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PROFESSIONAL TECHNICAL GROUP ON AUDIO

IEEE TRANSACTIONS ON AUDIO

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The Editor's Corner

WOOFERS, TWEETERS, HI-FI? BAH! GIVE ME OLD FAITHFUL*

SITTING IN THE next room as I scrawl these words with my goosequill pen, which I dip in a mixture of lampblack and water, is an old portable record-player, which cost like 60 bucks over 10 years ago and has never bust once.

It has withstood the importuning, if not the downright onslaught, of variable Spanish electric power, for all these years, without blowing either its top or its tubes. It still spins with great good will at both 33 and 45 rpm. I understand this machine. It will sort of come when I call.

Stacked in odd corners are other machines, of much more recent vintage and much higher prices, looking like the carcasses of dead mastodons. These record players will do everything but cut your hair and walk the baby, supposedly. The only trouble with them is that they're always in hospital for further surgery. They start fast, but they're no good in the mud, and they always manage to cast a shoe or lug out at the turns.

And this is simple stuff. I am not an esoteric music lover. All I ask is a comfortable amount of noise to drown out the screams of the wounded at cocktail parties, or to help me over a tough evening of copyreading. The electric power in Spain is mainly on the wane, when it is not on the wax, and is prone to make a highly strung gadget nervous.

But even with a regular power flow, I would never attempt to compete in the modern record-playing race. You do not simply play a record today. You have to take a two-year course at M.I.T. before they will sell you a really hip modern noise-maker. This machine cannot be bought whole, like an automobile. It must be put together, or other people's children sneer at your children.

I have before me a prospectus for an "XAM high fidelity loudspeaker system," which was just flown in by two unfrocked astronauts. At random, I quote you some technical terms:

"Tweeter, woofer, rumble filter, speaker impedance switch, 50–15,000 CPS cartridge, mode selector, phase switch, stereo preamplifier," plus some old-timey things like watts and amps. I would delve further but it reels the mind.

From other information I see that "some of the microphones used, representing the best of all manufactures available, are the RCA-44BK, Telefunken U-47, Western Electric 1142A, Altec 639B." Furthermore, "the edited tape is re-recorded into a master disc from a Fairchild type machine through Pultec equalizers and a McIntosh 200-watt amplifier to a specially built cutting head mounted on a Scully automatic lathe."

Strictly between me and my flux density, if you won't rat on me to the people that hang out with the center-pole diameters in cone resonance headquarters, I am going to lie down on my wire-wound L-pad and turn on Old Faithful again and play a little scratchy Bing Crosby. Old Faithful may not have much in the way of sound-pressure-level characteristics—you know, the woofer SPL of 90 DB at 3 feet when energized by a 2V signal in its frequency range in an anechoic environment—but it plays in the rain and can hit with men on base.

As a boy who remembers that you used to have to crank the things so that Victor's fox terrier could hear his master's voice singing "Three O'Clock in the Morning," this new-fangled gobbledegook is as meaningless as the average briefing from Washington, and uses much the same language.

Mama didn't raise her boy to be no electrical engineer, and I'm too old to go back to college to learn how to spin a record which will play "Houn' Dawg" right back to me, except for the scratch in its defective surface.

My cross-over characteristic will never bridge my gap depth, and if they throw any more of this stuff at me, back to radio I go, and wreck the record industry. Play, Don.

ROBERT C. RUARK

We are grateful to Benjamin F. Meissner for calling attention to the above, which he spotted in the "Miami Herald."

PETER W. TAPPAN, Editor

PTGA Awards

The Awards Committee on the Professional Technical Group on Audio is proud to announce the following for the year 1962.

IEEE-PTGA Achievement Award

To honor a member of the PTGA, who, over a period of years, has made outstanding contributions to audio technology documented by papers in IEEE publications. A certificate and $200 have been presented.

JOHN K. HILLIARD

John K. Hilliard (A'25–M'29–SM'43–F'52) was born in Wyndmere, N. Dak., on October 22, 1901. After receiving the B.S. degree in physics from Hamline University, St. Paul, Minn., in 1925 he did graduate work in electrical engineering at the University of Minnesota, Minneapolis, Minn. In 1951 he received the D.Sc. degree in engineering from Hollywood University, Calif.

He spent fourteen years at MGM Studios, working on the development of recording and reproducing film and tape equipment and designing microphones and loudspeakers for theaters. Also, he was Project Engineer at the Radiation Laboratories, M.I.T., Cambridge, Mass., working on radar. For ten years he was engaged with high-intensity sound environmental equipment. He designed a microphone for measurement of nuclear blast, high-speed boundary layer measurements, high-intensity environmental equipment to simulate jet and missile engine noise to evaluate fatigue of electronic equipment and air frame structures, microphones to pick up heart sound, communication equipment for telephone systems and anti-submarine warfare equipment. From 1943 to 1960 he was with the Altec Lansing Corporation, Anaheim, Calif., first as Vice President in the Advanced Engineering Department, working with transducers and communication equipment, and recently as Director of the LTV Research Center, Western Division, Anaheim, Calif.

Dr. Hilliard is a Fellow of the Acoustical Society of America, the Audio Society, and SMPTE; a member of Eta Kappa Nu, Armed Forces Committee on Hearing Bioacoustics and the Institute of Environmental Engineers. He is also an Acoustic Consultant at the Brain Institute, UCLA Medical School, Calif.

IEEE-PTGA Senior Award

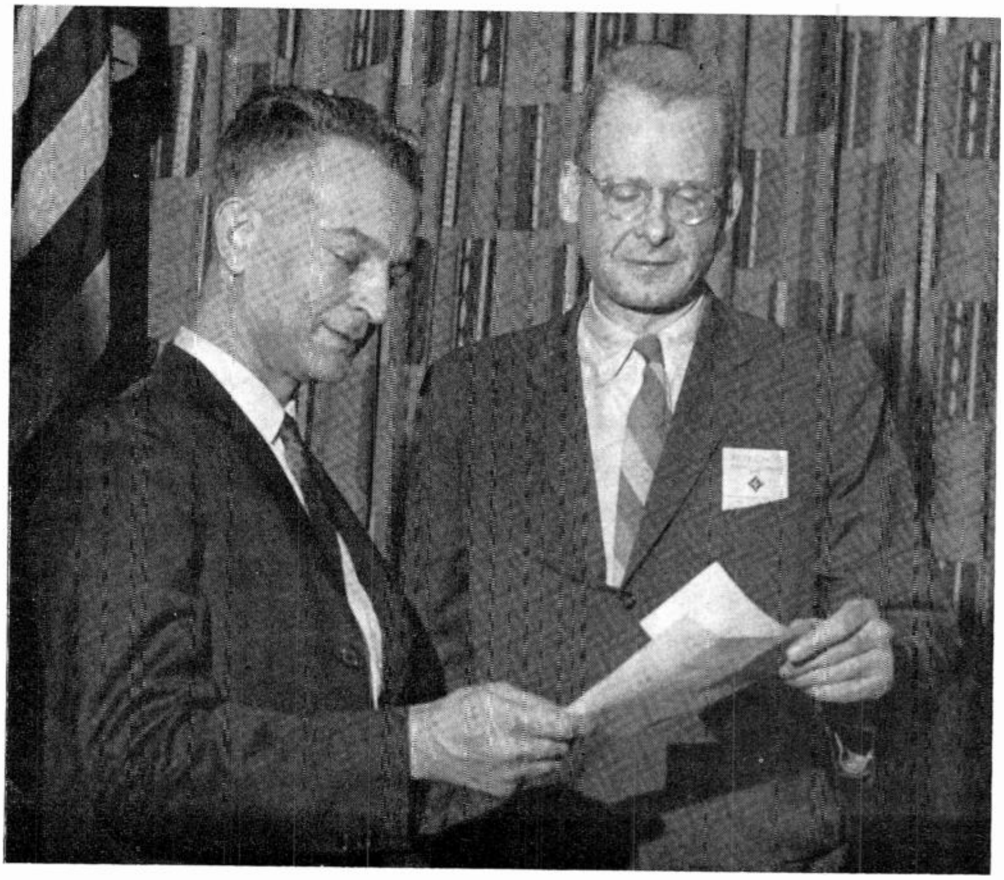
For the paper "Absolute Measurements of Magnetic Surface Induction," published in IRE TRANSACTIONS ON AUDIO, *Vol. AU-10, pp. 64–78; May-June, 1962. A certificate and $100 award were presented by Kendall Preston, Jr., Vice Chairman of the Connecticut Section at their December 10 meeting.*

F. A. COMERCI
Chairman
1963–1964

Frank A. Comerci (SM'55) was born in Newark, N. J., on January 18, 1920. He received the B.S.E.E. degree from Newark College of Engineering in 1943.

From 1943 to 1946 he served in the U. S. Army as a Communications Officer, installing and maintaining cryptographic speech communications systems. He joined the Rangertone Corporation in 1946, where he worked on the design of the first high-quality magnetic tape recorder built in the United States. In 1947 he became affiliated with the Navy Material Laboratory, Brooklyn, N. Y., where he was in charge of their Acoustics and Communications Section from 1950 to 1959. He was later employed by Audio Devices, Inc., Glenbrook, Conn., as Senior Electronic Engineer. At present he is Manager of the Magnetics Branch of Columbia Broadcasting System Laboratories, Stamford, Conn., and has responsibility for fundamental and applied research on magnetic materials and magnetic recording techniques. He has written several papers on magnetic recording and flutter.

Mr. Comerci is a member of the Acoustical Society of America and the Audio Engineering Society, serving on the Editorial Board of the *Journal of the Audio Engineering Society* for several years; he is also a member of the Sound Committee of the Society of Motion Picture and Television Engineers. He is Chairman of the IEEE Recording and Reproducing Committee and serves as IEEE representative to ASA Section Committee Z-57 on Sound Recording.



FRANK A COMERCI (*left*) RECEIVING AWARD FROM KENDALL PRESTON, JR.

PTGA News

NATIONAL NEWS

The PTGA sponsored a tutorial session at the National Electronics Conference in Chicago on October 28. The program was as follows:

1) "The State of the Art in Vocoder Techniques"—Dr. Edward E. David, Bell Telephone Laboratories, Murray Hill, N. J.

2) "Class B Audio Power Amplifiers"—Professor A. B. Bereskin, University of Cincinnati, Cincinnati, Ohio.

3) "Disk Records, A Review and Evaluation"—Dr. Abraham M. Max, Radio Corporation of America, Camden, N. J.

4) "Magnetic Recording"—Marvin Camras, IIT Research Institute, Chicago, Ill.

CHAPTER NEWS

Chicago

The Chicago Chapter has held three meetings this fall, all joint with the Chicago Acoustical and Audio Group. Papers were as follows:

September 13: David Arnold and James White, Jensen Manufacturing Company "Testing Horn Drives with Acoustic Lines."

October 16: Basil Bonk, General Radio Company, "Reciprocity Calibration of Microphones."

November 13: George Lang, WGN, "Description and Tour of WGN Radio and Television Studios."

New Haven

On December 10, Benjamin B. Bauer, Vice President of CBS Laboratories, Inc., discussed the latest developments in high fidelity and demonstrated the use of professional test records.

Considerations in High-Fidelity Moving-Coil Earphone Design

JAMES V. WHITE, STUDENT

Summary—The theory of simple moving-coil earphone behavior is briefly described. Some considerations concerning the measurement of sound pressure response and harmonic distortion are mentioned, with special reference to difficulties encountered at high frequencies. Several design objectives are presented, followed by a discussion of design techniques including two new approaches: 1) the introduction of a damping grease into the air gap of the magnetic structure for increased power handling ability, smoother sound pressure response and improved transient performance near the fundamental resonance; 2) a resonant back-loading cavity for improved low-frequency performance when air leaks are present.

I. Introduction

A SCIENTIFIC approach to the creation of a marketable earphone intended for high-fidelity applications involves much thought and experimentation. Engineering considerations encountered in the design work for an earphone of the moving-coil type are the subjects of this paper. Section II is a brief exposition of what can be expected theoretically from simple moving-coil earphones. The discussions of specific design objectives and the measured performances of actual earphones are then presented, including the results of two new design techniques. Some information which has already appeared in the literature is included, because amplifying the content with observations heretofore scattered about in different papers permits a presentation suited for a diverse audience of audio engineers.

II. Theory

An earphone as worn by the listener "looks" into a small volume of air bounded by the listener's ear canal, ear drum and head, and the earphone's diaphragm, housing and ear cushion. As the earphone diaphragm vibrates, the other boundaries also vibrate, and air alternately leaves and enters through any leaks that exist in the boundaries.

First consider the theoretical frequency response of a simple[1] moving-coil earphone as shown in Fig. 1. The frequencies in region A are low enough to preclude the formation of standing waves. For simplicity assume that the boundaries are rigid and air-tight. The sound pressure is then directly proportional to the amplitude of diaphragm motion as shown in the Appendix. Amplitude is virtually independent of frequency below the fundamental resonant point for a Q of about one, in a mechanical system with one degree of freedom. Thus, we can expect flat frequency response below resonance for a correctly damped, perfectly sealed earphone.

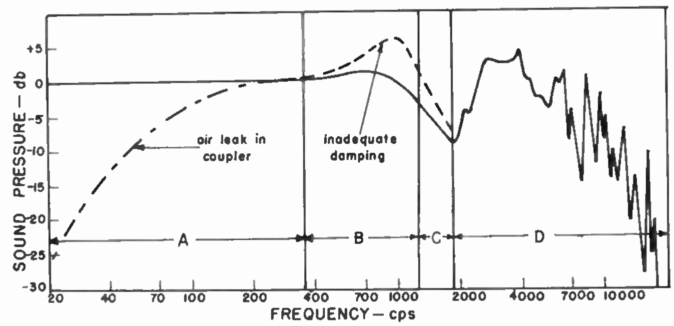


Fig. 1—Coupler sound pressure response of a theoretical moving-coil earphone.

With the introduction of an air leak, the direct proportionality of sound pressure to diaphragm excursion does not hold. Severe dropping off of the sound pressure response in the low-frequency region occurs with increased diaphragm movement and possibly an attendant increase in distortion due to suspension nonlinearities.

A peak in the frequency response curve (region B) will occur due to the increased diaphragm excursion at resonance when the moving system has a Q greater than one. Moving-coil earphones are usually under-damped unless special precautions are taken. The magnitude of this peak is often greater than 6 db, falls in the 500- to 2000-cps range and can be expected to produce a characteristic "honking" color in the sound.

The frequency response of the earphone will fall off asymptotically to the rate of 12 db per octave above resonance (region C). This is a result of the inverse relation between amplitude and frequency for single degree systems above resonance.

Standing waves form and the diaphragm ceases to act as a perfect piston when the wavelengths approach the air volume dimensions into which the earphone "looks" (region D). An exact analysis of chamber modes and diaphragm break-up would be very complex, but two generalities can be sifted from experience.

1) The frequency response becomes erratic with sudden peaks and troughs.

Manuscript received June 13, 1963.

The author is with Jensen Manufacturing Co., Chicago, Ill.

[1] A *simple* earphone is one whose mechanical resonant system has only one degree of freedom and is mathematically describable in terms of one independent variable. This is opposed to a *complex* earphone which requires two or more independent variables for a complete lumped element representation of its multiple degrees of freedom.

2) The sound pressure response trend does not generally follow the 12-db per octave roll off noted above.

Distortion due to Doppler effects and nonlinearity in the suspension should be very low since a diaphragm of 3-cm radius "looking" into a sealed air volume of 50 cm^3 will produce an rms sound pressure of +100 db with a peak-to-peak sinusoidal excursion of only 2.5×10^{-5} cm. This follows from (5) in the Appendix referred to a reference pressure of 0.0002 microbar.

III. Acoustic Tests

Having presented what can be expected from the theory of simple moving-coil earphones, we should consider some common measurements which are used in the design and evaluation of earphones.

The coupler is an apparatus basic to virtually all acoustic tests of earphones. It is a rigid chamber with a volume of from 2–80 cc or more, which accepts one or more microphones and an earphone unit. The coupler provides a means of standardizing the coupling between the microphones and the earphone during testing.

One common coupler has an internal volume of 6 cc and is designed to use a Western Electric 640-AA condenser microphone.[2] The performance of this particular coupler has been investigated to a greater extent, perhaps, than any other described in the literature.[3] A very important fact becomes evident from these investigations. The coupler measurements of absolute sound pressure above 3000 cps are invalid, because cavity modes are excited when wavelengths approach cavity dimensions, and "interactions" take place between microphone and earphone.

Effects similar in nature to those described in the literature for the 6-cm^3 coupler can be expected from other coupler designs. The coupler we have used extensively is designed to accommodate earphones typical of those intended for high-fidelity applications. Many of these earphones have large circumaural[4] ear cushions which provide a good air seal without irritating the pinna (external ear). Because these ear cushions rest against the bony part of the head rather than directly against the ear, the volume contained by them is in excess of 6 cm^3. Consequently, the 6-cm^3 "standard" coupler was not used. Rather, a 640-AA microphone was flush mounted on a rigid plate. The earphone being tested was placed face down on the plate with the ear cushion intact (Fig. 2). The resulting volume of this flat-plate coupler varied from earphone to earphone, being dependent upon their individual geometries. This more closely simulated the actual working conditions

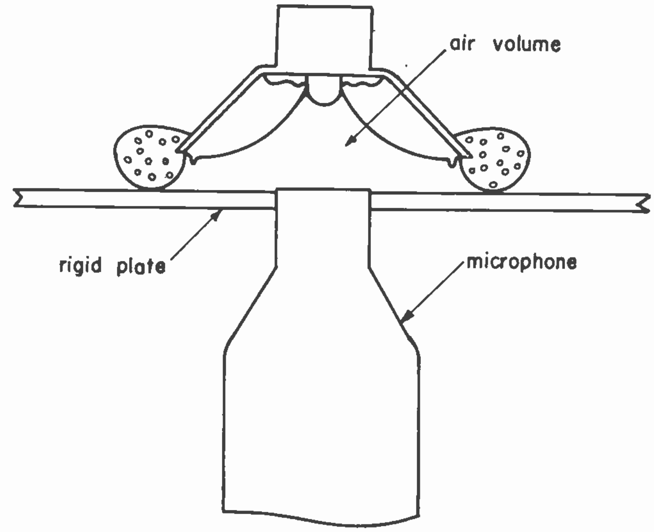


Fig. 2—Air volume of flat-plate coupler depending upon the earphone's geometry.

for earphones of this type and was much more convenient to use than the 6-cm^3 coupler. The earphone-coupler combination could be made air-tight by the liberal use of petroleum jelly.

A simple experiment was performed to demonstrate how much the coupler design influenced the sound pressures produced inside of it at high frequencies. A doughnut of clay was fashioned—slightly smaller than the earphone diameter, and about one half inch thick. The frequency response of an earphone was measured with and without this ring inside of the coupler. Fig. 3 shows the gross differences between the two resulting curves in the frequency range above 2000 cps, illustrating how coupler measurements at high frequencies can be affected by coupler design.

It may appear that high frequency measurements are of little value; actually they are quite useful. Although absolute measurements are extremely difficult, comparisons between different earphones can be made with acceptable accuracy. For instance, the effects of modifications on a basic design can be followed, using coupler measurements and listening tests. The high frequency coupler data, by itself, is misleading to the inexperienced, but through careful listening tests one discovers a correlation between the objective data and how the earphone sounds. Eventually, by working enough with a single coupler design, it is possible to predict with reasonable assurance how an earphone will sound by examining comprehensive coupler data, even though this data is itself not absolutely correct. Therefore, coupler measurements by themselves (divorced from any other information) are not a valid means for the absolute evaluation of high frequency response. They are useful only as a means for comparing units with each other in this frequency range.

Coupler measurements of absolute sound pressures

[2] American Standards Association specification Z24.9; 1949.

[3] K. C. Morrical, J. L. Glaser and R. W. Benson, "Interactions between microphones, couplers and earphones," *J. Acoust. Soc. Am.*, vol. 21, pp. 190–197; May, 1949.

[4] Circumaural implies that the ear cushion rests against the head instead of the outer ear, circumscribing the auricle.

at low frequencies, however, can be made with good accuracy because of the simplicity of what happens inside the coupler with long wavelengths.

Fig. 4 serves to illustrate several points thus far discussed. The moving-coil earphone used for these measurements had a diaphragm of $\frac{1}{2}$-inch radius. First we observe the relative smoothness of the curve below 2800 cps. This indicates that the wavelengths were long enough so that no standing waves were set up in the coupler. Above 2800 cps, the curve is poorly behaved; standing waves in the coupler and break up of the earphone diaphragm produce very sharp peaks and troughs.

Secondly, it can be seen that the damping of the moving system at the fundamental resonant frequency (1500 cps) is insufficient to prevent a 7-db rise in output.

Lastly, the response is very flat down to 40 cps. The falling off of sound pressure below this point is due to slight air leaks between the coupler and earphone.

The total harmonic distortion and noise which is produced in response to a single-frequency sinusoidal input is another aspect of earphone performance about which meaningful quantitative information can be obtained. In the frequency range where the sound pressure response is well behaved, total distortion and noise can be measured using a distortion and noise analyzer such as the Hewlett-Packard 330C Analyzer. Such measurements will truly represent the performance of the earphone, since coupler design has no effect on the relative sound pressures developed in the low frequency region. The main difficulty usually encountered is excessive ambient noise. This noise can easily produce measured distortion and noise values of 4 per cent in cases where the actual noise and distortion is less than 0.3 per cent. The ambient signals which produce this intolerable mud level are extremely low frequency mechanical vibrations in the coupler due to slight (but common) tremors in the laboratory floor. A sharp cut-off rumble filter is therefore a necessity when total distortion and noise are measured.

In view of the complete lack of a one-to-one correspondence between coupler pressures and the sound pressures developed in the ear at high frequencies, distortion measurements in this upper frequency region cannot be expected to be very accurate. Rather, they are useful in showing general trends which may exist in the distortion characteristic of the earphone. Of course, if the distortion is extremely low for a particular unit, then the distortion vs frequency plot can be expected to be quite smooth in spite of the gross raggedness of its frequency response curve. To clarify this, consider an earphone producing a very small amount of 2nd harmonic distortion with a given input frequency. Assume that at the 2nd harmonic of this frequency there is a 20-db peak in the sound pressure due to a standing wave in

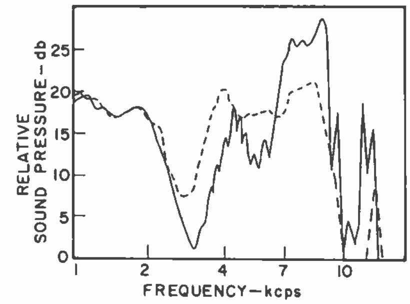


Fig. 3—Response of an earphone measured with (dotted) and without (solid) clay doughnut inside the coupler.

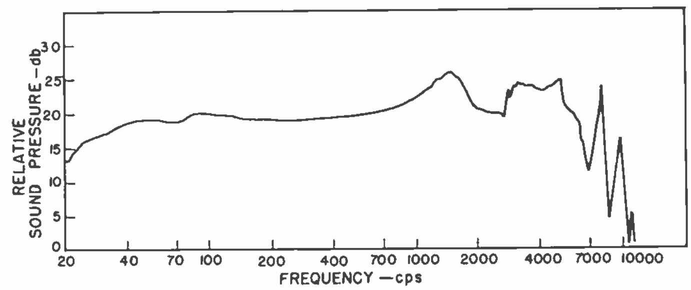


Fig. 4—Response curve exhibiting several characteristics typical of moving-coil earphones.

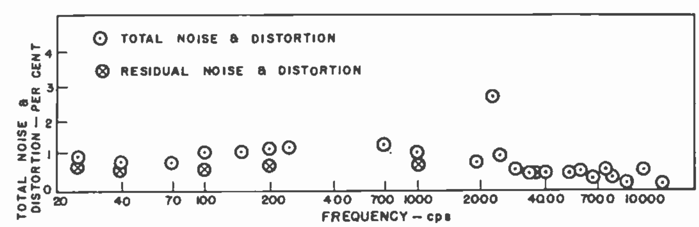


Fig. 5—Although the sound pressure response of the earphone was very irregular above 2000 cps, this plot of total noise and distortion for the most part smoothly follows the residual noise and distortion in the test equipment.

the coupler—a phenomenon purely the result of an arbitrary coupler design and having no necessary counterpart for human ears. Assuming further that the actual harmonic output of the earphone is 80 db below the fundamental, this results in an actual distortion value of 0.01 per cent. When the 20-db peak of the coupler is superimposed on this earphone output, the measured distortion would be 0.1 per cent with perfect test equipment—ten times the actual value, but still quite low, especially when it is realized that residual distortion values run in excess of 0.1 per cent for most test equipment. In the limiting case of an earphone producing no distortion, the measured distortion will be only the residual value of the test equipment regardless of the frequency response irregularities. Consequently, smooth distortion curves can be expected in spite of coupler effects when distortion is very low. Fig. 5 shows the total noise and distortion which was measured on a high quality earphone driver producing 112-db sound pressure with respect to 0.0002 microbar at 1000 cps in the coupler.

IV. Design

Objectives

Sections II and III of this paper provide a background for understanding the behavior of simple moving-coil earphones. In this section some design objectives are presented, followed by some suggested techniques for approaching these objectives. As in Section II, the total frequency range will be divided into parts. In this manner the particular design difficulties characterizing each range can be dealt with in an orderly fashion.

Referring again to Fig. 1, in the low frequency range (A) flat response is obtained provided there are no air leaks, and the air volume is enclosed by boundaries which are both rigid and air-tight. This means that the earphone diaphragm must be made of a material which does not admit air, that the earphone assembly is air-tight and that the ear cushions provide a perfect fit to the listener's head. Another requirement (to be described later) concerns the ear cushion compliance and damping. Usually the requirement of perfect ear cushion fit cannot be consistently realized in practice. There are two main reasons for this. First, many listeners are ignorant of the necessity of adjusting the earphones for perfect fit, and secondly it is generally expensive to fashion a cushion which seals well and at the same time is comfortable for prolonged periods of listening. Some individuals are made uncomfortable by the "suffocating" effect of a "perfect seal." For practicality in a product designed for the commercial market, it is the author's opinion that a compromise must be made between seal, cost and comfort. The first design objective follows: Low frequency performance should not be adversely affected by slight air leaks.

The fundamental resonance of the moving-coil earphone, when worn for listening, lies in the region of middle frequencies (region B). The transient response of the earphone deteriorates, and its sound pressure response peaks up when this resonance is insufficiently damped. It is shown in the Appendix that a simple earphone with a Q equal to one, gives virtually flat steady-state response up to this resonance frequency. A second objective follows: The Q of the moving system should be about one at the fundamental resonance point.

The two upper frequency regions (regions C and D) occupy the range above the fundamental resonance frequency. Exact statements regarding the desired response trend at high frequencies are fraught with danger because of the measurement difficulties at short wavelengths. There is no comprehensive data available which provides a rigorous means of quantitatively correlating on-the-listener high frequency performance with coupler measurements of circumaural earphones. The tastes and prejudices of the designer probably rear their ugly heads to greater heights here than in other aspects of earphone response evaluation. One defensible objective regarding measurable performance in this range is that the response should extend comparatively smoothly to as high a frequency as possible.

Sensitivity, by itself, is of little significance in most high-fidelity applications where the earphone is driven from a typical power amplifier. The maximum sound pressure which can be produced without introducing overheating or excessive distortion is of importance, however. Many individuals seem to tolerate much greater sound pressures when listening with earphones than to loudspeakers. Therefore, a desirable minimum for this peak sound pressure might be 110 db with respect to 0.0002 microbar.

The ability of the earphone to exclude ambient noise is important but not overwhelmingly so, as is the case in some industrial applications. The requisite attenuation for domestic environments is usually obtained without resort to any sophisticated techniques; the simple encasing of the ear by a rigid earphone structure provides acceptable results.

Techniques

Let us now consider some specific design techniques for circumaural earphones, keeping the above design objectives in mind. First of all, it may seem desirable that the earphone driver diaphragm be made small (perhaps $\frac{3}{4}$-inch diameter) so that the moving mass may be kept as low as possible for extended high frequency response. This is a reasonable approach, but one which is frequently not capable of producing good clean bass unless a near perfect air seal is maintained, because of the excessive diaphragm excursion required under such conditions. It is the author's experience that a three-inch diameter diaphragm can provide a satisfactory compromise between the low-mass demands for extended high frequency performance and the substantial air-moving ability necessary for good bass in the face of air leaks. The design of good earphone diaphragms is more of an empirical quest than a mathematical exercise when break up is relied upon to carry the high end to beyond 10,000 cps. Assuming that the earphone driver unit does have substantial air-moving ability in reserve for the difficult times when air leaks appear, the question arises as to how to induce the diaphragm to increase its excursion in response to small air leaks (in such a way that the low frequency response does not drop off drastically). Obviously, simple single-degree-of-freedom earphone designs cannot be expected to meet this problem (see Fig. 1). A novel solution to this problem has been introduced by James F. Novak. It consists of having the rear of the earphone diaphragm radiate into a simple vented cavity—a Helmholtz resonator tuned to about 300 cps. Above this frequency the diaphragm moves essentially as if the cavity were sealed. In a narrow range of frequencies centered around the cavity resonant point the diaphragm sees a high mechanical impedance resulting in decreased diaphragm excursion. (This is the same situation that arises in a bass-reflex loudspeaker system when the speaker diaphragm excursion falls off drastically at frequencies in the vicinity of the tuned enclosure's resonance.) The earphone sound pressure response falls off somewhat as a result of this decreased excursion, the exact amount depending upon the amount of damping in the vent of the rear cavity. Below about 150 cps, the mechanical impedance presented to the diaphragm by the rear cavity greatly diminishes, virtually disappearing as the damping in the cavity vent goes to zero. The rear of the diaphragm thus becomes relatively "unloaded," resulting in increased diaphragm excursion and an attendant increase in sound pressure response. Diaphragm excursion is now much more responsive to changes in the impedance seen by the front of the diaphragm, for instance, those drops in impedance brought about by air leaks. Fig. 6 shows the improved sound pressure response obtained by the use of this tuned cavity when a controlled air leak is introduced in the coupler. Success with this approach demands that the driver unit have a sufficiently great air-moving capability so that distortion does not become troublesome.

A sharp dip is present in the sound pressure curves of Fig. 6 at 65 cps. This is the result of vibrations of the ear cushion.[5] If the cushion material is not adequately damped this sharp drop off appears near the frequency where resonance occurs between the ear cushion compliance and the earphone mass. It is desirable to have this resonance occur at a very low frequency when damping is inadequate. Fig. 7 shows the performance of a typical underdamped ear cushion. The more extended of the two curves was obtained by adding 33 oz of mass to the earphone housing so that the ear cushion resonance was lowered about one octave. A potentially well damped (though expensive) circumaural ear cushion which provides a particularly good air seal is the fluid-filled type consisting of a leak-proof plastic bladder of appropriate shape for a good fit with the head. The earphone of Fig. 4 was fitted with a fluid-filled cushion.

The damping of the earphone moving system at the fundamental resonance frequency is a troublesome problem. Earphones are often used with series resistor attenuators so that electromagnetic damping, which plays an important part in loudspeaker damping, cannot be relied upon. Even if it were available in earphones, large magnets and relatively massive voice coils would then be called for—both undesirable because of their weight. An effective means of supplying the necessary damping is to put a stable grease in the air gap of the magnetic structure so that the voice coil

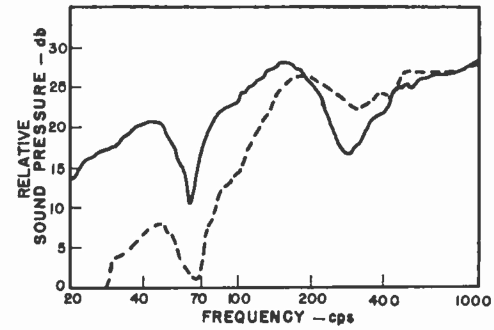


Fig. 6—An intentional air leak introduced in the coupler. The dotted curve is from an earphone with one degree of freedom, while the solid curve is from the same earphone with the introduction of a resonant cavity loading the rear of the diaphragm.

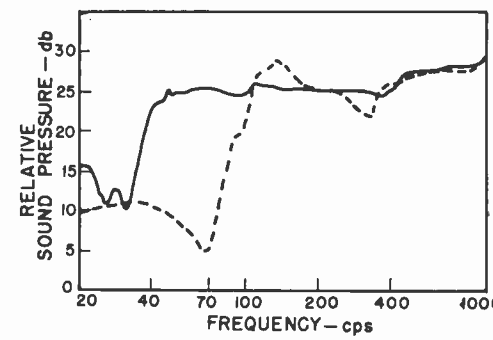


Fig. 7—The low frequency response drops off suddenly near the frequency of ear cushion resonance, because cushion damping is inadequate. Both curves are of the same ear cushion; the solid curve was obtained with 33-oz mass added to earphone housing thus lowering the cushion resonant frequency.

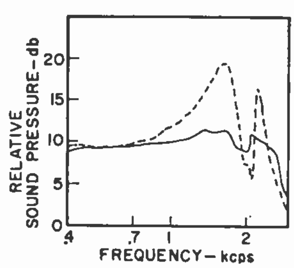


Fig. 8—The dotted curve is from an untreated driver, whereas the solid curve shows the performance with silicone damping at the air gap.

is partially immersed. Silicone grease is very stable (serviceable to 400°F), has excellent creep properties, supplies the requisite damping and makes a relatively good conductor of heat. This good heat conduction allows a very fragile voice coil employing fine gauge wire to be used for extended high frequency response without fear of failure due to electrical burn out. More than 6-watt rms continuous power can be safely dissipated by a "fluid damped" voice coil employing No. 38 aluminum wire—thus providing a substantial margin of safety against accidental burn out. Without this silicone compound the voice coil would be likely to overheat to the point of failure. Using a three-inch diaphragm, 6-watt input power easily produces a low-distortion sound pressure level of over 130 db at 1000 cps in a flat-plate coupler with circumaural ear cushions. The

[5] E. A. G. Shaw and G. J. Thiessen, "Acoustics of circumaural earphones," *J. Acoust. Soc. Am.*, vol. 34, pp. 1233–1246; September, 1962.

effectiveness of this damping technique is shown in Fig. 8. The earphone without silicone damping runs a $Q \doteq 3.5$ where as with the damping grease, $Q \doteq 1.2$.

Extending the frequency response to as high a frequency as possible can be accomplished first by keeping the mass of the voice coil very low (using fine aluminum wire), and second, by using a very light diaphragm with appropriate break-up characteristics. This approach succeeds in carrying the response out to beyond 15,000 cps when properly handled.

A different approach based upon a wave model of the earphone instead of the lumped element description used in this paper is to use a multiple chamber design with a driver unit specifically designed to avoid break-up.[5] Relatively flat response is achieved with this approach to about 10,000 cps, and occasionally somewhat higher. As may be recalled from Section II, above resonance the sound pressure developed in a coupler of fixed dimensions falls off at the rate of 12 db per octave when diaphragm break-up is avoided and a lumped element model is used. In the multiple chamber (wave model) approach two or more air chambers loading the front of the diaphragm are filled with sound absorbing material and coupled to each other by means of an annular slot. The effective volume seen by the earphone driver looking into this chamber complex varies with frequency so that as the frequency increases the effective volume it sees decreases. When the situation is handled properly, the tendency for the response to fall off because of diminishing diaphragm excursion is counteracted by the tendency for the sound pressure to increase as the effective chamber volume decreases. This is an elegant approach to the high-frequency problem, capable of relatively smooth response. Unfortunately the requirement that the diaphragm not break up usually demands that its diameter be one inch or less. Such small units do not usually possess sufficient air-moving ability to cope with the difficulties that arise in the low-frequency region when air leaks occur.

V. Conclusions

The behavior of a theoretical moving-coil earphone comprised of a linear mechanical system with one degree of freedom working into a perfectly sealed, rigidly enclosed air volume, acts as a useful starting point when considering the actual performances of physically realizable units. Such a model has flat steady-state sound pressure response down to any arbitrarily low frequency. Above resonance the response falls off asymptotically to the rate of 12 db per octave. At resonance the response peaks up if the Q of the mechanical system is greater than one.

With simple earphones, the low frequency performance deteriorates when the air volume boundaries are leaky or nonrigid. The resulting undesirable effects on low-frequency performance can be minimized by introducing a second degree of freedom in the form of a resonant chamber loading the back of the diaphragm.

The midrange "honking" effects characteristic of under-damped earphones are avoided by "fluid damping" the voice coil with silicone grease. The resulting Q, being approximately equal to unity, provides excellent acoustic performance. Also, the grease in the air gap of the magnetic circuit results in good heat conduction from the voice coil to the metal parts of the housing and magnetic structure thus decreasing the probability of electrical burn-out.

Response out to 15,000 cps without sacrificing good air-moving capability (necessary for low-distortion bass response when air leaks are present) is obtained by using a three-inch diameter diaphragm possessing the "proper" break-up characteristics. "Proper," in this context, unfortunately has the very unprecise import that the resulting high-frequency performance sounds satisfactory to the designer and his associates. This criterion of "properness" is forced upon the engineer because of the uncertainties regarding developed sound pressures which arise as wavelengths approach the cavity dimensions associated with couplers and human ears.

Appendix

At low enough frequencies the earphone diaphragm can be considered a rigid impenetrable piston looking into a chamber of volume V containing a constant mass of ideal gas at pressure P. At audio frequencies, the vibrating diaphragm compresses the air adiabatically. Thus, the following equation of change applies (where k is the ratio of specific heats and c is a constant):

$$PV^k = c.$$

Differentiating gives

$$PkV^{(k-1)}dV + V^k dP = 0,$$

or

$$dP = b\, dV \tag{1}$$

where

$$b = -kP/V.$$

We can make the approximation of equating differentials to small changes since the diaphragm displacement is much, much smaller than the volume of the coupler. Therefore, if p is the incremental change in pressure due to motion of the diaphragm, v is the corresponding incremental change in volume, and b is essentially a constant, the following comes directly from (1):

$$p = bv \qquad \text{with } v \ll V.$$

If the displacement of the diaphragm from its equalibrium position is x, and A its area, then $v = Ax$, and it immediately follows that

$$p = bAx. \tag{2}$$

Sound pressure is thus seen to be directly proportional to diaphragm excursion.

The mechanical system formed by the earphone driver has a single degree of freedom and is described completely by the familiar equation (where M is the moving mass, R is the damping coefficient, S is the stiffness and $f\cos\omega t$ is the sinusoidal forcing function),

$$M\ddot{x} + R\dot{x} + Sx = f\cos\omega t. \tag{3}$$

Using complex phasor notation the steady-state solution of (3) can be directly written

$$-\omega^2 MX + j\omega RX + SX = F \tag{4}$$

where, for instance,

$$X = (x_{\max})2^{-1/2}e^{j(\omega t+\phi)}.$$

Eq. (2), when written in phasor form, becomes

$$P = bAX. \tag{5}$$

Combining (4) and (5) produces the expression for steady-state sound pressure,

$$-\omega^2 MP/bA + j\omega RP/bA + SP/bA = F,$$

or

$$P = FbA/(S - \omega^2 M + j\omega R). \tag{6}$$

P_{ref} is a convenient reference pressure obtained by letting ω equal zero in (6);

$$P_{\text{ref}} = FbA/S.$$

The relative sound pressure level (RSPL) is defined as $20\text{Log}_{10}|P/P_{\text{ref}}|$;

$$\text{RSPL} = 20\text{Log}_{10}\,|S/(S - \omega^2 M + j\omega R)|. \tag{7}$$

Eq. (7) can be put into a much more useful form by normalizing it. The following ratios have physical significances:

$\omega_0 = (S/M)^{1/2}$ angular frequency of natural resonance
$g = \omega/\omega_0$ forced frequency ratio defining the frequency of the driving function
$Q = \omega_0 M/R$ a common measure of system damping.

Substituting these symbols in (7) produces the following normalized expression for the RSPL:

$$\text{RSPL} = -10\text{Log}_{10}\,[(1 - g^2)^2 + (g/Q)^2]. \tag{8}$$

Fig. 9 shows (8) plotted for different values of Q. The relative sound pressure response of a simple earphone is completely determined by stating the Q at resonance.

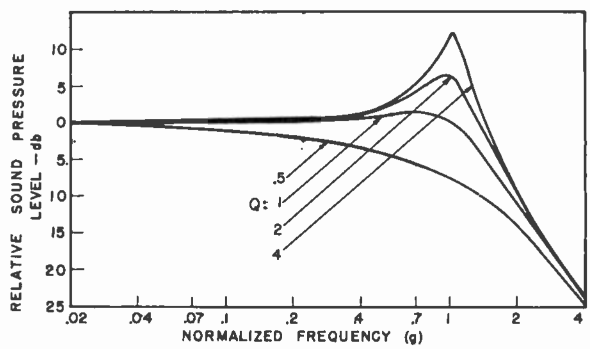


Fig. 9—Eq. (8) is plotted for different values of Q (the damping constant).

A Versatile Tone Control Circuit and Preamplifier

PAOLO SOARDO

Summary—A tone control circuit is described which allows a large choice in the high- and low-frequency attenuation or boost and in the position of the tone compensation curves over all the acoustic spectrum. These two controls, which are completely independent, are obtained with structurally simple RC networks. This tone control is included in a preamplifier for music reproduction. The compensation and equalization curves are reported in the paper and the characteristics of the complete preamplifier are described.

Introduction

MODERN SYSTEMS of music reproduction are usually equipped with tone controls which are intended to compensate for room acoustics and system defects, or simply to introduce at the will of the listener attenuation or boost in the low- or high-frequency ranges of the acoustic spectrum. A good tone compensation can only be obtained with two separate and independent tone controls (bass and treble). Of course, the introduction of these compensations must leave unaltered the signal level in the midfrequency range. A flexible tone control circuit must allow a large choice in the high- or low-frequency attenuation or boost, and in the position of the compensation curves over all the acoustic spectrum. Some papers[1,2] have already described tone controls of this kind, but the circuits utilized were generally somewhat complicated.

In this paper it will be shown that it is possible to design comparatively simple circuits featuring the above characteristics. This aim can be achieved while the number of capacitors is minimized: in particular, an RC circuit with a single capacitor allows one to obtain, with various resistance valves, bass boost or cut as shown in Fig. 1 (page 196), all curves having the same turnover or roll-off frequency. Moreover, a change in the capacitance value leaves unaltered the attenuation or boost, but introduces a displacement of the complete "fan-shaped" set of curves of Fig. 1 along the frequency axis; that is, a displacement of the roll-off or turnover frequencies, that we shall here after simply call "critical frequencies." The same holds of course for the treble case (Fig. 2).

For a fixed critical frequency attenuation or boost can be varied in discrete steps (Figs. 1 and 2). For its accuracy, this kind of selection was preferred to a continuous one for professional equipment. In the preamplifier here described, the complete set of curves for a fixed critical frequency is composed of fifteen curves about 3 db apart, giving up to 21 db of attenuation or boost. A choice is allowed among three critical frequencies for both bass and treble compensation.

These tone controls were included in a preamplifier for music reproduction to be placed between a variable reluctance pick up (output level about 10 mv) and a power amplifier. which requires 200-mv input for full power output.

Tone Control Characteristics

In order to prevent interaction between bass and treble controls, these controls were placed into two separate stages. Each stage is composed of a cathode follower supplying a triode feedback amplifier (Fig. 3). Provided that the gain without feedback is high, the gain with feedback can be written, apart from a minus sign, as follows:

$$K = \frac{Z_1}{Z_2}. \tag{1}$$

The mid-frequency gain, where compensation is not required, was chosen to be equal to one; consequently in this frequency range Z_1 and Z_2 must be equal.

One can also observe that the desired boost and attenuation curves (Figs. 1 and 2) are symmetrical about the 0-db axis; it follows that a boost curve can be obtained from the corresponding attenuation curve by simply interchanging Z_1 and Z_2.

The Bass Control

Critical frequencies must be shifted at will, this requirement can be met if Z_1 and Z_2 are RC networks with only one capacitor in the two impedances. In this way one can vary attenuation or boost by changing some resistances and obtain frequency shifting by simply replacing one condenser, the two operations being completely independent. Moreover, these networks have a single zero-pole pair, and this meets the requirements of the tone control.

After some trials, the network shown in Fig. 4 was chosen. In this network, impedances Z_1 and Z_2 do not appear at a first glance, but they are immediately evident after a Δ/Y transformation. Referring to Fig. 3, one gets

$$Z_1 = \frac{R_1}{pC(R_1 + R_2) + 1}, \quad Z_2 = \frac{R_2}{pC(R_1 + R_2) + 1} \tag{2}$$

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[1] F. Langford-Smith, Ed., "Radiotron Designer's Handbook," Wireless Press, Sydney, Australia, pp. 660–661; 1953.

[2] R. H. Rose, "An adjustable shelf-type equalizer with separate control of frequency and limiting attenuation and amplification," IRE Trans. on Audio, vol. AU-9, pp. 112–117; July/August, 1961.

where p is the complex frequency. The third impedance of the Y network appears in series with the grid and is then of no matter. The voltage gain K_b of the bass control can then be written as

$$K_b = \frac{R + Z_1}{R + Z_2} = \frac{p + \dfrac{R + R_1}{CR(R_1 + R_2)}}{p + \dfrac{R + R_2}{CR(R_1 + R_2)}}. \tag{3}$$

With the substitutions

$$w_n = \frac{R + R_1}{CR(R_1 + R_2)}, \qquad w_d = \frac{R + R_2}{CR(R_1 + R_2)}, \tag{4}$$

one gets

$$K_b = \frac{p + w_n}{p + w_d}. \tag{5}$$

In the mid- and high-frequency range ($w \gg w_n$, $w \gg w_d$), the gain is one as required; in the low-frequency range we obtain attenuation or boost if respectively $w_n < w_d$ or vice versa. The low-frequency gain can be written as the zero-pole ratio

$$K_{bb} = \frac{w_n}{w_d} = \frac{R + R_1}{R + R_2}. \tag{6}$$

Let us examine the boost case. The fan-shaped set of curves can be obtained by changing the resistance values so as to satisfy (6) for the required gains, but w_n is to be held constant. These requirements can be met by setting $R = R_2$, from which one can deduce

$$w_n = \frac{1}{RC}. \tag{7}$$

From (6) one then gets immediately

$$R_1 = (2K_{bb} - 1)R \tag{8}$$

where K_{bb} is the desired boost.

The attenuation case is met by setting $R = R_1$, from which we get

$$w_d = \frac{1}{RC}. \tag{9}$$

From (6) we then obtain

$$R_2 = \left(\frac{2}{K_{ba}} - 1\right)R \tag{10}$$

where K_{ba} is the desired attenuation. If in (10) we make $R_2 = R_1$ and $1/K_{ba} = K_{bb}$, we obtain again (8). This means only that the same values of attenuation or boost can be obtained simply by interchanging impedances Z_1 and Z_2 as was already mentioned in the previous section.

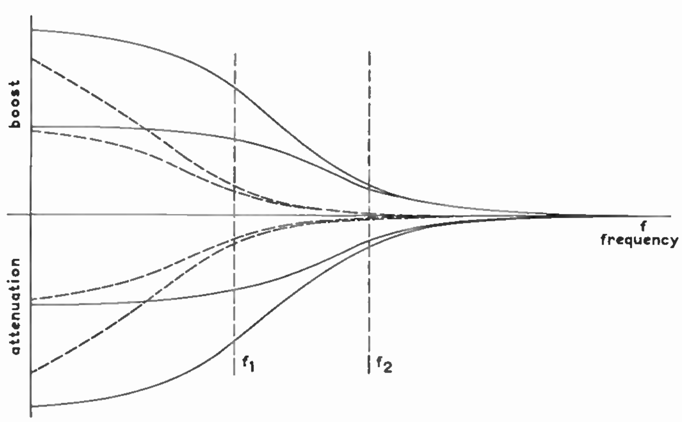


Fig. 1—Typical bass control curves. f_1 and f_2 are two critical frequencies.

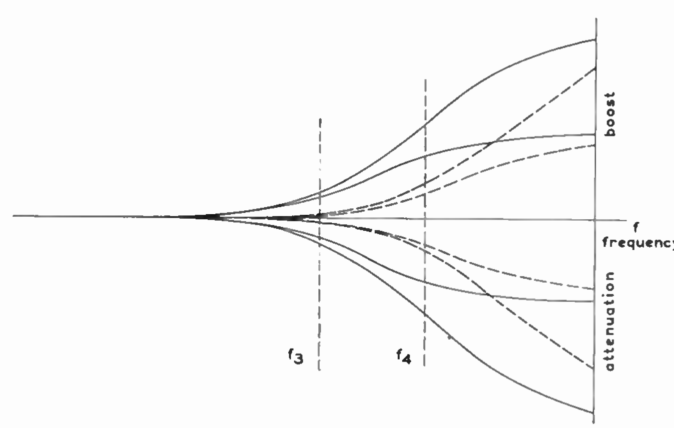


Fig. 2—Typical treble control curves. f_3 and f_4 are two critical frequencies.

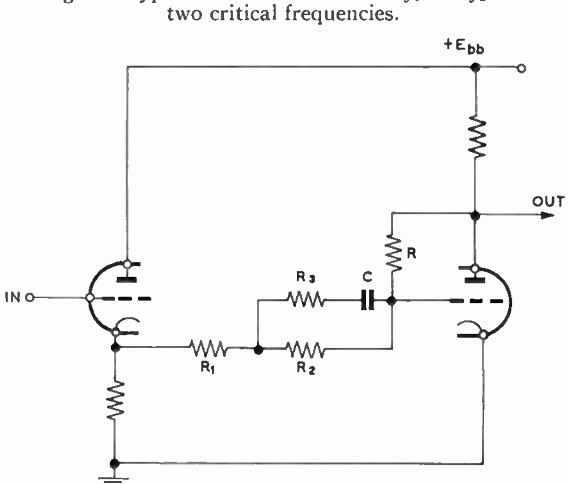


Fig. 3—Simplified diagram of the tone control circuits.

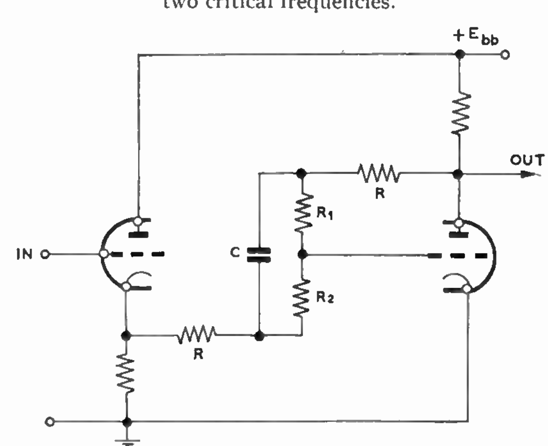


Fig. 4—Simplified diagram of the bass control circuit.

Eqs. (7), (8), (9) and (10) allow the determination of the circuit components. Fig. 5 shows the circuit diagram of the complete bass control, R is equal to 43 kilohms. The critical frequencies are 900, 440 and 200 cps, and attenuation and boost can be varied in 3-db steps. Figs. 6 and 7 show the complete set of experimental curves measured for two critical frequencies, 900 and 200 cps.

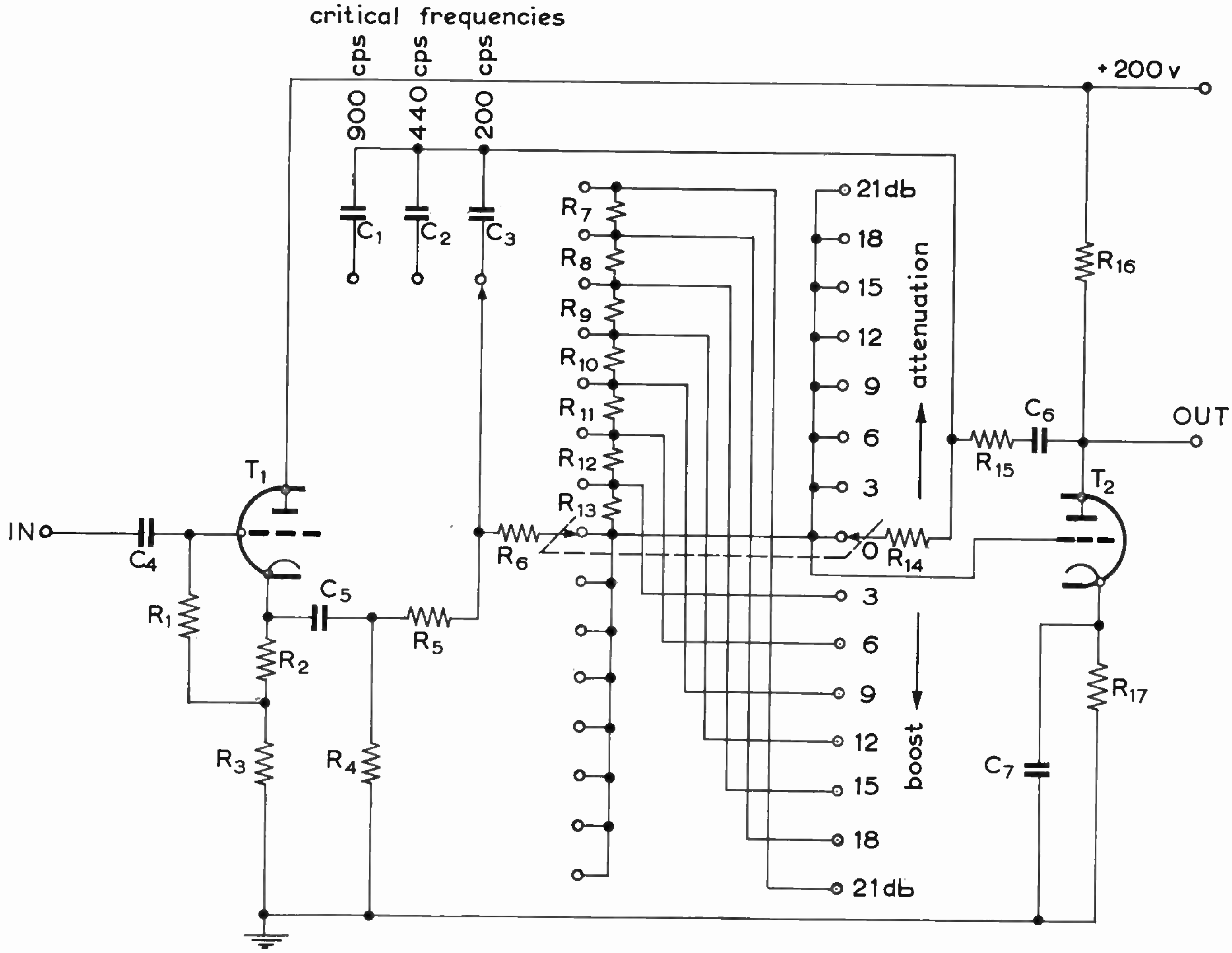
The Treble Control

The treble control was designed again according to the diagram of Fig. 3. Now low-frequency gain must be one, the other general requirements being the same as for the bass control.

A simplified diagram of the treble control for the boost case is shown in Fig. 8, one of the two feedback impedances is a pure resistance. One can write

$$Z_1 = R,$$

$$Z_2 = \left(R_1 + \frac{R_2R_3}{R_2+R_3}\right)\frac{p + \dfrac{R_1+R_2}{CR_1(R_2+R_3)+R_2R_3}}{p + \dfrac{1}{C(R_2+R_3)}}. \quad (11)$$



Components Values

$R_1=0,47$ MΩ	$R_{15}=43$ kΩ
$R_2=680$ Ω	$R_{16}=100$ kΩ
$R_3=100$ kΩ	$R_{17}=680$ Ω
$R_4=0,22$ MΩ	
$R_5=43$ kΩ	$C_1=4,7$ nF
$R_6=43$ kΩ	$C_2=10$ nF
$R_7=300$ kΩ	$C_3=22$ nF
$R_8=200$ kΩ	$C_4=2,2$ nF
$R_9=150$ kΩ	$C_5=4$ μF
$R_{10}=100$ kΩ	$C_6=0,47$ μF
$R_{11}=75$ kΩ	$C_7=100$ μF
$R_{12}=51$ kΩ	
$R_{13}=36$ kΩ	$T_1=\frac{1}{2}$ ECC 83
$R_{14}=43$ kΩ	$T_2=\frac{1}{2}$ ECC 83

Fig. 5—Complete diagram of the bass control.

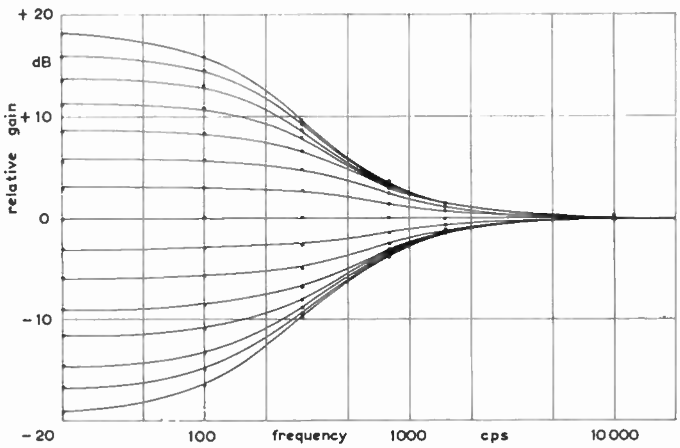


Fig. 6—Bass response curves. Critical frequency is 900 cps.

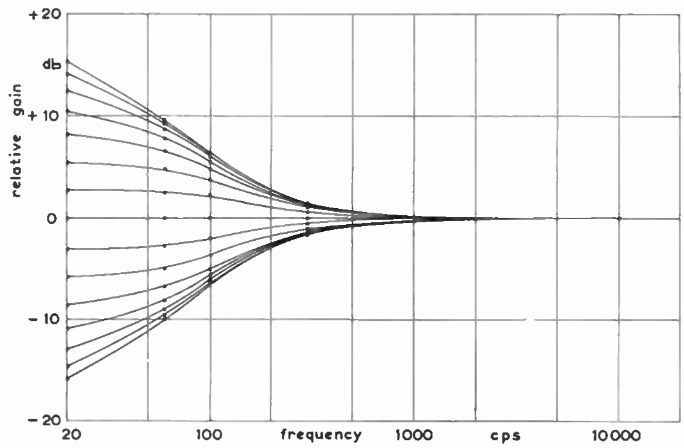


Fig. 7—Bass response curves. Critical frequency is 200 cps.

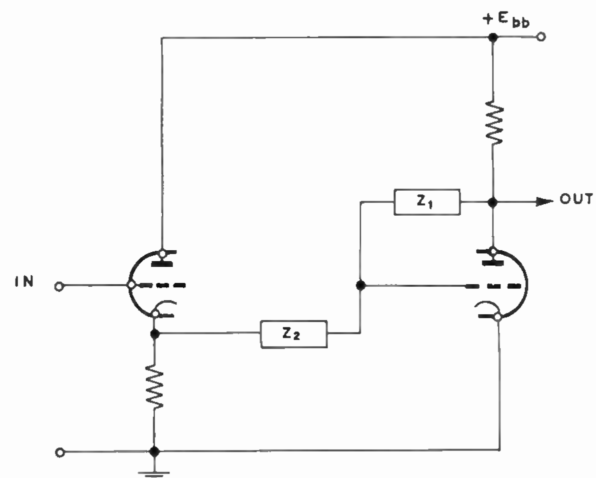


Fig. 8—Simplified diagram of the treble control circuit.

Then, in the boost case, the resulting gain is given by

$$K_{tb} = \frac{Z_1}{Z_2} = \frac{R}{R_1 + \dfrac{R_2R_3}{R_2 + R_3}} \frac{p + w_n}{p + w_d} \tag{12}$$

where

$$w_n = \frac{1}{C(R_2 + R_3)}$$

and

$$w_d = \frac{R_1 + R_2}{CR_1(R_2 + R_3) + R_2R_3}. \tag{13}$$

Low-frequency gain is one if $R_1+R_2=R=$constant.

On the other hand, the requirement $w_n=$constant can be met in (13) only if $R_2+R_3=$constant. We assume $R_2+R_3=R$; it follows also that $R_1=R_3$. The high-frequency gain can be written as

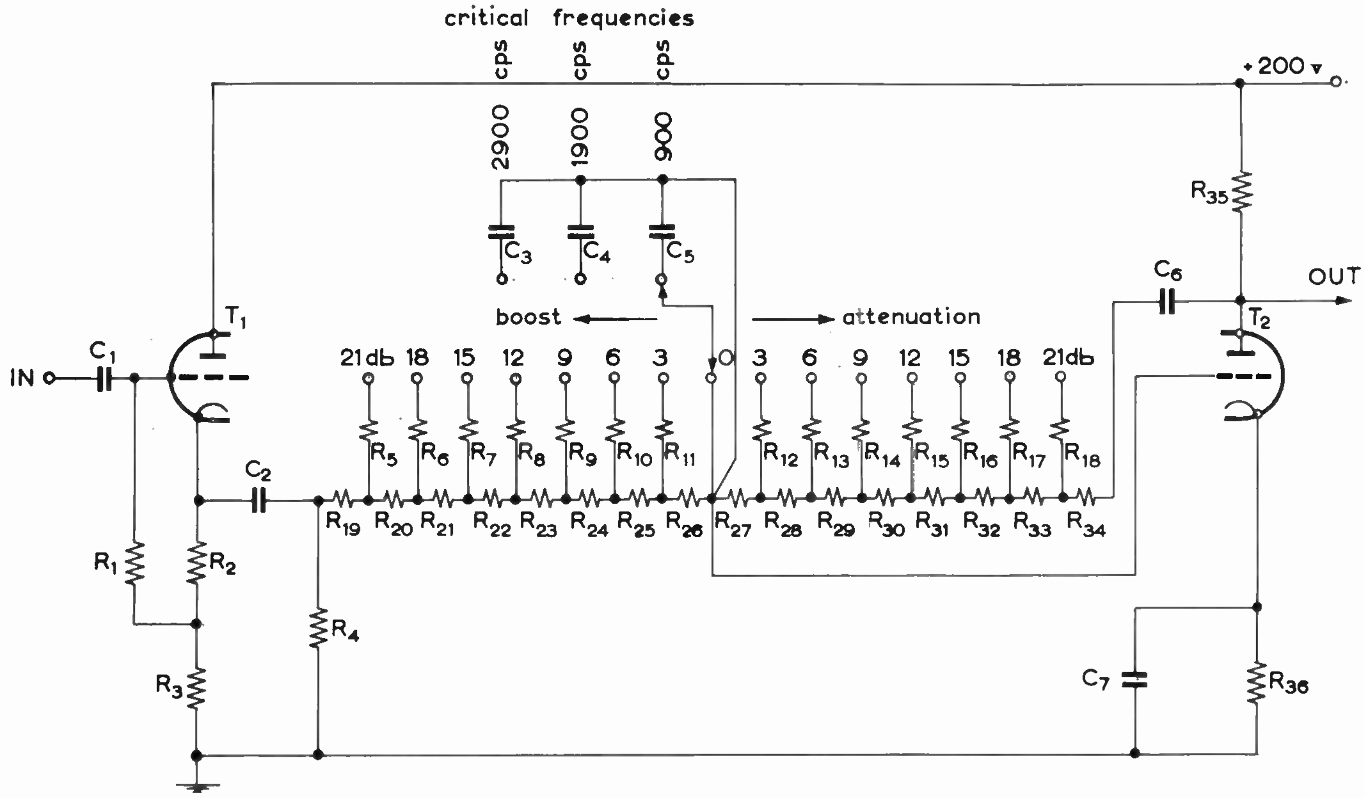
$$K_{tb} = \frac{R^2}{R_1(R + R_2)}. \tag{14}$$

With the valves $R_1=\alpha R$, $R_2=(1-\alpha)R$ one readily obtains

$$\alpha = 1 - \sqrt{1 - \frac{1}{K_{tb}}}. \tag{15}$$

This equation is not applicable to the treble attenuation, this case can be dealt with by simply interchanging impedances Z_1 and Z_2. This means that in (15) K_{ta} is substituted for $1/K_{tb}$, where K_{ta} is the attenuation required.

Eqs. (13) and (15) allow the determination of the circuit component valves. Fig. 9 shows the circuit diagram of the complete treble control, R is equal to 250 kilohms. The critical frequencies are 900, 1900 and 2900 cps, and attenuation and boost can be varied in 3-db steps. Figs. 10 and 11 show the complete set of curves for two critical frequencies, 900 and 2900 cps.



Components Values

$R_1=470$ kΩ	$R_{17}=16$ kΩ	$R_{33}=5,1$ kΩ
$R_2=680$ Ω	$R_{18}=11$ kΩ	$R_{34}=11$ kΩ
$R_3=100$ kΩ	$R_{19}=11$ kΩ	$R_{35}=100$ kΩ
$R_4=220$ kΩ	$R_{20}=5,1$ kΩ	$R_{36}=680$ Ω
$R_5=11$ kΩ	$R_{21}=6,8$ kΩ	
$R_6=16$ kΩ	$R_{22}=10$ kΩ	$C_1=2,2$ nF
$R_7=22$ kΩ	$R_{23}=16$ kΩ	$C_2=0,47$ μF
$R_8=33$ kΩ	$R_{24}=27$ kΩ	$C_3=220$ pF
$R_9=47$ kΩ	$R_{25}=39$ kΩ	$C_4=330$ pF
$R_{10}=75$ kΩ	$R_{26}=130$ kΩ	$C_5=680$ pF
$R_{11}=110$ kΩ	$R_{27}=130$ kΩ	$C_6=0,47$ μF
$R_{12}=110$ kΩ	$R_{28}=39$ kΩ	$C_7=100$ μF
$R_{13}=75$ kΩ	$R_{29}=27$ kΩ	
$R_{14}=47$ kΩ	$R_{30}=16$ kΩ	$T_1=\frac{1}{2}$ ECC 83
$R_{15}=33$ kΩ	$R_{31}=10$ kΩ	$T_2=\frac{1}{2}$ ECC 83
$R_{16}=22$ kΩ	$R_{32}=6,8$ kΩ	

Fig. 9—Complete diagram of the treble control.

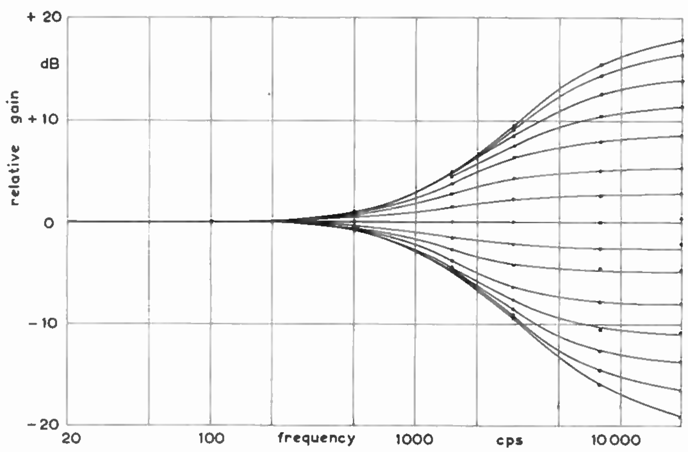


Fig. 10—Treble response curves. Critical frequency is 900 cps.

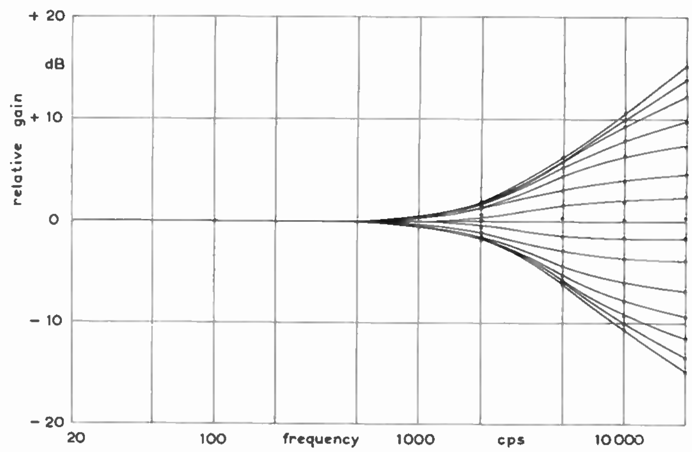


Fig. 11—Treble response curves. Critical frequency is 2900 cps.

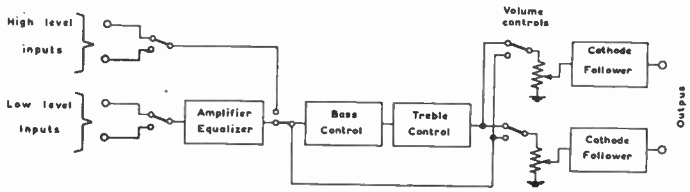
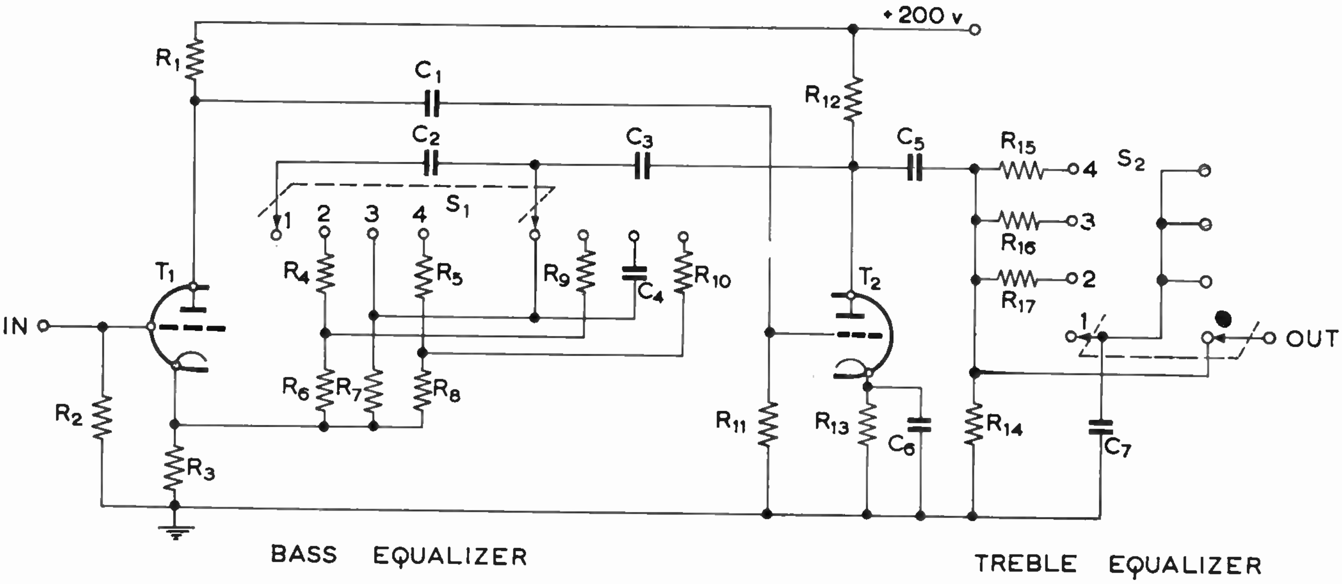


Fig. 12—Block diagram of the complete preamplifier.



Components Values

$R_1=100$ kΩ	$R_{15}=200$ kΩ
$R_2=39$ kΩ	$R_{16}=120$ kΩ
$R_3=1,5$ kΩ	$R_{17}=90$ kΩ
$R_4=24$ kΩ	$C_1=220$ nF
$R_5=22$ kΩ	$C_2=10$ nF
$R_6=5,1$ kΩ	$C_3=3$ μF
$R_7=30$ kΩ	$C_4=2,5$ nF
$R_8=10$ kΩ	$C_5=47$ nF
$R_9=270$ kΩ	$C_6=100$ μF
$R_{10}=150$ kΩ	$C_7=500$ pF
$R_{11}=0,47$ MΩ	
$R_{12}=100$ kΩ	$T_1=\frac{1}{2}$ ECC 83
$R_{13}=1,5$ kΩ	$T_2=\frac{1}{2}$ ECC 83
$R_{14}=1$ MΩ	

Fig. 13—Circuit diagram of the amplifier-equalizer.
Bass equalizer: position 1: flat
position 2, 3 and 4: see Fig. 14
Treble equalizer: position 1: flat
position 2, 3 and 4: see Fig. 15.

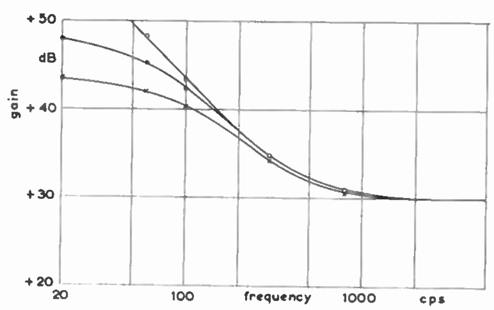


Fig. 14—Bass equalization curves: × position 2 (Fig. 13)
○ position 3
● position 4.

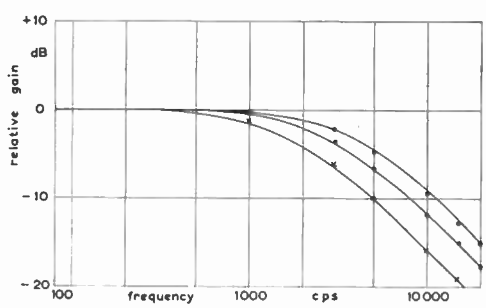


Fig. 15—Treble equalization curves: ○ position 2 (Fig. 13)
● position 3
× position 4.

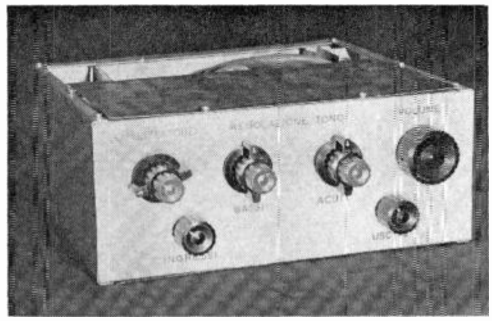


Fig. 16—Over-all view of the preamplifier. (Construction I.E.N.g.f.)

The Complete Preamplifier

The tone controls previously described were included in a preamplifier for musical reproduction, the block diagram of which is shown in Fig. 12. There are two inputs for low-level signals (gain 30 db), two "flat" inputs for high-level signals (0-db gain) and two separate outputs. The tone controls can be bypassed independently for both outputs.

The functions of amplification and equalization of low-level signals are performed by a double triode plate-to-cathode-feedback stage, the diagram of which is shown in Fig. 13. The equalization curves have been chosen according to current data, and, in order to achieve more flexibility, it was decided to control independentely the bass and treble equalization.

Bass boost is obtained with RC networks placed in the feedback loop. These networks employ as few capacitors as possible in order to minimize switching transients. The curves obtained are shown in Fig. 14; mid-frequency gain is 30 db.

Treble equalization is obtained through RC networks at the output of the second triode, the equalization curves are shown in Fig. 15.

Distortion, measured with a one-volt output which is five times the maximum signal required by the power amplifier, is less than 0.1 per cent with any setting of the various controls. Noise is more than 80 db below the maximum output level (200 mv).

Conclusions

The independent selection of attenuation or boost and critical frequency in the low- and high-frequency ranges of the acoustic spectrum gives a great flexibility to this tone control circuit. These features are obtained with structurally simple networks; however, owing to the large number of possibilities offered and to the accuracy given by the selection of attenuation and boost in discrete steps, the number of components is rather large but in any case not larger than for similar equipments. If a continuous variation of attenuation or boost is preferred, it seems possible to reduce substantially the number of components: the use of potentiometers in this circuit is now under consideration.

Acknowledgment

The author wishes to express his thanks to Prof. M. Soldi for his contributions during the design of the preamplifier and to Prof. C. Egidi for his valuable assistance.

A New High-Frequency Horn

PAUL W. KLIPSCH, LIFE MEMBER, IEEE

Summary—**A new horn was designed with reduced taper rate and mouth size as compared to an existing horn. Low-frequency response was improved by adding a flange or baffle. Comparison with the existing unflanged horn of similar length shows ability to radiate a wavelength 20 per cent longer.**

The reduced exit angle provided an unexpected gain in high-frequency range and smoothness of response. With a new wide-range driver, the horn produces smooth response at frequencies more than an octave higher than the old horn and driver. Thus, the bulkier horn with 500–5000 cps response and a tweeter may be replaced with a single horn having 260–17,000 cps response.

Old Horn

THE KLIPSCHORN type K-5 horn has been in production since 1946.[1,2] The best wide-range driver units available perform well with this horn only to about 5000 cps. With the low-end crossover at 500 cps, it is seen that the resulting range is a trifle over three octaves, and the horn-driver combination can function best as a midrange or "squawker"[3] in a woofer-squawker-tweeter speaker system.

Development of New Horn

To reduce design limitation on the bass horn or woofer, it was deemed desirable to attempt to reduce the low-end cutoff of the mid-range horn. It was reasoned that the original horn with a free mouth (not flanged) could be modified in three steps as shown in Fig. 1. Fig. 1(a) depicts the original horn; Fig. 1(b) shows a horn of the same taper and shape but cut off to form a smaller mouth (about half as much area) with the low-frequency response restored with a flange. Elementary theory indicates that the mouth area required to radiate a given maximum wavelength into a solid angle of 2π steradians is only half that required to radiate the same maximum wavelength into 4π steradians. In Fig. 1(c) the taper is elongated to restore the original length but provide a lower taper cutoff frequency.

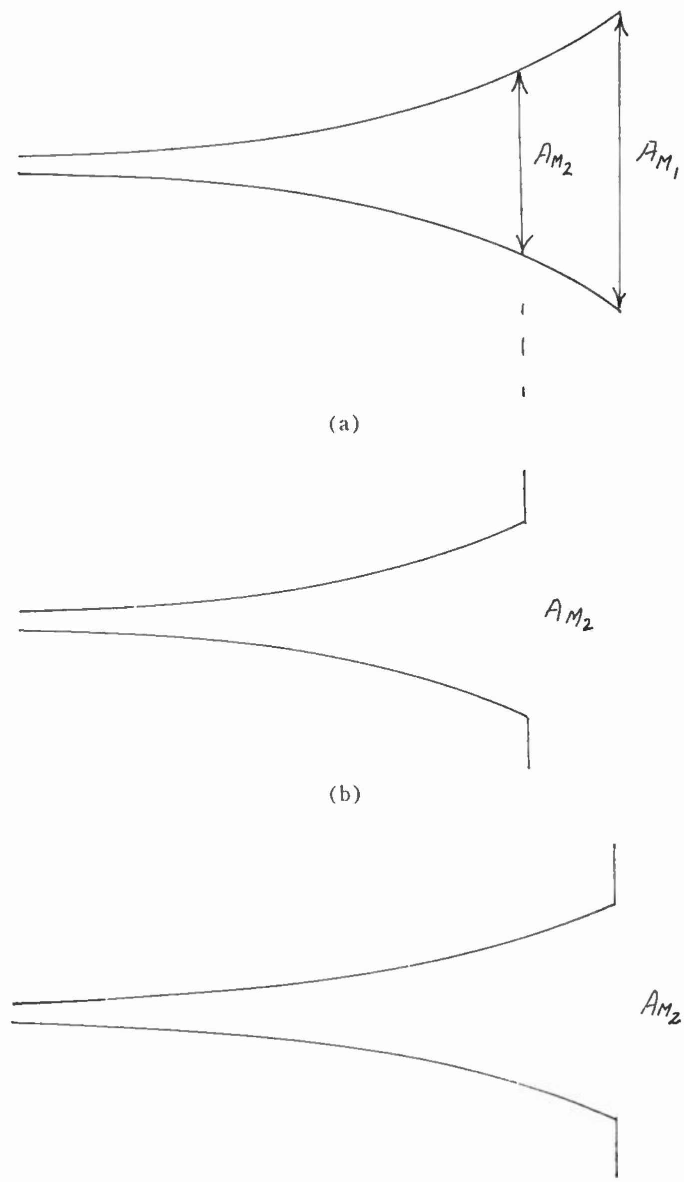


Fig. 1—Qualitative evolution of a flangeless horn to one of smaller size with flange and with improved low-end cutoff. (a) Original horn. (b) Original horn shortened and with flange added to reduce radiation angle and preserve low-range response in spite of smaller mouth. (c) Horn with decreased taper rate for improved low-end response.

New Horn

Such a horn was constructed. The old K-5-J horn doubles in area every 2.25 inches. As normally used, it occupies almost the entire top housing of the KLIPSCHORN® loudspeaker system, so that the housing

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The author is with Klipsch and Associates Inc., Hope, Ark.

[1] P. W. Klipsch, "A high quality loudspeaker of small dimensions," *J. Acous. Soc. Am.*, vol. 17, pp. 254–258; January, 1946.

[2] P. W. Klipsch, U. S. Patent 2,537,141; January 9, 1951.

[3] "Woofer" and "tweeter" have been accepted terminology in theater speakers since about 1928. The introduction of three-way speaker systems and this writer's production of such a speaker in 1950 suggested the need for another term. Because of the sound quality of a midrange speaker used alone, the onomatopeia "squawker" seemed appropriate. Since then four- and five-way speakers have been evolved with some claimed advantages but no contribution to nomenclature.

adds little or no baffle effect. As originally designed, the mouth was made larger than required for the lowest frequencies, so as to provide a straight taper section for control of radiation angle.

The new horn doubles in area in 2.7 inches and terminates in a mouth of about 0.4 times the area of the old horn. It is fitted with a wood flange of about three times the mouth area.

The new horn is designated K-400 for its designed lower crossover frequency.

Results

As expected, the low-end cutoff was reduced from 300 cps to about 240 cps. An unexpected result was a flattening of response above 3,000 cps. With a driver intended for midrange use, the droop with the old horn is nearly 10 db at 5,000 cps, whereas in the new horn the same driver offers response to beyond 6,000 cps. Fig. 2(a), next page, shows the response of the K-5-J horn with K-55-V drive unit. Fig. 2(b) shows the new horn with the same driver.

These results prompted the adaptation of the horn to a "wide-range" driver intended to cover both the squawker and tweeter range. Fig. 3(a) shows the result of this test.

Off-axis response for the new horn is shown in Fig. 3(b) for 15° off axis vertically, and in Fig. 3(c) for 35° off axis horizontally. There is less high-frequency deterioration off axis horizontally than with the K-5-J.

Off-axis response would desirably be compatible with normal toe in of flanking speakers for stereo, and the off-axis curves together with stereo listening tests indicate the same order of stereo geometry accuracy found for speakers using the K-5-J midrange horn.[4]

Fig. 4 shows the response of a squawker-tweeter combination or "two-way" top end using the K-5-J with K-55-V driver plus a K-77 tweeter, with a 6-db crossover network dividing at 5,000 cps. For comparison, Fig. 5 illustrates the response of a single top end employing the new horn in conjunction with a KLIPSCHORN bass unit. (The driver here was different from that used for Fig. 3.) Evidently, with the new horn and a wide-range driver it is possible to maintain a 10 db peak-trough ratio in a single unit covering a range from 260 cps to 17,000 cps.

The explanation for top-end improvement appears to be in increased concentration of high-frequency radiation on axis because of the reduction in the vertical exit angle. The old and new horns have exit angles in the horizontal plane of 90°; in the old horn, vertical angle was 60° vs 30° for the new horn. Off-axis response curves of the new horn are, however, quite satisfactory for the angular coverage intended.

The wide-range drive unit used to produce Fig. 3 was applied to the K-5-J and to a near copy by another maker; the on-axis curves are shown in Fig. 6. Note the 16- to 18-db peak-trough ranges, compared to 8 db for the same driver in the new K-400 horn.

Response curves were run at a distance from microphone to horn mouth of 24 inches. The environment was at a room corner (the horn being evaluated in conjunction with a corner-horn woofer) of a room of "ordinary" reverberation time deemed to be a good listening room. Slightly smoother responses can be measured outdoors but the differences are small.

All the response measurements were performed at a constant input of 2 volts corresponding to a nominal $\frac{1}{4}$ watt into a load of 16 ohms. Thus the sound pressures, well in excess of 100 db,[5] exhibit the high efficiency of the unit. Sensitivity at four feet with one watt input is 106 db with most of the drive units tested and measured with 500–2500-cps warbled tone. The power output capacity of all these speakers is in excess of three acoustic watts with negligible distortion. "Tolerable" FM distortion levels are assumed herein to be of the order of 0.1 per cent.

Mere smoothness of response or extended range is considered to be subordinate to freedom from distortion. For a given driver unit it is found that the horn affording the smoothest response also displays the least distortion.

Description of Horn

The new K-400 horn is of exponential taper from a throat of 0.70- or 1.00-inch diameter (depending on driver) to a mouth of approximately $5\frac{1}{2} \times 17$ inches. The area doubles in approximately 2.7 inches. The exit angle is 90° in one plane and approximately 30° in the other. Naturally, the expansion rates in the two planes are not equal, but the area expansion is still exponential. Since the mouth area is about 40 per cent of what was deemed adequate in the old horn, a flange of about three times the mouth area is used. Response curves with and without flange (Fig. 7, page 206) indicate the necessity of the latter to smooth the response in the bottom octave and to extend the cutoff some 50 cps. Note also the effect of the flange in eliminating the mouth-reflection dip at 470 cps.

Previous experiments with metal horns indicated an undesirable ring or hangover. The K-400 experimental prototype was of sheet iron, coated with damping compound on the outside. This did not prove satisfactory. A further experiment with cast aluminum exhibited the undesired ringing quality, but when the necessary flange was attached in the form of $\frac{3}{4}$-inch plywood board, the resonances disappeared and no evidence of hangover was detectable by ear or measurement.

Fig. 8 is a photograph of the horn with midrange-type

[4] P. W. Klipsch, "Stereo geometry tests," IRE Trans. on Audio, vol. AU-10, pp. 174–176; November–December, 1962.

[5] re 0.0002 microbar.

driver, on the same flange with a tweeter to form a wide-range top end.

Listening Tests

The evaluation is continuing, both objectively and in terms of listening quality. The comparisons indicate that the new horn with wide-range driver can supplant the two-way top end. A superiority of either has not yet been established. The interference pattern between 3000 and 6000 cps with the two-way top end has not been audible; its absence in the single top end at least produces a better looking curve.

Applications

The new horn is about the same length as a typical reflexed trumpet, but with a response which preserves precise articulation and affords intelligibility well above that of reflexed horn speakers. Its use would seem to be indicated wherever speech reproduction is depended upon for safety of life and property. A specific example would be in flight service stations where the customary direct-radiator speakers exhibit restricted frequency response and severe distortion at the usually high loudness levels employed. The author intends to use a smaller version (with a 500-cps cutoff) in his airplane where ambient noise levels of 103 db require high speaker output. (Some airplanes have been measured at 117 db with "flat" or "C-scale" settings of the sound level meter.) With restricted electrical power, the extra output afforded by a horn speaker plus the high articulation resulting from smooth, flat, extended response are expected to increase intelligibility by a wide margin.

An obvious application is of course for high-quality audio in the entertainment field for home or auditorium.

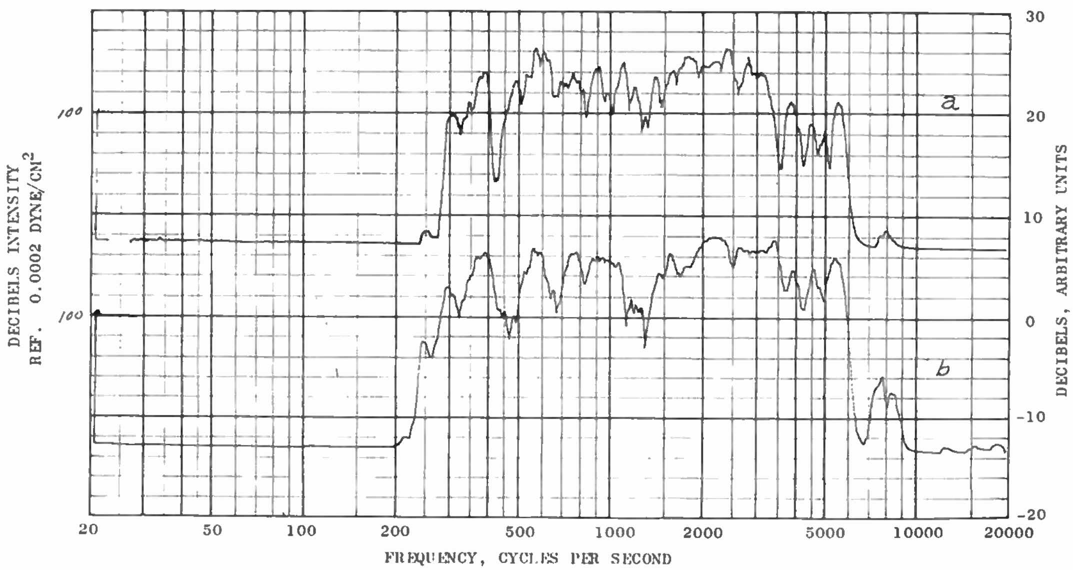

Fig. 2—Response of old and new horns. (a) K-5-J horn, K-55 drive unit; horn has 9×25 inch mouth, 20 inch length; area doubles in 2.25 inches. (b) K-400 horn, same driver unit; horn has flanged 5.6×17.3 inch mouth, 20 inches length; area doubles in 2.75 inches. Note improvement in response both below 500 and above 3500 cps.

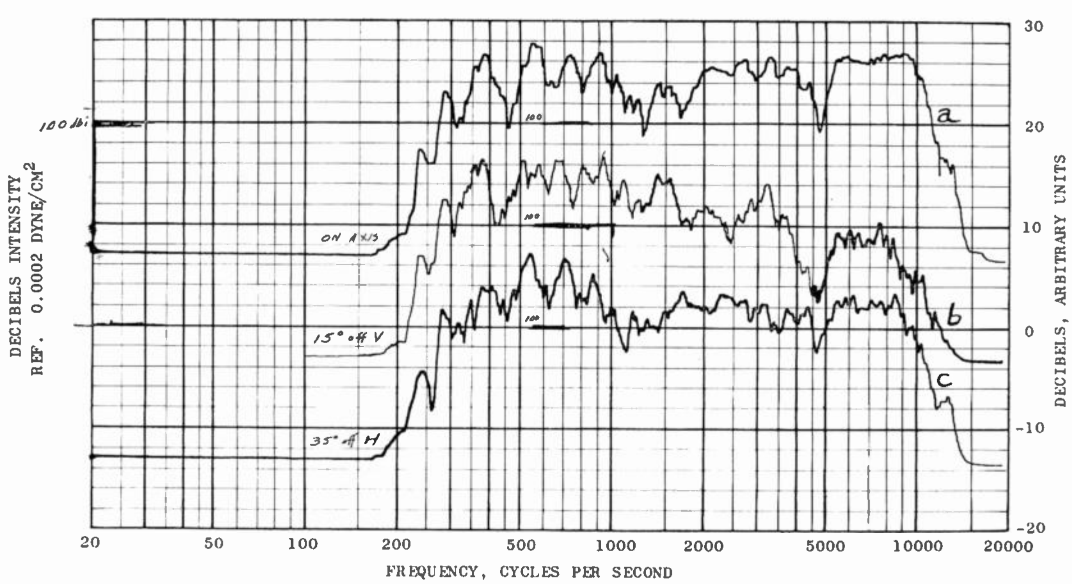

Fig. 3—Response of K-400 horn with wide-range driver unit suitable for two-way speaker system. (a) On axis. (b) 15° off axis vertically. (c) 35° off axis horizontally.

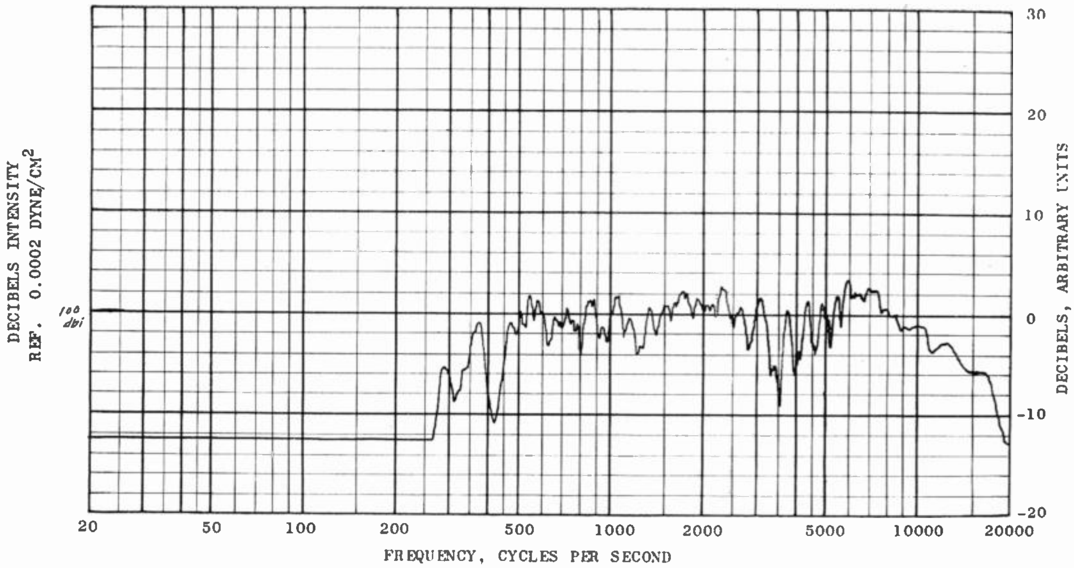

Fig. 4—Response of two-way "top end" for three-way speaker system for comparison with Fig. 3. The series of peaks and dips between 3 and 5 kc is due to crossover effect or the dipole effect of 2 radiators which cannot occupy the same space simultaneously. Response was on axis of coaxially mounted horn midrange and horn tweeter. The midrange horn was K-5-J.

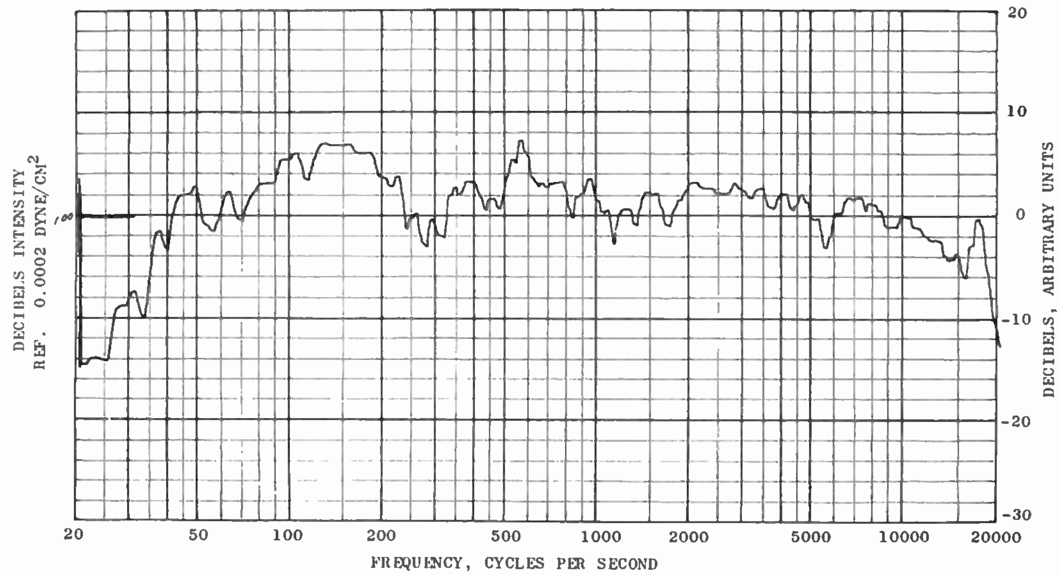

Fig. 5—Response of Klipschorn loudspeaker system with K-400 horn and wide-range driver unit. Treble attenuated 3 db in network. Differences between this curve and that of Fig. 3 are due to use of a different drive unit.

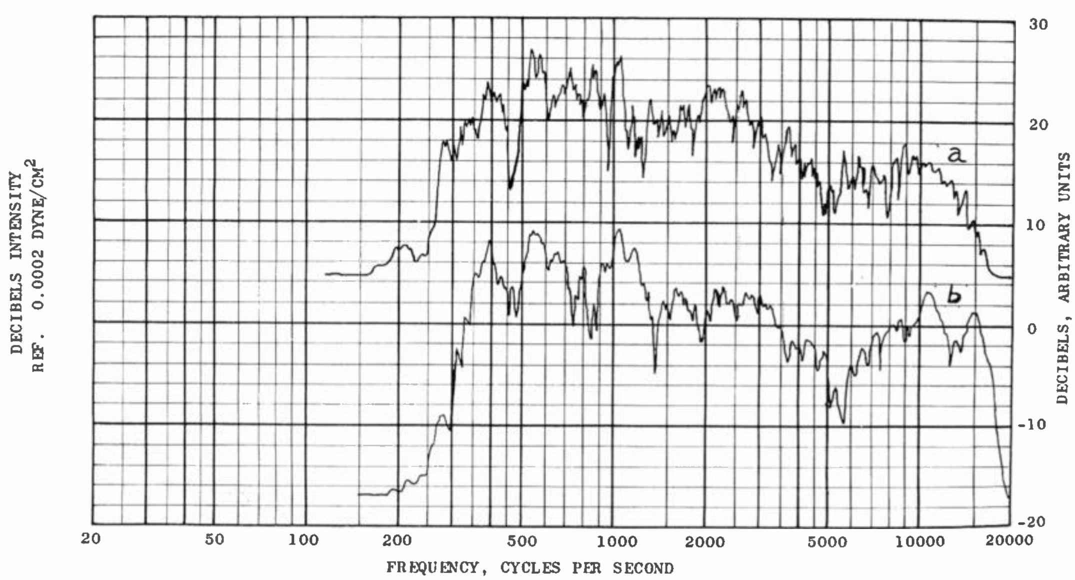

Fig. 6—Response of the wide-range driver of Fig. 3 on contemporary horns. (a) The K-5-J horn. (b) A metal horn of similar shape. Comparison with Fig. 3 indicates a peak-trough ratio of 8 db for the K-400 compared to 16 db for the K-5-J and 18 db for the metal copy.

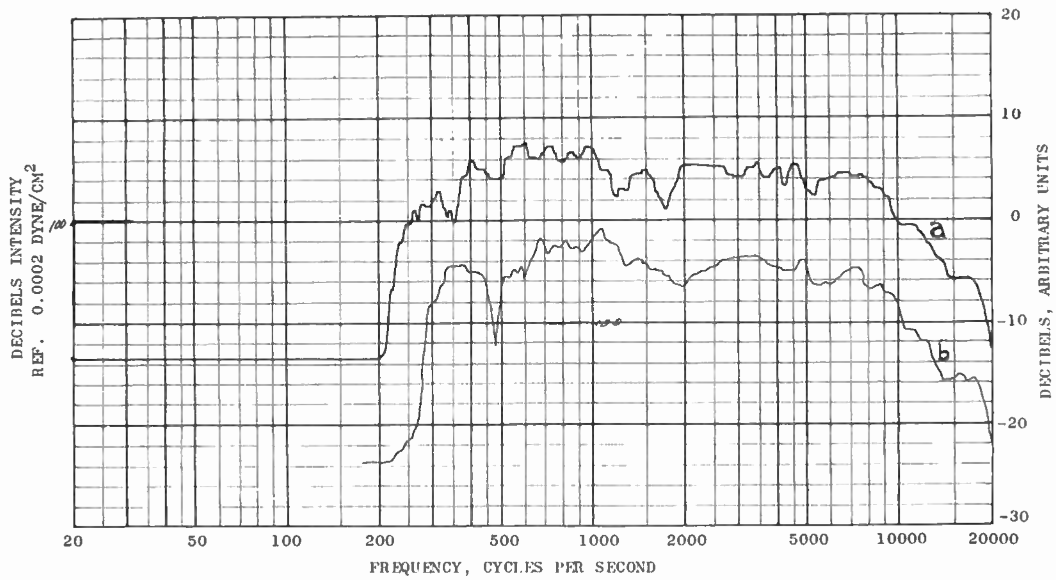

Fig. 7—Response of K-400 horn with and without flange. (a) With flange. (b) Without flange. Note 50 cps lower cutoff with flange, and reduction of mouth reflection dip at 480 cps.

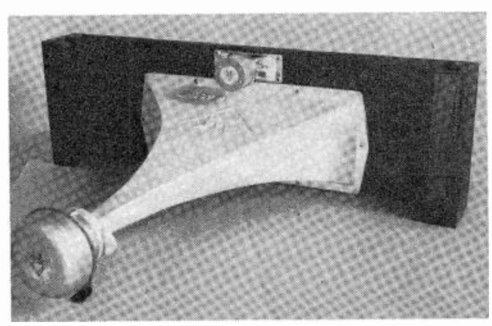

Fig. 8—New horn with midrange-type driver (K-55-V), mounted on baffle, with K-77 tweeter. This is typical assembly for wide-range "top end."

An Experimental 600-Foot Electrosonic Delay Line for the Navy Space Surveillance System

M. G. KAUFMAN, SENIOR MEMBER, IEEE

Summary—For many types of data transfer systems it is desirable that the data be delayed long enough so that the system is ready for the data. In the present Space Surveillance system it is desirable to delay data until the recorder is up to speed. The requirement for delaying the composite Inter-Range Instrumentation Group (IRIG) FM telemetering signals from a Space Surveillance site by one-half second was fulfilled by converting the phone-line frequencies into sound, which is then transmitted through a pipe long enough to provide the desired delay and then reconverted into electrical signals. The delayed output is essentially distortionless.

Introduction

The present instrumentation of the Space Surveillance system at the U. S. Naval Research Laboratory is designed to receive a satellite alert pulse, *via* telephone line, from each field station; this pulse can be used to start a recorder. This method of operating the recorders, as opposed to continuous operation, would result in a large reduction in the yearly budget for recording paper, but it does have the disadvantage that a small part of the satellite signal may be lost while the recorder is starting. However, if the signal could be delayed a few tenths of a second, the recorder would have time to start before the signal appeared on the record. The necessary delay can be obtained from a tape loop or a recording drum. Both methods have been tried but were rejected as unnecessarily complex for the task. It was found possible to obtain this delay by converting the electrical data signals at the field station into sound, transmitting them through a pipe long enough to provide the desired delay, then reconverting the sound to electrical signals for transmission over the telephone line in the usual manner. Fig. 1 shows a typical satellite recording after a delay of 0.53 sec. This system is simple, inexpensive and reliable and provides the required delay with negligible signal degradation.

The 0.53-sec delay was considered long enough to allow recording of the time of the event on the paper preceding the satellite pass signal itself. A special Universal Time readout technique, called parallel-time readout,[1] was developed for this purpose. Whereas the usual coded serial-time readout, located at the bottom edge of the record, took ten seconds of recording paper to display it unambiguously, the new parallel-time readout takes only 0.3 sec. Parallel-time readout is shown in Fig. 1 along the left-hand margin.

Laboratory Tests

The composite of eight data channels of standard IRIG FM telemetry signals (250 cps to 3 kc, Fig. 2) was speaker driven through a 600-foot long, 4-inch diameter fiber pipe which contained air at normal atmospheric pressure and ambient temperature (Figs. 3 and 4). These data were subsequently picked up at the far end of the pipe with a microphone, separated by filters, demodulated, recorded and then compared with the same composite signal similarly processed but not passed through the sonic delay. Tests showed that the method produced a delay of about one millisecond per foot of pipe length (Appendix I).

In the laboratory model of the delay line, several 180° bends were used to get the total length into one building. The turns caused no noticeable deterioration of the signals.

The effect of temperature, starting at 72°F, is more pronounced than that of humidity, being about one millisecond of delay decrease with each degree rise in temperature. However, by making the pipe longer than the required minimum delay, the decrease due to temperature was offset. Since "bulk delay" is required and not precision delay, the above technique was permissible.

The frequency response of the system was influenced markedly by the choice of the pipe's diameter. Even with a pipe diameter as large as four inches, it was found to fall off about 20 db per octave from a corner frequency of 400 cps. This response was compensated for by deemphasizing the strong lower frequencies with a simple high-pass RC filter having a corner frequency at 400 cps. This filter was inserted into the system ahead of the power amplifier.

Field Tests

Data from the Fort Stewart, Ga., Space Surveillance station were sent through the delay-line installation and then compared with nondelayed data (see Appendixes II and III for installation details). Each of the FM data channels was swept through its normal frequency-deviation range using a sawtooth signal waveform. All of the channels, in composite, were sent through demodulators and to the recorder. The results show good signal linearity and fidelity (Figs. 5 and 6, pages 209–210).

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The author is with the United States Naval Research Laboratory, Washington 25, D. C.

[1] The technique of putting time on a record normal to the paper's direction of travel, as opposed to serial-time readout. This timing technique was developed by Mr. L. O. Hayden of the Naval Research Laboratory.

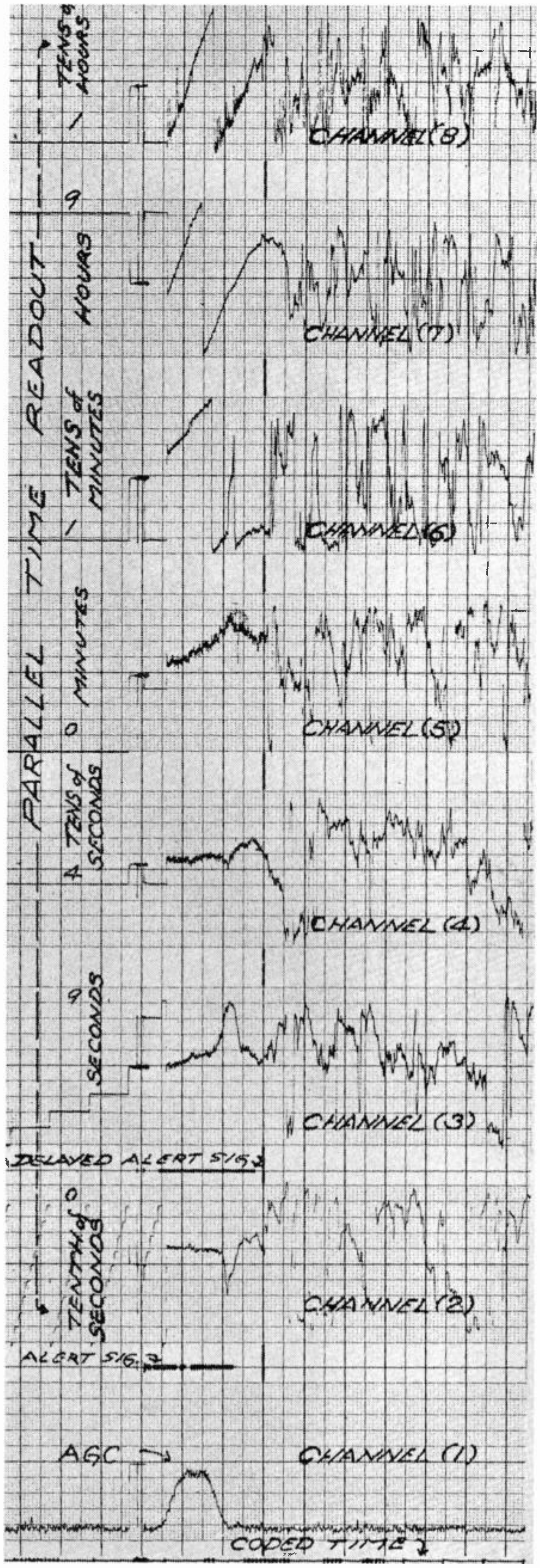

Fig. 1—Recording of a satellite signal delayed 0.53 sec, as determined by the difference between the extreme right ends of the "alert signal" and the "delayed alert signal" (shown above channels 1 and 2, respectively). Parallel-time readout indicates 19 hr-14 min-49.0 sec. Recorder speed was 10 mm per sec.

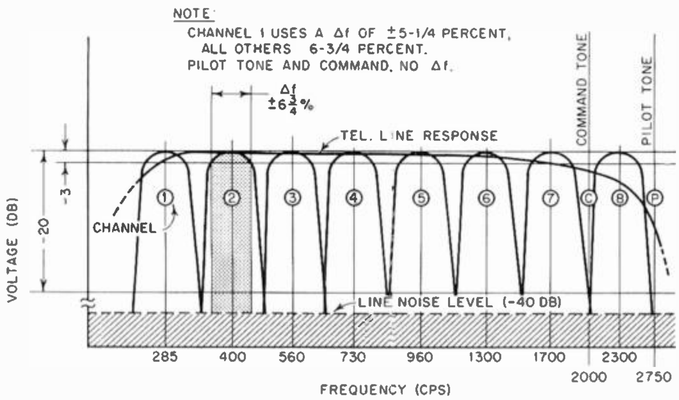

Fig. 2—Channel frequency allocations for one data line.

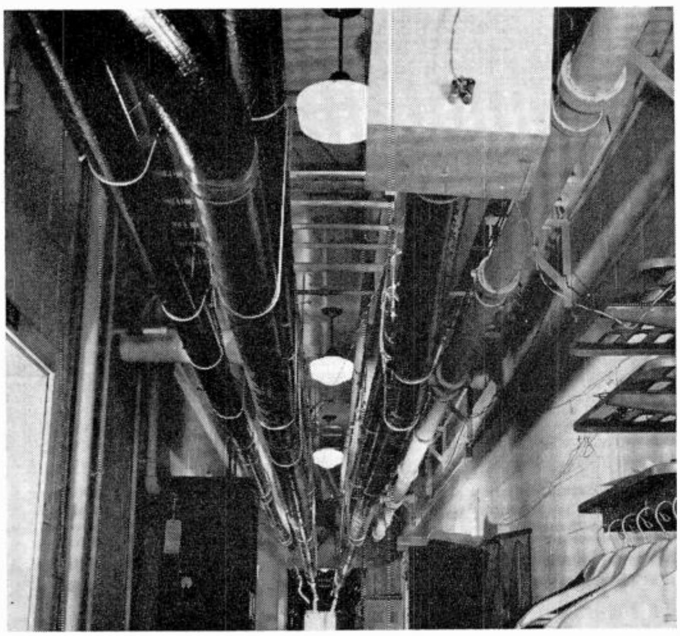

Fig. 3—Original laboratory prototype of sonic delay line, shown hanging from hallway ceiling. Plywood box contains driver unit.

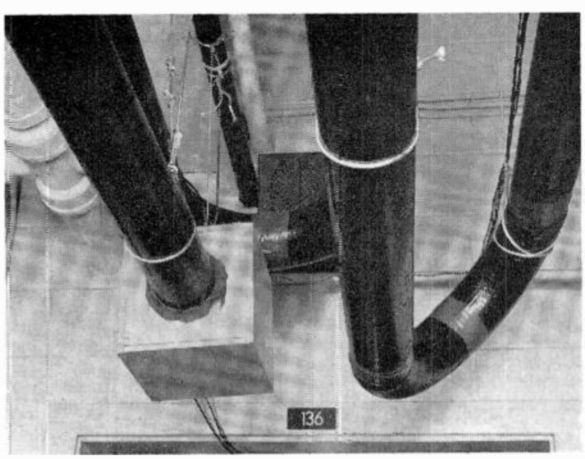

Fig. 4—Closeup of driven end of sonic delay line. Note that laboratory setup incorporated 180° bends in order to get full length into the corridor.

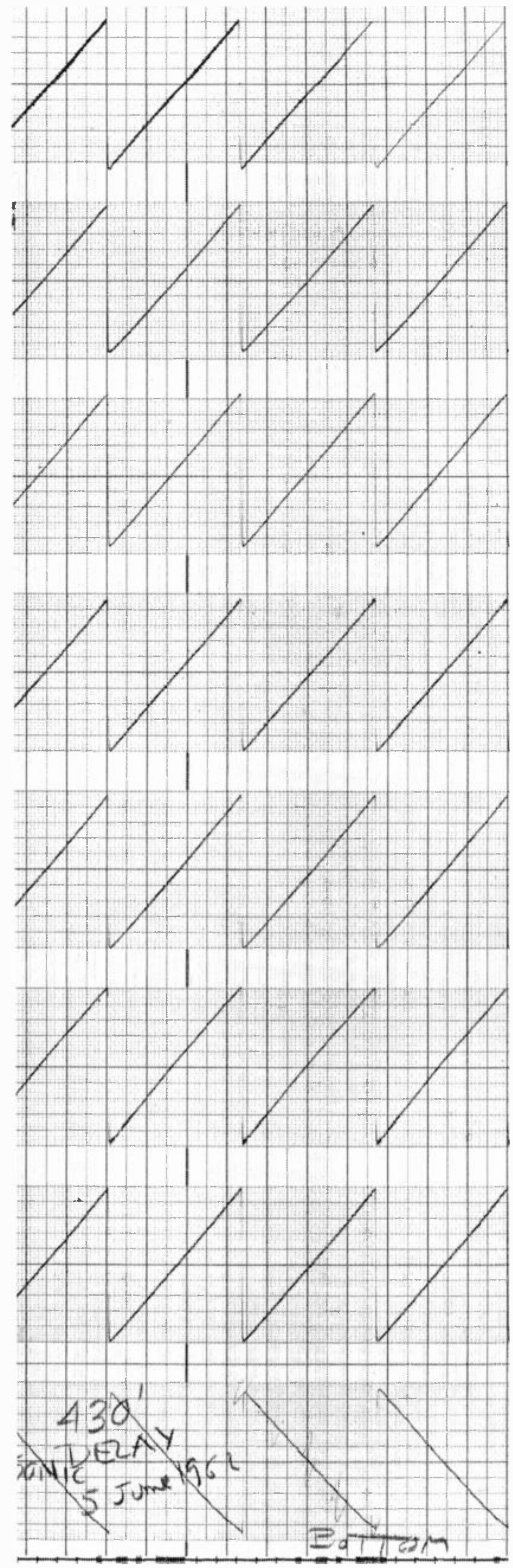

Fig. 5—Recording of first eight FM telemetry channels passing through pipe, showing each channel deviated over its normal range.

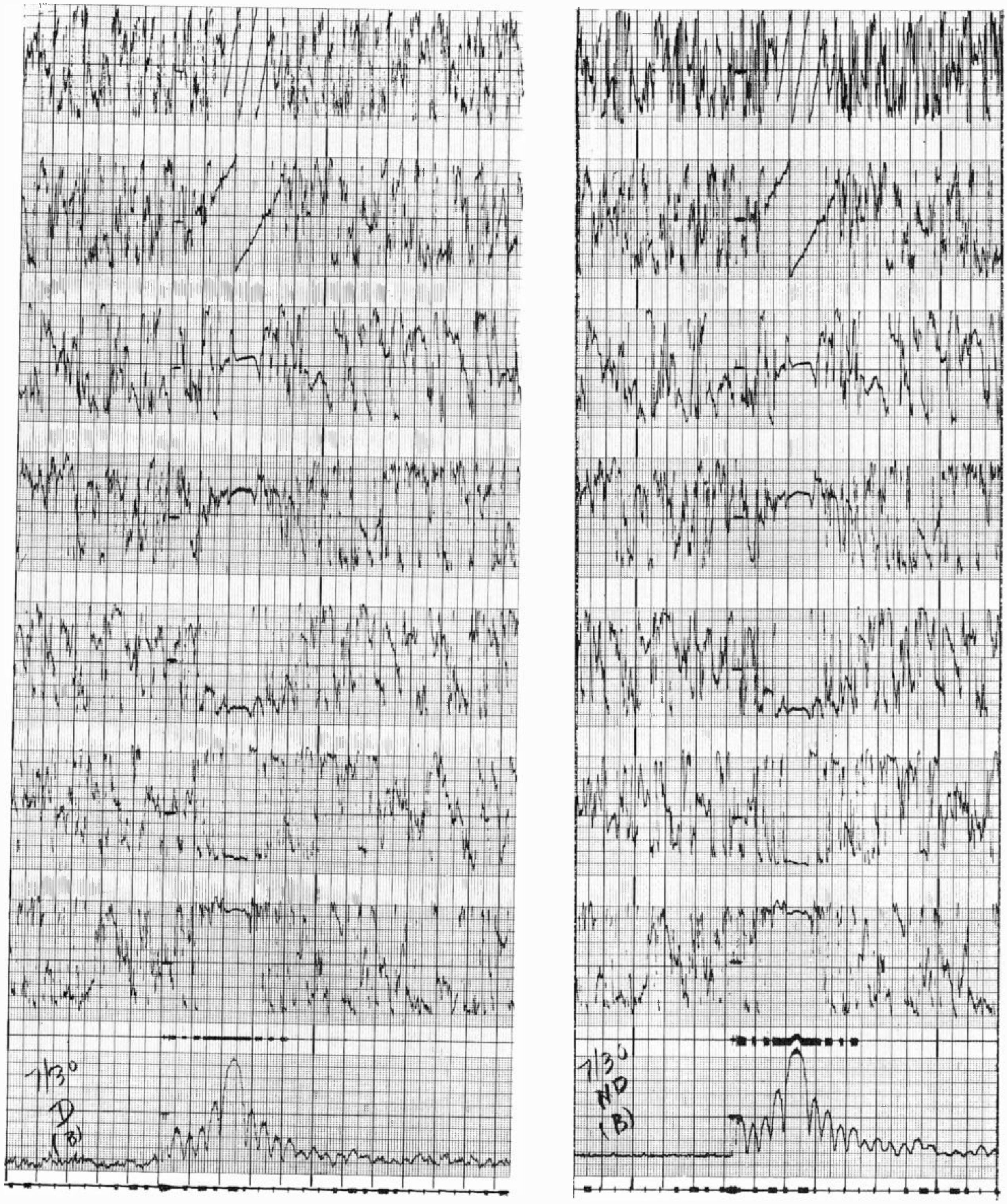

Fig. 6—Recordings of a satellite signal both delayed and undelayed. Note the resemblance of the two signals.

Remarks

Some interesting phenomena appeared during the experiment while using relatively short lengths of pipe. For example, the expected problem of reverberation, so handy for church-organ construction, was evidenced on the record of the low-frequency channel as distortion. This gradually disappeared through attenuation as the pipe length was increased, primarily because the signal traverses the length of pipe only once, whereas the "echo" must make three trips before it is back at the microphone end of the pipe again. Also, the very long pipe length decreases the echo resonance frequency to a low value (about 1 cps); consequently, an interference from the echo, or its crossproducts, becomes insignificant. The fact that FM and fiber pipe, instead of metal, were used made the system rather insensitive to external physical noise.

Fiber pipe is superior to metal for this application because it is less sound conductive, more impervious to outside noise pickup, lighter to handle, less responsive to physical impacts, rust and corrosion proof, inexpensive and easy to seal.

The proper working of the line can be attributed primarily to two parameters, namely: 1) The use of FM sound because the carrier made the line practically independent of amplitude variations. 2) The extreme length of the line reduced reflections to an insignificant amount because of the large attenuation sustained by any reflection.

Echoes showed up more prominently when using relatively short lengths of pipe. Even with a run of 250 feet of pipe, the echo was strong enough to show up as a slight deformation of the trace on channel 1. Channel 1, Fig. 2, is centered at 285 cps and is frequency deviated only ± 19 cps. Consequently, with such a low Beta[2] value, an interfering signal has more effect on the intelligence signals carried by the tones at the lower end of the telephone pass band. Since the line was long, the echoes were of a corresponding low frequency (resonance). Such "lows" pass through pipes with less attenuation than "highs." In order to equalize within the pipe, a foam rubber wedge was placed at the receiving end of the line. A simple RC high-pass filter was inserted electrically to help equalize the output of the microphone which also has a low-frequency preference. Combining the wedge and the high-pass filter with the frequency-response controls on the driver amplifier gave sufficient adjustments to compensate for the echo effect. The echo problem became less prominent as the line length was increased, becoming almost negligible when the length reached 600 feet (two football fields long). The resonance length was approaching dc and degradation due to echo practically disappeared.

The size of the foam rubber wedge was determined by trial and error, starting with a long wedge about two feet in length and reducing its length until the echo just started to appear. Such a shape seemed to allow the highs through but attenuated the lows. Both the driver and microphone were packed in putty and foam rubber to keep them physically isolated from the pipe itself and to prevent any sound signals or noise in the pipe material from getting into the data signals. No special techniques were required in mounting these items.

It was found that bends in the line had very little effect on the operation of the line. There probably was a slight loss as the wavefront propagated the turns; however, with 4-inch diameter pipe it was not noticeable in our tests. The line was laid straight out in the field. It had been folded back and forth in the laboratory because we did not have a 600-foot run.

Both a magnetic tape loop and a drum were tested and considered too bothersome for practical 24-hour automatic operation. The small signal levels, a dozen mechanical adjustments, tape-wear, maintenance, etc. deterred us from adding another rack of electronics unless it could be left free running.

Conclusions

The results of this experiment indicate that the method described is entirely practical for obtaining long signal delays with fidelity and simplicity. It is seen, therefore, that both the signal delay and the parallel-time-readout techniques, coupled with recorder start-stop action, can reduce the paper consumption substantially. Conservatively calculated, a saving factor of twenty is considered representative.

Appendix I

Delay and Resonance Frequency of a 600-Foot Pipe

The velocity of a sound wave in gases[3] is

$$V = \sqrt{\frac{rp}{d}}$$

where

V = velocity in feet per second
r = ratio of specific heats c_p/c_v, at constant pressure to that at constant volume
p = pressure in pounds per square foot
d = density in pounds per cubic foot.

For air under standard pressure at a temperature of 0°C,

$r = 1.40$
$p = 1{,}013{,}000$ dynes per cm^2
$d = 0.001293$ g per cm^3.

Substituting these values, the equation gives

[2] $B = \frac{\Delta f}{fm} = \frac{\text{Freq. Deviation}}{\text{Max. Mod. Freq.}}$.

[3] E. Hausmann and E. P. Slack, "Physics," D. Van Nostrand Company, Inc., Princeton, N. J.; 1948. See especially pp. 563 and 580.

$V = 33{,}130$ cm per sec
$= 1{,}087$ ft per sec.

The "delay" is calculated as follows:

$V = \Delta x/\Delta t$

where

V = velocity of the sound wave in ft per sec
Δx = distance in feet
Δt = delay time in seconds
$\Delta t = \Delta x/V$.

For

$\Delta x = 600$ ft long pipe
$V = 1{,}087$ ft per sec
$\Delta t = 600/1{,}087 = 550$ msec delay.

The velocity of sound is unaffected by changes in the pressure because density changes proportionately. On the other hand, changes in the temperature affect the density without influencing the pressure. This does cause a change in the velocity, which can be expressed as

$$V_2 = V_1 \sqrt{\frac{T_2}{T_1}}.$$

The resonance frequency of the pipe, closed at both ends, can be calculated from

$$f = \frac{V}{\lambda} = \frac{nV}{2l}$$

where

V = velocity in ft per sec
λ = wavelength in ft
f = frequency
n = number of antinodes (harmonics)
l = length of pipe in ft.

For a 600-foot-long pipe and $n = 1$,

$$f = \frac{nV}{2l} = \frac{1087}{2(600)} = 0.91 \text{ cps.}$$

Appendix II

Installation of Driver and Pickup System

In order to convert from an electrical to an acoustic signal, a heavy-duty permanent-magnet hermetically sealed University ID-60T driver unit is used as the sending-end transducer. It contains a built-in multi-impedance line transformer with power taps and is designed for operation between 70 cps and 12 kc. This unit, driven by a 30-watt Sherwood power amplifier, Model S-1000-2, is mounted in such a manner as to seal off the end of the pipe. A rubber washer is used to form the seal joining the driver to the pipe in order to reduce any mechanical vibrations between the driver and the pipe (Fig. 7).

At the receiving end of the pipe an Electro-Voice 649B microphone is used. The unit is mounted with rubber standoffs inside a wooden box with the air space packed with high-density putty. The box is mounted directly onto the pipe and closes off the opening, completing a closed column of air. The putty is used to decrease mechanical vibrations to the microphone and also to reduce the air volume in the box, in order to raise the resonance frequency above the operating range of the wave train. An acoustic filter[4] is mounted within the pipe at the pickup end in order to pass the high frequencies with minimum attenuation but reduce the amplitude of the low-frequencies end. The low frequencies can produce an echo effect within the pipe, thus contributing to signal degradation. Fig. 8 shows the experimental sonic delay line installation at the Fort Stewart, Ga. Space Surveillance site.

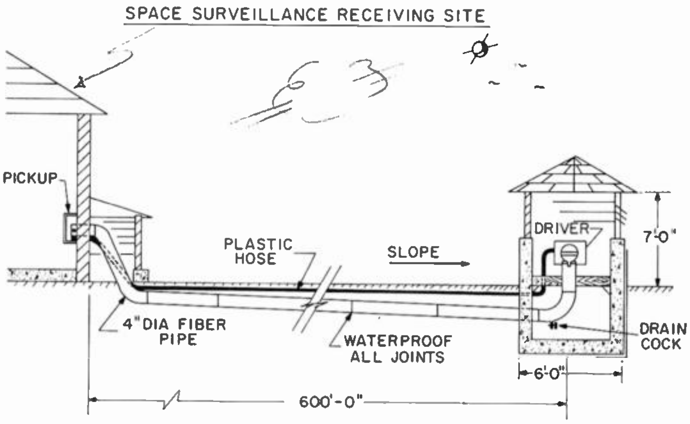

Fig. 7—Field-installation drawing of the sonic delay line. Plastic hose carries wire with IRIG data signals to driver unit.

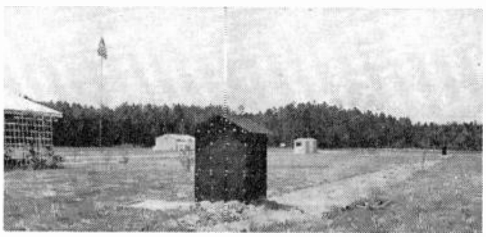

Fig. 8—Field-station delay line installation completed.

Appendix III

Acoustic Delay System Components

Item No.	Description	Number
1	Straight fiber pipe with couplings (each length 7 to 8 ft long)	600 ft.
2	Sherwood power amplifier, Model S-1000-2 (with case)	1
3	University ID-60T Driver	1
4	Electro-Voice Microphone, Model 649B	1
5	UTC transformer LS-14X (for mike matching)	1
6	HP preamplifier, Model 466A	1
7	Telephone wire	1500 ft.
8	Adhesive tape	3 rolls
9	Polyethylene 1-in-diameter hose	500 ft.

[4] Consists of foam rubber, wedge-shaped, 8 inches long and 4 inches thick at the base (dimensions determined by experiment).

ACKNOWLEDGMENT

The author is particularly indebted to his supervisor, R. L. Easton, head of the Space Surveillance Branch, for his support and advice, to R. Moore, the physicist who ran most of the laboratory tests, to the personnel at the Fort Stewart, Ga. Surveillance site, under the management of C. Fralick, for their excellent cooperation and to Capt. E. Van Ribbink, Cmdr. R. Carr and W. Schreech at the Naval Space Surveillance System Headquarters, Dahlgren, Va., for their cooperation.

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Correspondence

Comment on the History of the PTGA

The "Brief History of PGA, 1949–1952," which appeared in the May–June issue of the TRANSACTIONS, is very interesting. The PGA, however, persists in falsifying its age. This is the second time a report has appeared in the TRANSACTIONS which has omitted its earliest history.

In March of 1948, I suggested to the IRE Professional Groups Committee the need for a professional group in audio. An organizing committee was formed and a number of IRE members enlisted in the group. Although I no longer have a record of the organizing committee, I remember Ben Bauer, Leo Beranek, Art Curtiss and John Green in particular. The IRE Executive Committee accepted a petition to form an audio professional group on June 2, 1948.

The group was able to sponsor a symposium on high fidelity at the Radio Fall Meetings in Rochester, N. Y., November 10, 1948, and sponsored a session on audio again at the IRE National Convention in New York City, March 7, 1949. The organization was completed and officers elected in the spring of 1949. I was elected the first Chairman. Later that year, because of a change in my business, I found it necessary to resign the Chairmanship, and Leo Beranek was appointed by the Administrative Committee to fill the remainder of the term. As recorded in the "Brief History of PGA," Leo Beranek was elected Chairman at the March 9, 1950 meeting, but this was a reelection, not the first one.

In recalling that the Audio Professional Group is among the oldest, if not the oldest, of the professional groups, it is well to remember it was functioning even before its formal organization as a professional group in 1949.

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Manuscript received November 6, 1963.

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Mr. White is a member of Tau Beta Pi.

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ON

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INFORMATION FOR CONTRIBUTORS

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