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PGA News

CHAPTER NEWS

Chicago, Ill.

Karl Kramer, Chairman 1957–1958, reports as follows:

The 1957–1958 season proved a reasonably satisfactory one for the Chicago Chapter. Technical papers of high caliber were considered the most important objective, and a program of four such papers was presented to the membership. Two of these meetings were held jointly with the Chicago Acoustical and Audio Group (an independent Chicago club). One of these papers was in conjunction with a field trip. In addition, this chapter was invited to cosponsor a paper procured by another Chicago Group Chapter. The technical programs, with corresponding attendance figures, are listed below.

In addition to the technical programs, the Chicago Chapter acted in behalf of S. J. Begun, National PGA Program Chairman, in procuring a panel of five technical papers forming an Audio Panel for the National Electronics Conference to be held in October, 1958, in Chicago.

The Chicago Chapter on Audio, according to the minutes of the Chicago Section Executive Committee meeting of September 24, 1957, is second largest (Circuit Theory—233, Audio—202). The most recent official "strip list" indicates our membership at 208. The March 4, 1958 issue of *PGA Chapter* (a newsletter) shows the Chicago membership at 230—third largest in the national setup (both Boston and Philadelphia exceed 250). Total national PGA membership is approximately 4300.

Four meetings of the Chapter Executive Committee were held to plan and coordinate activities. At least five of the six members of this committee were in attendance at each meeting. A budget request of $50.00 was not fully used. Total expenditure was $24.54 for mailing of ballots, notice of May 21 meeting, and return postcard.

Mr. Kramer, the outgoing chairman, has expressed his appreciation to the Chapter Executive Committee members listed below, and to the companies with which they are associated, for their cooperation during the past season; their activities have made this successful season possible.

Technical Program 1957–1958

October 11, 1957—"Modern Studio Recording Techniques," *W. T. Putnam, President, Universal Recording Corp.;* 45 attended.

December 13—"High-Powered Droppable Sound System," *F. C. Fischer, Cook Research Laboratories;* 25 attended.

January 10, 1958—"Some Practical Considerations Concerning Reliability and Quality Control of Transistors Used in Audio Applications," *A. T. Kundrotas, Beltone Hearing Aid Co.* (jointly with R and QC who arranged program).

March 14—"Stereophonic Disk Pickup," *J. F. Wood, Electro-Voice Inc.* (jointly with CAAG); 150 attended.

May 21—"The NBC Compatible Color TV System," *W. C. Prather, WNBQ;* including an inspection tour of the NBC Chicago facilities (jointly with CAAG); 91 attended.

New Officers—Executive Committee 1957–1958

Robert J. Larson, Jensen Manufacturing Co., *Vice-Chairman*;

Leonard G. Eckmann, General Telephone Laboratories, *Secretary-Treasurer;*

William H. Ihde, General Radio Co., *Program Chairman;*

James S. Aagaard, Northwestern University, *Membership Chairman;*

Sid Solomon, Allied Radio Corp., *Publicity Chairman*

Cincinnati, Ohio

Seventy members and guests attended an inspection trip to Crosley's "Voice of America" at Mason, Ohio, and witnessed a demonstration of the new transmitter at WLW. This program, on June 24, 1958, was the fourth meeting of the year for the Cincinnati Chapter. The speaker was R. J. Rockwell, Vice-President and Director of Engineering of the Crosley Broadcasting Corporation, who is responsible for the new cathanode transmitter design.

The meeting started with an inspection of the Crosley-operated Voice of America facilities which included inspection of the antennas. Mr. Rockwell then gave a brief description of the basic circuits used in WLW's new cathanode transmitter. After the talk, the new transmitter was switched off the air and the old one placed in service. The output of the cathanode transmitter was connected to a 50,000-watt artificial load, and at this time high-quality records were used to modulate the transmitter. An A, B, listening test was conducted so that samplings of the input and of the demodulated output were compared alternately. The audio power amplifier used to drive the Jensen speaker was itself a cathanode type unit. An inspection of the WLW transmitters and shop preceded the closing of this interesting meeting.

For 1958–1959 new officers for the Cincinnati Chapter are:

William C. Wayne, Jr., *Chairman;*
Clyde G. Haehnle, *Vice-Chairman;*
John P. Quitter, *Secretary.*

Dayton, Ohio

In "Thumbing through the Chapters," the *Dayton Wave Guide* reports:

The Professional Group on Audio has completed a banner year, a success which required diligent effort and team work. Most responsible for scheduling a fine array of speakers and demonstrations was our vice-chairman, Jack Stanton.

PGA has chosen the following officers for the 1958–59 season: *Chairman*, Jack Stanton; *Vice-Chairman*, Ed Valentine and *Secretary*, Bob Stanford.

Programs presented this year included: "Problems Associated with Bass Reproduction," Edgar Villchur; "New Developments in VHF Drives," Howard Souther; "Your Telephone of 1965," Mr. Riefenstahl, Ohio Bell and "Review of Current High Intensity," Mr. Liebich, Stromberg-Carlson Sound Systems Developments.

In addition to these regular meetings at the Engineer's Club, also presented were: Tape-scripts and Slides on the Basic Fundamentals of Disc and Tape Recording, obtained from National PGA Headquarters. Meeting at SREPCO Auditorium; Audio Listening Session, home of Jack Stanton and a film—"The Strange Case of Cosmic Ray," from the Bell Telephone Laboratory, shown at Comm. and Nav. Laboratory, WADC.

Seven meetings in one season, is a new season record for us. We emphasize, not only quantity, but quality of our programs.

WESCON

Audio papers were included in the WESCON program for Wednesday, August 20, 1958, at 9:30 A.M., in the Venetian Room of the Ambassador Hotel in Los Angeles.

Session Chairman: *J. C. Webster, Naval Electronic Laboratories, San Diego, Calif.*

"Experiments with Speech Using Digital Computer Simulation," *E. E. David, Jr., M. V. Mathews, and H. S. McDonald, Bell Telephone Laboratories, Murray Hill, N. J.*

"A Survey of Speech Bandwidth Compression Techniques," *S. J. Campanella, Melpar, Inc., Falls Church, Va.*

"The Four-Track Stereotape Magazine for Home Hi-Fi," *R. J. Tinkham, Ampex Corp., Redwood City, Calif.*

"A Versatile Compressor-Limiter Audio Amplifier for Studio Use," *E. W. Templin, Westrex Corp., Hollywood, Calif.*

"Audio Characteristics of Piano Tones," *J. P. Quitter, The Baldwin Piano Co., Cincinnati, Ohio.*

WITH OTHER ACOUSTICAL AND AUDIO SOCIETIES

The April, 1958 issue of *The Journal of the Acoustical Society of America* contains its usual array of articles on pure and applied acoustics. Many of these papers will be of interest to members of the IRE Professional Group on Audio.

In a study "On the Minimum Audible Angle," A. W. Mills of the Psycho-Acoustic Laboratory, Harvard University, describes a number of measurements of the difference limen for the azimuth of a source of pure tones as a function of the frequency of the tone and the direction of the source. Tone pulses between 250 and 10,000 cps were sounded in the horizontal plane around the head of a subject seated in an anechoic chamber. The smallest angular separation that could be detected between the sources of two successive tone pulses (the minimum audible angle) was determined for each of three subjects. These threshold angles were analyzed in terms of the corresponding threshold changes in the phase, time, and intensity of the tone at the ears of the subject.

The fact that we hear what we *want* to hear is well brought out in an article "On the Effect of Practice and Motivation on the Threshold of Audibility," by A. Zwislocki, F. Maire, A. S. Feldman, and H. Rubin of the Harvard University Psycho-Acoustic Laboratory. The thresholds of audibility for 100 and for 1000 cps have been measured on five groups of originally naive listeners by various experimental techniques. All of the experiments showed improvement of the threshold with practice. The improvement was greater at 100 cps than at 1000 cps. Pretraining at 1000 cps did not affect the threshold change at 100 cps. The improvement of the threshold with practice was enhanced considerably by "reward" and feedback.

A new instrument for creating measurable sound pressure is described by Josef Merhaut and Miroslav Vlcek of the Research Institute of Telecommunications, Prague, in an article entitled "Pistonphone with Differential Piston." The article first examines the various sources of error in absolute calibration of standard microphones in a pistonphone. A new differential pistonphone has been developed which enables absolute calibration of microphones with the accuracy 0.1 db. The design and properties of the instrument are described.

"Limits of Direct Speech Communication in Noise" is a contribution to the problem of intelligibility in noisy surroundings by J. M. Pickett of the Operational Applications Laboratory of Bolling Air Force Base. Person-to-person tests of sentence intelligibility were carried out in low frequency and white noise at noise levels ranging from 85 to 118 db. Talkers attained shouting levels of vocal effort but the maximum tolerable noise levels for 90 per cent sentence intelligibility and 1 m between talker and listener were estimated to be 95 db for white noise and 105 db for low-frequency noise. Another article dealing with intelligibility at high noise levels by Irwin Pollack of the same organization will be of interest to many readers.

In Letters to the Editor, R. W. Young suggests methods for simplifying noise reporting. In another letter on the "Priority in Invention of the Ultrasonic Metal Horn," R. W. Wick of the Bell Telephone Laboratories, Inc., Murray Hill, N. J. points out that W. P. Mason has invented this horn as specified in Patent No. 2,514,080, and not Lozinski and Rozenberg, as claimed by the Russians.

This issue contains the review of Acoustical Patents by R. W. Young.

BENJAMIN B. BAUER

Time and Frequency Scaling in Magnetic Recording*

FRANCIS M. WIENER†

Summary—**An electrical signal recorded on magnetic tape can be relatively easily transposed in frequency by reproducing it at a tape speed different from the tape speed at which it was originally recorded. Transposition into the audio-frequency range is most frequently employed for convenience because of the ready availability of indicating and analyzing equipment in that frequency range. Examples where this technique has been helpful are cited.**

Introduction

A HIGH DEGREE of perfection has been achieved in magnetic tape recording techniques in recent years. Since early use as a recording medium in the *audio*-frequency range, great strides have been made in perfecting this technique for recording signals in the kilo- and megacycle range on the one hand and in the fractional cycle range on the other. Although it is very frequently desirable to reproduce the recorded signal unchanged, it is often advantageous to transpose the signal into a different frequency range on reproduction by using different tape speeds for recording and reproducing the signal. Unless tolerances on permissible flutter are unusually severe, this is easily accomplished in many cases by interposing a simple arrangement of additional rollers and rubber idlers between the drive motor and the capstan driving the tape. By exercising a measure of ingenuity, the tape speed change can be accomplished by throwing a lever or pushing a button. Several designs are available commercially. Most commonly, the scaled signal ends up in the audio-frequency range; analyzing and indicating equipment is handy in the laboratory and easily calibrated. It must be remembered, however, that a frequency shift is traded for a change in time. A 100-kc signal of one-second duration becomes a 1-kc signal lasting one hundred seconds on scaling it down by a factor of 100 in frequency.

There is no claim on the part of the author to originality or even novelty of this approach. The main purpose of this note is to illustrate the technique of time and frequency scaling by citing examples of perhaps more than passing interest, and to draw attention to the fact that, by the use of this technique a nasty instrumentation problem may frequently be avoided.

Mechanical Vibrations

In the investigation of mechanical vibrations of equipment, signals containing contributions from fractional cycles to several hundred cycles per second are frequently encountered. The spectrum below 20 or 30 cps is most easily investigated by recording the signal on magnetic tape at slow speed and reproducing it at a higher tape speed. If a speed ratio of 30 is employed, contributions down to about one cycle per second are easily analyzed by conventional filters operating in the audio-frequency range. Fig. 1 shows an acceleration spectrum obtained in a railroad car for a fixed set of operating conditions.[1] The output signal generated by an accelerometer was recorded on magnetic tape, and the results were plotted in terms of acceleration level in decibels relative to 1 inch/sec² in third-octave bands. The open circles were obtained by using a tape speed ratio of 30:1 on reproduction; the solid circles were obtained with a 1:1 speed ratio. Note the generally good agreement in the overlap region between 40 and 150 cps after proper calibration of the systems.

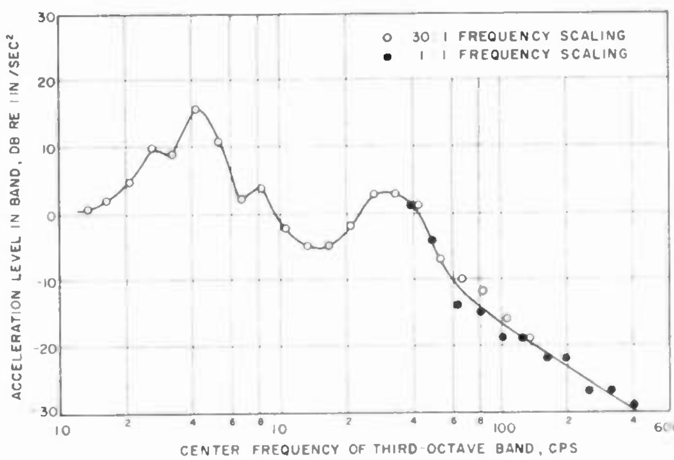


Fig. 1—Acceleration spectrum in a railroad car (after Dyer).

Low-Frequency Pressure Fluctuations in Wind Tunnels

Air supply systems for supersonic and transonic wind tunnels of present design frequently exhibit considerable pressure fluctuations in the frequency range below approximately 10 cps. These fluctuations are generally undesirable and before taking remedial measures it is necessary to determine their spectra. Fig. 2 shows a spectrum of the pressure fluctuations in the settling chamber of a blow-down wind tunnel operating in the supersonic range.[1] The output signal of a special microphone system, responsive to low-frequency pressure fluctuations was recorded on magnetic tape. A tape speed ratio of 30:1 was used on reproduction, and the spectrum was obtained by means of a conventional third-octave filter set.

* Manuscript received by the PGA, April 25, 1958.
† Bolt, Beranek, and Newman Inc., Cambridge 38, Mass.

[1] I. Dyer, Bolt, Beranek, and Newman Inc., private communication.

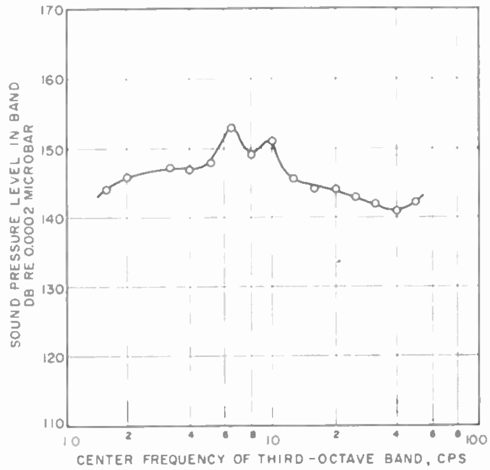


Fig. 2—Pressure fluctuation spectrum in the settling chamber of a blow-down wind tunnel (after Dyer).

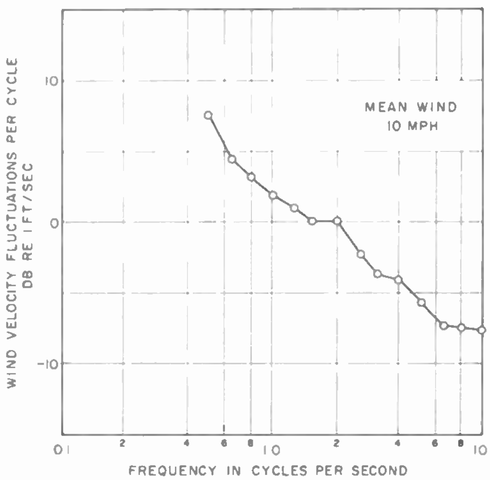


Fig. 3—Turbulent velocity spectrum in the atmosphere over open level terrain (after Keast).

Small-Scale Atmospheric Turbulence

The investigation of the velocity fluctuations in the open atmosphere near the ground, or in a laboratory wind tunnel for that matter, has recently received a good deal of attention. These fluctuations outdoors are of importance not only in their own right, but also in the study of the diffusion of atmospheric contaminants and the propagation of sound outdoors. Fig. 3 shows a spectrum of the velocity fluctuations obtained over open level terrain for a mean windspeed of about 10 mph. The output signal obtained from a small bead thermistor anemometer was recorded on magnetic tape. A tape speed ratio of 100:1 was used on reproduction, and the spectrum was obtained by means of a conventional third-octave filter set. The ordinate shows the velocity fluctuations in decibels relative to 1 foot/second reduced to bands one cycle wide.[2]

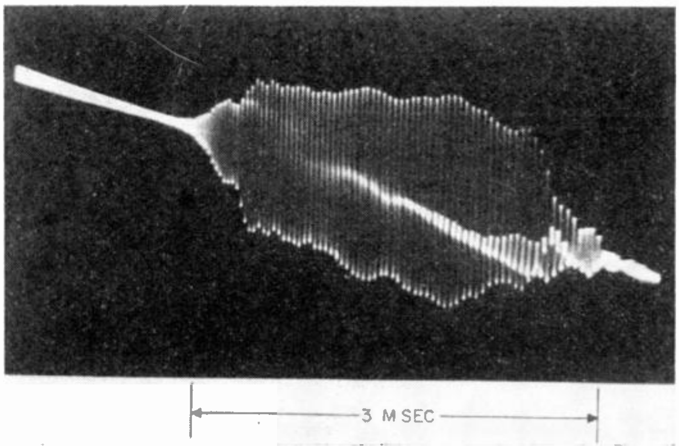


Fig. 4—Frequency modulated ultrasonic cry of a Panamanian bat (after Griffin).

The Ultrasonic Cries of Bats

It is well known that certain species of bats emit bursts of ultrasonic energy to avoid obstacles by echo location, and to locate and pursue their insect prey.[3] The frequency spectra of these pulses range from tens to hundreds of kilocycles. Very often the frequency of the ultrasonic oscillations changes markedly throughout the duration of the pulse. Fig. 4 is a photograph of the trace of a frequency modulated pulse on an oscilloscope screen.[4] For further study, the output signal obtained from a microphone responsive in the ultrasonic frequency region was recorded on magnetic tape. By employing a tape speed reduction of 1:10 or 1:20 the pulses emitted by a bat were made into a startling auditory display, and a frequency analysis was carried out conveniently. Moreover, the spacing between pulses, which changes markedly as the bat approaches an obstacle or his prey, could then be measured with a simple stop-watch rather than by the use of electronic timing circuits operating in the ultrasonic frequency range.

Conclusion

The technique of time and frequency scaling of signals by means of magnetic recording techniques, although not new, has several advantages. This process lends itself to scaling up or down in frequency, and it deserves the attention of workers in many diverse scientific fields.

[2] D. N. Keast, Bolt, Beranek, and Newman Inc., private communication.

[3] D. R. Griffin, "Listening in the Dark," Yale University Press, New Haven, Conn.; 1958.

[4] D. R. Griffin, Harvard University, private communication. The photograph was taken with a moving film camera—hence the tilt of the base line.

Loudspeakers and Negative Impedances*

RICHARD E. WERNER†

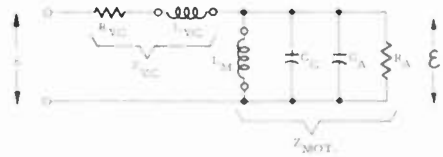
Summary—**A direct radiator moving coil loudspeaker driven by an amplifier whose output impedance approaches the negative of the blocked voice-coil impedance can be made to exhibit extended low-frequency response with reduced distortion. The results are not to be confused with the effects of a negative resistance source. In a typical case, neutralization of 70 per cent of the blocked voice-coil impedance completely damps the cone resonance, as well as substantially reducing the nonlinear distortion below resonance. When the amplifier is compensated for the falling radiation resistance at low frequencies, uniform output can be obtained to any arbitrary low frequency, limited only by the ultimate power-handling capability of the amplifier and speaker. In this system, no additional amplifier power is required at frequencies down to the speaker resonance; additional power is required below that point.**

Introduction

Direct radiator, moving-coil loudspeakers are basically inefficient transducers. The influence of the mechanical impedance upon the electrical input impedance is very slight as is typical of most "wide-band" electromechanical transducers. Even the magnitude of a mechanical resonance is often strongly masked by the electrical impedance. Because the electrical impedance of the blocked voice-coil is large compared to the average reflected mechanical impedance, the transfer characteristic of the transducer is largely influenced by the nature of the mechanical impedance.

A commonly used equivalent circuit for a direct radiator, moving-coil loudspeaker is shown in Fig. 1. Useful radiation is assumed to take place from one side of the cone as is the case wherein the loudspeaker is mounted in a totally enclosed box. The reflected radiation resistance, R_A, is inversely proportional to the square of the signal frequency for frequencies below that for which the diameter of the loudspeaker cone is approximately equal to a half-wavelength (the frequency of ultimate radiation resistance). For low frequencies, the air load upon the cone becomes essentially that of a constant mass.

For acoustic output independent of frequency, it is necessary that the voltage across R_A be inversely proportional to frequency at low frequencies. Therefore, the compliance of the moving system, L_M, is made very large so that its resonance with C_A and C_C occurs at the lowest possible frequency. Unfortunately, for loudspeaker cones and cabinets of convenient size, this resonance appears within the range of musical frequencies; and, by virtue of its lack of resistive loading, is insufficiently damped to avoid "ringing" on transient signals. Below the resonant frequency, the loudspeaker cone becomes stiffness-controlled and the acoustic output falls at a rate of 12 db per octave.



E = Input voltage
$\mathcal{E}$ = BLV

where:

B = Magnetic flux density
L = Length of voice-coil wire
V = velocity of voice-coil motion.

Reflected Motional Impedances

$$L_M = B^2L^2 \frac{C_S C_B}{C_S + C_B}$$

where:

C_S = Compliance of cone suspension
C_B = Compliance of air load in box.

$$C_C = \frac{M_C}{B^2L^2}$$

where:

M_C = mass of cone and voice-coil.

$$C_A = \frac{M_A}{B^2L^2}$$

where

M_A = mass of air load on loudspeaker.

$$R_A = \frac{B^2L^2}{r_A}$$

where:

r_A = radiation resistance presented to loudspeaker.

Electrical Impedances

R_{VC} = resistance of blocked voice-coil
L_{VC} = inductance of blocked voice-coil.

Frictional losses are assumed negligible

Fig. 1—Common equivalent circuit for direct radiator moving-coil loudspeaker.

In addition to frequency and transient distortions, the direct radiator loudspeaker is subject to considerable nonlinear distortion at low frequencies. Below the resonant frequency, where the motion of the cone is determined principally by the compliance of the system, the nonlinearity of the compliance produces distortion in the radiated sound. There are other factors contributing to nonlinear distortion in a loudspeaker but the nonlinearity of the compliance is the principal offender.

There appears to be an unlimited variety of ways to modify the performance of a loudspeaker. The major effort has been concentrated on the design of the

* Manuscript received by the PGA, July 17, 1958; revised manuscript received, August 19, 1958. This paper is based on an article which appeared in *J. Acoust. Soc. Amer.*, vol. 29, pp. 335–340; March, 1957.

† RCA Missile Test Project, Patrick Air Force Base, Fla.

speaker enclosure, some of which have taken rather bizarre forms. Reflex cabinets, multistage reflex cabinets, column resonators, labyrinths, folded horns, and "semihorns," even cabinets with vibrating walls, have made their appearance. Some of the more serious designs have produced really noteworthy improvements in performance over the somewhat ill-defined "basic loudspeaker."

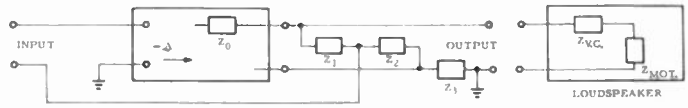
Recently, a design has become very popular in which the loudspeaker is provided with a heavy moving system so as to obtain a low-frequency resonance with its small enclosure. The compliance of the suspension is sufficiently high that the enclosure stiffness is the controlling element below resonance. This virtually eliminates the distortion caused by a nonlinear suspension. Unfortunately, the heavy moving system reduces the sensitivity of the loudspeaker and also requires that a separate speaker be employed for high-frequency reproduction. These limitations are unimportant in some applications but for general use the cost is prohibitive.

Some improvement in low-frequency performance can be obtained by acoustically damping the typical loudspeaker. If this is accomplished by stuffing the enclosure with fibrous material the speaker sensitivity is reduced only in the vicinity of resonance. This method is difficult to control accurately and the results are not always predictable if heavy damping is attempted. Once effective damping is achieved acoustically, the loudspeaker will become deficient in low-frequency response. In this case the amplifier must be equalized and the low-frequency power rating increased, which may not be economically feasible.

If the loudspeaker can be damped electrically by the output impedance of the driving amplifier, unnecessary losses are avoided and a more economical system results. This has long been realized, but so too has the fact that the blocked voice-coil impedance presents an effective barrier, isolating the motional impedance from an external "short circuit." If the blocked voice-coil impedance can be "reduced" by subtracting from it the output impedance of the amplifier, a more effective short circuit of the nonlinear, resonant motional impedance should result.

Within the last ten years the interest in negative-output impedance amplifiers has grown and died a number of times. The idea looked all right but there was always something overlooked and as a consequence the publicity has been largely unfavorable.[1–4] Perhaps a good close look at the problem will stimulate new interest in "negative damping factors."

[1] W. Clements, "Loudspeaker damping," *Audio Eng.*, vol. 35; August, 1951.
———, "Positive feedback," *Audio Eng.*, vol. 36; May, 1952.
[2] U. J. Childs, "Loudspeaker damping with dynamic negative feedback," *Audio Eng.*, vol. 36; February, 1952.
———, "Positive current feedback," *Audio Eng.*, vol. 36; May, 1952.
[3] "A new hi-fi speaker system," *Radio & TV News*, vol. 57, pp. 54–55, 82; April, 1957.
[4] N. H. Scott, "Power amplifiers for music reproduction," *J. Audio Eng. Soc.*, vol. 3, pp. 138–142; July, 1955.



$$Z_1 + Z_2 \gg Z_0$$

$$A' = -\frac{A}{1 + \dfrac{Z_2 A}{Z_1 + Z_2}}$$

$$Z_0 = \frac{Z_0 + Z_3 - \dfrac{Z_1 Z_3 A}{Z_1 + Z_2}}{1 + \dfrac{Z_2 A}{Z_1 + Z_2}}$$

$-A$ = open circuit voltage gain of basic amplifier.
Z_0 = output impedance of basic amplifier.
A' = open circuit voltage gain with negative impedance circuit.
Z_0' = output impedance with negative impedance circuit.
If A is very high, and

$$\frac{Z_1}{Z_2} = \frac{Z_{VC}}{Z_3}$$

(a bridge balanced against Z_{VC}), then $Z_0' = -Z_{VC}$. Bridge may take form of:

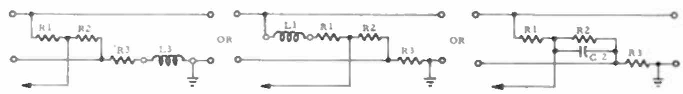


Fig. 2—Basic negative impedance circuit.

Basic Circuit

A particularly suitable circuit for obtaining an output impedance composed of a negative resistance and inductance is shown in Fig. 2. In this circuit, positive current feedback and negative voltage feedback are combined in a bridge and fed in a common feedback path through the amplifier. This system avoids complications resulting from gain and phase shift variations which may be encountered if separate feedback paths are employed.

The equations for operation of the circuit indicate that the output impedance is quite independent of the amplifier gain and phase shift when the loop gain is high and the phase shift not severe. With the bridge circuit balanced against the blocked voice-coil impedance, there will always be a net negative feedback at audio frequencies. At the extremes of the frequency spectrum where the reflected motional impedance becomes zero there will be no feedback produced by the circuit. The balanced bridge will, therefore, produce a stable negative output impedance which in no way detracts from the quality of the basic amplifier.

A close look at Fig. 2 will disclose the identity between negative-output impedance amplifiers and "motional feedback" or "motion control" systems. With the bridge of Fig. 2 accurately balanced against the blocked voice-coil impedance, the feedback voltage is proportional to the "generated back EMF," *i.e.*, the motional impedance. For the more likely condition of some degree of unbalance in the feedback network, the feedback voltage is a mixture of either negative voltage, or positive current feedback plus the "motional feedback."

The impedance concept offers certain advantages over the other analogies when it comes to equivalent circuitry or when studying the amplifier and loudspeaker individually, and is preferred by the author.

Practical Considerations

There are a few fundamental precautions which merit special attention, not because they are particularly complex but because they are vital to success.

Loop Gain

It is desirable that the loop gain of the circuit be as high as possible. For conventional high-quality amplifiers employing considerable feedback, this necessitates the use of additional stages of amplification which may result in stability problems at the frequency extremes when the loudspeaker is disconnected and the feedback voltage is maximum. If oscillation so results, the amplifier may be damaged and such a potential disaster is to be avoided if possible.

Perhaps the simplest solution to this problem is to restrict the circuit to a two-stage amplifier with little, if any, inherent feedback. The quality of the basic amplifier may be poor compared to a feedback amplifier but its performance with a loudspeaker can actually be superior. Improved results are to be obtained, however, if the basic amplifier is provided with some inherent feedback. The circuit of Fig. 3 is illustrative of one of the many successful approaches. The two-stage circuit provides a loop voltage gain of approximately 20 for the negative impedance circuit. Rated output power was achieved with less than ¼ per cent distortion at 40 cps in a model employing this circuit. This performance has been achieved through the use of positive feedback around the low-level stage which raises the gain of the main feedback loop to essentially infinity. The feedback factor of 1/20 is slightly more than sufficient to reduce the gain of the circuit to its initial value under loaded conditions. The distortion of the low-level stage is thus reduced slightly and the distortion and output impedance of the final stage are reduced essentially to zero.[5]

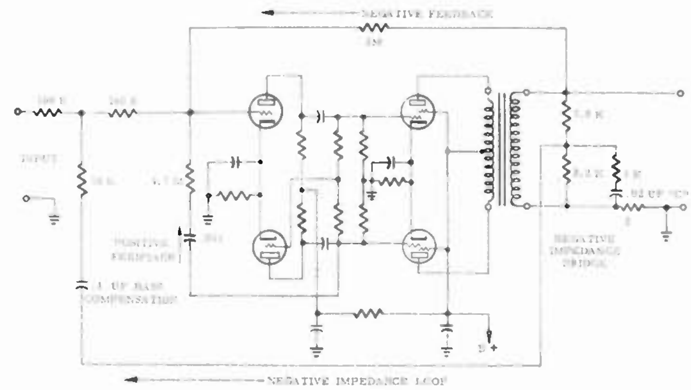


Fig. 3—Practical negative output impedance amplifier.

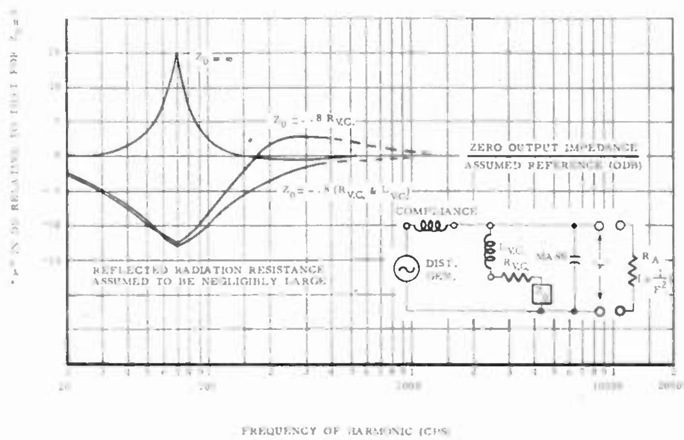


Fig. 4—Effect of output impedance on distortion of cone velocity relative to distortion for zero output impedance.

Voice-Coil Inductance

A factor requiring especial attention is the nature of the loudspeaker's blocked voice-coil inductance. This inductance is far from ideal in nature, being influenced by assorted hysteresis and eddy-current losses associated with the mass of iron structure surrounding it. Accurate cancellation of an impure inductance such as this may involve considerable complication of the bridge circuit. And, since the actual motion of the voice-coil has little precise bearing upon the radiation of higher frequencies, accurate cancellation of the voice-coil inductance is of doubtful value. However, neutralization of a large percentage of the voice-coil resistance alone raises the Q of the voice-coil inductance and this is almost certain to result in disappointing performance. The new "High Q" voice-coil inductance in parallel with the reflected motional stiffness resonates with the reflected mass of the moving system accentuating the response and transient distortion in the mid-low-frequency range. At higher frequencies the response will be weakened as is expected from a look at the equivalent circuit (Fig. 1). Some negative output *resistance* circuits bypass the feedback at middle and high frequencies to compensate for this loss, but the resonance remains as evidenced by a more or less sharp knee which incidentally makes equalization for the reduced bass response difficult. Although the effects on response may be more noticeable, the lack of inductive cancellation can result in significantly increased nonlinear distortion. Redrawing the equivalent circuit of a loudspeaker to show the effect of the amplifier output impedance upon the distortion yields the circuit of Fig. 4. The distortion components produced in the nonlinear stiffness pass through a low-pass filter before appearing across the radiation resistance of the loudspeaker. With the resistance in the circuit reduced, the filter accentuates the harmonics appearing in the vicinity of the cross-over frequency. The curves of Fig. 4 illustrate the effect of output impedance upon the magnitude of the distortion components as compared to the reference condition of zero output impedance. The hypothetical speaker illustrated is

[5] J. M. Miller, Jr., "Combining positive and negative feedback," *Electronics*, vol. 23, pp. 106–109; March, 1950.

typical of full range speakers. An efficient low-frequency "woofer" having a heavier moving system and more inductive voice-coil will exhibit this effect to a more pronounced degree and at a considerably lower frequency. It is then possible that a tone as low as 20 cps may show increasing distortion as the source *resistance* is lowered from a high positive value through zero into negative values. Such a speaker can be expected to benefit more from the use of a negative *inductance* source than from the occasionally specified positive *resistance* source.

Sufficient inductive neutralization is produced in the circuit of Fig. 3 to provide reduced distortion at all frequencies and some treble boost is incorporated to maintain the high-frequency response of the speaker within one db of the constant voltage reference. The treble boost is inherent in the bridge circuit employed in Fig. 3 because the capacitor, C, reduces the amount of open circuit feedback at high frequencies resulting in an open circuit response which rises at high frequencies. In practical applications, the amount of negative inductance and treble boost can be chosen to correct for certain general trends in the high-frequency characteristics of the loudspeaker. In applications involving coaxial or "woofer-tweeter" speakers, neutralization of the combined voice-coil impedance can be very complicated. A convenient remedy in the case of simple capacitor coupled tweeters, however, is to neutralize for the low frequency speaker and connect the tweeter through its capacitor directly to the output transformer—avoiding the series current feedback resistor. More complicated multi-speaker systems may require individual amplifiers for each speaker.

Bass Equalization

If neutralization of the voice-coil impedance is sufficient to reduce the Q of the fundamental resonance to less than unity, some bass boosting will generally be desired. Optimum flatness of response is obtained when a 6 db per octave bass boost is employed in the region where the motion is resistance controlled and 12 db per octave for the lower frequencies where the cone resumes stiffness control. Usually the additional boosting in the stiffness control region is unnecessary with the available program quality and places a needless strain on the amplifier's output power at rumble and "compressor-thump" frequencies. Often it will be desirable to limit the amount of the 6 db per octave boost to prevent the amplifier from assuming too heavy a low-frequency burden. If the damping is beyond critical ($Q<0.5$) the turnover of the usual bass boost circuit will complement the speaker response nicely. All of this assumes that the radiation impedance of the loudspeaker is as predicted for infinite baffle, free field conditions. The use of the loudspeaker in a living room may considerably modify its radiation impedance at low frequencies as well as the propagation of its sound. These effects are usually beyond the control of the equipment manufacturer[6] and compensation for them is best left for the listener to attempt with tone controls and selection of speaker location. Presuming for the moment that perfect low-frequency response is obtainable directly from the loudspeaker by application of these principles, it follows logically that additional modification of the response by enclosures which allow radiation from the rear of the cone must necessarily result in a less perfect response. It is, however, conceivable that reflexing an enclosure at a very low frequency can further improve the performance of a modestly designed negative impedance system.

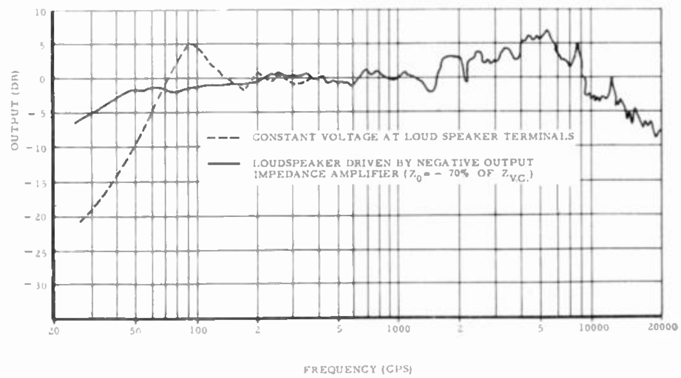


Fig. 5—Response-frequency characteristics on ℄ of RCA SL-12 loudspeaker in 3-cubic-foot box.

In the circuit of Fig. 3, the bass compensation for rising reflected radiation resistance is accomplished in the negative impedance loop, but the circuit parameters are chosen so that there is negligible effect on the efficiency of the negative impedance circuit. The bass boost is effected by frequency-variant loading of the input circuit to the main amplifier.

PERFORMANCE

Loudspeaker in a Large Box

The loudspeaker chosen for these tests is a high quality 12-inch single-cone unit mounted in a totally enclosed 3-cu-foot box. Resonant frequency in the box is 89 cps.

Frequency Response: The response-frequency characteristic of this loudspeaker is shown in Fig. 5. The negative impedance amplifier is seen to level and extend the low-frequency response.

Distortion: The loudspeaker distortion was measured with a ribbon microphone of essentially flat response above 40 cps and a total rms distortion meter. The microphone's sensitivity was low at frequencies below about 30 cps resulting in an apparent noise level comparable to the distortion. Therefore, the distortion measurements below 30 cps are approximate. The distortion characteristics of the loudspeaker are shown in Figs. 6

[6] A different form of feedback loudspeaker in which the feedback voltage is derived from the acoustic output (by a microphone) rather than from the voice-coil motion can minimize the effects of variations in radiation impedance upon the radiated power.

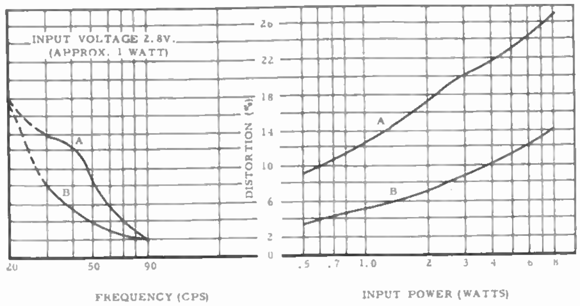


Fig. 6—Total harmonic distortion of RCA SL-12 loudspeaker in 3-cubic-foot box vs (a) frequency at 1 watt, and (b) input power at 40 cps. A = loudspeaker alone, B = loudspeaker driven by negative output impedance amplifier ($Z_0 = -70$ per cent of Z_{VC}).

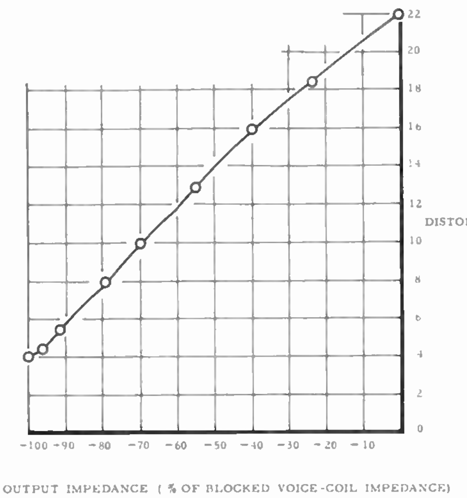


Fig. 7—Total harmonic distortion of RCA SL-12 loudspeaker in 3-cubic-foot box at 4 watts of 40 cps input vs amplifier output impedance.

and 7. In Fig. 6, the distortion is plotted vs frequency at 1 watt and vs power at 40 cps. Measurements were not recorded for powers above 8 watts because the loudspeaker began "ticking" the limits of its maximum excursion. Above 8 watts at 40 cps, the measured distortion did not nearly indicate the increase in listener annoyance due to the "ticking." Driving the loudspeaker with a negative impedance amplifier results in a substantial reduction in low-frequency distortion.

Another possible source of distortion in a loudspeaker is the nonlinearity of the flux density in the voice-coil gap. If distortion due to this cause were of important magnitude, there might be a limit upon the amount of desirable impedance cancellation. Fig. 7, however, shows that the loudspeaker distortion continues to decrease as the cancellation is increased to 100 per cent. The percentage cancellation indicated is true only for the resistive component of the voice-coil impedance. The inductance was somewhat less effectively cancelled. The values of cancellation above 80 per cent were accomplished in this amplifier by unbalancing the negative impedance bridge, under which condition the amplifier

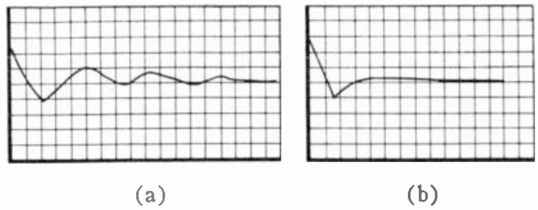


Fig. 8—Response to step function input, RCA SL-12 loudspeaker in 3-cubic-foot box. (a) Loudspeaker terminals shorted. (b) Loudspeaker connected to negative output impedance amplifier ($Z_0 = -70$ per cent of Z_{VC}).

experienced a net positive feedback from the bridge. Therefore, Fig. 7 illustrates, in part, the excellent immunity of the circuit to critical adjustment.

Transient Response: The response of the loudspeaker to a step-function signal is shown in Fig. 8. To obtain these patterns a dry cell was connected directly to the loudspeaker in one case and in series with the amplifier output terminals in the other case. The battery was shorted to obtain the step function input. The radiated sound was detected by a condenser microphone, amplified, and fed to a long-persistence screen oscilloscope. No attempt was made to correct for the effects of successive differentiation by the uncompensated radiation impedance of the loudspeaker, microphone, amplifier, and oscilloscope responses; or for possible room reflections.[7] Ringing is evident at the resonant frequency for the shorted voice-coil condition but not for the negative impedance condition. Adjusting the amplifier output impedance showed aperiodic damping to occur at about 54 per cent cancellation of the blocked voice-coil impedance.

Loudspeaker in a Small Box

Another loudspeaker of the same type was mounted in a totally enclosed box of $\frac{1}{2}$ cubic foot. The resonance of the speaker in this box, which was just large enough to hold the loudspeaker, occurred at 200 cps. The negative impedance amplifier used to obtain the following frequency and transient responses was an inexpensive 10-watt unit possessing no feedback other than that due to the negative impedance loop. The amplifier is typical of the type used in home and automobile radios. The net negative feedback due to the negative impedance loop is about 6 db at midaudio frequencies; and the distortion at 100 cps is about 2 per cent at 5 watts. This distortion is of the same order of magnitude as that due to the loudspeaker. For this reason, the distortion measurements for the loudspeaker in the small box were made using the previous amplifier (similar to Fig. 3).

Frequency Response: The response-frequency characteristics of the loudspeaker mounted in the small box are shown in Fig. 9. The improvement obtained with the negative impedance source is more pronounced for

[7] Measurements of the actual cone motion indicate a nearly perfect square wave response of a loudspeaker driven by a negative output impedance amplifier. (R. E. Werner, "Loudspeaker performance as affected by a negative output impedance amplifier," unpublished.

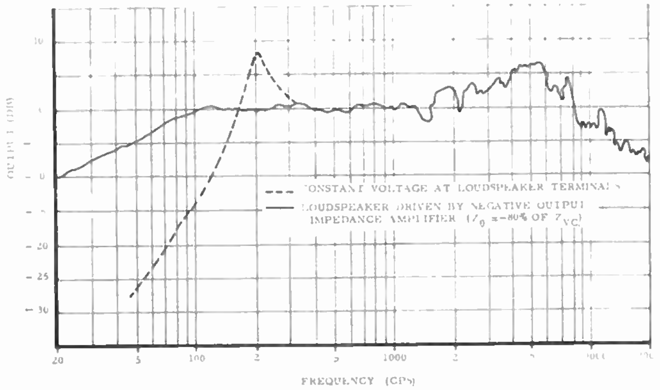


Fig. 9—Response-frequency characteristic on ℄, RCA SL-12 loudspeaker, ½-cubic-foot box.
--- = loudspeaker alone
— = loudspeaker driven by negative output impedance amplifier ($Z_0 = -70$ per cent of Z_{VC})
0 = 66 cps
x = 100 cps.

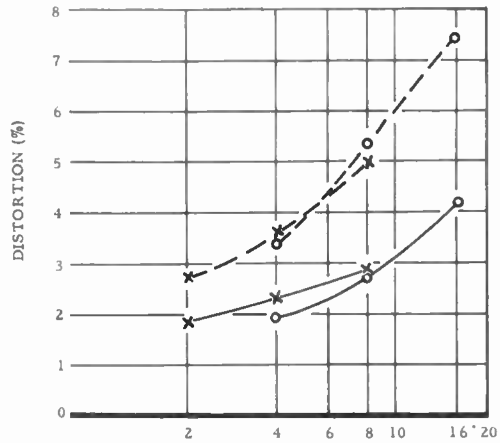
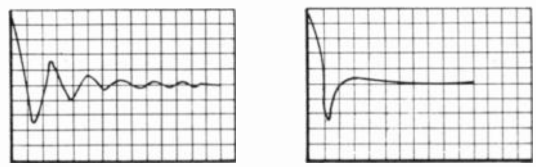


Fig. 10—Total harmonic distortion RCA SL-12 loudspeaker, ½-cubic-foot box.



(a) (b)

Fig. 11—Response to step function input RCA SL-12 loudspeaker in ½-cubic-foot box. (a) Loudspeaker terminals shorted. (b) Loudspeaker connected to negative output impedance amplifier ($Z_0 = -84$ per cent of Z_{VC}).

the small box than for the larger box as one may have anticipated. It is unlikely that any form of acoustical treatment of a ½-cubic-foot box could produce such a response from a 12-inch loudspeaker.

Distortion: The distortion characteristics of the loudspeaker in the small box are shown in Fig. 10. The distortion produced by the loudspeaker is again materially reduced by the use of a negative output impedance amplifier.

Transient Response: The response of this system to a step-function input signal is shown in Fig. 11 and is similar to that measured for the loudspeaker in the larger box. In this case, however, the Q of the resonance is higher and aperiodic damping is achieved at about 73 per cent cancellation of the blocked voice-coil impedance.

Subjective Tests

Subjective tests are a necessary assessment of value in a consumer product, particularly an acoustical product. In order to subjectively evaluate the performance of a sound system affecting only the lower frequencies, it is necessary to precondition inexperienced observers so that their attention will be adequately focused on the generally unobserved low-frequency accompaniment. This assumes, of course, that music is chosen for the program material. The use of selected noises and sound effects may be more effective; but the end use is with voice and music and the system must be so evaluated. About the two systems herein presented: it is difficult to find an adequate music recording to evaluate the performance of the loudspeaker in the 3-cubic-foot box with an inexperienced audience, but equally difficult to find unsuitable program material for evaluating the performance of the loudspeaker in the ½-cubic-foot box.

The system was demonstrated before the Delaware Chapter of the Acoustical Society of America on June 7, 1956. A number of loudspeakers were employed including a 4-inch "Drive-In Theater" loudspeaker in its normal enclosure and a 15-inch "Woofer-tweeter" loudspeaker in a 3-cubic-foot box. In all instances, an improvement over the normal system was noticed by all members present at the meeting. The effect was most startling in the case of the "Drive-In Theater" loudspeaker, as would be expected. The 15-inch loudspeaker was made utterly flat to below 20 cps by its negative impedance amplifier and is an excellent example of a situation in which a reflex or horn type of enclosure would be undesirable.

The power-handling capability of a sound system is one characteristic which demands subjective listening tests. Because the use of a negative impedance amplifier in no way modifies the efficiency of the loudspeaker, no additional power is required in the spectrum above the loudspeaker resonance. The amplifier is required to deliver additional power below the resonance determined by the amount of equalization desired. Subjective tests reveal that a 10-watt amplifier is more than adequate for home use even with the 12-inch loudspeaker mounted in the ½-cubic-foot box. Musical program material appears to have a power-frequency distribution such that the 10-watt amplifier overloads at the same volume whether wired normally or with a negative output impedance equalizing to 80 cps.

Over 8 years of subjective evaluation have convinced the author that the low-frequency performance of a loudspeaker can be made superior to that of the majority of available program sources.

Conclusions

The results of the foregoing measurements encourage the following conclusions:

1) The use of a properly designed negative-output impedance amplifier will greatly extend the low-frequency response, reduce the nonlinear distortion, and eliminate the resonant frequency hangover of a direct-radiator moving-coil loudspeaker.

2) The improvement in frequency and transient response obtained with this system is most dramatic when the loudspeaker is mounted in a small box wherein the resonant frequency will be higher in the music spectrum.

3) The music power-frequency spectrum is such that no increase in amplifier power capability will be required in using this system for most applications.

4) The improvement in loudspeaker performance obainable with this system allows the use of less expensive loudspeakers and smaller speaker enclosures in high-quality sound systems.

5) Because of the poor quality of loudspeakers in comparison with present-day amplifiers, the use of a negative impedance circuit can provide improved performance even with a reduction in quality (and cost) of the amplifier.

6) The use of negative output *resistance* amplifiers with loudspeakers is to be avoided unless very careful analysis indicates that the results will be as desired.

7) Claims regarding the performance of a loudspeaker with a resistive source impedance should not be carelessly extrapolated to the conditions of reactive source impedances, in particular negative impedances.

8) A loudspeaker driven by a negative output impedance amplifier should always be mounted in a totally enclosed box unless careful attention reveals that a different type of enclosure will augment rather than detract from the performance.

Correspondence

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A Compatible Method of Recording and Reproduction of Stereo Sound*

There is currently much interest among audiophiles concerning the introduction of stereodisks. Reports from various audio shows indicate that the 45 degree–45 degree orthogonal groove system is quite acceptable. At least one other method of recording two audio channels on disks has been described, namely that of recording the second channel as a high-frequency FM signal superimposed on a normal audio channel.

One of the shortcomings of the orthogonal groove method is that it is not compatible with existing monaural records and audio systems since a special stylus is required. This difficulty is overcome with the FM method with the addition of rather complicated and critical equipment. The following describes another method of recording stereophonic sound which largely eliminates the above mentioned difficulties.

More than one audio channel can be recorded on a single information channel by simply heterodyning the additional audio channels into frequency bands above the normal audio spectrum but still within the range of existing recording and reproducing equipment. The audio channels thus combined are recorded and then reproduced by separating and demodulating the heterodyned channels. Drift-free operation of the additional channels is possible by also recording the heterodyning local oscillator frequencies and using these same frequencies for local oscillator signals in the demodulating process. Such a method is compatible with existing monaural records.

This system is in the process of evaluation by an independent group of engineers. Tests are not yet complete. However this subject is timely and it seems advisable to present a detailed description of operation at this time.

Fig. 1 shows a block diagram of the recording system for a two-channel stereo program. Channel 1 is a normal audio frequency spectrum, as is channel 2. The local oscillator frequency fo is slightly greater than the highest frequency in either channel 1 or channel 2. The resultant sum frequencies are selected by a high-pass filter and combined with the audio of channel 1 and the local oscillator fo in an adding circuit. Thus at the input of the recorder a signal consisting of audio channel 1, $f'1$ to $f'2$, the local oscillator frequency fo, and the sum signal obtained by heterodyning the local oscillator and the audio of channel 2, $fo+f1$ to $fo+f2$ is presented to the recording device, either a tape recorder or disk cutter.

* Received by the PGA, August 4, 1958.

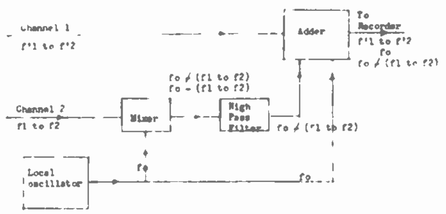

Fig. 1.

Fig. 2 shows the separation of these frequency spectra as presented to some recording media. Currently available recording equipment is not capable of recording two truly high-fidelity channels at standard disk or tape speeds combined according to this method since $fo+f2$ would then be about 35,000 cycles. However there is reason to believe that the recording art is advancing such that this can soon be accomplished.

Fidelity comparable to that currently being broadcasted on AM-FM stereo programs is possible on currently available equipment by this method of recording. Furthermore, such stereo broadcasts are possible on a single FM channel. In this case typical parameter values of Fig. 1 would be $f'1=16$ cps, $f'2=13{,}000$ cps, $fo=14{,}000$ cps, $f1=100$ cps, $f2=6000$ cps, $fo+f1=14{,}100$ cps, $fo+f2=20{,}000$ cps. Then, upon reproduction as described below, two stereo channels would be obtainable having frequency ranges of 16 to 13,000 cps and 100 to 6000 cps, respectively. This is approximately equivalent to fidelity obtained from conventional AM-FM stereo broadcasts.

Fig. 3 shows the block diagram of the circuit used to reproduce the original two channels of Fig. 1. The combined output of

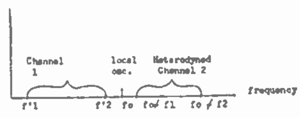

Fig. 2.

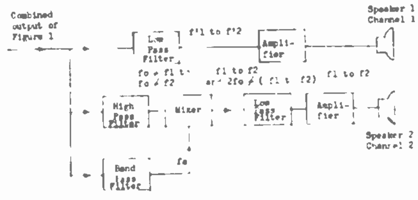

Fig. 3.

Fig. 1 is reproduced by a cartridge or tape head the output of which is connected to a filter system. A low-pass filter separates the frequencies of the original channel 1, $f'1$ to $f'2$, which are then amplified and reproduced by speaker 1 in original form. A band-pass filter isolates the local oscillator frequency and a high-pass filter separates the sum channel $fo+f1$ to $fo+f2$. These two are mixed, resulting in sum and difference frequencies. The difference between fo and $fo+f1$ to $fo+f2$, which are the difference frequencies, corresponds to the original channel 2, $f1$ to $f2$. These are separated from the sum frequencies by a low-pass filter and then amplified and reproduced by speaker 2.

A unique feature of this method is the recording of the local oscillator frequency. This eliminates the necessity of a critical frequency local oscillator in the recording system. In addition, no local oscillator is needed in the reproducing system. Any drift of the local oscillator in recording is exactly repeated in the reproducing system, thus cancelling any undesirable features of local oscillator drift or alignment.

The quality of the stereophonic reproduction is a direct function of the quality of the filters. This system is compatible with existing monaural recordings by utilizing only channel 1 of the reproducing system or by bypassing the filter system completely. Just as important is the fact that conventional monaural audio systems are compatible with recordings made by this heterodyne method since no special stylus or other equipment is necessary. Thus it would be possible to collect and play stereodisks recorded by this method on conventional monaural audio systems, with no danger of damage to the disks, until conversion to stereo is made by adding the required filters. Currently available recording and reproducing equipment requires that the frequencies fo and $fo+f1$ to $fo+f2$ be in the upper audio frequency range. Thus a conventional monaural reproducing system would require a whistle filter which would attenuate frequencies above and including fo. When equipment capable of recording two "high-fidelity" channels is available, allowing fo to be above 15,000 cps, reproduction of high-power, supersonic frequencies of the heterodyned channel 2 by the conventional monaural audio reproduction equipment may have undesirable psychological effects on the listener. Addition of a whistle filter for frequencies above and including fo might still be necessary.

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Richard E. Werner was born on June 1, 1928, in Ithaca, N. Y. He received the B.E.E. degree from Cornell University, Ithaca, N. Y. in 1950 and thereafter served three years in the U. S. Navy.

In 1953 he joined RCA where he became engaged in the development and design of microphones and assorted transducers. Early in 1958 he transferred to the audio communications activity, Missile Test Project, at Patrick Air Force Base, Fla.

Mr. Werner is a member of the Audio Engineering Society and an associate member of the Acoustical Society of America.

F. M. WIENER

Francis M. Wiener (SM'52) was born in 1911 in Prague, Czechoslovakia. After graduating with honors in communication engineering from the Technical University in Prague in 1934, he worked in industry as a designer of loudspeakers and electro-acoustic measuring facilities. From 1938 to 1941 he was at Harvard University, Cambridge, Mass. where he received the S.M. degree in 1939 and the S.D. degree in 1941. He was an Eveleth Scholar and an instructor in physics and communication engineering in 1939–1940, and a McKay Scholar in 1940–1941.

During the war years he served as a special research associate at the Electro-Acoustic Laboratory at Harvard and as a research fellow at the Psycho-Acoustic Laboratory from 1945 to 1946. From 1946 to 1952 he was a member of the technical staff of the Bell Telephone Laboratories and engineering section head at the Spencer-Kennedy Laboratories from 1952 to 1954. He was a member of the faculty of the Brooklyn Polytechnic Institute from 1948–1952 before joining Bolt, Beranek, and Newman Inc. in Cambridge. Mass.

Dr. Wiener has published many papers in the field of theoretical acoustics, psycho-acoustics, acoustic measurements, and atmospheric acoustics. He is a Fellow of the Acoustical Society of America, and a member of Sigma Xi and the AAAS. He is currently Chairman of Writing Group S1-W25 of the American Standards Association.

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