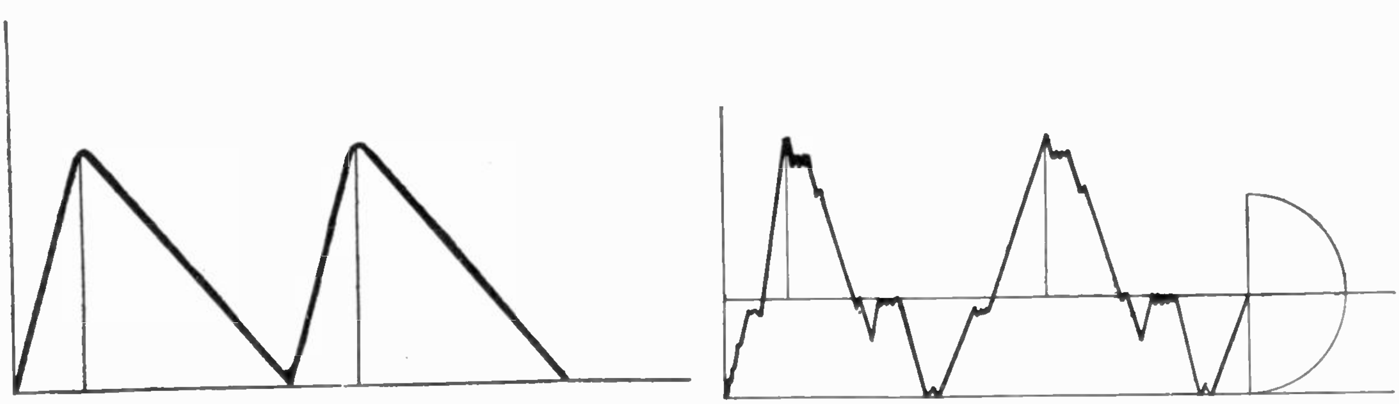
(a) (b)

Fig. 1—(a) 50 cps saw-tooth signal applied to amplifier input. (b) Amplifier output.



(a) (b)

Fig. 2—(a) White noise signal (uniform energy per cycle from 40 to 16,000 cps) applied to amplifier input. (b) Amplifier output.



(a) (b)

Fig. 3—(a) Wave shape of vowel "E" at the vocal chords. (b) Wave shape of vowel "E" at the lips.

imposed modulation. Much the same applies to many musical instruments. All practical sound systems handle two general groups of sounds, speech to convey information and music for entertainment. It is advisable, therefore, that the basic quantities involved shall be fully considered; *viz.*, frequency range, spectral content with energy per cycle throughout the distribution, dynamic range, rise time, die-away time, polar pattern.

To express the performance of electroacoustic equipment physical tests must take all these factors into detailed consideration. The ear can, and does detect minute changes in the constituent parts of complex sounds and quickly assesses the result to register approval or disapproval. As an example, modern manufacturing methods should insure that speakers are made within ± 2 db of a standard over the entire operating range expressed as a frequency response built up with single-frequency sine waves, but it is difficult to obtain two speakers which sound identical when reproducing complex sounds. Thus the ear has every quality necessary to permit a precise judgment to be made but it has a poor memory.

This introductory section serves to show that an exposition of the developments to come must be made on the basis of information-bearing tests which are related in a practical way to the problem in hand. A system does not, in practice, take single-frequency sine wave sound pressures in air, translate them into single frequency sine wave currents and voltages which, amplified, actuate a loudspeaker and so cause single-frequency sine wave sound pressures in its environment. It handles speech or music and the analysis must be on that basis.

Appreciation of the Problem

To appreciate the problems in audio engineering it must be realized that most sounds are extremely complex and therefore investigations may be made more thoroughly under complex conditions. Furthermore, thought must not be centered on any one part of the system which, in general, comprises the following:

1) The source of sound.
2) The sound medium and the contours of solid material by which it is enclosed.
3) The sound-to-electrical transducer.
4) The transducer in 3) in its coupling to the amplifier.
5) The amplifier in its over-all concept.
6) The transducer in its relationship to the output of the amplifier.
7) The electrical-to-sound transducer.
8) The sound medium and the contours of solid material by which it is enclosed.
9) The psychoacoustic elements of the ear.
10) The perception of sound.

Of these, the problems of the transducers and the ambient acoustical conditions are by far the farthest from being solved satisfactorily. Our knowledge of the ear is increasing considerably, but much is still unknown.

The two essential transducers are the microphone and the loudspeaker. The latter comes in for much criticism and quite justifiably so but, whereas it must be engineered for quantity production and to sell at a low price, the microphone is almost invariably hand-made at a cost which is high compared with that of a loudspeaker.

An important aspect in the use of microphones lies in the manner in which the original calibration is carried out, *e.g.*, if the microphone is close to a sound source, say a 12-inch speaker, then the waves are as on the surface of a sphere, whereas, at distances of about ten wavelengths, the waves may be considered plane. A microphone which is valid in its operation with a plane wave behaves altogether differently with a spherical wave. This is particularly noticeable on potential-gradient or ribbon microphones where the low frequency response to a sound source in close proximity is very different from that calibrated with a plane wave. As evidence of this, the National Physical Laboratory at Teddington, England, offers three aspects of calibration for microphones: free field, pressure, and close talking. At the lower frequencies, the respective calibrations give considerably different results.

The ambient acoustical conditions also militate against the successful handling of low frequency sounds. Rooms in general have a most unpredictable response to low frequencies and, even with anechoic chambers difficulties are experienced at 40 and sometimes at 60 cps.

Purchaser's Requirements

It is the usual practice for the purchaser of a speaker to seek information about its resonant frequency, its highest frequency, and its power-handling ability. However it is not possible to give a simple unqualified statement regarding these properties.

Resonant Frequency

The resonant frequency of any system is dependent upon the mass of the entire vibrating system, and the compliance, or reciprocal of stiffness, of the suspension system. One very important constituent of the mass is that of the air set in motion and therefore the efficacy of coupling the piston action of the cone to the air controls the resonance quite markedly, *i.e.*, the resonant frequency is affected by the method of mounting.

There are also five indicators of resonance and, in actual practice, except at low amplitudes, each may indicate a different value for the resonant frequency. The five indicators of resonance are as follows:

1) The frequency at which the current in the voice coil is a minimum in a constant voltage system.
2) The frequency at which the voltage across the voice coil is a maximum in a constant current system.
3) The frequency at which the numerical value of the impedance of the voice coil is a maximum.
4) The frequency at which the sound-pressure is a maximum.
5) The frequency at which there is a change in the phase angle.

An expression of resonance, therefore, requires an exposition of the conditions under which resonance is measured.

High Frequency Limit

The highest frequency, or top-cut, is usually expressed as that frequency at which the sound pressure level falls by 6 db below that line which is the mean through the 200 to 1000 cps response. In even the best speakers, that region gives variable performance and so the mean in that region is generally a matter of personal opinion, and with it, the expression of top-cut.

This aspect is also very much controlled by the microphone in use since considerable turbulence and diffraction around the microphone often cause inaccuracies in the measurements in the very region where precise interpretations must be made.

Power Output

The power-handling ability is almost impossible to express accurately. As an example, a unit which was tested to withstand 15 watts of 100-cycle square wave power for 15 hours, showed absolutely no sign of charring of the voice coil and its former, nor did any portion of the cone or the suspension system show signs of physical damage. Yet the same speaker produced about 8 per cent distortion of a single-frequency sine wave at its resonance with a power of 5 watts. Here again, only much relevant information associated with the statement can convey any real meaning to the expression of power handling ability.

Interpretation of Numerical Values

The expressions "15 watts of 100 cps square wave form power" and "5 watts at resonance" used above are quoted frequently in a rather meaningless manner since the current and voltage in even a wholly resistive circuit are difficult to express where nonsinusoidal wave forms are present. All measuring instruments in general use are calibrated by reference to sine waves of a single frequency and errors up to 12 db can be expected where a peak-reading voltmeter so calibrated is used to measure the quantities associated with a complex wave such as white noise.

As an example, the wave form of the oboe at 262 cps shown in Fig. 4 may be examined. The rms value of this wave, based on the integrated quantity of power expended in a resistance when such a current flows would be only about 0.4 times peak amplitude whereas in a sine wave it would be 0.707. Unfortunately, rectifier-type meters and tube voltmeters would give any reading between 0.4 and the peak value depending upon the value of the multiplier and certain other circuit constants. A further examination will show that the positive and negative peaks differ by the ratio of 5 to 4 so that a peak-reading voltmeter would give no indication of grid-swing limitations.

Therefore, an expression of performance of any equipment, whether amplifier or speaker on a single-frequency sine wave basis is entirely misleading. In the case of

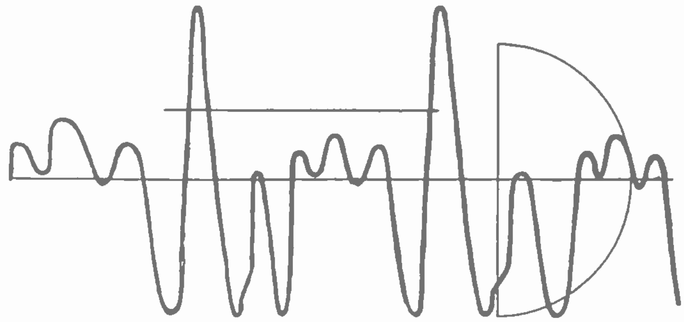


Fig. 4—Waveform of an oboe note at 262 cps.

the speaker, the position is complicated further by the marked change in impedance with the complexity of the wave form and even the frequency of sine waves. Thus, to put a voltmeter across the voice coil at resonance and to state that the power is 5 watts as in the case above is entirely erroneous. The best that could be stated under these conditions is that "there is available about 5 watts from the source."

Performance Limitations

In order to express the performance of a speaker realistically, it is necessary to know:

1) The change of sound pressure with frequency.
2) The extent of the generation of even-harmonic distortion.
3) The extent of the generation of odd-harmonic distortion.
4) The extent of the generation of intermodulation products.
5) The extent of the inability to handle transients. This involves the inability of the unit to take up a wave front as well as its inability to cease as the wave train ceases.

Frequency Response

The change of sound pressure with frequency, expressed quantitatively as a frequency response curve is important in that the harmonic components of a complex tone may be changed in their relative proportions. It is even more important to appreciate that the maxima and minima in such a curve are indications of cone break up or departure from true piston action.

Harmonic Distortion

Even-harmonic distortion results mainly from faults in the construction of speakers such as when the cone does not move as far backwards as forwards or vice versa. This may be caused by poor location of the voice coil with respect to the pole plate or by an inherent feature of design which does not allow movement one way as readily as the other.

Odd-harmonic distortion is present when excursions in cone movement withdraw too many turns of the voice coil out of the field and the amplitude of movement is not proportional to the current causing it. This statement applies equally to forward and backward movements and the distortion is symmetrical.

To overcome symmetrical distortion, it is necessary to use a voice coil which is long with respect to its corresponding pole plate so that, even with large movements (say 0.2 inch in a 12-inch speaker), the same number of turns is always under the pole plate. From this aspect alone a loss of the order of 4 db may result compared with a voice coil equal in length to the pole plate. A loss of this amount could pose serious limitations on the available output of the amplifier.

This raises the matter of the sensitivity of a speaker which is a very involved subject. If a speaker has low sensitivity then, for a given sound level, more power must be delivered from the amplifier but most present-day amplifiers have low distortion up to a critical power output after which the apparatus overloads. Thus a vicious circle of conflicting conditions arises.

While no clearly defined sound pressure level (spl) can be expressed for domestic reproduction, it is certain that a considerable power output must be available to handle peaks, and distortion may exist in either the amplifier or speaker or both. It appears that after taking all factors into consideration, a speaker should be as sensitive as possible. This is important also in the matter of damping.

At the present time, damping of a speaker is a very controversial subject and there have been many statements made from many points of view. However, it can be shown quite definitely that, regardless of the flux density, the spl at resonance always reaches the same value if a constant sine wave voltage is applied to the voice coil. This means that the lower the flux density the higher the relative Q of the resonant circuit. Every effort should be taken to eliminate such a condition, particularly since a high Q means that disturbances at varying frequencies can excite vibration in the cone at the frequency at which the system is sharply tuned.

Intermodulation Products

It is a well-established tenet of musical theory that there are certain specific combinations of tones which are pleasant or consonant, some tolerable, but all others are totally unbearable or dissonant and much of the poor quality in musical reproduction may be explained on this basis. A cone, driven backwards and forwards by the motor action of the driver acts as a piston up to a frequency at which its wavelength may be described completely on the cone body. Under this condition a wave may "stand" on the cone and a disturbance may exist in the frequency response curve. At the frequency at which the "standing wave" occurs the cone assembly has a fairly high Q and it may be excited by shock waves which have only a small component at the frequency at which the wave stands. Thus, if a wavelength corresponding to 1300 cps can "stand" on the cone, then any complex wave, frequency components of which have sums or differences near although not necessarily at 1300 cps can have a spurious component at that frequency. A clamped plate will break up at ratios 1.6, 2.1, 2.3, 2.7, 2.9, 3.5, 3.6, and 4.2 times some fundamental. The important fact is that only two are harmonically related and therefore the harshness must be accentuated.

The development of intermodulation products may be investigated in various ways as follows:

1) Straight-out combinations such as 70 and 2000 cps or 400 and 4000 cps each in proper amplitude proportions. These combinations are satisfactory for amplifiers but not necessarily for speakers.
2) CCIF and SMPTE tests.
3) The use of discrete bands of noise with close investigation of performance outside the pass band.

As an indication of the type of information which may be derived from such tests, details of a particular case involving the method outlined in 3) will be discussed.

A white-noise generator was applied to a low-pass filter with a top-cut frequency of 700 cps. The output of the filter, correctly terminated, was applied to an amplifier which fed the speaker under test. A calibrated microphone adjacent to the speaker was connected directly to a wave-analyzer with a pass band 4 cps wide throughout its range. Up to 700 cps the analyzer gave a reading in accordance with the frequency response of the speaker and then fell but rose again at 1300 cps to a value only about 10 db below that in the pass band as illustrated in Fig. 5. Supplementary tests showed that the second and third harmonic distortion of single-frequency sine waves in the 400 to 700 cps region was only about 2 per cent or 34 db down in each case. The rise in spurious generation at 1300 cps could therefore be due only to intermodulation, particularly as the response curve shows that a rise occurs in the spl at 1300 cps.

When considering the results of such a test, it is of interest to investigate the implications in a specific case. For example, Fig. 4 shows the wave form of the sound of an oboe played mf at 262 cps and Fig. 6 shows the spectral analysis of that note. It will be seen that the fifth harmonic of that sound occurring at 1310 cps is some 9 db below the fundamental in amplitude, but the combination of the fundamental 262 cps (17 db) with the fourth harmonic 1048 cps (18 db), and second harmonic 528 cps (12 db) with the third, 786 cps (11 db) would give an enhancement to the fifth harmonic so much out of its perspective as to change the nature of the sound.

The sound frequency 262 cps is close to that of middle C, C3 of the scale of equal temperament. Suppose now that the note of B3 of 247 cps is to be played. The intermodulation product of 1300 cps will still result even though the only consonant sound in that spectrum is 1235 cps. It is this very principle that really distinguishes one speaker from another, because two speakers would differ in the frequency and extent of generation of such intermodulation products.

Obviously, if different bands of noise were used in the test described, other intermodulation artifacts would be

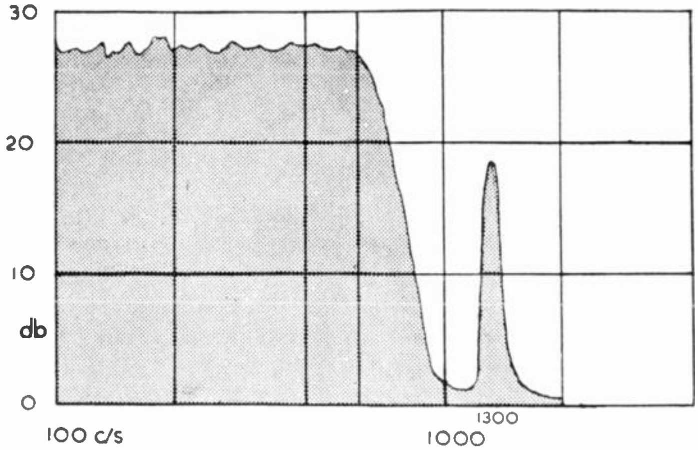


Fig. 5—Output spectrum of speaker driven by white noise signal with cutoff above 700 cps. Note spurious intermodulation product at 1300 cps.

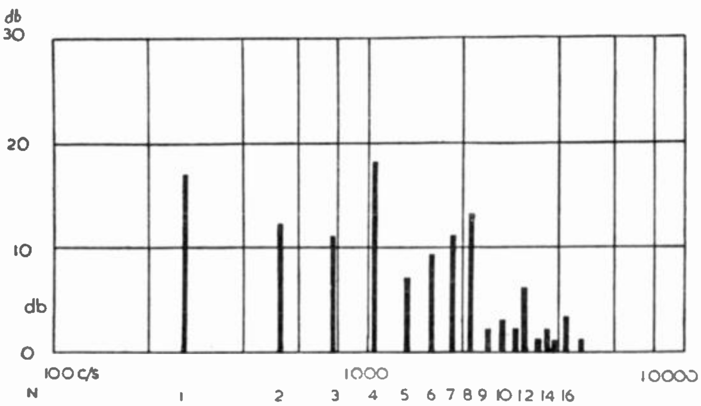


Fig. 6—Spectral analysis of an oboe note at 262 cps.

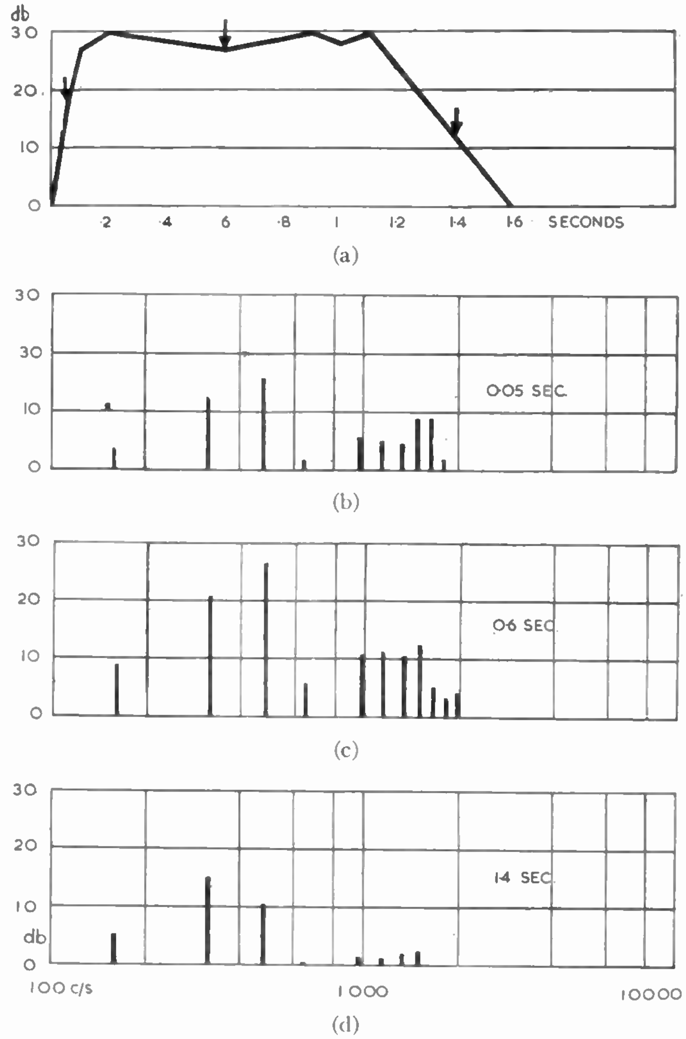


Fig. 7—(a) Variation of sound pressure level vs time for the vowel sound "ah." (b)–(d) Spectral analysis of "ah" at 0.05, 0.6, and 1.4 seconds.

discovered in the same speaker, but the same principle would apply.

Transient Response

Sounds are intermittent; they must start and stop but, more important still, they convey their meaning by changes within the period of their existence. For example, Fig. 7 shows the variation in spl and the changes in spectral content over the duration of the vowel sound "ah" as in "father."

Sensitivity and its Importance

It is insufficient to express sensitivity in terms of low-level amplitudes because one speaker may have high sensitivity but little ability to handle high-amplitude excursions. Furthermore, it is not at all practicable to use single-frequency sine waves for the following reasons:

1) All frequencies do not contribute uniformly to the appreciation of sensitivity on the part of the ear and therefore it would not be wise to use an index at say 400 cps.
2) An examination of an accurately depicted frequency-response curve shows that in the region from 300 to 1200 cps there are rapid changes, sometimes as much as a gradient of 6 db in 20 cps. Thus if the sensitivity of two speakers is being compared a small drift in frequency during the two tests could introduce unwieldy errors.
3) Unless measured in an anechoic chamber an intended impression of sensitivity would really be an indication of room conditions because, in most rooms, loops and nodes of 10 db difference are the rule, and a constantly moving speaker and microphone are necessary in any ordinary room.
4) Straight-out program material is also an unreliable source as it is constantly changing in nature and intensity.

There is available as a result of researches by Hopkins and Stryker[1] a noise spectrum shaped so that each frequency is present to the extent of its importance in the assessment by the ear of loudness. It has the advantage of a constant and accurately known source and, being of poly-frequency composition, the ambient acoustical

[1] H. F. Hopkins, and N. R. Stryker, "A proposed loudness-efficiency rating for loudspeakers and the determination of system power requirements for enclosures," Proc. IRE., vol. 36, pp. 315–335; March, 1948.

conditions and the position of the microphone are not controlling factors. It is quite definite that comparisons using that source may be made to within 0.5 db; *i.e.*, a loudness difference of only 0.5 db between two speakers may be accurately measured.

The permissible excursion in relation to the generation of distortion, as already mentioned, is definitely the controlling factor and therefore an increase in loudness should be obtained without increasing the excursion. It would appear therefore that the only alternative is to increase the area of the radiating surface, and a 15-inch diameter speaker would have a better index of performance than one of 12 inches. This reasoning appears justified when it is considered that RCA's type LC1-A speaker could have been of any diameter, since it was designed to be the ultimate expression of the acoustical engineers' knowledge in 1946 and its designer chose a diameter of 15 inches. There are other contributing factors but the use of as large a diameter as possible seems desirable.

Application of Design Factors

Up to this point there has been given an exposition of the factors which a designer must consider and it is now proposed to show how design data are incorporated into speaker applications.

Twin Speakers

It was shown that the handling of speech and musical waves in a common acoustical impedance, *viz.*, the complex structure of force on voice coil (generator), mass (inductance), compliance (capacitance) and resistance, the last-named being made up of equivalent friction and radiation, sets up the conditions for intermodulation in varying degrees of severity. Furthermore, as cone break-up, being a major contributing factor, is controlled largely by the geometry of the cone structure, it is apparent that there is a line of demarcation between frequencies where the cone can break up and those where it can not. There is thus a clearly defined line where frequencies may be diverted into one or other of two speakers and a convenient value is about 800 cps.

It is quite definite that a single unit even though it may comprise several parts mechanically linked together suffers from the effects of cone break-up more than a speaker comprising two or more separate units. The only possible exception is that where the hf unit is a horn, the extension of which is the lf cone. In this case, movement of the lf cone causes discontinuity in the expansion or rate of growth of horn area and as a result, the production of the high frequencies has a modulation of the low-frequency on the high-frequency envelope. A further problem in the use of multiunit speakers is the time-distance factor between the centers of the respective cones. If, for example a speaker of 12 inches diameter is used with one of 4 inches diameter, and they are mounted side by side on the same baffle, the minimum distance between centers cannot be less than 10 inches and this is the half wavelength in air of a frequency of about 670 cps.

It is current practice to eliminate most of these disadvantages by mounting the speaker units concentrically or coaxially so that the respective centers coincide. By this arrangement there is an insignificant time distance factor between the units. Two methods present themselves, one of which is to use a large diameter voice coil for the lf speaker and to build the hf unit within the voice coil of the former with the same cone angle in both units. In appearance the unit looks like a large cone with a small voice coil in the center and some marked discontinuity on the cone at a diameter of about three inches. It is claimed that because the contour of the lf cone is not part of the radiation network of the hf unit the intermodulation product by virtue of that discontinuity is not significant.

The second disposition results in a hf unit being mounted within the volume occupied by the lf cone. If the hf unit is totally enclosed in a balanced housing and it is located symmetrically with respect to the lf cone, then considerable absorption can be effected in the cavity so developed and the response curve would have a severe valley or dip at certain frequencies. This weakness is improved by mounting the hf unit off-center with respect to the lf unit. A very important advantage of coaxial mounting is that it may be used with 12-inch units of small voice-coil diameters. It would be difficult to mount a 3-inch voice-coil in a 12-inch speaker using the former method.

To sum up, there is no doubt that a speaker of two units mounted in accordance with either method offers a definite improvement over the older spatially separated units. Attention must be paid to the relative sensitivities of all multiunit assemblies and again, instrumentation is of paramount importance. To start with, a frequency response curve may give an indication, but, even so, much care must be taken in its interpretation.

Cross-Over Arrangements

The cross-over network required with a group of speakers is a very important item and considerable caution should be exercised in its design and application. It is primarily a filter and as such it is controlled by the impedances on the respective sides. Furthermore, even when ideally terminated, there must be an insertion loss and, with a practical unit which has considerable losses and mismatched impedances, its presence must be a distinct disadvantage. Even without the considerable departures from a smooth frequency-response curve at the cross-over point, the insertion loss must be considered carefully because for every decibel lost in the cross-over network the available net performance of the amplifier is reduced accordingly and considerable trouble may arise at critical distortion levels.

The practical method now coming into general use is to incorporate the cross-over features in the respective units as an integral part of the cone performance. Thus it is possible to provide a lf unit with a top-cut of about 1200 cps and with adequate attention to the design of the cone, break-up may be minimized up to this value. The reduction of the top-cut to this value may be effected by the choice of the material of which the cone is made as well as a judicious use of body filters in the design. In other words, the mass of the cone, which is the acoustical version of the inductance in a filter, is used in conjunction with a series of compliances which are capacitances to make a multisection low-pass filter of accurately predictable performance and no insertion loss.

Much the same performance may be built into the hf unit, by incorporating a high-pass filter comprising the capacity which is the volume between the cone and the housing completely without air leaks and the mass of the cone, again an inductance. It is sufficient to use a capacitor in series with the hf unit to isolate the latter from the effect of large-amplitude low-frequency currents and the resulting unit is quite satisfactory and much more economical than one requiring the classical and specific filters.

One very important advantage lies in the greater uniformity of the over-all impedance at the high-frequency end. An examination of the impedance vs frequency curves shows that at 2000 cps the impedance of large diameter speakers may be twice the nominal value, but if a further shunt circuit, such as an hf unit voice-coil with a capacitor in series, is connected across the lf unit, then the two in shunt must provide an impedance lower than that of either element. Although the impedance is not constant with frequency it becomes more nearly so than does a single unit.

It is conventional practice to use a high impedance voice-coil for the hf unit thereby reducing the value of the capacity in series with the hf voice-coil. For best results, it is most important to pay strict attention to the impedance of the hf unit at the proposed cross-over point. As an example, a case may be quoted where a multielement speaker sounded unnatural in the middle frequency register. A careful analysis of the analogous circuitry showed that, because the measured impedance of the hf unit was so different from the nominal, the use of a capacitor suited to the nominal value of the hf unit effectively caused a serious overlap in the middle-frequency region, *i.e.*, over an octave both speakers were operating simultaneously. Reduction of the series capacitor to the proper value in terms of the measured impedance effected a marked improvement.

Diaphragm Loading

A piston-type speaker cannot effectively couple its low-frequency excursions to the air itself. It must be provided with some auxiliary surfaces or enclosures to provide sufficient radiation. The baffle board has largely been supplanted by enclosures but a relic of the baffle is still in current use where the speaker is mounted in a wall. This seems quite satisfactory but conditions behind the cone do introduce some measure of control.

One of the earliest enclosures designed for the home is the vented or reflex baffle, even though it is not a baffle in the form of a flat surface, but a box, It is capable of a great improvement in low-frequency operation when properly designed. The cone-type of motor comprises the equivalent of an inductance, the mass of the cone and the air in motion a capacitance, the compliance of the suspension elements and resistance all in series. As a speaker has a Q value between 8 and 18, the amplitude at resonance is quite large. To overcome the high sharply-tuned output at the net resonance, it is possible to incorporate in shunt with the series resonant circuit a further series circuit comprising capacitance expressed in terms of a volume, in series with an inductance expressed in terms of a duct. The coupling between the two inductances and the nature of the circuit results in the response to tuning effected by a double-tuned over-coupled circuit in which slight rises occur in response on each side of what is the resonant peak of a single circuit element.

Use of the vented or reflex baffle results in a greater uniformity of operation down to low frequencies. This uniformity of operation includes both a more nearly uniform frequency response and a very much reduced variation in impedance with frequency because at frequencies below the net resonance the particle velocity through the vent is in phase with that direct from the cone. At frequencies just a little above resonance there is some cancellation, but the over-all effect is the following:

1) A leveling of the hump in the frequency response characteristic always associated with resonance.
2) An extension of the frequency range in the lower-frequency spectrum even by seven cycles.
3) A more nearly uniform loading on the amplifier.

A practical result of these advantages lies in reduction of various forms of distortion but particularly that known as frequency doubling and the marked tendency of a sharply resonant circuit to oscillate at its natural frequency even when stimuli are applied at frequencies away from the true resonant frequency. The latter effect is observed in musical reproduction when below a certain value an instrument always seems to be playing the same note and this is generally accompanied by considerable distortion.

Problems of Low Frequencies

The problems described above lead to the very vexed question of how low a frequency is required at resonance even assuming that it is measured and expressed accurately.

A careful analysis of the spectra of musical instruments shows few occasions when a frequency below 40 cps could, let alone would, be played. Furthermore, at any frequency at or below 40 cps, the performance of the microphone used in the sound field is of questionable performance and the ambient acoustical conditions are almost always inimical to the proper handling of such notes. But worst of all the ear is changing so violently in its perception in this frequency region that the slightest generation of distortion results in a greater perception of the higher-frequency components, *viz.*, the harmonics, than the fundamental. This means a complete upset to the tonal balance of the sound. The lower the resonant frequency the greater the distortion, the worse the rim break-up, and the worse the frequency doubling. Such an approach serves no useful purpose, with the single exception of satisfying the listener who requires the full range of the organ and even then, assuming that the only recital to which he will listen has the longest pipe played repeatedly.

Provided that a resonance down to 50 cps is accepted, a vented enclosure seems to offer to the home listener a reasonable solution to his requirements, because a speaker with 50 cps resonance in air will give quite adequate results down to 43 cps in such an enclosure. Many varieties of enclosure have been suggested.

Survey of Enclosures

One recent suggestion for housing a speaker is the RJ enclosure. It is based upon developing an equivalent acoustical circuit in which inductance, capacitance, and resistance are present in such proportions as to be aperiodic. Much publicity has been given to this arrangement and the latest official information from the sponsors themselves contains the statement, "Of course a bigger enclosure would give better results." Yet the main advantage claimed for the RJ enclosure is that despite its small volume, *viz.*, about 4000 cubic inches, it gives results equivalent to the larger volume of the conventional vented enclosure.

With speakers of very low resonant frequency the Helmholtz enclosure may be used. Very few types of speakers are suitable for such an application and, compared with other more practical types of reproducing systems, are usually outstanding.

The FAS Air-Coupler is also a suggestion but here again precise analysis does not show any marked advantage as it requires critical location and it is not free from the low-frequency effects of frequency doubling.

The Klipsch-horn is based upon good sound principles and has a great deal to commend it. It has been designed as the outcome of the need to reproduce the low notes of an organ. Therefore it is basically outside the scope of the remarks about the lowest desirable frequency for domestic purposes. It is understood that the Klipsch organization was originally founded to produce electronic organs and found that the speakers available constituted the limiting factor in over-all performance. Development was undertaken and the result was a sectionalized folded horn system which eventually uses the walls of the room as the final section of the increasing cross-sectional area. Here again, the location is critical; it is essential that it be placed in a corner. Whether the discontinuity in boundary conditions between the hard wood surfaces and the room finish is significant is not known. It is certain that the performance of horns generally is readily affected by changes in boundary conditions. However, it can be said that this particular unit performs admirably.

To sum up on the matter of enclosures for domestic applications, it must be remembered that a home is not an acoustician's playground and an ideal enclosure should be a flexible and harmonious unit in the furnishing scheme and an integral part of the sound generation system. In this regard a unit which has some bands of frequencies from one point and other bands from another point will not be as realistic as one where the source is relatively fixed. The vented enclosure allows the hf speaker to be mounted with its centroid to lie on the same point as that of the lf speaker and this makes a very convenient and satisfactory arrangement.

Future Considerations

Many observers feel that the loudspeaker, as we know it today, cannot be any better than a poor makeshift which must be suffered until some organization can produce a really scientific piece of equipment. The whole acoustic world is waiting anxiously for further details of the French development using ionic phenomena. It appears that this is a point source and therefore a low-efficiency radiator requiring a horn. As such, it is impracticable in a home quite apart from the associated high-voltage equipment required.

From whatever angle the problem is approached, sound can be conveyed to the ear only by compressions and rarefactions in a fluid medium between the source and the ear and some form of piston is essential. Just that is embodied in the present-day loudspeaker and future developments might well be along the lines of improving the existing design. This may be carried out more effectively by a more complete realization of the nature of the equipment used. The unit is referred to as a loudspeaker. It is that and more—it is a highly complex transmission system.

If a communication authority were simply to string a wire indifferently to trees between Sydney and Melbourne for instance, no useful information could be received at either end. The communication authority takes into account every minute detail of the line even to the extent, in critical cases, of accounting for the junctions of antennas to cables for entry to a repeater, yet the acoustical counterpart has, at its best, been little more than a paper surface driven backwards and forwards at its center under the influence of a force developed by the interaction of magnetic fields. Modern concepts envisage the speaker as an integrated transmission system with a clear picture of the function of its interrelated parts and with the tools to investigate its performance. The future for an extension of the present system looks bright, mainly because so much is becoming known of its true significance.

Because much attention has been paid to the high-priced speakers it should not be thought that the smaller speakers used in mantel and portable receivers have been considered unworthy of notice. Much of the basic knowledge on detailed performance has been gleaned in endeavors to improve such speakers. As an example, a portable broadcast receiver must be fitted with a sensitive speaker of light weight and small over-all dimensions. Moreover, the nature of a portable receiver is such that it is taken into areas of low signal strength and the problem of obtaining a good signal to noise ratio is consequently very real. By attention to details of cone behavior it has been found possible to improve the signal to noise ratio by 6 db or more, making speech readily discernible where, previously, it was impossible to detect sufficient syllables to gain any perception of the intended information. Also, the lower background noise from the improved speaker provided a better appreciation of a musical program.

It can be stated quite definitely that sound reproducing systems in the home are being improved constantly and, as tools in the form of information-bearing tests are put into use and greater attention is paid to the performance and capabilities of the ear, so further improvement can be made. Even in systems now considered good, there is considerable room for improvement and I feel certain that this will eventuate.

Discussion

Mr. W. N. Williams: The term "ultralinear" conveys the impression of overshadowing triodes whereas in fact ultralinear amplifiers only approach triode performance, although admittedly retaining to a considerable degree the plate circuit efficiency of pentodes.

Mr. F. Langford-Smith: I agree with Mr. Williams. I have attempted unsuccessfully to popularize the alternative term, "partial triode" operation.

Mr. A. Robertson: The main advantage of ultralinear operation is that it provides amplifiers of reasonable fidelity and adequate output with small output tubes.

Mr. Teale: A plate circuit efficiency, including output transformer losses, of 38 per cent has been obtained in the output stage of an ultralinear amplifier.

Mr. Lackey: The ultralinear circuit was used in Australia as early as 1933.

Mr. J. Moyle: Amplifier distortion is significant only as it contributes to the distortion of the whole system of recording, amplifying, and reproducing. I feel that perhaps a reduction of amplifier distortion from 1 per cent to 0.5 per cent might not be significant.

Mr. Langford-Smith: The change from a Williamson amplifier to another with 1 per cent distortion can be detected aurally.

Mr. A. McLean: The ear can detect minute changes in distortion as measured by conventional methods. I refer to a test in which a speaker with an input of 1 watt reproduced a 200 cps signal with a distortion of 3.2 per cent as measured with microphone and wave analyzer. Distortion equivalent to a speaker rattle was then introduced, with disastrous results from the listeners' point of view. However the measured distortion rose only to 3.6 per cent although harmonics up to the twentieth could then be detected.

Mr. E. Benson: The importance of 0.1 per cent to 0.5 per cent distortion is doubtful when 50 per cent distortion can be introduced into the waveform delivered from a speaker enclosure.

Mr. R. Tremlett, Chairman: The sensations experienced during the recording of, for example, a sixteen-piece orchestra producing sound intensities at the threshold of feeling, are entirely different to those of the listener to the playback at a lower level. The problem of defining distortion under such different circumstances is formidable.

Mr. W. Buckland: Although I cannot quote distortion figures on typical recordings, nevertheless, tests on the whole chain of recording equipment give intermodulation figures not greater than 3 per cent for standard 78 rpm discs.

Mr. Moyle: In tests on a series of amplifiers of different design using the same records, pick-up and loudspeakers, none of a number of listeners could detect any difference between the amplifiers, and because of this I doubt whether small changes in amplifier distortion were noticeable.

Mr. McLean: The concepts of distortion expressed by some of the speakers are too prosaic. Existing methods do not measure the "aberrations" in the reproduction which the ear finds so distressing. For example, inadequate control of the loudspeaker by the amplifier can produce "aberrations" and frequencies not present in the original can be created, thereby "coloring" the reproduction. Similar effects can occur elsewhere in the chain record-pick-up-amplifier-loudspeaker.

Challenged by Mr. Benson, Mr. McLean drew a diagram of a typical loudspeaker response to a single pulse and illustrated the finite rise time, overshoot and protracted decay with ringing at inharmonic frequencies, for example, 1.0, 1.6, 2.2, 3.6, etc., times, that would probably occur.

Mr. Benson: The response would almost certainly pass through zero several times and if to Mr. McLean's picture were added cancellations, rejections, and interference at the speaker, in the enclosure, and in the room, some idea of the problem would be obtained.

Mr. McLean: The effects of reverberation should be added to Mr. Benson's list. No standard method of measuring reverberation time had been agreed upon universally.

Mr. A. Campbell: Would the effect of the various aberrations mentioned be subject to rms addition?

Mr. Langford-Smith: They would not.

Mr. D. Lindsay: Different types of distortion have to be considered. Some types of distortion make the reproduction different from the original but are not unpleasant. Perhaps Mr. Moyle's amplifiers produced distortion such as this which needs comparison with the original to allow detection. However, other types of distortion introduce aberrations such as ringing and other spurious effects and only when the objectionableness of this type of distortion can be expressed in figures will it be possible to measure quality of repoduction.

Mr. Rothwell: I agree with Mr. McLean that the listener's reaction is the only real criterion of amplifier quality.

Mr. McLean: The importance of dissonance in any spurious responses created in a reproducing system must be emphasized. A response at a level of 0.1 per cent can produce a very unfavorable listener reaction if it is dissonant. However, although listening tests are important, they are essentially comparative and a basic requirement exists for a set of absolute measurements to define amplifier quality.

Mr. Robertson: In the absence of a sonata for tuning forks, sine wave measurements leave something to be desired. Dissonance is difficult to define since, with training and familiarity, some dissonances may become consonant to a particular observer.

Mr. Teale: What is the relative importance of amplitude and intermodulation distortion?

Mr. Langford-Smith: All forms of distortion are symptoms of a fundamental nonlinearity. Intermodulation distortion seems to give the best indication of listener discomfort but the rate of increase of intermodulation is not proportional to the effect on the listener. An attempt is being made to determine a method of measurement of nonlinear distortion which would agree with subjective effects.

Mr. E. Hodgkinson: What improvement can be expected from feedback from a separate winding on the loudspeaker voice coil?

Mr. Langford-Smith and Mr. McLean: Apart from the practical difficulties involved in such a separate winding we know that the mechanical movement of the voice coil is reflected electrically into the output transformer. However, one of the main problems in a loudspeaker is that the cone does not follow faithfully the movements of the voice coil and does not even vibrate as a single unit at higher frequencies.

Mr. Williams: What method was used to adjust speakers of different frequency response and sensitivity to within $\frac{1}{2}$ db as previously recommended by Mr. McLean for comparative testing?

Mr. McLean: Random noise frequencies weighted in a manner proportional to the importance of each frequency may be used in assessing loudness.

Mr. E. Moss: The resonant frequency of the cone could be important in tonal "coloring."

Mr. McLean: The resonance of a speaker is not necessarily an indication of its performance at low frequencies. To have a low resonance, *e.g.*, 90 cps in a 5-inch speaker, the rim or corrugations must be very thin. As such, the rim breaks up very markedly when handling complex tones and so much dissonance is developed that the low frequencies are masked or rendered less apparent. In many cases the use of a speaker with a higher resonance results in a better appreciation of the bass.

Mr. Teale: I have tested a number of speakers and found a resonance of about 2250 cps in each of them. Would this be due to staples in the cone as previously mentioned by Mr. McLean?

Mr. McLean: The sensitivity of the ear is most important in determining subjectively the response of any speaker. No matter how flat the response, as measured by physical means, a pronounced peak in the region of 2000 to 3500 cps results on listening. This effect is so pronounced that it renders unimportant any purely physical characteristics of the speaker.

Mr. Gonda: Is amplifier overshoot noticeable if the gain peak which caused it occurred at an inaudible frequency?

Mr. Langford-Smith: Such a peak might occur at a frequency even as high as 200 kc where it could be neither reproduced nor heard. However, such peaks could cause overloading and thus distortion and a low frequency peak could cause damped oscillation on transients giving rise to "muddy" reproduction.

Mr. Hunt: Can improvements be expected in loudspeaker design?

Mr. McLean: Improvements in speaker design have definitely resulted already from an appreciation of the significance of the entire speaker assembly. It is an axiom of communication theory that the generator, transmission line, and receiver must balance with one another in their respective properties and it is just the same with a speaker where the mechanical force developed by the voice coil carrying current in the magnetic field is the generator and the resulting movement is conveyed by the cone as a transmission line to the air as a receiver. The mechanical considerations in the entire speaker network can be set out in the form of an electrical circuit with every complexity accounted for. Therefore an appreciation of the entire problem indicates to the development engineer the lines along which progress may be made. Progress is being made in this field.

Some Practical Aspects of High-Fidelity Design*

ROY S. FINE†

Summary—**Modern day practices in designing high-fidelity equipment make necessary certain knowledge concerning the characteristics of components and complete reproducing systems. Some of these characteristics are often overlooked and many times some are overemphasized. Since a listener can hear music reproduced on nothing less than a complete system of some kind, the system characteristics should be those of primary concern. Component evaluation should be based primarily on the considerations involved in placement in a system, and certain parameters of components should maintain their proper relationships to each other. Much criticism today is directed toward basic theory of operation instead of the shortcomings of existing designs.**

An attempt is being made here to place some of these characteristics in their proper perspective and to point out certain fallacies that exist in the thinking of some people who have not had the opportunity to delve into the practical aspects of high-fidelity design problems. Everything is treated with the view of eventual placement in a complete system. Correlation of measurements and listening tests is an important factor.

MOST ENGINEERS appreciate the fact that high-fidelity is a subject of much controversy and such discussions usually hinge on the particular preference of an individual. The particular desires of one person can be satisfied by custom designing equipment to suit specific listening tastes. From an economic standpoint, however, such deluxe high-fidelity systems are not usually within the means of the average listener nor are they ideally suited to acoustical properties existing in the average home.

This article is intended entirely as an informative treatment of the problems involved in the design of practical, economical, high-fidelity instruments for the home, rather than an advanced treatise on the art of high-fidelity. The high-fidelity considerations discussed in this paper not only include the design problems involved in striking an ideal compromise but in giving consideration to all stages which combine to make up the high-fidelity system from "records to loudspeaker."

The primary objective of any design of a high-fidelity system should be concerned with the quality of sound reaching the ears of the listener; the method of reaching this objective is incidental. Concern over the quality of the individual components comprising a system should be dictated by the extent to which they affect the overall sound and not necessarily by figures of merit relating to the particular components. Naturally, any system is made of components which must be treated individually, keeping in mind the above mentioned statements.

Both components and systems will be considered and it is assumed that the reader is familiar with the basic principles of voice and music reproduction. The components in which we are interested are records, pickups, record players, preamplifiers, tuners, tape recorders, power amplifiers, and loudspeaker systems.

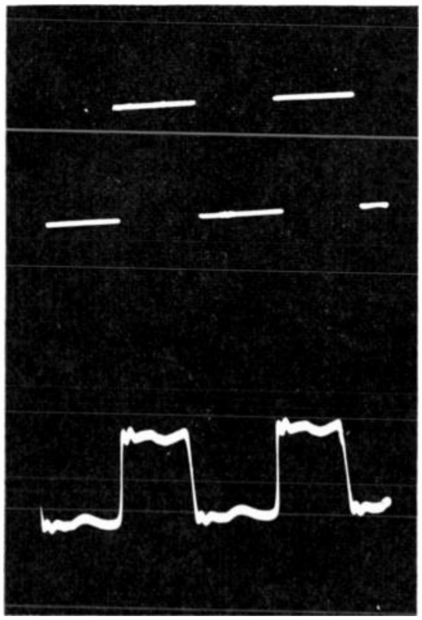


Fig. 1—1000-cycle square wave response of a good high-fidelity pickup vs an ideal response from a square wave generator. The slight raggedness of the pickup response from the record is probably due to the inability of the recording stylus to cut a clean square wave in the recording medium.

Records

At the present time there are two commonly used record types: disc and tape. The current high quality level of both media has been instrumental in increasing the interest in obtaining good reproducing equipment. Concomitantly, adherence to the same recording frequency characteristic standard by all major disc record companies has produced much greater uniformity of reproduction from record to record. As is well known, RCA introduced and began using the "New Orthophonic" recording characteristic in 1952. By 1954, all major companies adopted it under either of the terms Record Industries Association of America (RIAA) or National Association of Radio and Television Broadcasters (NARTB). Of course, microphone placement, room acoustics, and instrumentation in the original recording session vary, but in general most present-day records show a marked degree of uniformity of performance. Furthermore, they have extremely low distortion, wide-frequency range, and good signal-to-noise ratio.

Concerning tape records, there is an NARTB recording characteristic for tape and it is used by most companies making prerecorded tapes. The superiority of tape records over disc records, however, has been gener-

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† RCA Victor Radio and "Victrola" Division, Cherry Hill, N. J.

ally exaggerated. The one marked advantage of tape is the lack of deterioration with use, but the notion of better signal-to-noise ratio is a misconception. One major prerecorded tape manufacturer claims only equal signal-to-noise ratio on tapes and new discs. It should be borne in mind that these statements are made with respect to duplicated tape records and not to an original recording made on virgin tape, in which case the noise is somewhat lower.

Pickups

The four basic types of pickups in use today are magnetic, crystal, ceramic, and capacity. It is emphasized that with proper design, a high-fidelity phonograph pickup can be made from any of the above types, and that the much maligned Rochelle salt crystal is not inherently a low-fidelity device. It's true that a crystal pickup is more subject to variations and deterioration due to high temperature and humidity, but this has not been the basis for criticism. A crystal pickup can be designed to have wide range (0–20 kc), smooth response, good tracking capabilities, low-distortion, and low noise if its designer specifically attempts to obtain these qualities. Generally in practice, high-output voltage and low cost have received prime consideration necessitating a sacrifice in other attributes more closely associated with high-fidelity requirements. Also, the temperature and humidity effects can be minimized to the extent that satisfactory performance can be maintained for years in most parts of the United States and many other countries.

The prime reason for the unsavory reputation held by crystal pickups is simply that good high-fidelity designs have not been available on the market, but ordinary ones have. The same story is true, to a lesser extent, of ceramic pickups, except that the temperature and humidity effects are negligible. Some efforts have been made to produce a design suitable for high-fidelity application.

The situation is a little different when it comes to magnetic type pickups. There have been many good designs available for several years, but this is true primarily because they have been designed specifically for high-fidelity applications. As a matter of fact, some of these pickups enjoy a reputation not completely deserved. One magnetic type which has a reputation as a good high-fidelity device, actually is deficient in the extreme high-frequency range. Another type, sold in considerable quantity, has low compliance and results in poor tracking and high-needle talk.

Because of the foregoing it is unfair to judge the merit of a phonograph pickup for a high-fidelity application only on the basis of its principle of operation, *i.e.*, whether it is magnetic, crystal, etc. On the other hand, on the basis of pickups currently available, the magnetic designs in general are superior to crystal and ceramic types.

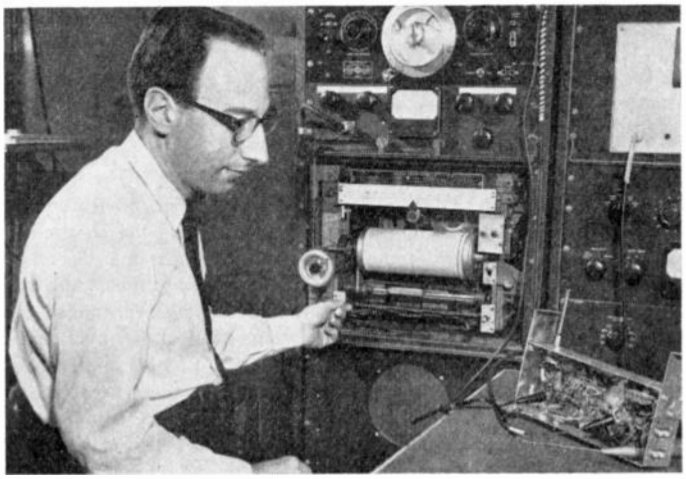

Fig. 2—The author is shown using automatic frequency response tracing equipment on a tape recorder chassis. The development work on each set produced requires literally dozens of such curves, finally resulting in a composite product, each component of which is matched to all others. Thus permanent performance records of each stage of design are on hand for future reference and as tools for use in the design of subsequent instruments.

Pickup stylii also receive a large share of attention from those interested in high fidelity. The three types in use today are osmium, sapphire, and diamond, with the last considered far superior to the others. Assuming that the shape of the stylus is initially correct, the most important aspect of its quality is its ability to maintain that shape with use. Because it is the softest material, osmium has the shortest usable life and is used only where low cost is of primary importance. Sapphire is somewhat more expensive and considerably more satisfactory; and of course diamond stylii show the smallest amount of wear with time. However, this picture is not a black and white one; the use of diamonds does not necessarily imply high quality stylii. Diamond being an extremely hard material is correspondingly difficult to shape and grind to a smooth finish. It has been found that some diamond stylii are so rough that a record can be ruined with 25 plays by them.

These rough diamonds, incidentally, represent products that are actually for sale as diamond stylii pickups. Furthermore, although satisfactorily ground diamonds are available, the best is not as smooth as the sapphire versions available from the same sources. Although diamond stylii, when right, are unquestionably superior, sapphires are capable of giving many hours of playing time before damage is done to the record or distortion is heard. In short, a sapphire stylus definitely has a useful application in high-fidelity work, especially since replacement is easily effected.

Record Players

Two types of record players are used in high-fidelity work; record changers, and manual transcription type players. The latter are generally preferable, where cost and the lack of operating convenience is not prohibitive for several reasons: 1) lack of changer mechanism makes instrument more dependable; 2) turntable flywheel effect more easily achieved; 3) permits use of better tone arm and bearing; and 4) lack of changer mechanism allows less needle force to be used.

In general, therefore, manual transcription type players are preferable primarily because they represent an easier engineering design job. This does not mean that no record changers are available which perform satisfactorily. Conversely, many manual players have performance very much inferior to the best changers, even though the makers claim otherwise.

However, most record players in use today are of the changer type and a brief discussion of these follows. In general, it is desirable to have as simple a mechanism as possible and one which requires a minimum of torque from the motor to operate and cycle. To minimize rumble, motor shafts, turntable rims, and idlers are ground to smooth concentricity. Motors are also dynamically balanced. Weight added to the periphery of the turntable helps, and of course reduces flutter and wow. Further, if motor excitation is kept to a minimum, 120-cycle torque pulses, a factor in rumble, will be minimized. The use of a four-pole four-coil motor is not an absolute requirement in a high-fidelity changer, but since the hum field is considerably less than that of a two-pole two-coil motor, the hum induced in a magnetic pickup is lower in the four-pole type.

Preamplifiers

Preamplifiers may or may not be contained on the same chasis as a power amplifier, but problems involved in making them separate or integrated are minor. The most satisfactory preamplifiers are those which give necessary flexibility with a minimum of complexity in operation and circuitry. When designing a completely integrated instrument, it is not necessary to make a preamplifier as versatile as if it were a separate component since most of its associated equipment is provided and matched. However, some versatility of input signal selection and control is necessary. For example, in an instrument having AM/fm radio and phonograph services, an input for a tape recorder and/or television sound is generally sufficient. Of course on less expensive sets where ac/dc circuits are used on phonograph only instruments, it is generally advisable to omit provisions for external attachments because of additional cost and engineering problems, such as hum, Underwriters' Laboratories approval, etc.

One feature often found on high-fidelity instruments is a record equalization selector switch, considered to be of prime importance by many. It has been found, however, that the differences involved in changing from one characteristic to another are, on listening, relatively minor; much smaller, for example, than the differences due to microphone placement, recording techniques, and record manufacture from one record to another within the same equalization pattern of a single manufacturer. Further, the general over-all effect achieved by equalization switching can be almost exactly accomplished by slight adjustment of the tone controls. Therefore, it is felt that a single playback characteristic is completely satisfactory and the characteristic should be the "New Orthophonic" which is very nearly an average of most other characteristics. At this writing about all record companies are using it, either under the above name or as RIAA or NARTB equalization characteristics; they are identical.

The subject of volume controls vs loudness controls is of interest in our consideration. A volume control is one which changes level uniformly at all frequencies; a loudness control varies the frequency response with level setting.

It is the feeling here that a loudness control is desirable in a composite instrument since the manufacturer has control of the frequency response at average listening levels. However, experience has shown that loudness controls closely approximating the action dictated by the Fletcher-Munson curves are grossly inadequate, the changes at room listening levels being insignificant to most listeners. Satisfactory loudness sensations are obtained with considerably more increase of bass response at decreasing settings of the control. Also, the Fletcher-Munson information indicates that no compensation is necessary at high frequencies, the distance between curves being generally constant. However, some designers find it desirable to incorporate some high-frequency boost with reduction in loudness control setting to overcome the masking effects of room and other extraneous noises.

An adequate tone control system is a necessity in a high-fidelity preamplifier with separate continuously variable bass and treble controls being mandatory. However, it is felt that graph type precision is not a requirement if satisfactory action is observed on listening. Adequate boost and cut, and uniform action are the important factors, but the elaborate circuits required to give ideally precise action do not justify themselves on on the basis of listening tests.

If a preamplifier has a monitor output for making tape recordings, it should be ahead of both volume and tone controls so that it can be maintained flat regardless of the shaping required to meet particular listening requirements at the time the recording is made.

A phonograph preamplifier presents an opportunity for a very interesting application of transistors: the first amplifier stage. This is a case where transistors are much superior to tubes and their use can be justified entirely by the exceptional performance obtainable. Of course, low noise units are required, but hum problems are solved virtually automatically. The cost of low-noise transistors for first stage use is very little higher than tubes designed for the same application and is not considered a prime factor. Undoubtedly the use of transistors will become almost universal in high-grade preamplifiers, resulting in a general improvement in performance.

Tuners

The subject of radio tuners will be dealt with only superficially here since they are themselves subject to a lengthy and involved discourse. It is taken for granted that fm services are capable of full high-fidelity performance but AM radio can be made quite capable of giving enhanced performance. Generally the requirements dictate broadening the IF response, inclusion of a 10-kc "Whistle" filter and full attention paid to a design capable of minimum distortion.

Although fm is quite capable of providing optimum high-fidelity performance, full satisfaction from use of a tuner can be obtained only if certain features are provided. An accurate indication of tuning is a necessity. The combination of a tuning eye or meter and automatic frequency control is unbeatable; very satisfactory results are obtained with an indicator and good temperature compensation of the local oscillator. AFC alone is somewhat less satisfactory since distortion occurs at extremes of the lock-in range even though the station is brought in strongly.

The usual quality factors in general radio performance are of course mandatory: good sensitivity, selectivity, signal to noise ratio, capture ratio, limiting, avc, etc., as well as good audio frequency response and low distortion.

Tape Recorders

Tape recorders, too, are a subject worthy of lengthy and involved discussion, and will be dealt with here only inasmuch as they affect high-fidelity reproduction. All but the most expensive recorders leave something to be desired for high-fidelity performance, indicating that the return for costly improvements is small, however necessary. Good frequency response at this time is readily obtainable; it is simply a question of adequate design. At present, there are specifications for playback response outlined by the NARTB for a tape speed of 15 inches per second. This specified response curve has been adopted for 7½ inches per second by RCA and other manufacturers of prerecorded tape. It is therefore essential that the playback system of any high-fidelity recorder conform to this curve, and the degree of conformity is considered to be an important figure of merit factor in the evaluation of such a machine. It is understood that the output voltage for application to a preamplifier be flat from a frequency tape such as RCA test tape 12-5-61T. On a complete tape reproducer, the playback compensation should be adjusted for the best acoustic output from the loudspeaker system.

The recording characteristic of a recorder should be adjusted so that a composite frequency response measurement, *i.e.*, a tape made and played back on the same machine, is flat from a flat input signal.

Fig. 3—Special tapes have been made to test loudspeakers. The author is shown using recordings of tone bursts to test the transient response of several loudspeakers. Other tapes contain specially selected passages, such as organ music, for testing low-frequency response, and cymbals and triangles for use in high-frequency tests.

Good signal-to-noise ratio is another extremely important factor in tape recorder performance, and judging from the performance of even high quality machines it is not easily achieved. Two types of noise are generally prevalent; hum and "white noise."

Hum can be minimized to inaudibility by good design. Some factors here are adequate shielding of heads and reduction to a minimum of the hum induced in first or first and second stages of the playback amplifier, generally achieved by using transistors, or dc on the tube heaters.

White noise elimination is more of a problem and comes from transistors, tubes, and the tape itself.

It is generally minimized not only by good design of the amplifier itself, but by proper selection of the best components available especially tubes and/or transistors, resistors, and the tape itself. Care must also be taken in the engineering design to prevent switching and other transients from magnetizing the heads, a factor which also gives rise to noise. In all deference to existing designs however, it must be pointed out that where a single tube or transistor phonograph preamplifier stage is generally required, from two to four tubes or transistors are necessary to amplify the head output signal to the same level.

The third important factor in tape recorder performance is flutter and wow, and good results are obtained primarily by precision machining of mechanical parts. Care must also be taken to insure that irregularities in reel motion are not transmitted to the tape as it travels over the head.

In general, the problems of good tape recorder performance have not been successfully solved especially in inexpensive machines, but it is definitely felt that good engineering application by a company with sufficient resources can soon bring about a much improved product.

Power Amplifiers

In the over-all picture of high-fidelity equipment, power amplifiers are probably the strongest link in the chain. It is relatively easy to design and build a good amplifier. The problems then become those of economy vs performance. The performance factor most often associated with the quality of a power amplifier is power output vs distortion. Some requirement and specifications have gotten beyond all reason currently. The industry apparently forgets that it is the distortion at normal listening levels which is important. It is felt here, that if the harmonic distortion is kept below 2 per cent at 10 watts output from 30 cycles to 15,000 cycles, any amplifier will perform satisfactorily with any combination of components and that the limiting factor in the performance of such a system will never be the amplifier.

It is true, of course, that higher powered units will more completely insure good performance. Although some feel that an increase of two to one in the rated power output of an amplifier makes a tremendous improvement, they are perfectly willing to ignore loudspeaker sensitivity differences which can easily be in the order of three decibels, and can effectively result in cutting the power output of the amplifier in half. It is recognized, however, that a brute force amplifier design job is perhaps the safest course, and that the technological progress of other components still has a goal to attain. Another aspect of amplifier and/or system design that is often overdone is the meticulous attention to matching of load to amplifier, especially when done on the basis of a specified loudspeaker impedance. Since the impedance of a speaker is not constant and is normally specified at the frequency of lowest impedance within its working range, the amplifier invariably works into a properly matched load only at that one point; there is a mismatch at all other frequencies. In a good amplifier, therefore, a load variation of as much as four to one should not materially affect distortion. The closer the amplifier is to being a constant voltage source, the better it is in this respect.

With respect to over-all amplifier design, it is felt that the methods of obtaining the desired results are unimportant; it makes little difference if triodes, tetrodes, or pentodes are used, if and how much feedback is used, etc. The important parameters such as frequency response, intermodulation and harmonic distortion, output impedance, input sensitivity, damping factor, noise level, etc., will determine the performance results obtainable.

Loudspeaker Systems

The design and construction of loudspeaker systems is again a complex subject; in fact engineering carrers are built around such acoustic work alone. In designing

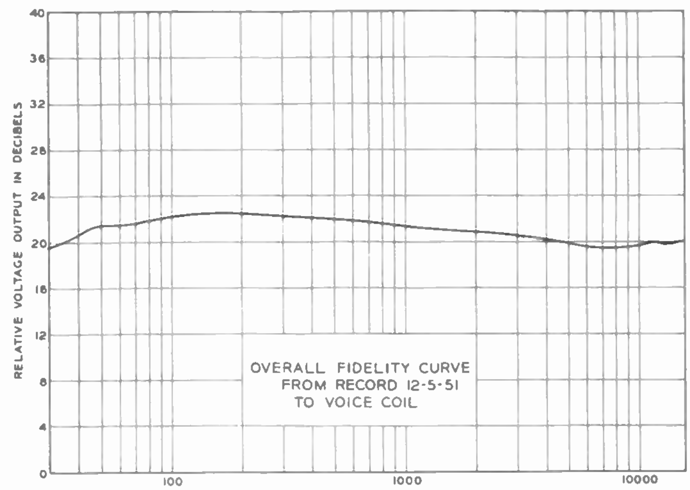

Fig. 4—Actual response curve of a high-fidelity instrument from record to voice coil. This response is considered excellent and experience has proven that such a curve is rare in high-fidelity installations and instruments.

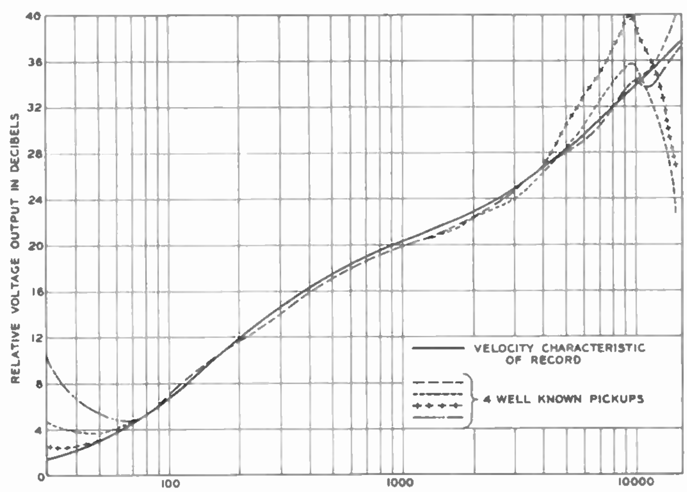

Fig. 5—Response of four well known magnetic pickups with good reputations for high-fidelity performance. Note the high-frequency response deviations, with two pickups rapidly falling off above 10 kc. One of the other units is such that tone arm resonance produces a pronounced low-frequency peak, Therefore, only one pickup in this group is considered to be capable of excellent frequency response in the tone arm in which they were tested. In spite of the apparent shortcomings of some of these pickups, proper compensation can enhance their performance.

a good high-fidelity instrument, it should be emphasized that, besides a sound background of acoustic engineering, elaborate, expensive anechoic chambers (sound rooms) and complex test equipment are necessary.

For example, a good loudspeaker and a good cabinet cannot always be combined to provide good results. Further improvements can invariably be made after the two are combined. Small changes may be measured although not heard, but the cumulative effect of several such small increments of improved performance can make startling differences upon listening to the system.

The quality of sound from a high-fidelity system can be almost entirely dependent upon the performance of its loudspeaker system. The so-called "presence" invariably stems from the speakers, and lack of "presence" can make an otherwise excellent system quite unsatisfactory from a listener's point of view. The same is true of the distribution pattern within the listening area. Since high-fidelity loudspeaker systems are invariably placed against a wall or in a corner, it is desirable to have adequate sound distribution over a large arc in front of the speaker. It is felt that any steps to achieve uniform distribution of sound behind the instrument are unnecessary.

Many persons are unaware that the cabinet affects chiefly the low frequency end of the audio spectrum. Tweeters and mid-range speakers may be placed at any convenient location and do not have to be within the confines of the speaker enclosure. The problems of directivity are also frequency sensitive; the distribution angle is somewhat inversely proportional to frequency and is dependent on the size and shape of the radiator.

There are more problems in matching a loudspeaker to a system than the one of impedance matching. It is well known that most speakers do not have absolutely flat frequency response, and irregularities can many times be compensated in the amplifier. One method of compensation commonly used, with small enclosures, is to tilt up the low and high frequency ends of the amplifier to a considerable degree to compensate for speaker system deficiencies in these regions. Of course, the speaker itself should have certain characteristics to make this compensation possible; it should be capable of sufficiently large excursions with low frequencies and should have a gradually sloping high-frequency roll-off rather than sharp cutoff.

It is also possible to compensate for deficiencies in the "presence" region by choosing a phonograph pickup with a broad but low amplitude resonance rise over this region.

General

In order to design a good high-fidelity system, it is necessary to have not only a thorough background of knowledge, but extensive and complex test equipment. Part of this test equipment is the environment, *i.e.*, a sound room and a listening room. The final results are the impressions that a listening panel observes on an evaluation of the over-all system from signal source to acoustic output. In building a composite system for sale to the general public, an average must be attained. Conversely, a system being built for an individual must please only one person; an infinitely easier job since there is no average individual, only an average group. It is imperative to have correlation between electro-acoustic measurements and the results of listening tests. Only then, can a composite system be satisfactorily designed to meet the requirements necessary for sale to the discriminating general public.

Contributors

R. D. Stewart

Ronald D. Stewart was born in Sydney, Australia in 1920 and attended Sydney Technical College where he received his diploma in electrical engineering with honors in 1940 and in radio engineering in 1942.

He joined Amalgamated Wireless (A/sia) Ltd. in 1936 as an engineering trainee, and from 1940 to 1945 worked in the Receiver Laboratory. For the next five years he was engaged in loudspeaker development. In 1950 he became head of the Components Development Laboratory.

Since 1955, Mr. Stewart has been Superintendent of the Quantity Production Laboratory.

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F. Langford-Smith

Fritz Langford-Smith (M'36–SM'43) was born near Sydney, Australia in 1904. He received the degrees of B.S. and B.E., the latter with honors from Sydney University in 1904.

From 1928 to 1929 he was with Metropolitan Vickers, Manchester, England, and from 1929 to 1932 was factory and development engineer for tubes at Cosmos Lamp Works under Associated Electrical Industries. He joined A. W. Valve Co. Pty. Ltd. in Sydney in 1932. From 1935 to 1950 he was editor of *Radiotronics*, to which he continues to contribute.

Mr. Langford-Smith was also editor of the "Radiotron Designer's Handbook," first published in 1934, and revised in three later editions. He is an associate member of the Institution of Engineers Australia.

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A. McLean

Arthur McLean was born in Sydney in 1909. In 1928 he joined the International Correspondence Schools as an instructor in engineering. In 1943 he became a member of the staff of the Acoustic Research Laboratory where he handled problems of high noise levels in the armed services. Immediately following the war he was engaged in the field of manufacture of electronic equipment for neuro-physiological research.

In 1948 Mr. McLean was employed by Amalgamated Wireless (A/sia) Ltd. where he is now an acoustical engineer at the Ashfield plant in Sydney.

He is an associate of the Acoustical Society of America.

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R. S. Fine

Roy S. Fine was born in Nashua, N. H., in 1922. He received the degree of Bachelor of Science in engineering from Brown University in 1944. He spent three years in the Signal Corps, and for the majority of this time was stationed at the Signal Corps Engineering Laboratories, Fort Monmouth, N. J., working on radar and microwave communications.

In 1946 Mr. Fine joined RCA as a student engineer and worked in the advanced development section of the RCA Victor Home Instruments Division until 1951. He then transferred to the Radio and "Victrola" section.

Mr. Fine is now working on product design of high-fidelity instruments and tape recorders.

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