IRE Transactions

on AUDIO

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IRE TRANSACTIONS on AUDIO

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Complexity and Unreliability in Electronic Equipments*

G. H. SCHEER†

This invited technical editorial, by the chief of the Communication Branch of the Communication and Navigation Laboratory, pertains to a vital subject of importance to all audio engineers engaged in development and design of military audio systems and components.—*Editorial Committee*

IN THIS DAY and age, complexity is an integral part of our everyday lives because we demand automaticity. We have relatively simple appliances such as automatic toasters. We go through the gamut of automatic record changers, automatic gas furnaces, automatic washers, broadcast receivers, automobiles, and even color television receivers. We seldom give a thought about what really happens when we dial a telephone. In spite of this complexity all about us, the reliability is usually acceptable if we pay more than a minimum price, and maintenance is relatively infrequent. With all of this background, why is there a problem in military electronic equipment?

The answer is not straightforward or simple, and no attempt will be made here to make it complete. If a component fails in a television receiver, all that results is inconvenience until it can be replaced. If a flat tire occurs, it is inconvenient, but putting the spare on remedies the fault. Seldom, if ever, is a life at stake. However, in military equipment, the failure of a single vacuum tube may be a real hazard. Even if an aircraft is not lost, the mission may abort. If this aircraft is carrying a nuclear weapon, it becomes of paramount importance because our very existence may depend on its delivery as scheduled. Generally, the demand for automaticity has increased rapidly in the past few years as more and more demands have been placed on electronic equipment to do more remembering and even considerable thinking. In airborne equipments this has been caused by reduction in numbers of crew members, which eliminated the radio operator as an exclusive individual. Flying higher, faster, and farther requires more attention from the crew than it did ten years ago.

Increased automaticity naturally increased the complexity; it takes more circuitry and more tubes to replace the former manual functions. Ground equipments now include computers and data transmission systems which must furnish new information almost instantaneously in guiding inhabited and uninhabited aircraft, some of which are out of line-of-sight in a few seconds.

In airborne equipments it has been necessary to put more electronic equipment into even smaller spaces than existed heretofore. This meant subminiaturization and the consequent rapid development of a whole new line of components, including vacuum tubes. The state of the art had advanced beyond the capabilities of the industry to furnish tried and true subminiature components which were completely "debugged" through years of production refinement and improvement.

Several precautionary measures were taken from the very start and have since proven their worth; both concern the maintenance of complex electronic equipment and are not limited to subminiaturization. First, circuitry was arranged in several separable packages commonly called subassemblies. These plug into a main frame to make a complete, operable equipment. There are many advantages in such a scheme. First, when a failure does occur, it is not necessary to do the repairing in order to make the equipment operable. Since subassemblies of a kind are interchangeable, it is necessary only to replace the defective subassembly with a good spare. The repair can be done later and even at a different geographical location. The use of subassemblies requires fewer spare complete equipments and also fewer bits and pieces because the latter need be available at only a few designated places of repair.

Having provided for subassembly or "unit" construction, the problem of quickly locating the fault had to be solved. All that was required was to provide each subassembly with a test socket, accessible without removal of the subassembly. At this socket the important voltages and currents were made available. A simple multi-meter with a multi-position switch is connected successively to these test sockets to isolate the faulty subassembly on a "go, no-go" basis. Limit readings are furnished for each switch position for each test socket in the cover of the multimeter.

There are many ways in which the fundamental design of the equipment itself can increase the potential reliability. It may be possible to exploit to almost any degree the idea of "parallel" rather than "series" use of components. If one headlight burns out or one sparkplug becomes fouled, you can still operate your automobile, even though not as efficiently as before. An example is the use of several transmitter output tubes in parallel. The failure of one tube need not cause a complete shutdown if it can be cut out of the circuit and the remaining tubes continue operating, even though the total output may be reduced. In airborne electronic equipments this approach usually means added size and

* Manuscript received by IRE January 28, 1955.
† Wright Air Dev. Center, Ohio.

weight which are so costly that the design becomes prohibitive. In ground equipments more use should be made of this philosophy. It goes almost without saying that all components should be operated well below their design limits.

After reliability and maintainability have been designed into electronic equipments to the best of our ability, such factors remain as quality of components, quality control, and workmanship. Special treatment has been found necessary in each of these departments, in order to get reasonable or even acceptable operating life between failures in very complex equipments. There is no intent to point fingers or establish blame for unreliability in military equipments. Rather, by frank admission of problem areas and concerted effort to determine cause and cure in every instance, we can greatly expedite improvement in reliability which, in the normal course of events might gradually cure itself over a period of years. This is borne out by a recent experience with a large, complex equipment in which the reliability was increased at least thirty fold in only three months. Such action would have been impossible, of course, if the fundamental equipment design had not been good.

Use of the Rand formula shows that a good portion of unreliability could be avoided if the problems are considered extremely important from the very start. "Hoping" that problems would cure themselves merely through elapsed time has proven to be fallacious. Any untoward sign must result in immediate corrective action or at least thorough investigation. No potential contributing factors may be ignored at any time. Consider a hypothetical case of an equipment with 2,000 components in its active circuitry. 99 out of 100 equipments are to have no early life failure. Substituting,

$$\frac{1 - .99}{2,000} = 0.000005.$$

This indicates that, statistically, we can meet our requirements if *each* and every component used has a failure rate of *not more than five per million units!* It has been indicated that this formula is a bit pessimistic in actual practice, but it does indicate the correct order of magnitude. This points out clearly that "garden variety" components, even JAN components, cannot achieve the desired results.

The above drastic requirements would not be unreasonable if there were a *positive* means of determining, in advance of use, the parts which would fail in early life. For the type of component which is particularly susceptible to early life failure, such as vacuum tubes, the longer the burn-in time (within limits) the more likely these tubes will be culled out. However, if one wishes to test quality into something like a tantalum capacitor, which is likely to fail because of deterioration of the acid seal with time only, no amount of testing can be substituted for elapsed time. These are but two classes wherein it is indicated that quality cannot be "tested" into components, and neither can it be "tested" into a complete equipment. Quality must exist in *every component* before it can exist in a complete equipment.

Assuming that satisfactory components can be assured, quality of the final product is still dependent on quality of workmanship. Even after many complete equipments have been manufactured, constant vigilance is required to re-educate shifting personnel, principally, and to impress workers who may relax because familiarity breeds contempt. When automatic manufacturing (automation) is completely established, the variable human element will be eliminated and constant, high-quality production will be maintained automatically.

As with components which may exhibit early-life failures, a relatively long run-in time for the complete equipment has paid dividends. The military requirements for 200 hours operating without any failure (except vacuum tubes and we do not want to except them indefinitely) is often exceeded in testing before the article is submitted for government acceptance. This does weed out many of the early-life failures, even the vacuum tubes which have recently been burned in for longer periods of time than heretofore. The first reaction is that running an equipment for 200 hours decreases the total life of the equipment by just that much. Actually, there is no definite time when end life can be considered reached. Some of our equipments, in use over 15 years, are still operating and repairable. Fig. 1 illustrates the value of a long factory test time.

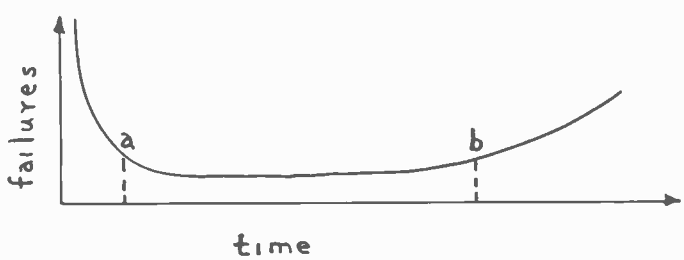

Fig. 1

It is quite obvious that it is advantageous to the customer to begin use of the equipment not before point "a" is reached in the life of the equipment. From "a" to "b" we have the highest reliability and lowest maintenance. Where "b" is located in the life span of the equipment, that is, when is it not economical to meet rising outages and maintenance cost, is not too well defined at this time, but, as more experience is gained with complex equipments, this region can be estimated for any particular equipment. It is much the same problem as when to trade in your automobile.

The most important factor in improving reliability in the finished product was found to be superlative quality control at every step of the manufacture of the component, subassemblies, and the finished equipment. Ordinary sampling methods were completely discarded, and all tests were made on a 100 per cent basis. Even today, sampling is not employed until it is proven beyond a doubt that the component or the complete equipment is a highly satisfactory article. The least indication of slippage and 100 per cent inspection is invoked. It is now obvious that the usual garden variety of component cannot fill the need, and a new quality level of component has been born to provide the reliability required in a complex military equipment.

It was through concerted effort and co-operation of the equipment manufacturer and the component industry that the new order of components was born, and born remarkably rapidly. However, the ultimate has not as yet been reached, and effort must continue until equipments operate consistently, without maintenance, for at least 2,000 hours. At the end of this period, the worn parts are either replaced or the entire equipment is discarded. On the surface, the latter procedure sounds extravagant, but it would not be. At present, the military services pay about ten times the original cost of the equipment in maintenance over the useful life. Actually, then, throwing away an equipment which had operated 2,000 hours without failure would be very inexpensive indeed.

We may be guilty of "Monday morning quarterbacking" but, in analyzing what had to be done to most components to improve their quality to an acceptable, if not superlative level, the faults to be overcome should have been obvious in the original design. Here are a few such examples.

Sealed relays were, and still are, rather bad offenders. One of the early faults was found to be that a volatile hydrocarbon impregnant was used. In a sealed relay there was no escape for the vapor from the hydrocarbon. Sparking at the contact points resulted in the deposit of free carbon on the contacts and, after a few operations, the contact resistance increased to a prohibitive value. The cure was simply not to use a volatile hydrocarbon as an impregnant.

In the case of ceramic capacitors, there were two outstanding faults. First, there was trouble with silver migration, principally through very fine cracks in the ceramic. Secondly, the silver was not symmetrically placed on the insulator which, in turn, prevented an adequate seal at the edges where the silver was closest, purely a quality control problem.

In tantalum capacitors, the acid seal was unsatisfactory and deteriorated with time. Many more examples could be given, but they fall in the same category. Significant is the fact that there has been very little trouble with resistors.

Subminiature vacuum tubes, developed especially for the Air Force subminiature equipments, were an enigma. Quantities of these tubes run off on a development basis were phenomenally good. Failures were practically unheard of in more than a year of usage. Yet, when they were produced in quantity production, the failure rate was worse than either miniatures or regular tubes. Normally, vacuum tubes are more uniform and better in all respects when automatically produced. Special efforts have been expended on subminiatures to reduce the causes for catastrophic failures. Also the dynamic limits have been tightened. These improved tubes are now available under what is known as "A" specifications, and only these subminiature tubes should be used in any military application.

It is expected that all of the valuable lessons learned on reliability, including detailed component treatment, will be available to the entire electronic industry when the compilation of data is complete. The lessons learned may be most vital to airborne applications, but will pay big dividends in reduced number of outages and less maintenance in *any* application. It may be that the new order of quality will find its way to JAN specifications, perhaps as a special quality level for uses where reliability requirements are severe and paramount.

The natural conclusion is that the superlative components will be prohibitive in cost. Also, that component parts manufacturers would rather sell a product which can compete, pricewise, with the product of a competitor. In the case of the subminiature vacuum tubes, there is little or no difference in cost of the better quality tubes. In fact, there is no reason for the existence of the earlier design, and it may never be a problem. Regarding other components, however, there is a definite problem. It is hoped that the answer will be simply that the demand for better quality components for all applications will be so great that inferior quality items will be forced off the market by lack of demand. This would result in price decreases as competition entered the picture as a normal consequence.

While quality cannot be "tested" into either a component or finished equipment, it can serve to precipitate early life failures *before* delivery to the customer if it is extensive enough. Here, again, it pays to run type tests continuously, on a few equipments, through at least 200 hours of operation, even after the quality level is high and sampling is used. One test, recently developed and adopted, is not very scientific, but it has proven of great value in showing up poor workmanship such as cold solder joints, rosin joints, poor mechanical connections, etc. This test consists of brute-force vibration of the assembled equipment, *without shock-mounting*, on a rather makeshift shaketable connected to a motor through an eccentric. The test is run for 48 hours, on every equipment, with the vibration applied ten minutes during each hour. Listening tests are performed during the vibration periods. Intermittents and outright failures are easily ascertained. One manufacturer considers this test indispensable because of the faults which it discloses. This test, as described, is directly applicable to equipments with an audio output. It should be adapted to other equipments, with a different display, where there can be no audio signal.

In summary, complex electronic equipments can be made adequately reliable by special quality control measures, carefully exploring every potential trouble area before it develops, using special quality components, maintaining high quality workmanship, utilizing special test methods, and maintaining an active rather than passive attitude at all times. Victory is far from complete at this time, and there must be no slackening of the effort to improve reliability in electronic equipment until we have obtained at least 2,000 hours of trouble-free operation in the most complex of electronic equipments.

Connecting Piezoelectric Pickups to Magnetic-Pickup Amplifiers*

B. B. BAUER†

THE MAJORITY of phonograph pickups and pickup amplifiers in use today fall into two classes: (1) Velocity-responsive (magnetic) pickups operating directly (or with the aid of a transformer) into high-gain amplifiers with input impedances of 20,000 ohms or more; and (2) displacement-responsive (piezoelectric) pickups operating into low-gain high-impedance amplifiers, with input impedance of the order of 1–5 megohms.

There is a general belief that pickups of Class 1 cannot be connected to amplifiers in Class 2, and vice versa. This belief is only partly correct. Magnetic pickups generate voltages of 10–30 millivolts at 1 kc on standard test records and require a degree of gain and compensation not available in amplifiers of Class 2. However, pickups in Class 2 often generate voltages of .2–.8 volts or more, and by use of simple networks can be used satisfactorily with amplifiers of Class 1.

The way this situation may come about can be seen by the following considerations:

Let the record groove be sinusoidally modulated whereby the displacement $\boldsymbol{a}$ is given in phasor form by

$$\boldsymbol{a} = ae^{j\omega t}, \tag{1}$$

where a is the amplitude of modulation; the velocity $\boldsymbol{v}$ is given by

$$\boldsymbol{v} = d\boldsymbol{a}/dt = j\omega ae^{j\omega t} = j\omega\boldsymbol{a}. \tag{2}$$

An amplitude-responsive pickup generates a voltage e_a proportional to the instantaneous displacement of the stylus, hence

$$e_a = k_a\boldsymbol{a}, \tag{3}$$

and a velocity-responsive pickup generates voltage e_v proportional to instantaneous velocity of the stylus, hence

$$e_v = k_v\boldsymbol{v} = j\omega\boldsymbol{a}k_v. \tag{4}$$

In (3) and (4), k_a and k_v are constants of proportionality.

A method for converting an amplitude-responsive pickup into a velocity-responsive pickup is shown in Fig. 1, which portrays the electrical side of a piezoelectric pickup. The pickup is represented by an equivalent Thevenin generator producing a voltage e_a in series with a condenser C (which is the total capacitance of the piezoelectric element and the leads). Let a relatively low resistance R be connected across the pickup terminals and choose

$$R = j\omega C \tag{5}$$

at the highest frequency at which the velocity-responsive operation is desired. The voltage across R is:

$$e = k_a\boldsymbol{a}R/(R + 1/j\omega C), \tag{6}$$

and since throughout most of the operating range $R \ll 1/j\omega C$, eq. (6) reduces to:

$$e = j\omega CRk_a\boldsymbol{a}. \tag{7}$$

But (7) is of precisely the same form as (4), except for the constant terms. Therefore, by the expedient of choosing a terminating resistance R in accordance with (5), the operation of a piezoelectric pickup can be made identical with that of a magnetic pickup, and identical amplification equipment may be used.

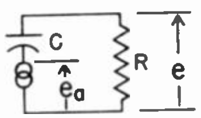

Fig. 1—Equivalent circuit of the electrical side of a piezoelectric pickup and terminating resistor.

Let us analyze some practical cases. A typical ceramic pickup and its leads may have a capacitance of approximately 500 mmfd. If velocity-responsive operation is desired to 15 kc, from (5) $R = 20{,}000$ ohms. As a first approximation, therefore, such a ceramic pickup may be connected to a magnetic-pickup amplifier with an input resistance of approximately 20,000 ohms, and it should provide satisfactory operation.

However, there are certain objections to this simple arrangement. First, certain piezoelectric pickups connected in the stated manner may have considerably more output than magnetic pickups, thus overloading the first stage of the amplifier. Second, the results will depend upon the amplifier input impedance and will vary with different amplifiers. Third, it has been found in practice that moderate departures from the simple resistor network provide a better velocity-responsive characteristic. In Fig. 2, for example, is shown a network designed by Roger Anderson, of our Development Department, for a typical ceramic pickup cartridge. The resistive voltage divider provides velocity-responsive performance and an output voltage of 30 millivolts at 1 kc on the Cook 10 LP test record, and the RC branch improves the balance between "highs" and "lows" by about 2 db. The "ideal" velocity response on the Cook 10 LP test record is indicated by circle dots, the pickup response with the suggested network is shown in the solid line. One can readily develop networks

* Manuscript received by the IRE, March 10, 1955.
† Shure Brothers, Inc., Chicago, Illinois

to suit the requirements of other pickup cartridges.

The pickup and network indicated in Fig. 2 may be connected to practically every magnetic-pickup amplifier and will provide performance indistinguishable from that of the better grade magnetic pickups. Needless to say, the use of magnetic-pickup amplifiers for piezoelectric pickups is advantageous only when it is desired to utilize existing equipment. In the design of new equipment employing piezoelectric pickups it is much more economical to use low-gain high-impedance amplifiers.

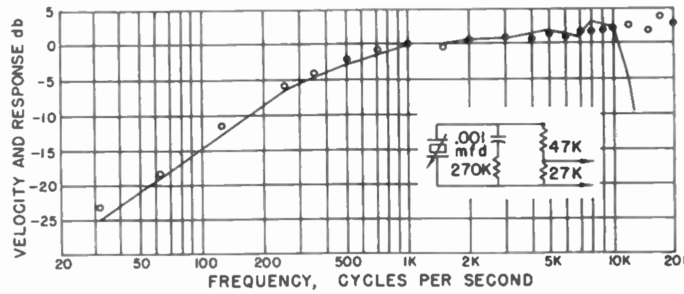

Fig. 2—Circle dots = velocity calibration of Cook 10 LP test record
Solid line = response of ceramic pickup and network.

PGA News

IRE-PGA ELECTION AND CONVENTION SUMMARY

NEARLY half of the membership of the IRE Professional Group on Audio participated in the election of officers for 1955–56. The results of this enthusiastic response to letter ballot follow.

Chairman for the year 1955–56, and member of the Administrative Committee for three years—Winston E. Kock, Bell Telephone Laboratories, Murray Hill, New Jersey.

Vice-Chairman for the year 1955–56, and member of the Administrative Committee for three years—Murlan S. Corrington, Radio Corporation of America, Camden, New Jersey.

Member of the Administrative Committee for three years—Andrew B. Jacobsen, Phoenix Research Laboratory of Motorola, Inc., Phoenix, Arizona.

Dr. Kock is Director of Acoustics Research at Bell Telephone Laboratories, and a Fellow of the IRE. In addition to his work in acoustics he is noted for research in electromagnetic lenses and electronic organs. He is also a Fellow of the Acoustical Society of America and was a member of its Executive Committee.

Mr. Corrington, who was re-elected both as Vice-Chairman and Administrative Committee member, is Manager of Audio, Acoustics and Antennas for the Advanced Development Section of RCA Victor Television Division. He is also a Fellow of the IRE. In addition to his work in audio and acoustics, Mr. Corrington has made notable contributions to circuit synthesis and tables of mathematical functions.

Mr. Jacobsen, who is a project leader in transistor applications at Phoenix Research Laboratory of Motorola, Inc., is the Chairman of the PGA Tapescripts Committee. The widespread use of tapescripts in PGA chapters is in large measure the result of his personal efforts in procurement and distribution of tapescripts. Before joining Motorola in 1953, Mr. Jacobsen was an instructor in the Department of Electrical Engineering at the University of Washington, where he also supervised a project on electronic instrumentation for research.

The Administrative Committee of the Professional Group on Audio met at the Waldorf-Astoria on Monday night, March 21, in order to review 1954–55 progress, to hear committee reports, to authorize a budget for the coming year, and to transfer responsibilities of PGA operation as required by the election and new appointments. B. B. Bauer was reappointed Secretary-Treasurer, and A. B. Bereskin was appointed Chairman of the Editorial Committee.

Both daytime audio sessions in the large Sert Room at the Waldorf-Astoria were attended by overflow audiences. The general audio session in the morning featured a tape recorded demonstration of an electronic music synthesizer by H. F. Olson and Herbert Belar of RCA Laboratories. During an intermission in the technical session, Vincent Salmon, retiring chairman of the IRE Professional Group on Audio, announced the results of the election, and turned over the gavel to W. E. Kock, incoming chairman. The chairman of the Awards Committee, J. K. Hilliard, outlined the new Awards plan (described in the March-April issue of the IRE TRANSACTIONS ON AUDIO. He announced that the first recipient of the PGA Award (to an author under 30 contributing to IRE a meritorious paper on audio) is Kenneth E. Goff of the Acoustics Laboratory at M.I.T., for his paper, "The Development of a Variable Time Delay," presented at the 1953 IRE Convention. The first recipient of the PGA Achievement Award was B. B. Bauer of Shure Bros. Inc., in recognition of many excellent audio papers appearing in IRE publications over a period of years.

The afternoon session was a symposium on the subject, "Music, High Fidelity, and the Listener," with Leo L. Beranek of Bolt, Beranek and Newman as chairman. The session was characterized by the large number of factors discussed by the panel members, on which additional research and factual information are needed. Active audience participation in the discussion period supported this main theme.

The evening session at Kingsbridge Armory was also very well attended. A panel headed by S. J. Begun of Brush Electronics Company presented a seminar, "Magnetic Recording for the Engineer." Particularly impressive was the evidence that the field of magnetic recording has expanded far outside the limits of audio. A lively discussion period closed with predictions for the future of magnetic recording.

Congratulations to the new IRE-PGA officers and to the award winners! And thanks to the retiring officers, authors of convention papers on audio, chairmen of sessions, and especially the IRE-PGA Program Committee, headed by Phil Williams with the particular cooperation of Michel Copel, eastern regional program chairman.

A. B. BERESKIN NEW EDITOR FOR PROFESSIONAL GROUP ON AUDIO

With the July–August issue, Alexander B. Bereskin, Professor of Electrical Engineering at the University of Cincinnati, will become the new Editor of the IRE Transactions on Audio. Daniel W. Martin of the Baldwin Piano Company in Cincinnati, who has served as Editor for two years, will continue as a member of the Editorial Committee for another year. Dr. Martin has resigned because of increased IRE responsibilities in the Cincinnati Section during the coming year.

Professor Bereskin obtained both the degree in Electrical Engineering and the M.Sc. degree in engineering from the University of Cincinnati. In addition to teaching he has worked in the engineering fields of power transmission, electrical arc and spot welding, hearing aid development, radar field service, and various projects as a consulting engineer. He has also published articles on electronic power factor meters; regulated power supplies; special video amplifier compensation circuits; and high efficiency, high quality audio power amplifiers. His unpublished works treat special RC oscillators, KW range audio power amplifiers, and selective audio-frequency transmission circuits.

The work of editorship is time-consuming and involves considerable responsibility. The Professional Group on Audio is grateful to Prof. Bereskin and to the University of Cincinnati for support of this activity.

FORMATION OF SYRACUSE IRE-PGA CHAPTER

According to W. W. Dean, Chapter Secretary Pro Tem, an all-time-high attendance record for the Syracuse Section was established at the January 6, 1955 chapter formation meeting, when some two-hundred members and guests turned out for the simultaneous launching of three Professional Group Chapters. As speaker for the audio portion of the program, Norman Cromwell of the General Electric Radio and TV Department discussed and demonstrated the benefits that can be achieved from the use of low-frequency rumble filters in home phonograph equipment.

Nearly fifty people have now expressed positive interest in the Syracuse Audio Chapter, either by their signature on the petition sheets circulated at the January 6 meeting, or by response to an earlier Section questionnaire. Organizational and activities plans are being formulated under a temporary slate of officers consisting of D. E. Maxwell, Chairman, Pro Tem, and W. W. Dean, Secretary, Pro Tem, who will function until the regular Section elections in May.

The first meeting was held on March 3, at the Syracuse Museum of Fine Arts. Over 80 people attended. The speaker was one of today's most outstanding authorities on disc recording, William S. Bachman, Director of Engineering and Development for Columbia Records, Inc. Mr. Bachman, who spoke on "Mechanical Considerations of Disc Recordings," was a major contributor to the development of the long-playing record, and was primarily responsible for getting it into production. His many other outstanding technical contributions to disc recording and reproducing, include the original development of the G-E variable reluctance phonograph pick-up and viscous damped reproducer arm.

Mr. Bachman's talk was enthusiastically received and the question and answer period was lively and most informative. A surprise feature was a drawing to determine recipients of five long-playing record albums contributed by Columbia Records, Inc.

DAYTON IRE SECTION ORGANIZES AUDIO CHAPTER

A. B. Henderson reports that the Dayton Section of the IRE held a regular section meeting, January 6, 1955, on "High-Fidelity." Julius Knapp of the United Transformer Corporation presented a paper entitled "Low Distortion Operation of Subminiature Triodes." (Ed. Note: It will be published in the next issue of IRE Transactions on Audio.) The regular section meeting was followed by an unofficial Audio Chapter meeting at which the tapescript "How Much Distortion Can You Hear?" was used. This tapescript was produced by the Cincinnati Chapter IRE-PGA last year. There is now a petition by the Dayton Section IRE, signed by the required number of section members, to form an Audio Chapter in Dayton. Formation of the chapter has been approved by the Administrative Committee of PGA.

IRE—PGA CHAPTER NEWS

Albuquerque, N. M.

The Albuquerque-Los Alamos Chapter met on October 23 at the Lovelace Clinic. Frank McIntosh of the McIntosh Corporation described and demonstrated the McIntosh 200-watt audio amplifier.

Boston, Mass.

On March 10, the Boston Audio Chapter heard a paper by Edgar Villchur and Henry Kloss of Acoustic Research, Inc. in Cambridge. The paper, "A Compact Loudspeaker System for Low Frequency Reproduction," described the "acoustic suspension" system that the authors have used to obtain linearity of loudspeaker cone-suspension stiffness at low frequencies with a small back enclosure. The cone suspension is purposely made deficient in mechanical restoring force, in order to achieve the desired results.

Chicago, Ill.

D. E. Weigand, Senior Physicist at Armour Research Foundation, was the speaker at the November 19 meeting of the Chicago Chapter at the Western Society of Engineering Building. Dr. Weigand's subject was "A Flux Sensitive Head for Recording and Playback."

Cleveland, Ohio

In addition to seven meetings in 1954, the Cleveland Chapter was instrumental in preparing eighteen stereophonic radio broadcasts and two television-radio stereosimulcasts over Cleveland stations.

Houston, Tex.

A. M. Wiggins, Vice-President of Electro-Voice, Inc., spoke on the subject "High-Fidelity Components" at a Houston IRE Section meeting sponsored by the Audio Chapter on January 18. The meeting was held in the new Research Center Auditorium of the Humble Oil Company.

Philadelphia, Pa.

Chairman H. E. Roys of the PGA Philadelphia Section arranged an interesting and varied program for the last three meetings of this season. The first meeting on the Spring agenda was a joint session with the Professional Group on Airborne Electronics. Mones E. Hawley, RCA, Camden, N. J., presented a paper, "Speech Communications in Aircraft," on February 9. On March 17, 1955, E. W. Kellogg, RCA (retired), spoke on "Experiences and Observations Along the Road to Improved Sound Reproduction." The final meeting of the season on April 13 was a joint session with the Philadelphia Section of IRE. A. A. Janszen of Harvard University delivered a paper, "Electrostatic Speaker System," which he accompanied by a demonstration.

San Antonio, Tex.

A petition from the San Antonio Section of IRE to form a chapter of the Professional Group on Audio has been approved by the PGA Administrative Committee.

San Diego, Calif.

New officers of the Audio Chapter in San Diego have been elected. The new Chairman is W. B. Bernard, Vice-Chairman, S. Q. Duntley, Program Chairman, S. H. Sessions, and Tapescript Chairman, H. L. Crispell.

At the September meeting, the tapescript "Time Compression and Expansion of Speech" was used. Written by Everitt, Fairbanks, and Jaeger of the University of Illinois, it was reviewed by J. C. Webster from the Psychophysics Branch of the U. S. Navy Electronics Laboratory. Dr. Webster also discussed related research in speech bandwidth reduction.

W. B. Snow, consultant in acoustics, spoke at the November meeting in San Diego. The program included slides, 16-mm motion pictures, and a 3-channel stereophonic demonstration. Recordings using three identical record-playback channels on 35-mm magnetic film were demonstrated. Comparisons were made of band music, sound effects, and dialog as recorded on single channel and stereo. The effect of reproducing stereo with reduced high frequencies was compared with wide-band single-channel recording. Stereophonic musical recordings reproduced with a 4,500-cycle low-pass filter were voted superior to wide-band single channel recordings of the same materials.

Tulsa, Okla.

The Tulsa IRE Section had an experiment in the use of tapescripts on March 17 at the Petroleum Science Building of Tulsa University. The tapescripts used were, "Push-Pull Single-Ended Audio Amplifier," by Arnold Peterson and D. B. Sinclair of General Radio Company, and, "The Physics of Music and Hearing," by Winston E. Kock, Director of Acoustic Research at Bell Labs.

Perceptibility of Flutter in Speech and Music*

F. A. COMERCI†

Summary—**The perceptibility of flutter at various rates, in recordings of speech and music, was investigated in relation to the development of a flutter meter which will give a direct indication of the effect upon programs as judged by listeners.**

SEVERAL YEARS AGO the Society of Motion Picture and Television Engineers proposed a standard method for measuring flutter.[1] The method proposed a flutter index based on empirical formulas derived from data on flutter threshold for pure tones at various flutter rates. This data was obtained in the Bell Telephone Laboratories and published previously by Albersheim and Mackenzie.[2] Subsequently a similar standard was presented by the IRE for American Standards Association approval. The standard in its present form (ASA 257.1, approved 16 March 1954) includes the flutter index in the form of an appendix for information only, because a good deal of controversy arose over the use of data obtained for pure tones as an indication of what effect flutter might have on speech and music.

Judging from the absence of precise information of the effect of flutter on speech and music, except for individual and sometimes contradictory opinions not supported by experimental data, there is little wonder that a controversy should arise over the use of a flutter index.[3] A series of subjective experiments, connected with the development of an instrument for measuring flutter, has been conducted at the Material Laboratory for the Navy Bureau of Ships. The results of these experiments indicate that not only can the information on pure tones be used to predict the effect of flutter on music programs, but that a weighting network can be incorporated in a flutter meter to provide a single reading of a flutter index, which will bear a direct relation to its effect on music programs as judged by listeners. These experiments were found necessary because there was an almost complete void of information in the literature relative to the effect of flutter on speech or music programs. The only information available was concerned with the threshold of perceptibility of flutter in pure tones alone. In this respect only one group of experiments at the Bell Telephone Laboratories plus a few measurements of Lautenschlager[4] could be found. Fortunately, several experiments to find the differential pitch sensitivity of the ear were conducted in such a manner as to be relevant to the flutter problem. Of these, the only experiments having any degree of reliability are those reported by Shower and Biddulph,[5] and by Knudsen.[6] An important group of experiments was conducted by H. Schechter at Massachusetts Institute of Technology,[7] but has not been published yet.

A preliminary study of the literature for information to guide the design of a realistic flutter-measuring instrument resulted in the following conclusions:

1. For frequencies up to about 500 cps the threshold of flutter perceptibility occurs for a constant frequency excursion, but above this frequency the threshold occurs at a constant per cent flutter except at the extremely high frequencies (10,000 to 11,000 cps) where a gradually increasing per cent flutter is required for perceptibility. It can be considered that a large portion of the energy of any speech or music program lies in the frequency range between 500 and 2,000 cps, where perceptibility is highest. Hence, the frequency response of a system would not be an important factor insofar as flutter perceptibility in program is concerned.

2. Flutter perceptibility increases with sensation level up to a level of 40 db (ref. threshold). Above a level of 40 db the threshold of perceptibility is essentially constant. Because listening levels lower than 40 db are practically never used, it could be considered that level would not be an important factor.

3. Although flutter perceptibility is slightly higher for diotic or bone conduction than for monaural hearing, the difference is not practically significant. However, when listening by means of a loudspeaker in a reverberant auditorium the flutter perceptibility is much greater. The flutter thresholds found at Bell Telephone Laboratories for listening in a live auditorium were about one tenth as high as those reported by Shower and Biddulph[5] for earphone listening. It is generally agreed that this effect results from standing wave patterns which produce amplitude as well as frequency modulation in a reverberant auditorium. If such is the case, the greater thresholds should only occur for pure tones or sustained tones. Because music and speech are continuously changing in both amplitude and frequency, standing-

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[3] E. W. Kellog, Comments on flutter standards," *Trans. IRE*, vol. PGA-2, no. 4, pp. 99–102; July-August, 1954.

[4] F. Lautenschlager, "Über die subjective Wahrnehmbarkeitsgrenze und eine Methode zur objective Verstimmung von Filmtransportstörungen beim tonfilm," *Elec. Nacht. Tech.*, vol. 11, p. 409; December, 1934.

[5] E. G. Shower and R. Biddulph, "Differential pitch sensitivity of the ear," *Jour. Acous. Soc. Amer.*, vol. 3, pp. 275–287; October, 1931.

[6] V. O. Knudsen, "Sensitivity of the ear to small differences of intensity and frequency," *Phys. Rev.*, vol. 21, p. 84; January, 1923.

[7] H. Schechter, "Perceptibility of Frequency Modulation in Pure Tones," Doctoral thesis, Mass. Inst. Tech.; 1949.

wave patterns, except for rare instances, may not occur. On the other hand, there are those who believe that the direct and reflected waves may carry two tones, differing in frequency by the amount of the flutter excursion, to the listener's ears, resulting in beat notes of a low-frequency, growling nature. If this is the case, then the flutter perceptibility should be much greater for those flutter rates whose period is equal to twice the time differential between the direct and reflected waves. The findings at Bell Telephone Laboratories do not indicate such phenomena, but the data do not appear to be complete enough to warrant discarding this theory, especially since the increasing perceptibility with increasing tone frequency (found constant with earphone listening for those tone frequencies considered) seems to support this concept. Certainly, here is an avenue for future investigation.

4. The effect of flutter on tones appears to be experienced subjectively in three different ways. For flutter rates up to about 5 cps the listener experiences a tone varying in pitch in accordance with the flutter amplitude and rate. For flutter rates between about 5 and 15 cps the listener seems to experience a change in loudness rather than frequency. Above a flutter rate of 15 cps the listener ceases to experience a changing pitch or loudness. Instead, he hears a low-frequency background tone which appears to coincide in pitch with the frequency of the flutter rate. The flutter perceptibility has been found to reach a maximum for a flutter rate of between 1 and 3 cps decreasing at flutter rates above and below almost as a direct or inverse function of flutter rate. Above flutter rates of about 80 cps Schecter shows that the perceptibility begins to increase, the amount of increase depending on the listening level and the tone frequency. For flutter rates above 100 cps Schecter[7] has shown that the threshold masking curves can be predicted by considering the frequency modulation sideband components of the fluttered wave at threshold. For earphone listening the flutter threshold at maximum perceptibility was found at 0.15 per cent peak flutter. For listening in a reverberant auditorium the threshold was found at as little as one-tenth the percentage flutter obtained for earphone listening. There is some disagreement between the rate of decrease in flutter perceptibility above and below the flutter rates of maximum perceptibility but, judging from the variation in the flutter wave shapes employed and the uncontrolled level and auditorium acoustics employed in some experiments, these differences can be understood.

5. Often flutter in equipment occurs in a very complex waveshape. Many believe that the effect of flutter is a function of the rms value of the flutter excursion. Others believe that its effect is better indicated by the peak excursion of the flutter. Further experimentation in this direction is warranted.

In order for the Material Laboratory to design or specify a meter for the realistic measurement of flutter, one which preferably would give one reading which would be directly related to the obnoxiousness imparted to speech or music, the following information was required:

1. Whether information concerning the effect of flutter on tones could be applied to speech and music.
2. Whether the indicating meter should be an rms, peak, or other type of indicating device.
3. Whether the obnoxiousness imparted to speech and music would be affected by the acoustical characteristics of the listening space.

The experiments reported herein were designed primarily to answer the three items above. Absolute threshold values were not considered as a primary objective. In view of the fact that "Vibrato," a form of flutter, is purposely added to tones by the musician to enhance quality, it was considered that music containing certain flutter could be found more pleasing. Hence, in most of the subjective experiments conducted, the flutter was varied from an amount that was judged to be imperceptible to an amount that was considered to have a maximum effect on programs. In addition, these experiments were particularly useful for determining whether a particular flutter meter was able to rank the various flutter types encountered, in an order which agreed with a subjective ranking of the same flutter on the basis of its effect on program.

To cover every possible type of musical program that might be encountered would have been a hopeless task. Listening to various types of fluttered programs indicated the points listed below.

1. Certain musical instruments, such as the piano, the brasses, and bells and chimes, which ordinarily are not played vibrato, were more affected by flutter than were the stringed instruments such as the violin, the cello, or the guitar.
2. Although flutter was more perceptible in certain instruments, on sustained tones the relative effect of flutter rate was about the same.
3. For tones of short duration, flutter of low-frequency rate was not easily perceived, but flutter of high-frequency rate (say above 10 cps) was equally perceptible regardless of tone duration.
4. It was almost impossible to detract from the intelligibility of speech by introducing flutter. In speech, the low flutter rates up to at least 5 per cent peak amplitude could not be detected, but the high flutter rates were heard as a quivering. (It is interesting to note that this quivering disappeared when white noise was introduced in the speech.)
5. The effect of flutter on male and female speech was similar.

As a result of these observations, it was decided to restrict experiments to a few types of program such as a pure 1,000 cps tone, a sustained chord of eight tones, piano music, a symphonic orchestration and male speech. Musical programs were selected which contained both sustained and short tones from local station FM broadcasts or from high-quality test films.

Quality rankings of fluttered program were obtained by a subjective technique which was termed the Listener Preference Test. This test is a form of round robin quality comparison where, in view of the number of items and the exhaustive number of individual comparisons required, one of the items is selected as a standard to which all of the other items are compared. In the experiments reported herein comparisons were made in five categories on the basis of over-all program quality. The judges were asked to state whether the item in question was, compared to the standard, much worse, worse, the same, better or much better. Every attempt was made to randomize the experiments with respect to order of presentation of flutter variation, standard or unknown, and program selection. A "Don't Know" category was added in case the judge, because of inattentiveness or another reason, was unable to make a judgment. At least 12 judges were used for each comparison. It is conceivable that there could be a large variation in the judges' categorization; for instance, for a particular difference in quality one judge might systematically categorize the comparison as "much better" whereas another judge would categorize the same comparison as only "better." This difference, in order to permit averaging all the judgments for each item to obtain a preference ranking score, was minimized by a statistical technique of quantification. The data was then subjected to an analysis of variance, in order to determine the significant mean difference. The ranking scores themselves were used to plot listener preference against flutter measurements. This procedure could be questioned but, as will be indicated later, the listener preference ranking scores are found to have a surprisingly linear relationship to flutter amplitude for each flutter rate investigated, and hence indicate that the procedure is valid.

In a preliminary experiment the realistic approach was taken, using the sound sections of nine different motion picture projectors. Listener preference rankings were obtained for diotic earphone listening with and without inclusion of a 2-second echo chamber for various program selections from high quality test film. The preference ranking scores were compared with measurements of frequency response, signal-to-noise ratio, harmonic distortion and intermodulation distortion at a medium level, peak-to-peak flutter, and a flutter index calculated from flutter threshold for a 1,000 cps tone. The rankings were found to be related to the intermodulation distortion and flutter index data. The relationship was considerably improved by converting the IM distortion and flutter index measurements to a combination "Preference Index" by somewhat questionable means. The results of the preliminary run indicated the following points.

1. In order to obtain any information on flutter all other distortions should be confined to an insignificantly low value;

2. There is a possibility of obtaining a combination quality index for all types of distortion in an audio system, although continued investigation toward this object was beyond the scope of the project;

3. The preference rankings obtained for direct earphone listening were similar to those obtained when an echo chamber was used. However, there may have been a difference in the actual threshold of flutter perceptibility which was not revealed.

In a second experiment, eighteen different types of flutter were obtained by modifying the drive mechanisms of three tape recorders. Other distortion was held at a constant low figure by using a separate high-quality reproducing system. Fifteen-second selections of piano-cello, slow speech and relatively fast speech programs with the various flutters introduced were recorded on a high quality tape recorder and spliced together in a manner which would allow comparisons of each type of flutter with one of the eighteen selected as standards. Listener preference rankings for the eighteen flutter variations were obtained for earphone listening, with and without reverberation introduced by means of an echo chamber. These rankings were then compared against measurements of peak-to-peak flutter and a flutter index measured on a flutter meter. The meter was modified to give an rms meter reading, and to include a weighting network based on the variation of flutter threshold with flutter rate for a 1,000 cps tone. Results in Table I indicate that there is no significant difference between ranking scores obtained with and without approximately two seconds of re-

TABLE I

LISTENER PREFERENCE RANKING SCORES FOR EIGHTEEN FLUTTER TYPES

Flutter variation no.	Peak-to-peak flutter (per cent)	Flutter index no.	Listener preference ranking scores: Music, No Reverb.	Music, With Reverb.	Fast speech, No Reverb.	Fast speech, With Reverb.	Slow speech, No Reverb.	Slow speech, With Reverb.
1	0.65	65	28	27	42.8	51.7	52.8	54.5
2	1.0	85	29	18	38.3	37.2	60	61.2
3	0.8	80	36	31	48.4	42.8	34.9	40.0
4	1.5	100	37	35	44.5	43.9	43.4	33.4
5	1.8	160	42	48	36.1	47.3	34.9	48.4
6	3.4	315	61	51	51.7	66.2	42.8	34.9
7	3.0	300	77	67	42.3	47.8	31.5	46.2
8	2.4	800	43	61	77.3	94.5	91.7	93.4
9	3.6	540	33	16	41.7	39.4	32.2	31.7
10	1.9	160	35	26	46.2	37.2	46.2	57.8
11	5.3	400	43.8	55.5	42.8	43.9	47.3	35.0
12	5.8	500	77.3	89	49.5	43.4	52.8	35.0
13	6.8	420	36.7	57	45	40.6	44	43
14	7.1	880	74	75	92	82	93	93
15	8.4	790	71	69	54	43	60	53
16	7.0	590	58	61	64	47	43	51
17	7.6	420	85	78	43	49	48	48
18	8.8	1000	83	83	95	92	93	90
Significant mean difference			17	17	17	18	16	18

verberation. This does not suggest that flutter is not more perceptible in a reverberant room. The data merely indicate that, if one particular fluttered program sounds worse than another particular fluttered program in a nonreverberant room, they will compare similarly in a reverberant room.

The data of Table I, when plotted against peak-to-peak flutter and the flutter index, will indicate that the flutter index as measured with the modified meter has a better relationship to listener preference than does the peak-to-peak flutter measurement. It would appear that the modifications to the flutter meter made in this experiment are a step in the right direction, but a better relationship was desired. The complex flutter waveshapes employed in this experiment led to difficulty when it was attempted to assess whether most of the improvement in flutter measurement could be attributed either to the weighting network or to the rms indicator.

The next experiment was conducted using only sinusoidal flutter. Every possible combination of flutter having peak amplitudes of ½, 1, 2, 3, 4 and 5 per cent and flutter rates of ½, 1, 2, 3, 4 and 5 cps were introduced in the following types of program:

1. A 1,000 cps tone;
3. A major "C" chord consisting of tones C_2 E_2 G_2 E_5 G_5 and C_6;
3. A piano selection, "Love Walked In";
4. A symphony orchestra;
5. The speech of a male news commentator.

A flutter generator was borrowed from Schecter at the Massachusetts Institute of Technology for the purpose.

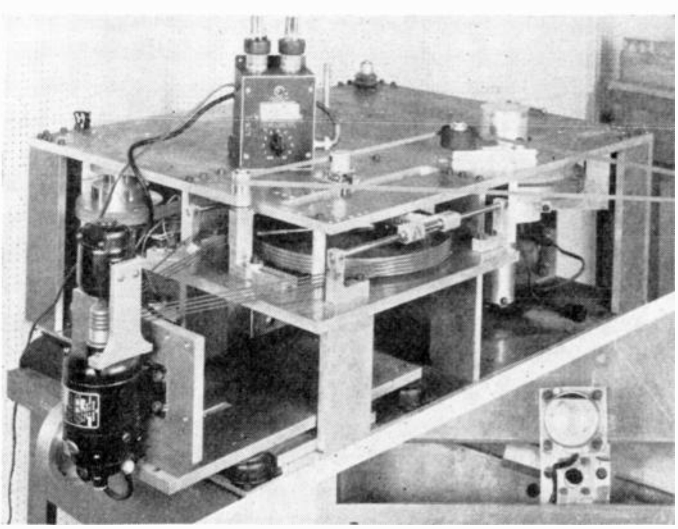


Fig. 1—Acoustics Laboratory, Massachusetts Institute of Technology. Flutter generator insert shows scotch yoke assembly.

This flutter generator shown in Fig. 1 was a magnetic-tape reproducing unit which introduced flutter by cyclically changing the position of the playback head in relation to the tape drive by means of a "Scotch Yoke." The amplitude of the head excursion was adjustable through a vernier adjustment, as shown in the insert on the figure.

Recordings of 10-second passages of these fluttered programs were then made and spliced together for a listener preference test. Listener preference ranking scores were obtained for loudspeaker listening in a relatively "soft" listening room. Unfortunately the flutter generator had an inherent 4-cps flutter of about 0.2 per cent rms amplitude. The frequency response of the over-all recording and reproducing process was limited only by the loudspeaker and included frequencies to 13,000 cps. Nonlinear distortion was held below 1.5 per cent harmonic content and signal-to-noise ratio was greater than 45 db.

The results showed that for the speech program, none of the judges was able to detect flutter except for the 4 and 5 cps 5 per cent peak-amplitude conditions. The listener preference ranking scores for the other four programs are shown in Table II, where it will be noted that the ranking scores for all the programs are similar. In

TABLE II

LISTENER PREFERENCE RANKING SCORES FOR TONE, CHORD, PIANO, AND ORCHESTRA

Flutter rate in cps	Flutter amp. per cent peak	Listener preference ranking score: 1,000 cps tone	Chord	Orchestra	Piano	Average
½	½	9	19	23	24	19
½	1	42	33	30	36	35
½	2	70	52	62	71	64
½	3	83	74	96	89	85
½	4	87	96	96	100	95
½	5	100	86	96	92	93
1	½	20	18	21	24	21
1	1	45	40	39	47	43
1	2	63	78	63	60	66
1	3	91	96	96	78	90
1	4	83	87	96	93	90
1	5	60	91	100	100	88
2	½	14	15	20	19	17
2	1	52	60	39	37	47
2	2	96	85	96	89	91
2	3	83	83	92	100	89
2	4	89	100	100	100	97
2	5	87	100	96	100	96
3	½	32	22	21	18	23
3	1	66	87	55	49	64
3	2	96	70	70	53	72
3	3	86	91	93	100	92
3	4	96	91	100	93	95
3	5	95	100	96	92	96
4	½	29	38	21	24	28
4	1	50	87	45	42	56
4	2	84	96	85	97	90
4	3	100	91	100	82	93
4	4	74	100	100	100	93
4	5	95	91	100	100	96
5	½	15	14	22	21	18
5	1	36	39	22	32	32
5	2	74	78	85	64	75
5	3	70	96	89	96	88
5	4	77	86	100	100	91
5	5	90	100	96	100	96
Significant mean difference		16	14	12	13	

Note: Lower listener preference ranking scores are associated with least degradation in quality.

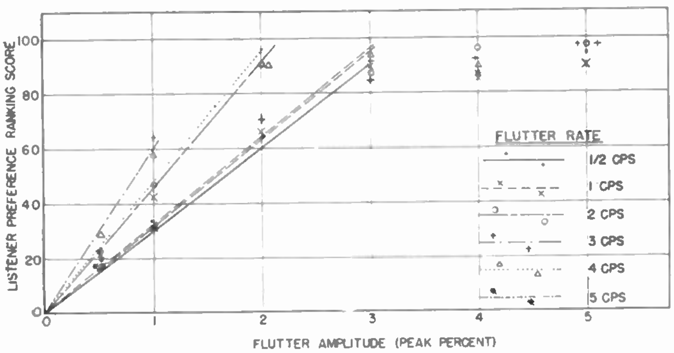


Fig. 2—Average listener preference ranking scores for tone, chord, piano, and orchestra vs flutter amplitude.

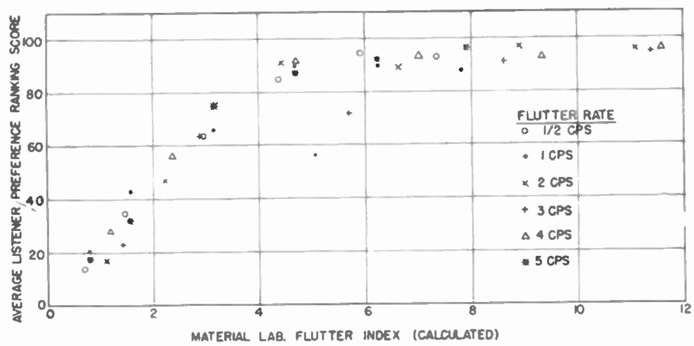


Fig. 3—Average listener preference ranking scores for tone, chord, piano, and orchestra vs Material Laboratory flutter index.

Fig. 2 the average ranking scores for all programs are plotted against flutter amplitude. The relatively smooth relationship between listener preference and flutter amplitude at each flutter rate indicates the validity of the listener preference ranking technique. For each flutter rate a straight line can be drawn through the data at the lower flutter amplitudes. The slope of these lines can be used to calculate the relative flutter-perceptibility thresholds at each flutter rate. The threshold values obtained in this manner, for an absolute threshold at 3 cps equivalent to the threshold reported by Schecter for a 1,000 cps tone, are shown in Table III.

TABLE III

FLUTTER PERCEPTIBILITY THRESHOLDS

Flutter rate cps	Flutter perceptibility threshold peak-to-peak per cent		
	Shower and Biddulph[5]	Schecter[7]	Material Laboratory
0.5	0.5 (by extrapolating)	1.4	0.68
1	0.33	0.60	0.64
2	0.28	0.36	0.45
3	0.29	0.35	0.35
4	0.45	0.38	0.43
5	0.70	0.42	0.64

Except for the flutter rate of $\frac{1}{2}$ cycle per second the relative thresholds for three experiments are similar.

A flutter index was calculated for each flutter variation by dividing the flutter amplitude by the appropriate Material Laboratory threshold flutter at each flutter rate.

In Fig. 3 the average listener preference ranking scores are plotted against the calculated flutter index, in order to show the improvement in relationship over that obtained for per cent peak flutter. Thus, for flutter rates from $\frac{1}{2}$ to 5 cps, a weighting curve dictated by a factor of the relative flutter thresholds could be employed in a realistic flutter-measuring instrument for program other than speech.

For flutter rates above 5 cps a subjective technique involving amplitude adjustment was employed, in order to find flutter perceptibility thresholds for the same five programs used in the previous experiment. Flutter was introduced by means of the flutter generator shown in Fig. 4, which was also a magnetic-tape reproducing unit, where relative motion between the playback head and tape drive was produced by mounting the head on a plexiglass tube cemented to the cone of a loudspeaker. The frequency range in these experiments was limited to a range of approximately 30 to 8,000 cps by the PDR-10 earphones used by the judges. Nonlinear distortion was again held to less than 1 per cent total harmonic content, and signal-to-noise level was maintained at better than 45 db. Inherent flutter was about 0.2 per cent rms at approximately 4 cps.

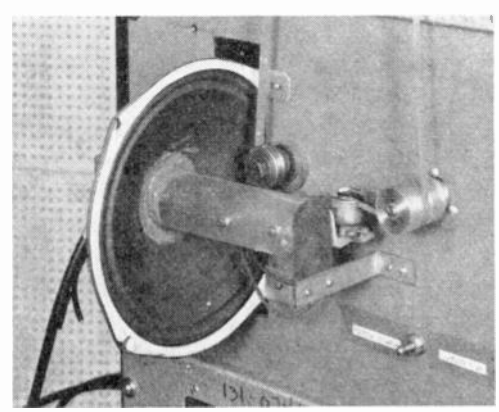


Fig. 4—Material Laboratory flutter generator.

The flutter perceptibility thresholds for each program are shown in Tables IV–VIII (pages 67–68). In spite of the variation between several trials for an individual judge and between judges, the average thresholds for a group of judges when plotted against flutter rate yield fairly smooth curves. This is shown (Figs. 5–9, pages 69–70) which do not vary appreciably (Fig. 10, page 70) among the five program selections employed. The index values found in the previous group of experi-

TABLE IV

FLUTTER PERCEPTIBILITY THRESHOLDS

1,000 cps tone, 90 db sensation level, diotic earphone listening

Judge	Trial	Per cent rms flutter threshold for various flutter rates 5 cps	7.5 cps	10 cps	15 cps	30 cps	50 cps	75 cps	100 cps
1	1	0.4	0.75	0.75	1.0	1.25	0.45	0.75	0.6
1	2	0.4	0.55	0.55	0.65	1.15	0.45	0.55	0.55
1	3	0.35	0.45	0.55	0.75	1.25	0.55	0.5	0.5
2	1	0.45	0.6	0.4	0.6	0.9	0.75	0.65	0.2
2	2	0.5	0.7	0.6	0.7	1.0	0.7	0.55	0.4
2	3	0.5	0.65	0.6	0.75	0.8	0.6	0.75	0.4
3	1	0.25	0.5	0.4	0.7	0.95	0.45	0.5	0.35
3	2	0.45	0.6	0.65	0.8	0.65	0.4	0.65	0.4
3	3	0.35	0.6	0.65	0.8	1.25	0.8	0.85	0.5
4	1	0.4	0.7	1.0	1.25	1.6	1.2	0.9	0.75
4	2	0.5	0.75	0.9	0.8	1.75	1.1	1.0	0.8
4	3	0.4	0.6	0.8	0.85	1.35	0.65	0.65	0.45
5	1	0.6	1.0	0.85	0.9	0.95	0.75	0.8	0.35
5	2	0.45	0.7	0.65	0.75	1.1	0.65	0.45	0.5
5	3	0.4	0.65	0.55	0.8	1.25	0.7	0.85	0.5
6	1	0.5	0.85	0.85	1.05	1.5	0.45	0.75	0.65
6	2	0.5	0.85	1.0	0.9	1.6	0.9	0.7	0.45
6	3	0.55	0.75	0.8	1.2	1.35	0.85	0.7	0.5
7	1	0.5	0.9	1.0	1.25	1.8	0.8	0.65	0.5
7	2	0.45	0.55	0.65	0.75	0.15	0.7	0.55	0.4
7	3	0.45	0.45	0.7	0.65	1.1	0.95	0.75	0.45
8	1	0.4	0.25	0.3	0.6	1.25	0.75	0.3	0.45
8	2	0.45	0.65	0.4	0.55	1.1	0.7	0.5	0.5
8	3	0.4	0.45	0.35	0.5	1.1	0.4	0.5	0.45
9	1	0.35	0.7	0.7	0.6	0.8	0.5	0.7	0.3
9	2	0.4	0.55	0.75	1.0	0.8	0.5	0.95	0.45
9	3	9.45	0.65	0.5	0.65	1.05	0.35	0.6	0.4
10	1	0.4	0.75	0.8	1.25	1.5	0.8	0.55	0.5
10	2	0.35	0.7	0.8	1.0	1.25	0.8	0.6	0.45
10	3	0.45	0.38	1.0	1.2	1.5	0.8	0.7	0.45
11	1	0.45	0.75	0.8	1.1	1.4	1.25	1.1	0.55
11	2	0.6	0.7	0.7	0.85	1.1	0.6	0.75	0.8
11	3	0.3	0.65	0.7	0.8	1.4	0.6	0.7	0.45
1	Av.	0.38	0.55	0.62	0.8	1.22	0.48	0.6	0.55
2	Av.	0.48	0.65	0.53	0.68	0.9	0.68	0.65	0.33
3	Av.	0.35	0.57	0.53	0.83	0.95	0.55	0.67	0.42
4	Av.	0.43	0.68	0.9	0.97	1.56	0.98	0.85	0.66
5	Av.	0.48	0.78	0.68	0.82	1.10	0.7	0.7	0.45
6	Av.	0.52	0.82	0.85	1.05	1.48	0.73	0.72	0.53
7	Av.	0.47	0.63	0.78	0.88	1.35	0.82	0.65	0.45
8	Av.	0.42	0.45	0.35	0.55	1.12	0.62	0.43	0.47
9	Av.	0.4	0.63	0.65	0.75	0.88	0.45	0.75	0.38
10	Av.	0.4	0.76	0.86	1.15	1.42	0.8	0.62	0.46
11	Av.	0.45	0.7	0.73	0.92	1.3	0.82	0.85	0.63
Group av.		0.43	0.66	0.68	0.85	1.21	0.69	0.68	0.48

TABLE V

FLUTTER PERCEPTIBILITY THRESHOLDS

Chord 90 db sound pressure level, diotic earphone listening

Judge	Trial	Per cent rms flutter for various flutter rates 5 cps	7.5 cps	10 cps	15 cps	30 cps	50 cps	75 cps	100 cps
1	1	0.55	0.8	0.8	0.95	0.75	1.0	0.65	—
1	2	0.55	0.65	0.8	0.9	0.45	1.2	1.4	1.0
1	3	0.7	0.8	0.95	1.25	0.85	0.85	1.15	—
2	1	0.65	0.6	0.65	0.75	0.3	0.8	1.0	1.15
2	2	0.6	0.7	0.6	1.0	0.3	1.2	1.1	1.2
2	3	0.6	0.65	0.9	1.05	0.65	0.7	1.2	1.4
3	1	0.6	0.75	0.95	1.05	1.2	0.7	1.1	1.2
3	2	0.55	0.8	0.6	0.7	0.5	0.9	1.5	—
3	3	0.8	1.0	1.1	1.45	1.5	1.25	1.95	1.15
4	1	0.7	0.8	1.0	1.45	1.55	1.5	2.25	—
4	2	0.95	1.1	1.0	1.15	1.25	1.5	2.1	—
4	3	0.85	1.0	0.85	1.15	0.95	1.25	1.85	—
5	1	0.55	0.7	1.0	1.15	1.5	1.25	0.9	0.65
5	2	0.7	0.6	0.8	0.85	0.5	0.95	1.15	—
5	3	0.45	0.55	0.6	1.0	0.5	0.75	1.0	—
6	1	0.7	0.75	1.0	1.15	0.75	1.7	2.1	1.45
6	2	0.5	0.85	0.95	1.15	1.25	1.1	1.15	1.8
6	3	0.85	0.7	1.25	1.45	1.0	1.3	1.4	1.55
7	1	0.45	0.55	0.65	0.8	0.5	0.9	0.5	—
7	2	0.6	0.7	0.8	0.75	0.5	0.9	1.4	1.35
7	3	0.6	0.7	0.7	1.0	0.8	1.05	1.25	1.0
8	1	0.3	0.45	0.6	0.8	0.9	0.75	0.9	1.0
8	2	0.2	0.5	0.6	0.85	0.7	0.6	1.0	1.2
8	3	0.3	0.55	0.65	0.65	0.6	0.8	0.75	1.2
9	1	0.6	0.75	0.95	1.3	1.4	1.05	0.85	0.65
9	2	0.75	0.75	1.75	0.05	0.35	0.8	0.95	0.8
9	3	0.65	0.75	0.8	0.95	0.6	1.5	1.1	—
10	1	0.4	0.75	0.75	1.1	1.2	1.0	0.9	0.85
10	2	0.55	0.7	0.6	1.0	1.25	0.95	1.05	1.2
10	3	0.6	1.0	1.1	1.15	1.25	0.9	1.0	0.95
11	1	0.8	0.85	0.7	1.05	0.75	0.9	0.85	0.9
11	2	0.9	1.0	1.0	1.0	0.7	1.5	1.55	1.75
11	3	0.5	0.85	0.85	1.2	1.0	0.9	0.85	0.9
1	Av.	0.6	0.75	0.85	1.03	0.68	1.02	1.07	1.0
2	Av.	0.62	0.65	0.72	0.93	0.42	0.90	1.1	1.3
3	Av.	0.65	0.85	0.88	1.07	1.07	0.95	1.52	1.18
4	Av.	0.83	0.97	0.95	1.25	1.25	1.42	2.06	—
5	Av.	0.62	0.62	0.8	1.0	0.83	0.98	1.02	0.65
6	Av.	0.68	0.77	1.07	1.25	1.0	1.37	1.55	1.6
7	Av.	0.55	0.65	0.72	0.85	0.6	0.95	1.05	1.18
8	Av.	0.27	0.50	0.62	0.77	0.73	0.72	0.87	1.1
9	Av.	0.67	0.75	0.83	1.1	0.78	1.1	0.97	0.78
10	Av.	0.52	0.82	0.82	1.08	1.23	0.95	0.98	1.0
11	Av.	0.73	0.9	0.85	1.03	0.82	1.1	1.08	1.18
Group av.		0.61	0.75	0.83	1.05	0.86	1.05	1.21	1.10

ments are used to complete the curve below 5 cps. In Fig. 5 the average thresholds obtained for tone are compared against those obtained by several other experimenters. All experimenters show approximately the same shaped curve with a minimum threshold at a flutter rate of approximately 2 cps. However, auditorium reverberation, as shown by the data obtained at Bell Telephone Laboratories and possibly that obtained by Lautenschlager,[4] greatly lowers the threshold. The higher thresholds obtained from the experiments conducted at the Material Laboratory may result from the inherent flutter in the flutter generator, or nonlinear distortion which tended to hide the true threshold. The break in the smooth curve at flutter rates of 50 and 75 cps was probably caused by an inherent 60 cps hum in the equipment which resulted in subjective beating.

The main purpose of the experiment has been fulfilled, however, by showing that at least for those types of

TABLE VI

FLUTTER PERCEPTIBILITY THRESHOLDS

Piano, 90 db average sound pressure level, diotic earphone listening

Judge	Trial	Per cent rms flutter for various flutter rates							
		5 cps	7.5 cps	10 cps	15 cps	30 cps	50 cps	75 cps	100 cps
1	1	0.4	0.7	0.6	1.1	0.7	0.9	1.9	1.35
1	2	0.4	0.75	0.85	0.8	0.8	0.8	0.85	0.85
1	3	0.5	0.45	0.45	0.6	0.45	0.4	0.4	1.1
2	1	0.6	0.7	0.75	0.9	0.8	0.9	0.8	0.9
2	2	0.45	0.65	0.8	0.9	1.0	0.7	0.6	0.85
2	3	0.45	0.55	0.65	0.85	1.0	0.85	0.8	1.0
3	1	0.7	0.85	0.95	1.0	1.2	1.05	1.25	1.15
3	2	0.6	0.75	0.95	1.25	0.6	1.45	1.2	1.1
3	3	0.6	0.6	0.9	1.1	0.6	0.7	0.8	1.05
4	1	0.5	0.75	0.95	1.0	1.5	1.35	1.55	1.15
4	2	0.65	0.45	0.75	0.8	0.6	0.9	1.25	—
4	3	0.65	0.65	0.85	1.05	0.9	1.1	1.15	1.15
5	1	0.55	0.8	0.75	0.75	0.5	0.8	1.1	1.2
5	2	0.45	0.5	0.4	0.55	0.65	0.45	1.25	1.25
5	3	0.6	0.6	0.65	0.55	0.85	0.7	1.15	1.2
6	1	0.65	0.7	0.95	0.8	1.7	1.3	2.1	1.15
6	2	0.55	0.5	0.75	0.8	1.25	0.8	0.9	1.05
6	3	0.45	0.65	0.8	0.8	1.5	0.75	0.9	1.25
7	1	0.4	0.35	0.5	0.75	0.8	0.75	1.15	1.15
7	2	0.3	0.5	0.55	0.65	0.75	0.9	0.95	1.15
7	3	0.25	0.35	0.35	0.65	0.5	0.75	0.85	1.15
8	1	0.55	0.55	0.55	0.7	0.35	0.3	0.65	0.8
8	2	0.35	0.45	0.5	1.0	0.45	0.5	0.75	0.95
8	3	0.3	0.55	0.5	0.55	1.05	0.55	0.85	0.8
9	1	0.5	0.85	0.8	1.0	1.25	1.0	0.9	0.9
9	2	0.5	0.5	0.55	1.0	0.6	0.9	1.1	—
9	3	0.5	0.3	0.45	0.5	0.45	0.3	0.35	1.2
10	1	0.5	0.55	0.7	0.9	0.8	0.45	0.4	0.75
10	2	0.85	0.8	0.7	1.0	0.75	0.65	0.85	1.05
10	3	0.55	0.25	0.4	0.55	0.45	0.4	0.45	0.65
11	1	0.55	0.75	0.95	0.95	0.6	1.15	0.95	1.35
11	2	0.45	0.3	0.4	0.35	0.4	0.35	0.85	0.8
11	3	0.5	0.6	0.5	0.5	0.55	0.65	0.75	1.05
1	Av.	0.43	0.63	0.63	0.83	0.65	0.70	1.05	1.10
2	Av.	0.5	0.63	0.73	0.88	0.93	0.82	0.73	0.92
3	Av.	0.63	0.73	0.93	1.12	0.80	1.07	1.08	1.10
4	Av.	0.60	0.62	0.85	0.95	1.0	1.10	1.30	1.15
5	Av.	0.53	0.63	0.60	0.62	0.67	0.65	1.17	1.2
6	Av.	0.55	0.62	0.83	0.80	1.48	0.95	1.30	1.15
7	Av.	0.32	0.40	0.47	0.68	0.68	0.80	0.98	1.15
8	Av.	0.40	0.52	0.52	0.75	0.62	0.45	0.75	0.85

TABLE VII

FLUTTER PERCEPTIBILITY THRESHOLDS

Orchestra, 90 db average sound pressure level, diotic earphone listening

Judge	Trial	Per cent rms flutter for various flutter rates							
		5 cps	7.5 cps	10 cps	15 cps	30 cps	50 cps	75 cps	100 cps
3	1	0.35	0.5	0.4	1.2	1.35	0.3	0.65	0.75
3	2	0.4	0.45	0.6	0.75	1.0	0.5	0.5	0.75
3	3	0.5	0.45	0.6	0.75	1.0	0.5	0.7	0.75
3	4	0.55	0.5	0.75	0.8	1.15	0.55	0.55	0.4
3	5	0.55	0.5	0.6	0.7	0.9	0.9	0.55	0.5
2	1	0.5	0.55	0.7	1.0	1.35	0.8	0.9	1.05
2	2	0.7	0.95	0.95	1.0	1.75	0.9	0.8	0.75
2	3	0.7	0.6	0.7	1.0	1.5	0.65	0.75	0.75
2	4	0.6	0.55	0.85	1.05	1.15	0.8	0.8	0.9
2	5	0.6	0.6	0.75	1.0	1.35	0.8	0.9	0.7
7	1	0.95	0.75	0.8	1.05	1.6	0.8	1.4	1.15
7	2	0.9	0.75	0.95	1.0	1.05	0.9	0.85	0.9
7	3	0.9	0.85	1.05	1.5	1.75	0.9	1.25	1.05
7	4	0.8	0.75	0.9	1.05	1.5	0.95	1.0	0.95
7	5	0.7	0.85	0.75	1.0	1.1	0.85	0.95	0.9
3	Av.	0.47	0.48	0.59	0.84	1.08	0.51	0.59	0.63
2	Av.	0.62	0.65	0.79	0.01	1.42	0.79	0.83	0.83
7	Av.	0.85	0.79	0.89	1.12	1.40	0.88	1.09	0.99
Group av.		0.65	0.64	0.76	0.99	1.30	0.73	0.84	0.82

TABLE VIII

FLUTTER PERCEPTIBILITY THRESHOLDS

Speech, 90 db average sound pressure level, diotic earphone listening

Judge	Trial	Per cent rms flutter for various flutter rates							
		5 cps	7.5 cps	10 cps	15 cps	30 cps	50 cps	75 cps	100 cps
3	1	0.8	0.7	0.95	0.85	1.5	1.2	1.85	1.5
3	2	0.6	0.55	0.95	1.15	0.95	1.45	1.25	1.4
3	3	0.7	0.5	0.95	0.9	1.0	1.0	1.25	1.15
3	4	0.8	0.6	0.85	1.0	1.0	0.75	1.25	1.45
3	5	0.6	0.7	0.75	0.85	0.95	1.3	1.4	1.0
7	1	0.8	0.8	0.95	0.9	1.05	0.95	1.4	1.1
7	2	0.8	0.85	0.65	1.0	1.1	0.9	1.25	1.3
7	3	0.7	0.6	0.85	1.15	1.05	0.85	1.4	1.2
7	4	0.8	0.85	0.75	1.15	1.25	1.25	0.9	1.0
7	5	1.0	1.25	1.2	1.3	1.15	0.9	1.5	1.1
2	1	0.75	0.55	0.8	1.0	1.25	1.25	1.4	1.1
2	2	0.65	0.5	0.65	0.8	1.05	0.9	1.25	1.0
2	3	0.65	0.55	0.65	0.75	0.9	0.7	1.15	1.05
2	4	0.65	0.65	0.65	0.9	1.25	0.7	0.85	0.8
2	5	0.65	0.5	0.6	0.9	1.25	0.45	0.85	1.1
3	Av.	0.7	0.61	0.89	0.95	1.08	1.14	1.4	1.3
7	Av.	0.82	0.87	0.88	1.1	1.12	0.97	1.29	1.14
2	Av.	0.67	0.55	0.67	0.87	1.14	0.80	1.10	1.01
Group av.		0.73	0.68	0.81	0.97	1.11	0.97	1.26	1.15

program used, the effect of flutter on music program is essentially the same as its effect on a 1,000 cps tone. The effect on speech is the same as tone for flutter rates above 5 cps. Below 5 cps, flutter as high as 4 per cent rms could not be detected. The difference between the thresholds obtained in these experiments, and those reported by other experimenters, led to the suspicion that the judges employed reacted differently to flutter than did those employed by previous experimenters. In order to settle this question, Schecter's experiment for 1,000 cps tone, 60 db sensation level, using an electronically warbled oscillator, were duplicated using three of the judges used in the previous experiments. The average thresholds obtained (see Fig. 11 on page 70) are in general agreement with Schecter's data.

A limited experiment was conducted to determine whether a realistic flutter meter should measure the peak or rms value of flutter excursion. In this experi-

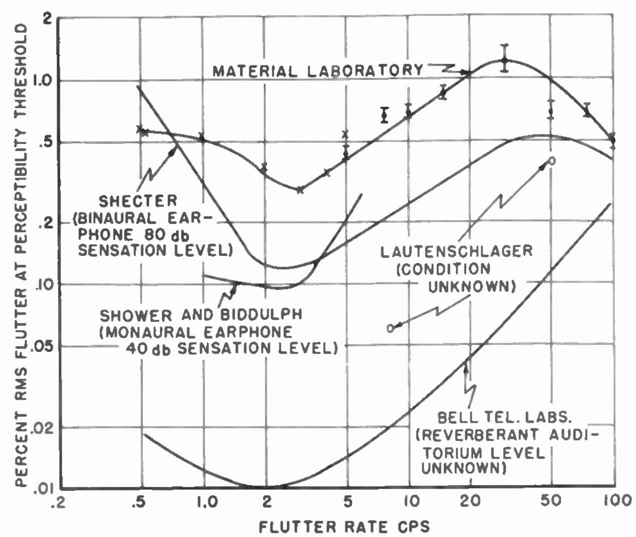


Fig. 5—Flutter perceptibility threshold for 1,000 cps tone vs flutter rate. Φ=average 33 observations, 11 judges, with 95 per cent confidence limit, 90 db sensation level, diotic earphone listening with inherent flutter of 4 cps—0.2 per cent and harmonic distortion of 1 per cent. x is determined previously from listener preference ranking scores.

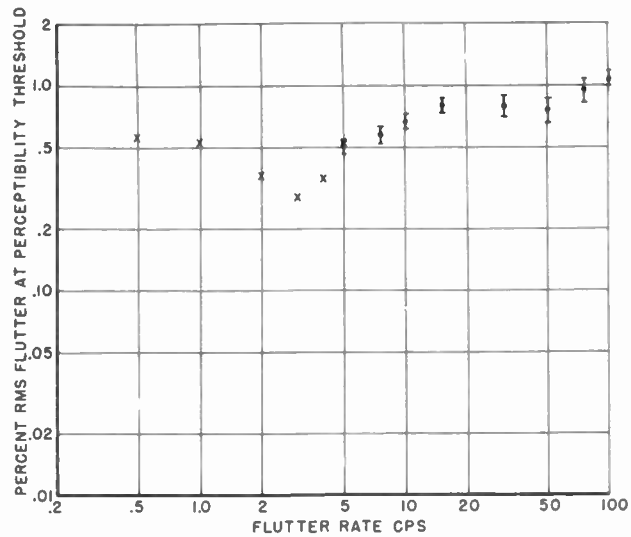


Fig. 7—Flutter perceptibility threshold for piano program vs flutter rate. Φ=average 33 observations, 11 judges, with 95 per cent confidence limits, 90 db level, diotic earphone listening with inherent flutter of 4 cps—0.2 per cent and harmonic distortion of 1 per cent. x is taken from listerner preference ranking scores.

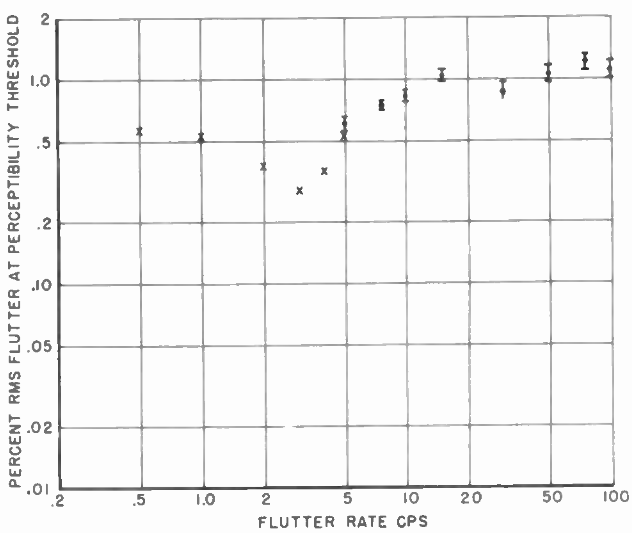


Fig. 6—Flutter perceptibility threshold for chord vs flutter rate. Φ=average 33 observations, 11 judges, with 95 per cent confidence limit, 90 db level, diotic earphone listening with inherent flutter of 4 cps—0.2 per cent and harmonic distortion of 1 per cent. x is taken from listener preference ranking scores.

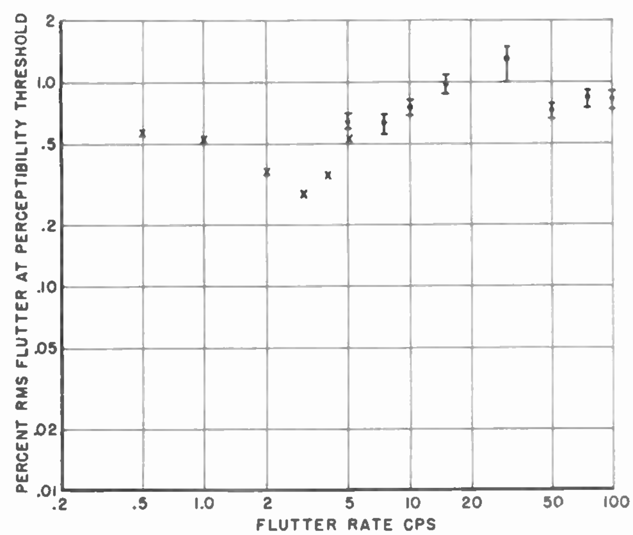


Fig. 8—Flutter perceptibility threshold for symphony orchestra vs flutter rate. Φ=Average 15 observations, 3 judges, with 95 per cent confidence limits, 90 db level, diotic earphone listening with inherent flutter of 4 cps—0.2 per cent and harmonic distortion of 1 per cent. x is taken from listener preference ranking scores.

ment, a 1,000 cps oscillator tone was electronically frequency-modulated with a 2 cps square wave of various degrees of unbalance. This resulted in a switching between two adjustable frequencies at a rate of 2 cps in such a manner that the duration of one tone could be adjusted from 10 to 250 milliseconds, while the other tone was present for the remainder of the 500 millisecond period. Perceptibility thresholds were obtained for various degrees of unbalance. The results are shown in Fig. 12 (next page) to agree well with the threshold curve calculated assuming that the relative thresholds were an inverse function of the rms content (reference level is equal to the average level) of the modulating wave. Therefore the flutter meter should be designed to have an rms indicating meter.

These experiments yield the following indications:

1. The subjective effect of flutter on the chord, piano, and orchestra selections is the same as that noted for a 1,000 cps tone.

2. The relative effect of flutter on program is not

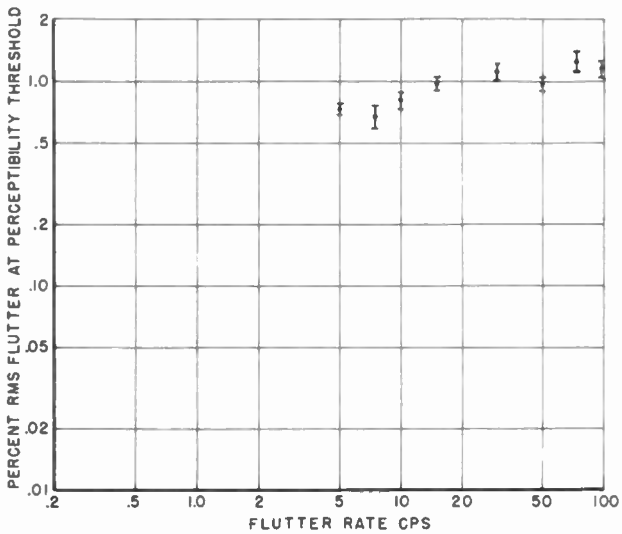


Fig. 9—Flutter perceptibility threshold for male speech vs flutter rate. Φ = average 15 observations, 3 judges, with 95 per cent confidence limits, 90 db level, diotic earphone listening with inherent flutter of 4 cps—0.2 per cent and harmonic distortion of 1 per cent.

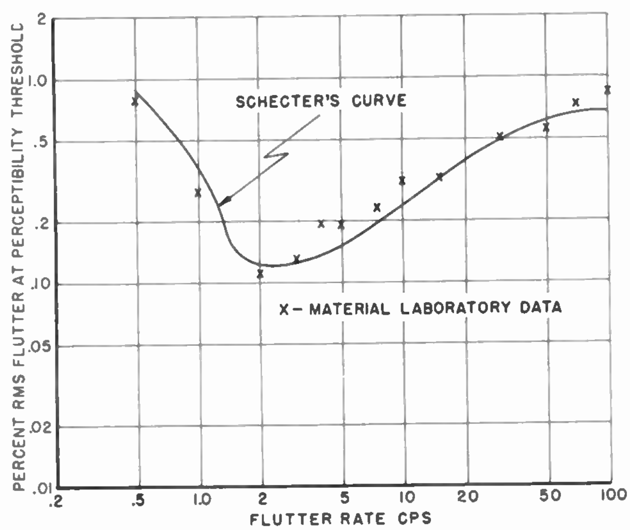


Fig. 11—Flutter perceptibility thresholds for 1,000 cps oscillator tones vs flutter rate. x = average of 11 observations, 3 judges, diotic earphone listening, 60 db sensation level.

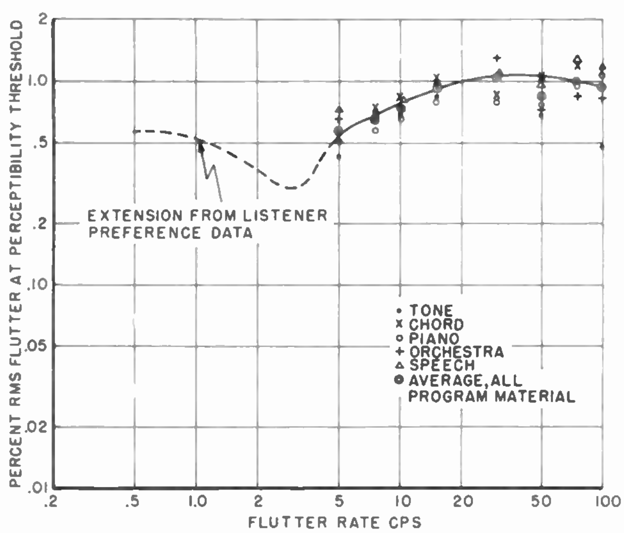


Fig. 10—Average flutter perceptibility thresholds for all program material vs flutter rate. 90 db level, diotic earphone listening, with inherent flutter of 4 cps—0.2 per cent and harmonic distortion of 1 per cent.

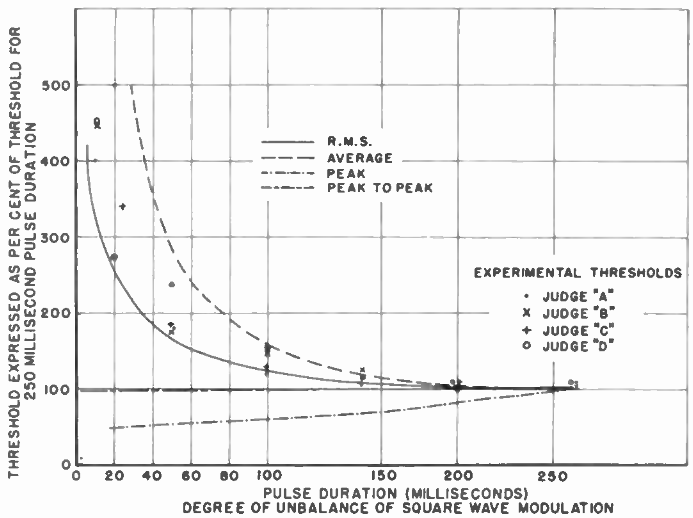


Fig. 12—Comparison of calculated and experimental perceptibility thresholds of unbalanced square-wave frequency modulation. Calculations on basis of threshold = inverse function of rms deviation from mean frequency; average deviation from mean frequency; peak deviation from mean frequency; and peak-to-peak frequency deviation. Experimental data was taken for 1,000 cps oscillator tone, 90 db sensation level, diotic earphone listening.

changed by introducing reverberation.

3. A flutter-rate weighting network dictated by flutter-perceptibility thresholds for a 1,000 cps tone, coupled with a well-damped rms indicating meter, can be used in conjunction with a flutter meter to provide a realistic measurement of flutter.

4. There is a possibility that all system distortion might be expressed in terms of an index, and combined to form a single quality index which would be a realistic indication of the system's merit.

In experiments now in progress at the Material Laboratory, listener preference rankings of music and speech program with 100 flutter variations introduced, and flutter index measurements of each flutter variation are being compared to show that a flutter index measurement is a realistic means of assessing the flutter content of equipment.

Acknowledgment

The loan of the Acoustics Laboratory M.I.T. Flutter Generator, and comments made by Dr. H. Schecter during the planning stages of the project, are sincerely appreciated.

Sound Measurements at Very High Levels*

ARNOLD PETERSON†

Summary—**The behavior of a number of microphones at high sound levels is described. Some of the problems encountered in making measurements at high sound levels are discussed.**

Introduction

As a result of the increasing use of jet and rocket propulsion, noise levels of 150 db and higher are becoming relatively common, and such levels are beyond the usual upper limit of commercially available sound level meters. This limit is usually 140 db above the standard pressure reference level of 0.0002 microbar; and it is set by the attenuation range in the instrument, but it is not possible to extend the range indefinitely by adding additional attenuation. The performance of the microphone is also important, since at these high noise levels the output voltage may no longer be linearly related to the sound pressure or the microphone may even be damaged. These limitations of a number of microphones used for measurement purposes will be described here. In addition, some of the problems that are peculiar to measurement at high levels will be discussed.

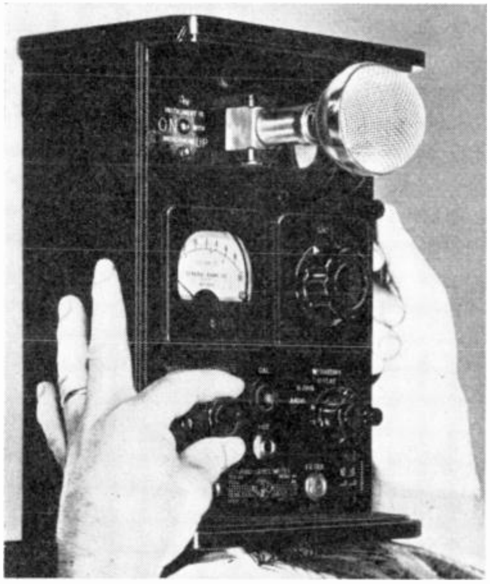


Fig. 1—Shure Model 98-98 Rochelle-Salt Microphone shown mounted on a sound-level meter.

Behavior of Microphones at High Levels

Rochelle-Salt Type

A Rochelle-salt crystal microphone, for example, the Shure Model 98–98, shown in Fig. 1, is regularly supplied with most sound-level meters. Although the Shure Model 98–98 is a sensitive microphone, tests on a number of samples of that model indicate that neither damage nor appreciable nonlinearity occurs at levels up to about 154 db. In particular, at that level the total harmonic distortion produced averaged less than 2 per cent. The distortion increases with level and the possibility of damage also increases. As a general rule, exposure to levels in the range of 160 to 170 db will result in damage, and some units may be damaged at somewhat lower levels.

Fig. 2—Altec Type 633 Dynamic Microphone shown mounted on a tripod with connecting cable to a microphone-to-grid transformer connected to the input of a sound-level meter.

Dynamic Type

The dynamic type of microphone is also used for sound-level measurements, particularly when the microphone must be separated from the instrument by an appreciable distance. The most generally used dynamic for measurement purposes is the Altec Type 633, shown in Fig. 2 (formerly made by Western Electric). The nonlinear distortion from a dynamic microphone is a function of both frequency and level. For the Type 633 the distortion at 300 cps is about 3 per cent at 140 db and 6 per cent at 150 db. Over the frequency range where the response is reasonably constant with constant pressure, from 50 to 8,000 cps. say, the pressure at which this distortion will occur is approximately proportional to the frequency. For example, at 75 cps, the 3 per cent distortion point should be reached at about 128 db *re* 0.0002 microbar.

The dynamic microphone can also be damaged at high sound levels, particularly when high levels are obtained at low frequencies. As a general rule, the maximum level should be limited to about 140 db, although

* Manuscript received by the IRE, January 31, 1955.
† General Radio Co., Cambridge, Mass.

it can usually be exposed to somewhat higher levels in the middle frequency range before damage occurs.

Condenser Type

The Western Electric Type 640-AA Condenser Microphone which is widely accepted as a standard microphone, can also be used at high levels. For example, measurements at 150 db showed a distortion of the order of 4 per cent, when used with a low-input-capacitance cathode-follower type of amplifier. Because it is an expensive microphone and because it is usually used as a laboratory reference microphone, however, it is not common to consider exposing this microphone to the possible rough handling that frequently accompanies field measurements.

The Kellogg Condenser Microphones, which are somewhat less sensitive than the W. E. 640-AA, should go to a corresponding higher level before serious distortion begins. Here, too, it is desirable to use a high-input-impedance low-distortion amplifier.

A number of Altec condenser microphones are available for sound-level measurements. A very high-input-impedance amplifier is essential for these microphones because of their low capacitance. Such a unit is shown with one of the condenser microphones in Fig. 3.

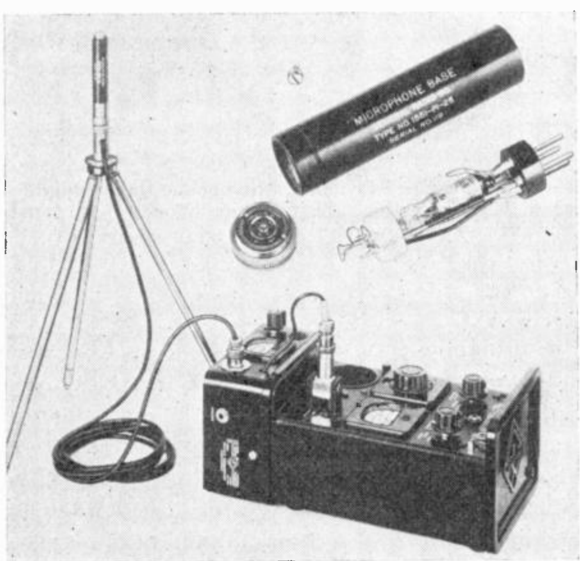


Fig. 3—Altec Type 21-BR Condenser Microphone shown as part of the General Radio Type 1551-P1 Condenser Microphone System, which includes a cathode-follower preamplifier and a battery-type power supply. The inset shows a close-up view of the microphone and the preamplifier disassembled.

Measurements on the Altec Type 21-BR-150 Condenser Microphone with the preamplifier shown in the figure indicate that the distortion will generally be less than 1 per cent at levels up to 135 db, increasing with level to about 10 per cent at 155 db. The Altec Type 21-BR-180 is about 15 db less sensitive than the Type 21-BR-150. Correspondingly, the level of distortion is below 1 per cent up to about 150 db and below 10 per cent up to about 170 db. For some applications even more distortion would be permissible for a measurement of a noise of high pressure level, and these microphones could be used since they are not usually damaged by high sound levels. The Altec Types 21-BR-200 and 21-BR-220 Condenser Microphones with even less sensitivity would be useable to still higher levels.

ADP Type

The Massa Model M-141 Standard Microphone uses ammonium-dihydrogen phosphate (ADP) crystals as the pressure-sensitive piezoelectric element. A stack of these crystals is mounted in a small, cylindrical, stainless steel housing, having the dimensions shown in the drawing of Fig. 4. This straightforward, rugged construction results in a microphone having very stable characteristics and that is rated for use to sound pressures of 200 db.

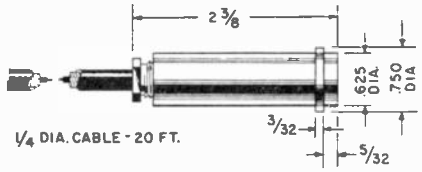


Fig. 4—Drawing of Massa Model M-141 Standard Microphone.

The front metal plate of the housing contacts the ADP crystals through a thermal isolation plate. When a sound pressure wave strikes this plate, a very small motion of the plate and a corresponding compression or expansion of the crystals results. Since the crystals are piezoelectric, the change in the dimensions of the crystals produces a voltage, without requiring any separate power supply. A 20-foot cable attached to the housing connects the microphone to the sound-level meter, and no high impedance preamplifier is needed.

The sensitivity of the Massa Model M-141B with its cable and including the input capacitance of the sound-level meter is 106 db below one volt for one microbar sound pressure. This sensitivity is about 50 db less than that of the regular Rochelle-salt microphones furnished with most sound-level meters.

The nominal range of the General Radio Type 1551-A Sound-Level Meter when using the Massa Model M-141B Microphone is then 74 to 190 db. The upper limit is set by the sound-level meter, and it is still within the rating of the microphone. To illustrate how high this upper limit is, we can compare it to atmospheric pressure. For example, if there were a periodic sinusoidal variation in pressure covering a peak-to-peak range from 0.1 atmospheric pressure to 1.9 atmospheres, the corresponding sound-pressure level would be 190 db. At such high levels, of course, the variation in pressure would normally be something different from a simple sine wave. We have not been able to produce a satisfactory sound pressure wave of sufficiently high level to test for distortion in the Massa Model M-141 Microphone.

As a consequence of the high-level use for which the microphone was designed, the minimum level at which it can be used is comparatively high. The basic limiting factor is the inherent noise level of the input circuit of the sound-level meter when the microphone, which has a capacitance with the attached cable of about 460 $\mu\mu$f, is connected across it. This noise level over the audio range in the Type 1551-A Sound-Level Meter is only about 4 microvolts; but, because of the low microphone sensitivity, the corresponding sound-pressure level is about 73 db. This circuit noise is negligible for levels of 80 db and higher.

The maximum safe operating temperature of the Massa M-141B Microphone is about 75 degrees C. (167 degrees F.), a limit set by the cements used in the construction of the microphone. The unit is sufficiently well sealed so that exposure to normal humidity conditions will not affect the operation, but it is recommended that prolonged exposure to relative humidities in excess of 94 per cent be avoided.

Useful Features of High-Level Microphones

These results of distortion tests show that the high-level microphones, such as the Massa Model 141 and the Altec Type 21-BR-180, are well suited for measuring sound levels in engine test cells and near high-powered airplanes and for measuring the blast pressures of gunfire and other explosions. But for most noise levels encountered in industry, the ordinary measurement microphones are satisfactory as far as distortion is concerned. Nevertheless, even for measuring noise levels appreciably below 150 db, these high-level microphones can be useful because of a number of features that they have. The small size of these microphones makes it possible to bring the microphone near to noisy parts of a machine, a procedure which is often invaluable in tracking down sources of noise. The small size also helps in exploring sound fields with a minimum of disturbance from the microphone. In addition, the Massa microphone, because of its exceptionally rigid construction, is particularly suited to direct measurements of pressure variations in pressure chambers by mounting the microphone to form part of the chamber wall.

Furthermore, these microphones have such good frequency-response characteristics that they are particularly useful for measuring noises characterized by strong high-frequency components, as, for example, turbine-blade noise, jet-engine noise, and textile-machinery noise.

A typical frequency response for the Altec Type 21-BR-180 Condenser Microphone is shown in Fig. 5. The response to 20 kc is seen to be very good, particularly for parallel incidence, that is, when the sound grazes the face of the microphone. In order to reduce the effects of unwanted disturbances in the sound field, it is often recommended that perpendicular incidence be used; and then the upper response curve of Fig. 5 applies. If it is desired to reduce the peak in response that occurs, the "C"-weighting network of the G. R. Type 1551-A Sound-Level Meter can be used, with a result for perpendicular incidence that is essentially the same as the curve shown for parallel incidence.

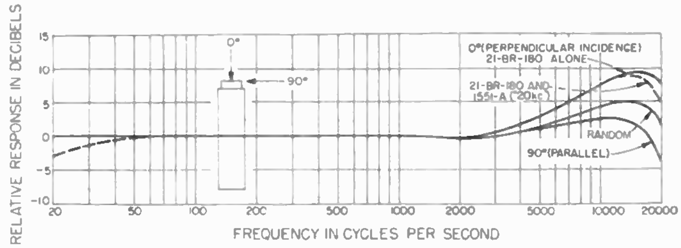


Fig. 5—Typical frequency response of the Altec Type 21-BR-180 Condenser Microphone for a plane wave of sound with the direction of propagation perpendicular (upper curve) or parallel (lower curve) to the face of the microphone. The curve labeled "Random" is an average response assuming all directions of propagation are equally likely.

The Type 21-BR-180 Microphone is not seriously affected by comparatively high temperatures, but the battery-operated preamplifier of the G. R. Type 1551-P1 Condenser Microphone System is limited by its vacuum tube to operation at temperatures below about 100 degrees C. (212 degrees F.). Also, because of the high electrical impedances involved, this microphone system is not recommended for operation at high humidities, although no permanent damage results from exposure to high humidity. High humidity can cause enough electrical leakage so that the required bias on the microphone cannot be obtained. This condition will be obvious in operation, since one will not be able to set the indicating meter on the power supply of the Type 1551-P1 Condenser Microphone System to the correct level. Usually the effects of humidity can be reduced to an adequate degree by heating the microphone well above the ambient temperature. If an ac supply is available, the Altec preamplifier[1] can be used; and this preamplifier operates at a sufficiently high temperature that the effects of humidity are avoided.

The frequency response of one of the Massa Model M-141B microphones at perpendicular sound incidence was measured, and the smoothed result is shown in Fig. 6 (next page) as the curve labelled "M-141B alone."

The relative levels shown beyond 15 kc should be regarded as approximate, since they are based on an extrapolated characteristic of a W. E. Type 640-AA Microphone, which had been calibrated at the National Bureau of Standards to 15 kc.

The first resonance of this microphone is at 34 kc, well beyond the audio range. The resonant rise in response is large, because it is impractical to dampen this rigid structure to any great extent. But some of this rise in response is a diffraction effect of the sound striking the rigid cylinder. This effect is less important for parallel incidence where the response is essentially uni-

[1] J. K. Hilliard and J. J. Noble, "The 'lipstick' condenser microphone system," *Trans. IRE*, vol. AU-2, pp. 168–175; November-December, 1954.

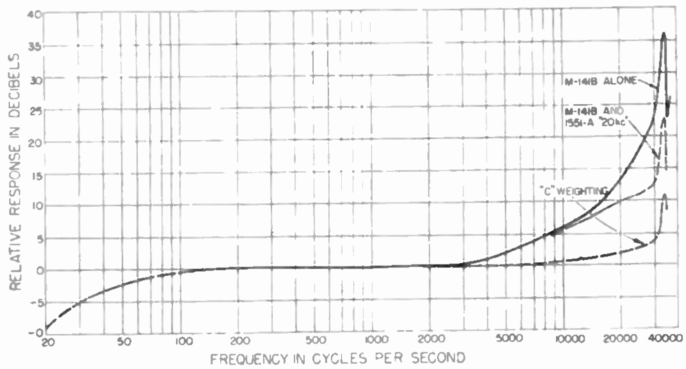


Fig. 6—Frequency response of a Massa Model M-141B Standard Microphone for a plane wave of sound propagated perpendicular to the face of the microphone (upper curve). The effect of the electrical response of the G.R. Type 1551-A Sound-Level Meter is also shown for the two different response characteristics labeled "20 kc" and "'C' Weighting."

form ("flat") at high frequencies up to 20 kc as shown in Fig. 7.

If the advantages of using the microphone with perpendicular incidence are desired, the "C"-weighting network of the Type 1551-A Sound-Level Meter can be used; and the resultant response, shown in Fig. 6, is essentially uniform to about 30 kc. Such a uniform response is valuable for many measurements. It practically eliminates the need for correcting the data for the effects of variations in frequency response of the microphone. For those measurements where such correction is impossible, a response as uniform as that shown will yield as reliable data as possible. It can also simplify the application of automatic recording systems for obtaining sound-pressure levels as a function of frequency in the testing of loud-speakers and other sound sources. The uniform response to 30 kc indicates that the transient response to blast or shock waves in the audio range will be exceptionally good.

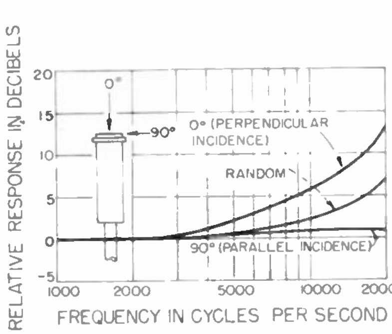


Fig. 7—Frequency response of a Massa Model M-141B Standard Microphone for a plane wave of sound with the direction of propagation perpendicular (upper curve) or parallel (lower curve) to the face of the microphone. The curve labeled "Random" is an averaged response assuming all directions of propagation are equally likely.

The capacitance of the Massa microphone and its associated cable is not so large as that of the Shure Model 98–98 Microphone regularly supplied with the sould-level meter, and the low-frequency response of the combination is somewhat poorer as a result of this lower capacitance, as shown in Fig. 6.

Additional Attenuation

The results of tests given above show that a number of common measurement microphones are satisfactory for use somewhat beyond 140 db. Above that level, however, it is usually necessary to add additional attenuation to the usual sound-level meter. This attenuation can be added between the microphone and the sound-level meter or in the instrument itself.

If the attenuation is added between the microphone and the instrument, care must be taken to avoid stray pickup from electric and magnetic fields, to avoid affecting the frequency response, and, in the case of the Rochelle-salt microphone, to avoid increasing the change in sensitivity with temperature.

Putting the attenuator in a completely closed metallic box with connections by shielded leads is usually adequate for avoiding stray pickup.

The important point with regard to frequency response is that most microphones do not have a resistive source impedance, and the attenuator design must be based on the actual source impedance of the microphone. For example, the Shure Model 98-98 Rochelle-salt microphone has a capacitive source impedance that varies with temperature from about 2,400 to about 600 $\mu\mu$f. An attenuator with a low resistive impedance would be unsuitable, because the over-all attenuation would be much greater at low frequencies than at high.

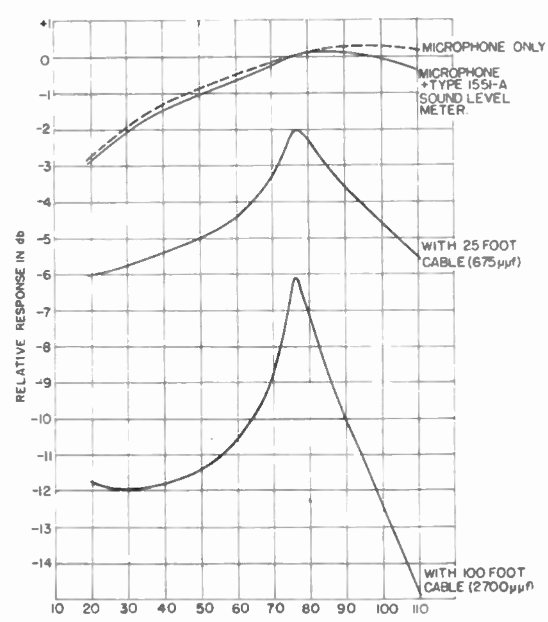


Fig. 8—The relative sensitivity of the Shure Model 98-98 Rochelle-Salt Microphone when connected directly to a sound-level meter, with a 25-foot cable intervening, or with a 100-foot cable. The relative sensitivity is shown as a function of temperature.

For the example of the Rochelle-salt microphone, adding capacitance in parallel with the microphone, such as occurs when a microphone cable is used, does give an attenuation that is independent of frequency. But, because of the large change in microphone capacitance with temperature, the sensitivity then also changes markedly with temperature. The effect is shown in Fig. 8.

One way of avoiding much of this undesirable behavior is to insert at the microphone a fixed series capacitor of about 100 $\mu\mu$f before the cable is connected. The cable will then produce a larger attenuation, but the attenuation will not be so dependent on temperature, because the small series capacitor limits the net variation in apparent source impedance to a moderate value. Or, in other words, the series capacitor and the cable form a comparatively high impedance attenuator.

Additional attenuation can usually be added in the instrument itself. If the level to be measured is only a few decibels above the nominal range, the normal gain control sometimes may be adjusted to a low value so that a useable reading can be obtained at these high levels.

A special attenuator pad is available for the G.R. Type 1551-A Sound-Level Meter that shifts the level up 20 decibels. This pad, shown in Fig. 9, is conveniently inserted between two amplifier stages. Not only does this pad extend the range of levels that can be measured, but it also helps to solve the microphonic problem (see below).

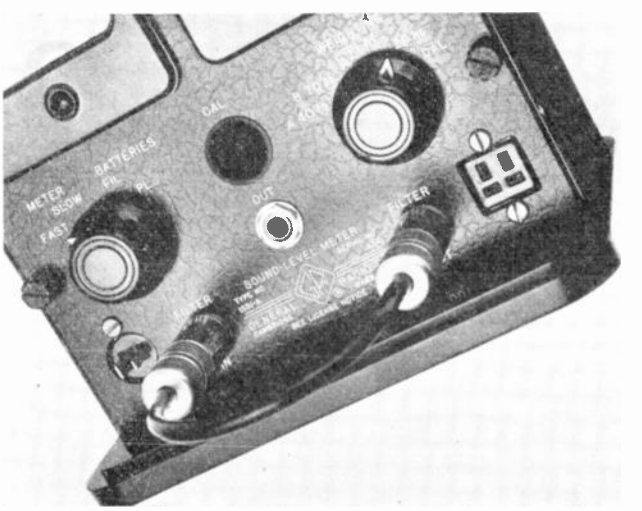


Fig. 9—As shown here, the Type 1551-P11 20-db pad plugs into jacks on the sound-level meter panel.

Special Precautions

When microphones of low sensitivity, such as the Massa Model M-141B and the Altec Type 21-BR-180, are used, the microphone should be mounted carefully to insure that a correct measurement is being made. In addition, when sounds of high level are being measured, any associated vacuum-tube apparatus should be kept out of the high-level sound field, if possible.

Method of Supporting Microphone

Mechanical vibration of a microphone will produce an output signal. It is necessary, then, to support a microphone so that the signal from existing vibrations is appreciably less than the desired signal from airborne sound. For microphones having high sensitivity to sound, this requirement is usually met even with fairly rigid mounting methods, but low-sensitivity microphones must be mounted carefully. These requirements can often be met by suspending the microphone by its cable, and the cable can be held in place by cords.

One way of detecting vibration errors is to note the effects on the measured levels of different microphone mounting methods.

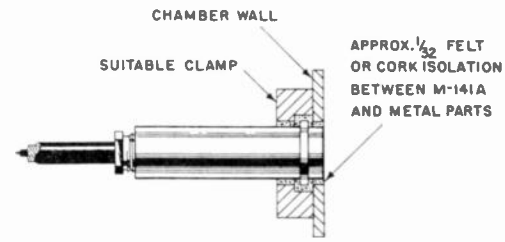


Fig. 10—Diagram of recommended installation of the Massa Model M-141B Standard Microphone in the wall of a chamber.

In general, very resilient supports should be used, with a low natural period of vibration, say below 10 cps. Looping the cable before the end support may also improve conditions, since it is particularly important to keep axial vibrations of the microphone to a very low level.

The small flange on the Massa Model M-141B Microphone simplifies the mounting of the microphone in the solid wall of a chamber. But here the isolation of the microphone from the solid-borne vibrations is particularly important, and an assembly such as that shown in Figs. 10 and 11 is recommended by the Massa Laboratories for such an application.

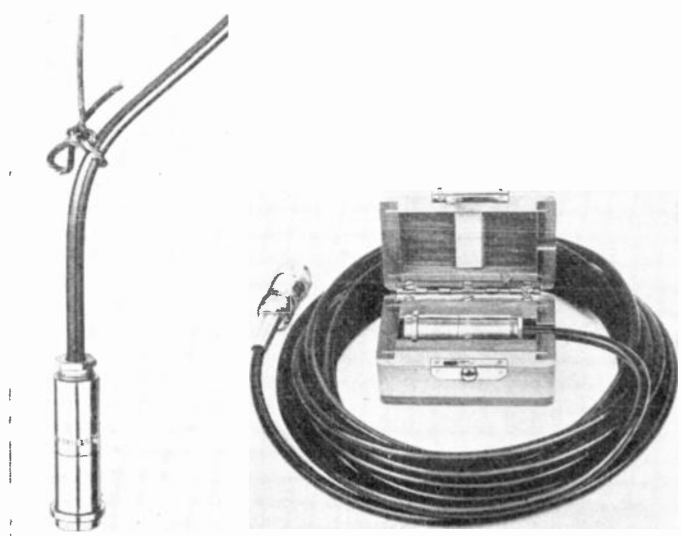


Fig. 11—(Left) Microphone suspended by cord to avoid the effect of mechanical vibrations. (Right) View of the Massa M-141B Microphone in its storage box, with 20-foot cable and connector.

Microphonics

A vacuum tube exposed to a high-level sound field will vibrate, producing an electrical output, called microphonics, that interferes with the desired signal. With the standard wide-range sound-level meter, the interference is usually unimportant until the sound level is well above 100 db. Microphonic effects can be detected by disconnecting the microphone and by then observing the reading on the instruments when exposed to the sound to be measured. When this test is made, the input terminals of the preamplifier should be shielded to prevent stray electrical pickup.

At high levels, the G.R. Type 1551-P11 20-db Pad can be used with the G.R. Type 1551-A Sound-Level Meter to reduce the effect of microphonics. This pad shifts the operating signal levels in the instrument so that the sensitive, early stages must operate with a higher signal, and the relative microphonic level is correspondingly lower.

In order to get adequate information for dealing with a noise problem, it is almost always desirable to use auxiliary equipment, for example, an octave-band analyzer, a tape recorder, or a cathode-ray oscillograph, with the sound-level meter. These instruments can also give incorrect indications because of microphonics, and they should be checked for this possibility. Such a check would usually consist of disconnecting the input to the instrument and observing its behavior when exposed to the noise under the expected test conditions.

The best solution to the microphonic problem is to keep the instruments using vacuum tubes away from vibrating surfaces and intense sound fields. But, of course, the microphone must be in the field to be measured, so that a cable must connect the microphone and the measuring instruments. The Massa Model M-141B Microphone includes a cable of about 20 feet, but a longer cable may be essential for adequately separating the microphone and the sound-level meter. Then a G.R. Type 759-P30 Extension Cable could be used, and 7 db should be added to the indicated level because of the cable capacitance.

When microphonics are troublesome, the most sensitive microphone that will handle the required level will usually give the most favorable signal conditions. But many other factors, for example, frequency response, size, temperature effects, are usually of so great importance that the very sensitive microphones cannot be used. The instruments should then be acoustically isolated from the sound field.

ACKNOWLEDGMENT

We are indebted to the Massa Laboratories, Inc., the Altec-Lansing Corporation, and Shure Brothers, Inc., for furnishing much of the general information on which this article is based, to G. Kamperman of Bolt Beranek and Newman for some of the data on distortion, and to E. Gross of the General Radio Company for some of the other measured data.

CORRECTION

T. J. Schultz, author of the paper "Triode Cathode-Followers for Impedance Matching to Transformers and Filters," which appeared in the TRANSACTIONS OF THE IRE, the March–April issue, 1955, has drawn attention to the following omissions in the text.

In Figs. 7–17 (pages 31–36), the titles which identify the tube types were inadvertently deleted. It is suggested that readers who anticipate use of these curves pencil the appropriate identification onto the figures. They should read as follows: Fig. 7—1/2-12AU7 "B" (except for GAIN, which is for 1/2-12AX7 "B"), Fig. 8—1/2-12AT7 "A," Fig. 9—1/2-12AT7 "B," Fig. 10—1/2-12AU7 "A," Fig. 11—1/2-12AX7 "B" (except for GAIN, which is for 1/2-12AU7 "B," Fig. 12—1/2-6BL7 "A," Fig. 13—1/2-6BL7 "B," Fig. 14—6AQ5 "A," Fig. 15—6AQ5 "B," Fig. 16—12AU7 "A" (both halves in parallel), Fig. 17—12AU7 "B" (both halves in parallel).

Note that because of the interchange of the GAIN data of Figs. 7 and 11, some of the data required in Example (1) of the text will be found in Fig. 7.

Electronic Organ Tone Radiation*

D. W. MARTIN†

Summary—**The principles of design for electronic organ tone chambers are outlined. The differences between the design goals for loudspeaker enclosures for organs and for other purposes are explained in fundamental terms. The construction of new organ tone cabinets for indirect radiation is described in detail. A few organ installation examples are given.**

Introduction

An AUDIO SYSTEM engineer generally plans to reproduce electrically whatever program material is provided or selected, with as uniform and smooth a response characteristic as possible, and expects that the loudspeaker engineer will not let him down at the terminal end of the system. A transducer engineer engaged in loudspeaker development is naturally concerned with uniformity and smoothness of response-frequency characteristic, on and near the axis of a loudspeaker, within the widest audio-frequency range which can be achieved. He measures the performance of the product of his efforts in an anechoic environment. It is well known in acoustics, however, that the typical living room environment for a sound reproduction system imposes severe stress upon the uniformity and smoothness of audio-system steady-state response. In spite of this, through the assistance of binaural listening and the non-steady nature of speech and music, the audio enthusiast likes the end product very much. In fact, the natural acoustical treatment (upholstery, drapes, wall-to-wall carpeting) which helps to approximate roughly the anechoic condition in which the loudspeaker yields the "ideal" performance characteristics, can be increased to a point where the reproduction is appreciated less rather than more, an apparent paradox.

In the field of electronic organ tone production the apparent engineering paradox seems even greater (but only superficially), because the loudspeaker installation techniques needed to obtain the desired acoustical effects depart even farther from free-field conditions. One of the inherent economic advantages of an electronic organ over a pipe organ is that the tones can all be radiated from a single source. Even a small pipe organ contains a relatively large number of individual sound sources, which together require an inordinate amount of building space, especially for domestic installations. Yet only part of the population is sufficiently conditioned by years of listening to radios to be completely satisfied with organ tone which comes from a small source, and which lacks reverberation. Reverberation is traditional in organ music and enhances one's appreciation.[1]

A substantial percentage of organ listeners, and probably a majority of the organists, prefer to hear organ music radiating in a somewhat diffuse manner from a source of large actual or apparent size, with greater actual or apparent reverberation than is typical of modern acoustically absorptive listening spaces. In order to provide these tonal effects for electronic organs while retaining the inherent economic advantages, rather unconventional loudspeaker installation techniques are used. One of the purposes of this paper is to describe these techniques.

In a previous paper[2] the author differentiated between the goals of research in sound *re*production ("high-fidelity") and music *production* systems. A second purpose of this paper is to illustrate how the same high-quality electroacoustic transducers used in the finest sound reproduction systems, through novel ***variations*** of the same well-established acoustical principles which serve as a foundation for high fidelity, have been adapted to the specific needs of organ tone production.

Apparent Physical Size and Reverberation

Reviewing fundamentals[3,4] with the help of Fig. 1(a) (next page), sound radiating at various angles from a loudspeaker in a room may be reflected many times by the walls, so that many image sources contribute to the sound received at a point distant from the loudspeaker. If the walls of the room are highly absorptive acoustically, the number of reflections before the reflected sound becomes inaudible is very small. However if the walls are highly reflective, and a window is provided in one wall [as in Fig. 1(a)], most of the sound will ultimately leave the room through the window, radiating into outside space in all directions. Whether a listener is located inside the room or in the outside space, the sound level produced by a loudspeaker tone keyed on and off will vary in time as in Fig. (1)b, depending upon the reflectivity of the room walls. The square curve represents a very high sound absorptivity. Moderate reflectivity permits later reflected waves to add to the direct wave, raising the level gradually to a somewhat

* Manuscript received ty the IRE, March 10, 1955.
† The Baldwin Piano Company, Cincinnati 2, Ohio.

[1] D. W. Martin, "The Enhancement of Music by Reverberation," 1954 IRE Convention Record, Part 6, "Audio and Ultrasonics," pp. 4–7.

[2] D. W. Martin, "High fidelity in musical tone production?", *Trans.*, vol. PGAU-2, pp. 102–104; July–August, 1954.

[3] F. R. Watson, "Acoustics of Buildings," John Wiley and Sons, New York; 1941.

[4] V. O. Knudsen and C. M. Harris, "Acoustical Designing in Architecture," John Wiley and Sons, New York; 1950.

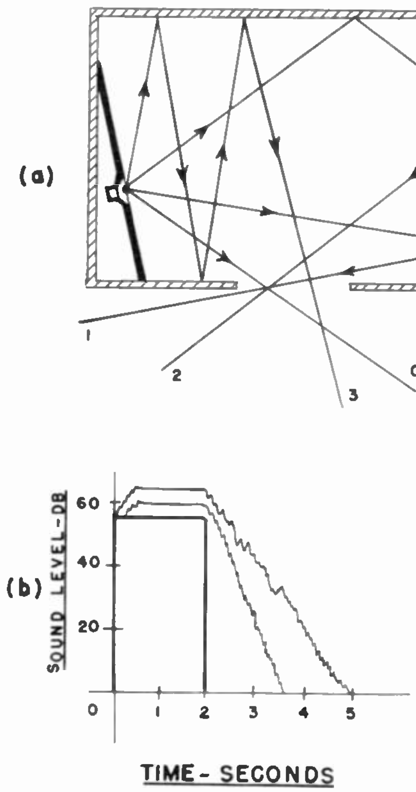


Fig. 1—(a) Multiple reflection of sound in a ported reverberant room, (b) Growth and decay of sound level in anechoic and reverberant environments.

higher steady-state value, as in the middle curve. Keying the tone off leaves many reflected tones yet to be heard from and finally to be absorbed locally or radiated away into space, yielding a reverberation time of about 1.5 seconds. The top curve is for higher reflectivity and a reverberation time of three seconds.

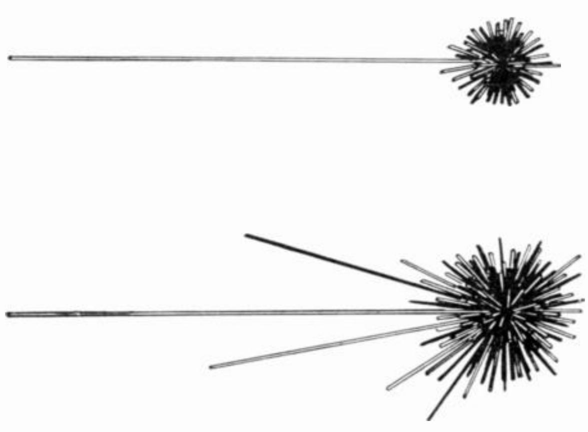


Fig. 2—Comparison of direct and reflected sound energy increments from various directions, in acoustically absorptive (upper) and reverberant (lower) rooms.

Fig. 2 will assist in visualizing more easily the manner in which direct and generally reflected sound approaches a listener in rooms having widely different acoustics. This graphical manner of showing sound-energy contributions to the listener from different increments of solid angle, integrated over time, was used recently by Meyer.[5] In both cases the sound source is at the left in the direction of the long spike and the listener is at the center of the "hedgehog." The length of each shorter spike represents the average energy of sound arriving from that particular direction. The upper example is for a room having walls with considerable absorption. In a truly anechoic room there would be only the one long spike representing the direct sound. A highly directional sound source would reduce the length of the side spikes relative to the direct spike, although it might make the rear spike larger unless the rear wall were highly absorptive. Under these conditions the binaural listener has little difficulty in locating the sound source, and the apparent source size is limited to the actual source size.

In a more reverberant room the lower example of Fig. 2 is illustrative. Some contributions from single reflections on highly reflective surfaces may be nearly as great as the direct sound. The total of all general reflections will exceed the contribution of the direct sound wave except at the instant the direct wave arrives. The less directional the sound source is made the greater the contributions from the sides, and the binaural listener gains the aural impression of a sound source of large size.

A general idea of the directivity of sound sources of different size and type can be gained from Fig. 3. For the background of this data, and for additional references, a textbook may be consulted.[6] In general, sound sources are increasingly directional, the higher the frequency, unless special methods are used to prevent this effect.

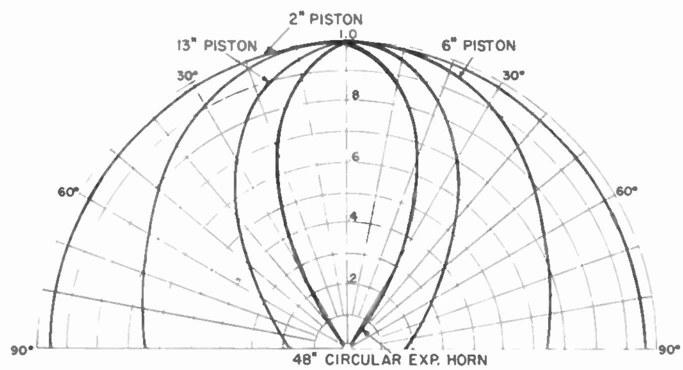


Fig. 3—Comparison of directional characteristics at 1,000 cps.

The directivity patterns of Fig. 3 are for a frequency of approximately 1,000 cps. The three outer curves are for circular pistons of different diameter in which the entire piston area is vibrating in phase. Although the latter condition holds only approximately for loudspeaker cones, a typical directivity curve at 1,000 cps for a fifteen-inch loudspeaker on a large flat baffle will resemble quite closely the curve marked "13-inch piston." If greater directivity were desired, or if greater efficiency

[5] Erwin Meyer, "Definition and diffusion in rooms," *Jour. Acous. Soc. Amer.*, vol. 26, p. 630; September, 1954.

[6] H. F. Olson, "Elements of Acoustical Engineering," D. Van Nostrand Co., New York, Ch. II, VI, and VII; 1947.

were sought irrespective of directivity, an exponential horn could be used. The highly directional inside curve of Fig. 3 is for such a horn, with conventional circular cross section and a rather large diameter of 48 inches. If instead a broad directivity pattern is the goal, it can be obtained from smaller sources, such as a six- or even a two-inch piston.

Four different methods have been used successfully, both individually and in combination, in order to create actual or apparent reverberation, and actual or apparent physical size for electronic organs heard in nonreverberant environment:

1. Installing the sound source in a moderately reverberant tone chamber from which the sound radiates into the auditorium;

2. Orienting the sound source in a manner which increases the ratio of reflected-to-direct sound;

3. Using multiple sound sources separated in space;

4. Adding artificial reverberation by accessory means. Each of these methods will be described.

Tone Chambers

The tone chamber method has probably been the most successful method which has been widely used. The majority of those cases in which pipe organs have been replaced by electronic organs have space already available for this purpose after the old instrument has been removed. In general, it is possible to get more reverberation from a chamber when loudspeakers are used than could be achieved when the pipes were installed. Originally the chamber was so crowded by pipes that the acoustical absorption within the chamber reduced the reverberation, even if the walls of the chamber were rigid and smooth. When all of the original contents have been removed from the chamber, and loudspeakers are installed on rigid reflective baffles, a very beneficial reverberation-chamber effect is achieved.

Tone chambers are generally recommended for installations of electronic organs in new buildings in which tone chambers are feasible from an architectural standpoint. The space required to build an effective tone chamber for an electronic organ is considerably less than that required to enclose the pipes of a pipe organ of equivalent power. Consequently the economic advantage in the installation of loudspeakers is only partially compromised by the addition of a tone chamber in a new building.

The optimum reverberation period for organ music is not a matter of general agreement. Some organists appreciate periods as high as five seconds. Certainly a loss in definition results. Baldwin practice is to recommend a minimum of two seconds for the tone chamber plus auditorium. When the auditorium reverberation period is under one second, the chamber period then needs to be nearly two seconds. This gives a good compromise with the reverberation requirements for speech or public address equipment.

Tone chambers have other advantages besides reverberation, and these advantages are sometimes of consequence in auditoriums where the reverberation and diffusion are adequate without a tone chamber. A tone chamber is more likely to be complementary to the church architecture than exposed tone cabinets would be, and it is more in keeping with organ tradition. Available spaces for tone chambers are usually elevated. This automatically raises the location of the sound sources well above audience level, thus improving the uniformity of distribution of sound, and preventing any nearby listener from receiving too high a percentage of direct sound. It also eliminates the floor-space requirements for tone cabinets within the auditorium or church sanctuary.

Flexible design practices have been established for organ tone chamber construction which have proven to be very useful. It is necessary to have working principles such as the following in order to take full advantage of the acoustical conditions, to avoid known acoustical defects, and to make the best acoustical compromise where necessary.

1. Use rigid acoustically reflective inner surfaces.

2. A minimum volume of 600 cubic feet is recommended.

3. Make opposite surfaces slightly nonparallel, and avoid identical, multiple, or extreme dimension ratios.

4. Baffles should provide adequate free-path length for good low-frequency response, and should be both reflective and rigid.

5. Make the port area 3 to 10 per cent of the total interior surface of the chamber. Grillwork should obstruct no more than twenty-five per cent of the port area.

6. Choose port locations as nearly equidistant as possible from all listeners in the audience, and which favor the console and choir.

Fig. 4 (next page) shows a tone-chamber installation of an electronic organ in a rather wide, medium-size church auditorium. The use of two tone chambers in these locations gives tone distribution well suited to this particular church, because the balcony areas are as well served as the main floor. A single tone chamber behind the choir would not have provided as good distribution in this case. The tone chamber on the left has two ports, one toward the audience and one opening into the choir space opposite the organ console, well above head level so that choir members under the grille opening will not receive a disproportionate level. Other examples of church and auditorium organ installations, and additional acoustical details on tone chamber design principles, are given in a separate paper.[7]

Fig. 5 (next page) shows two home organ installations. In residence A (left) a small tone chamber is build under the stairway with a port in the access door. The

[7] D. W. Martin, "Fundamental acoustics of electronic organ tone radiation," *Jour. Acous. Soc. Amer.* (To be published.)

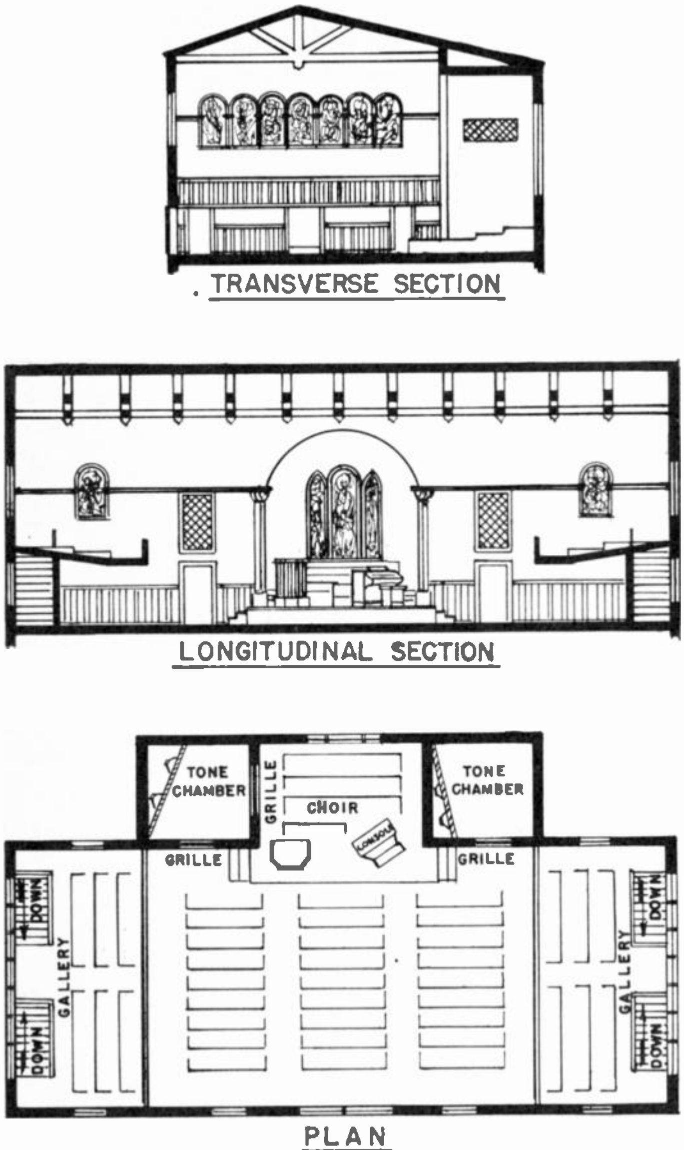


Fig. 4—Organ tone chambers in a wide church auditorium.

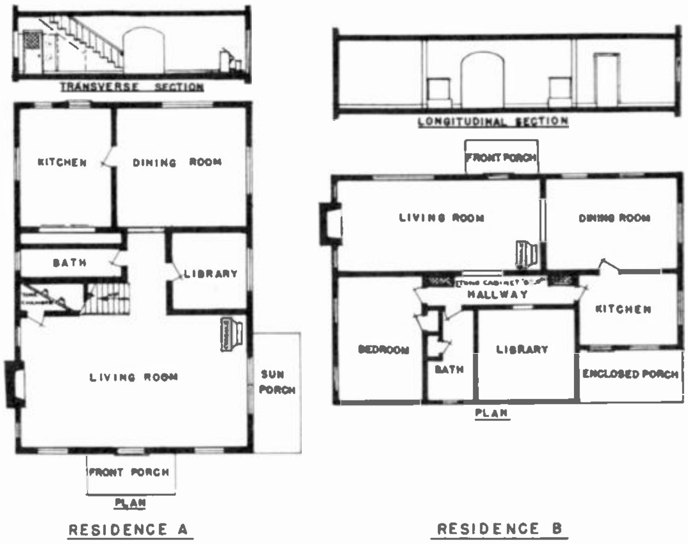


Fig. 5—Residential organ installations.

location tends to restrict the high sound level to the living room area. The console is situated to give the organist the full benefit of reflections at the fireplace end of the room.

In residence B a central hallway is used to advantage as a tone chamber. This places the tone cabinet sources outside the immediate vicinity of the organist and his audience in the living room. This has the possible disadvantage of disturbance to other members of the household engaged in other pursuits in the surrounding rooms. Preference for such an arrangement varies widely depending upon the organist, his ability, and the degree of musical appreciation of the rest of the family. However, where musical appreciation or tolerance is high this arrangement is considered to enhance the tonal effect acoustically.

Multiple Sources and Artificial Reverberation

One obvious answer to the acoustical limitations of the single-source method of organ tone radiation is the use of multiple sources. Multiple sources within a tone chamber offer little advantage except as they are needed for power-delivering capacity. More than one tone chamber is advantageous in some instances (see Fig. 4), especially for an "echo" organ, which is musically very useful for antiphonal effects. Moreover the combination of a "main" organ in one end of the auditorium, with an "echo" organ in the other end, improves the uniformity of organ tone distribution. This is a distinct advantage during congregational singing, for example. Multiple tone chambers (one for each of a large number of loudspeakers) are economically unsound, of course, because of the minimum volume required for each chamber to be effective in producing reverberation.

Multiple exposed loudspeaker units directed toward the audience are seldom very satisfactory for organ tone radiation, unless the auditorium is extremely reverberant. Regular geometric arrays of loudspeakers mounted on flat baffles in an exposed location are too highly directional.

If multiple exposed loudspeaker units are widely separated in the manner of a distributed speech-paging system, the directivity becomes less serious. However, distribution of the audio power in a concealed manner and fitting the loudspeakers into the architectural plan becomes a problem. Exposed loudspeaker housings are less acceptable in a church than in a railroad station, for example. Furthermore, the low-power loudspeakers which would be used in quantity in this type of system characteristically lack the response in the low-frequency range so important to organ music.

Although widely distributed multiple sources are not well adapted to reverberation augmentation by tone chamber, they can be improved by the addition of loudspeaker-accessory reverberation devices,[8] which provide

[8] D. W. Martin and A. F. Knoblaugh, "A loudspeaker accessory for the production of reverberant sound," *Trans. IRE*, vol. PGAU-2, pp. 95–98; May–June, 1954.

an aftersound effect simulating reverberation. Having the aftersounds radiate from scattered locations improves the illusion of true room reverberation.

The simulated reverberation devices have been most useful so far, however, in small acoustically "dead" locations. For example, the tone chamber of Fig. 5(a) is necessarily too small to add much natural reverberation. A reverberation accessory on one or more of the loudspeaker units would be recommended for a buyer who expects an organ in a living room to resemble one in a large auditorium.

Tone Cabinets

When tone chambers are not feasible, e.g., in temporary installations, in small homes, or in auditoriums where all available space is in use for other purposes, the most successful method of installation has been the use of indirectly radiating tone cabinets (i.e., loudspeaker enclosures). These cabinets are of unconventional acoustical design in that they orient the sound sources purposely to increase the ratio of reflected-to-direct sound in typical listener locations.

Another factor in the design of organ tone cabinets is the relative importance of the octave below 60 cps, a frequency range frequently sacrificed even in sound systems designed for general musical purposes. An artifice practiced in other branches of audio, i.e., letting the very low-frequency tones distort and hearing the fundamental by difference tone, is not applicable here because one of the tonal distinctions to be made is that between a very simple and a very complex tone. Dependence upon the sense of hearing to supply the "missing fundamental" does not work on an organ pedal bourdon tone which is chiefly the fundamental itself. An organ tone cabinet should produce actual fundamental tones down to nearly 30 cps. The acoustical principles for achieving solid low-frequency response are well established. Unfortunately, they require great wall rigidity and large baffle or cabinet size. Consequently a demonstration is sometimes required to justify the existence of the cabinet, or to show why certain principles are used to achieve good low-frequency response and power-delivering capacity. Fig. 6 is a special demonstration device used by the author for this purpose. (See Appendix for description and procedure.)

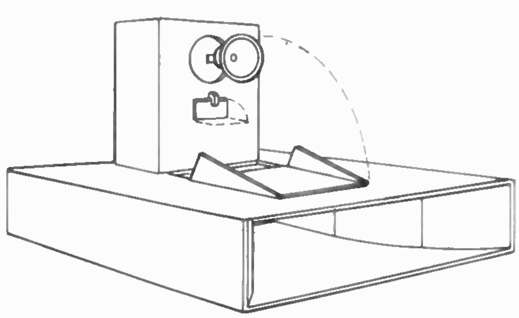


Fig. 6—Low-frequency tone-radiation demonstrator.

An example of a widely used organ tone cabinet using indirect radiation of medium- and high-frequency tones, with bass-reflex radiation of the very low-frequency tones, is shown in the line drawing of Fig. 7. Here two fifteen-inch loudspeakers are mounted in the top of a tall cabinet of reinforced construction, with the loudspeaker axes directed upward. A bass-reflex port tuned below 40 cps is situated near the bottom of the front of the cabinet beside the power amplifier shown (this also assists in ventilation). Fiberglas Aerocor blanket is installed on one inside face of the cabinet, and also on a horizontal frame approximately half way between the bottom and top of the cabinet. This material greatly reduces the interference effects of internal box resonances.

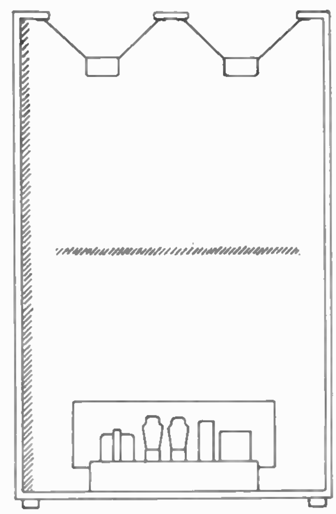


Fig. 7—Line drawing of Style Q organ tone cabinet—internal construction.

Thus the medium- and high-frequency ranges of the loudspeaker are radiated principally in an upward and outward direction, and in rooms with moderately reflective walls and ceiling the sound reaching listeners comes principally after multiple reflection. Thus the pertinent response-frequency characteristic of the loudspeaker system involves the integrated power output of the loudspeaker rather than the sound pressure on the axis or at some other point. The fact that the acoustical efficiency of a loudspeaker usually decreases with frequency throughout a large part of its effective frequency range, in contrast to the uniformity of the sound pressure on the axis, presents no major problem. This can be compensated in the design of the organ tone-color filters.

Audio engineers are probably better acquainted with another inherent limitation of the single sound source, a tendency for intermodulation distortion. An organist is in a much better position to detect this than are most users of sound equipment. Typical program material for a radio, TV, or phonograph listener is rapidly changing, and beyond the listener's control for the most part. By contrast, the tones of organ music are usually sustained, and the organist while practicing can control the duration of any combination of tones as long as he pleases. The tendency for intermodulation distortion, which will add foreign sounds to the combination of tones, can be markedly reduced, of course, by conventional frequency-range division in the electrical circuit, combined with multiple loudspeaker systems in which each loudspeaker is specifically designed or adapted to give maximum performance in the particular frequency range of signals supplied to it.

Frequency-range division and electrical equalization at low frequencies are both used in the small organ tone cabinet shown in Fig. 8, which is intended primarily for home organ use adjacent to a wall. The lower frequencies are radiated conventionally, because localization of the source at low frequencies is virtually impossible anyway.

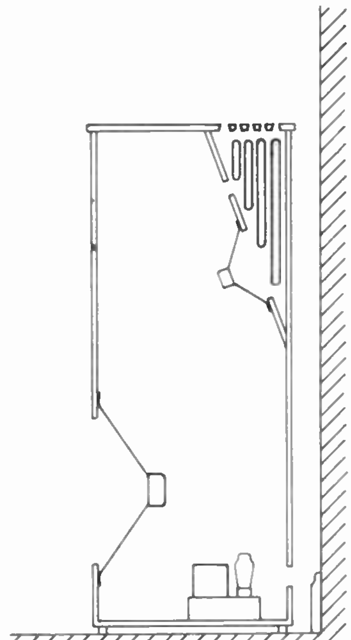


Fig. 8—Line drawing of Style J organ tone cabinet—internal construction.

The medium- and high-frequency tonal components are produced by a loudspeaker mounted on an inclined baffle in the upper rear part of the cabinet. The sound radiates from slots routed from the upper sidewalls and the rear topwall of the cabinet. Slightly over half of the available area is open. This leaves the major part of the top surface free for decorative devices deemed desirable in domestic design. Bass reflex slots are located near the amplifier in the lower back wall and above the level of the high-frequency unit in the inclined baffle. This assists in ventilation and augments the response below fifty cps. Larger port areas would provide greater overall loudness of tone, but would eliminate the tonal distinction between the pedal stops of the bourdon type and the more complex pedal tonal spectra. The acoustical effect of spreading of the high-frequency tonal components is to increase the apparent size of the source somewhat beyond the over-all dimensions of the cabinet. This is particularly advantageous for listeners near the cabinet, as they will frequently be in domestic environment. An acoustically reflective wall behind the cabinet is an asset. In fact, removal of the upper back panel is preferable under this favorable circumstance. However, a solid back panel is provided, because of the likely presence of drapes or other absorptive materials. The acoustical action in the mid-frequency range in the upper rear part of this cabinet, known as the Style J, is shown in the perspective view of Fig. 9. The wedge-shaped space between the inclined baffle and the rear wall of the cabinet approximates a horn in this frequency range,

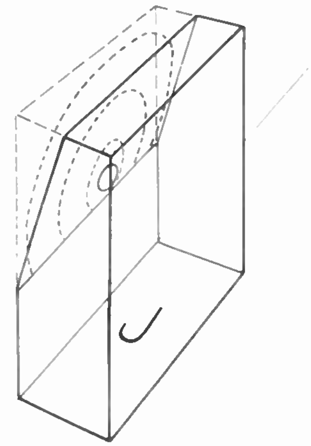


Fig. 9—Medium-frequency horn action in Style J tone cabinet.

with a mouth-to-throat area ratio of 5, and a nominal cutoff frequency of about 200 cps. The horn mouth is at its widest only five inches wide. The dashed curves lying in the plane bisecting the wedge angle show the progress of the wave front in the horn. Fig. 10 (opposite) shows the extrapolated progress of the wave outside the horn, neglecting wall reflection and diffraction effects.

It is customary practice in acoustical equipment engineering to improve the efficiency of loudspeakers by horn loading. However, horns tend to be highly directional. Horn loudspeaker systems have been used to achieve "presence" through the high ratio of direct-to-reflected sound which they provide. Organ tone cabinets employing horn loading would not give a sufficient spread to the sound source, if the horns were directed at the listener. On the other hand, if horns of conventional design were directed toward the ceiling, the sharp directivity would tend to waste the improved acoustical efficiency provided by horn loading, especially in auditoriums having acoustically treated ceilings.

Departing from usual horn design practice, a tone cabinet has recently been developed at Baldwin in which the width of the horn mouth, both for the high-frequency and low-frequency horns, is small relative to the wavelength throughout the major part of its operat-

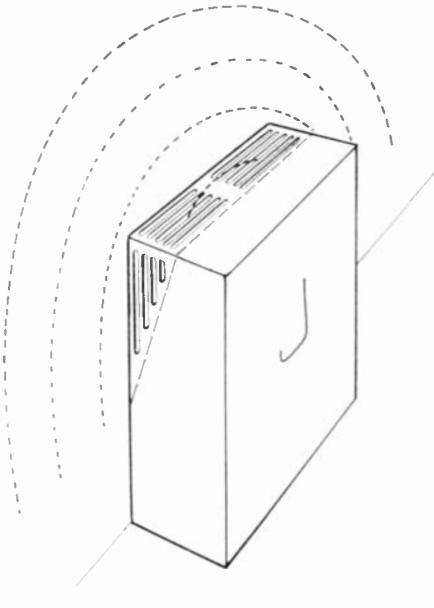


Fig. 10—Medium-frequency tone waves radiating from Style J tone cabinet.

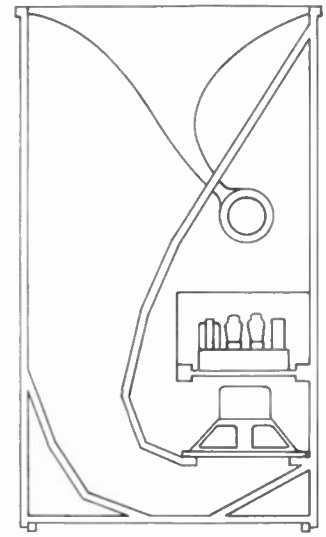


Fig. 11—Line drawing of Style K organ tone cabinet—internal construction.

ing frequency range. By this means loudspeaker efficiency has been increased greatly without high directivity. A line drawing of this tone cabinet design (known as Style K) is shown in Fig. 11.

The exponential low-frequency horn is curved within the cabinet to provide a nominal lower cutoff frequency of thirty cps, which lies just below the lowest fundamental frequency of the "sixteen-foot" organ stops. The relatively narrow mouth width, required to produce the broad directional characteristic desired in the bisecting vertical plane, permits the sidewalls parallel to the plane of the illustration to be flat and parallel, without having a cross-mode of vibration below the 400-cps frequency at which division of the frequency range occurs. This is an advantage in material, in construction, and in utilization of the space in the cabinet. The inner walls which define the taper of the horn add greatly to the cabinet rigidity so that a minimum of additional crossbracing is required.

The high-frequency horn is also curved. Thus both transducers, the amplifier, the crossover network, and all wiring can be reached through the same access door in the back of the cabinet. The high-frequency horn, which also uses the parallel-wall, narrow-mouth design principle, lies within the low-frequency horn and utilizes one wall of the cabinet. The axis of the high-frequency horn-driver unit is normal to the cabinet wall. A curved throat piece couples the driver to the horn throat. The amplifier enclosure is open to the atmosphere and is isolated acoustically from the back enclosure of the low-frequency transducer, in order to prevent loss of low-frequency range. The somewhat reduced efficiency at low frequencies, inherent in the size of the horn mouth relative to the wavelength near the horn cutoff frequency, is equalized from a frequency-response standpoint in the amplifier circuit. This efficient organ tone cabinet is intended particularly for large installations in which space is at a premium. Patent applications have been filed on features of both Style J and Style K tone cabinets.

The acoustical performance of the K loudspeaker system can be improved further at very low frequencies, when adequate space is available, by the addition of a horn extension, such as that in Fig. 12. The extension shown, which is twenty feet wide, is suitable for 4 K-tone cabinets in a cluster. The design is easily adapted to various numbers of cabinets, either more or less. The cabinets are staggered in orientation, so that extremely high-frequency tonal components are directed both to the left and the right. A horn mouth so large as this is more directional than the K cabinet itself, but in the huge auditoriums where clusters of efficient cabinets are required, the diffusion and reverberation are likely to be great enough to minimize the importance of directivity.

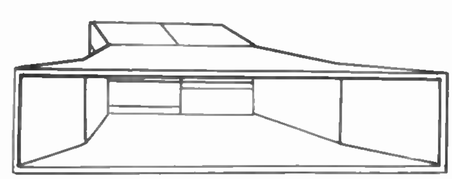


Fig. 12—Low-frequency horn extension for four Style K tone cabinets.

The efficiency of the Style K tone cabinet makes it a logical choice for installation in tone chambers where equivalent power delivering capacity in direct-radiator loudspeakers would require so much baffle material that baffle absorptivity would seriously reduce chamber reverberation. Fig. 13, on the next page, illustrates a manner of installing K cabinets in a tone chamber, adjacent to the grille of the chamber.

Conclusion

The acoustical radiation problems of electronic organ tone differ substantially from the problems of public address sound systems and general music reproduction systems. Although the underlying scientific principles are the same, their application to the solution of organ tone radiation problems results in different equipment forms and techniques.

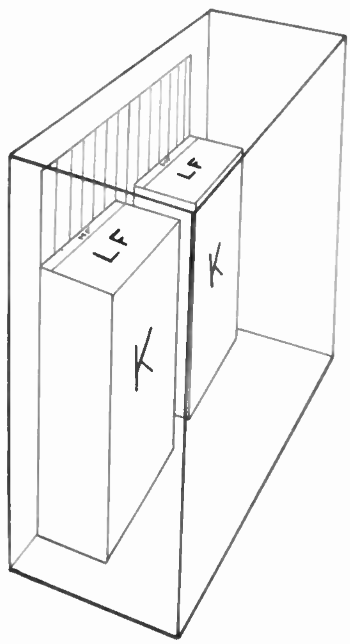


Fig. 13—Manner of installation of Style K tone cabinets in a tone chamber.

Appendix

Demonstration of Low-Frequency Tone Radiation Principles

See Fig. 6. A 15-inch loudspeaker is mounted on the end of a rod, well in front of a suitable bass-reflex cabinet, which in turn is firmly fastened to a platform containing a large exponential horn of low cut-off frequency. The mounting hole in the front of the cabinet is sealed shut by a plate which is also carried on the rod supporting the loudspeaker. This prevents the cabinet and mounting hole from serving as a Helmholtz resonator behind the loudspeaker and thus giving unseen assistance to the baffleless loudspeaker.

The electrical input should be turned off between different steps so that intermediate conditions will not confuse the listener. An electrically and mechanically quiet switching means is also essential to a clear-cut demonstration. All temporary surfaces of contact should be felted in order to prevent chattering and to provide a good acoustical seal. All permanent surfaces of contact should be glued and screwed tightly. Hinges should be rugged and yet work smoothly.

The loudspeaker is driven at a frequency of about 40 cps at a level which is just short of mechanical limiting action, so that this almost inaudible vibration can be clearly seen. Then the rod is pulled at the back of the cabinet and locked in a position which seats the back of the loudspeaker rim on the front of the cabinet. The reflex-port door below the loudspeaker is still clamped shut. The same tone is then repeated but this time quite audibly, showing the necessity for a baffle or cabinet.

Next the port door on the cabinet is opened, and the tone is played again to demonstrate this principle. Of course this requires proper tuning of the port-cabinet-loudspeaker combination. At this stage it helps the demonstration to place a lighted candle on the open door of the port, and extinguish it with the sound.

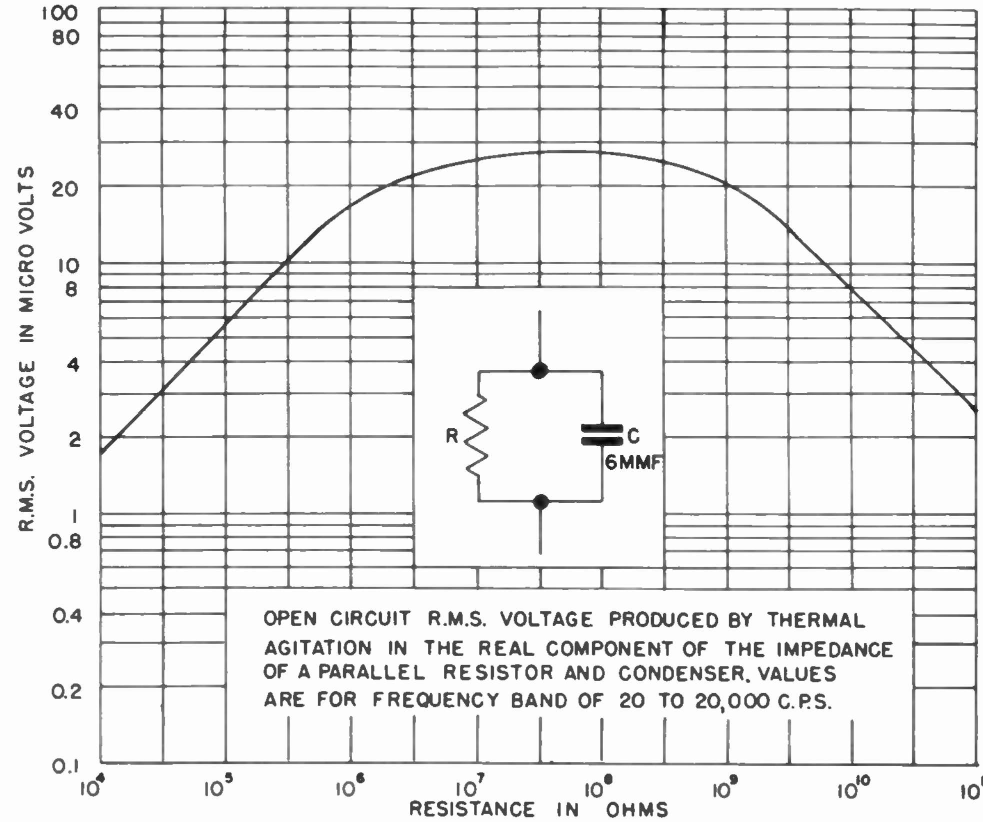
The final step, after closing the port door again, is to raise the hinged cover from the top of the platform until it engages the front of the cabinet and the top of the platform in a sealed relationship. The tapered cover couples the loudspeaker as a driver unit to the horn in the platform through a hole in the top of the platform, just in front of the cabinet. The horn axis is curved in this region. All of the horn taper is in the front-to-rear plane down through the curved portion of the axis, and in the horizontal plane thereafter. Sounding the tone again demonstrates the improved efficiency of horn loading.

Even more striking is the use of voltages which correspond to the maximum before audible overload in each case. It is necessary to demonstrate in this manner in a noisy crowd because a voltage which will make the cone rattle in the first step does not provide sufficiently audible tones in succeeding steps. This demonstrates the power-delivering capacity of each type of loading at low frequencies.

Acknowledgment

The encouragement of Mr. J. F. Jordan, chief engineer of the Baldwin Piano Company, and the assistance of a number of the writer's colleagues is gratefully acknowledged.

CORRECTION



An error in calculation has been brought to our attention by Clifford E. Berry of the Consolidated Engineering Corporation, Pasadena, California. The error appears in Fig. 4 of the paper, "Lipstik Condenser Microphone System," by J. K. Hilliard and J. J. Noble, on page 165 of the November–December, 1954 issue of TRANSACTIONS OF THE IRE, vol. PGA-2, no. 6. The correction has been made in the accompanying drawing which should be substituted for the original Fig. 4.

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