IRE Transactions on AUDIO

Volume AU-8 SEPTEMBER-OCTOBER, 1960 Number 5

Published Bi-Monthly

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PUBLISHED BY THE

Professional Group on Audio

IRE PROFESSIONAL GROUP ON AUDIO

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IRE TRANSACTIONS® ON AUDIO

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The Editor's Corner

AUDIO TERMINOLOGY

Audio Amplifier—A device for adding distortion to an audio signal.
Compliance—The result of weak resistance.
FM—A system for listening to a phonograph played at a remote location.
High Fidelity—What the sales department recently discovered in equipment you had been making all these years.
High-Fi—A phonograph; also see FM.
Stereo—A more expensive phonograph for playing more expensive stereo records through more loudspeakers.
Super-Fi—Your system, as compared to the one your friend owns.
Ultra-Fi—What your friend thinks of his system, compared to yours.
High Fi Record—Any phonograph record pressed later than the year 1926.
Loudspeaker—A transducer for converting electrical energy into noisy energy.
Program Equalizer—Means for modifying a program according to judgment of the music director.
Tone Control—Means for neutralizing the judgment of the music director.
Flat Response—Characteristic of an audio component which tends to flatten the pocketbook.

—MARVIN CAMRAS, *Editor*

STEREO COMMERCIALISM

Peter Tappan's cogent observations in the Editor's Corner for May–June 1960 must strike a responsive chord in all serious workers.

I think most old-timers in the audio game look with some contempt upon the machinations of the later-day entrepreneurs whose sole interest in the art seems to be to cash in and get out as quickly as expediency will permit. It seems a pity that nothing can be done to stem this tide toward moral bankruptcy. The audio "game" as Mr. Tappan puts it, is an art which is now being prostituted by industry. The company who interests itself in producing true quality equipment is rapidly being relegated to the brankrupt status, simply because most people do not know the difference—and the "smart" producers to which Mr. Tappan refers, realizing the situation, are quick to take advantage of it.

If equipment and records get much worse than they are today, with an occasional exception, those of us interested in the art will be ashamed to admit association with the industry. The present trend cannot do otherwise than leave a benighted and disillusioned public in its wake—to the detriment of all ethical workers and organizations.

E. D. NUNN
Audiophile Records, Inc.
Saukville, Wisconsin

PGA News

CHAPTER NEWS

Chicago, Ill.

The last meeting of the spring season featured Ladies Night, and was held on Wednesday, May 25, 1960, sponsored jointly with the Chicago Acoustical Audio Group. Dr. Dan Q. Posin, famous educator and TV personality, spoke on "The Age of Space."

On September 9, Carl G. Eilers of Zenith Radio Corporation, spoke on "Receiver Design Considerations for Stereophonic FM Multiplex Broadcasting." As reported in *Scanfax:*

"A receiver designed for a stereophonic FM multiplex broadcasting system requires the re-evaluation of the complete tuner. The receiver should have good sensitivity so that the stereophonic signal-to-noise ratio is acceptable in the fringe areas.

"AM rejection is improved when a limiter and ratio detector are used instead of a limiter and discriminator combination. The distortion characteristics of the receiver are improved when AGC and a wider IF-FM detector combination is used.

"Circuits within the receiver are analyzed relative to stereophonic cross-talk. Various circuit approaches for demodulating the subcarrier will be discussed and compared on the basis of complexity, stability, and performance. Effects of multipath and ignition pulse interference on stereophonic receiver performance will be described.

"Carl Eilers received his BSEE from Purdue University in 1948 and MSEE from Northwestern University in 1956. He joined Zenith's research department in 1948 and for the past 12 years has been working in subscription television systems research. He is now assistant supervisor of the department and also is engaged in stereophonic broadcasting systems research."

Dayton, Ohio

According to the June–July edition of the Dayton *Waveguide:*

"The PGA 1959–1960 season provided two interesting sessions.

"The first meeting of the year (December) presented a paper on 'Quality Factors in Magnetic Tape.' The speaker was Mr. Vernon Kuellner of the Ampex Corporation. The point was made that since magnetic tape is a component of delicate precision instruments it must be manufactured to instrument specifications. A Color film showing the manufacture and test of this tape demonstrated its title 'Objective Perfection.'

Our March meeting was to present the aspects of 'Efficient Transmission of Speech Information.' We did not anticipate the effect of old man weather since our speaker, Dr. James L. Flanagan of the Bell Telephone Laboratories, was weathered in at Newark airport and was unable to get here. A summary of his talk appeared in the March issue of *Waveguide*.

"The following officers were elected for the 1960–61 season:

"Chairman—Taulbee P. Mountz; Vice-Chairman-Program Chairman—Stanley E. Weber; Secretary—Albert P. Parker.

"These officers shold provide rigorous leadership for the chapter since they are all avid audio enthusiasts."

T. P. Mountz **S. E. Weber** **A. P. Parker**

Notice

PGA Chapter secretaries are asked to send announcements and reports of Chapter meetings directly to the editor of these Transactions, until the Chapters Committee is reactivated. We are grateful to J. Ross MacDonald for the fine job he did as chairman of this committee until his term expired last spring.

BACK ISSUES OF TRANSACTIONS WANTED

Electronic Communications, Inc., P. O. Box 12248, St. Petersburg 33, Florida, needs selected back issues of certain IRE TRANSACTIONS for its library. Those willing to sell or contribute please address "The Librarian" for details.

CALL FOR PAPERS

1961 IRE INTERNATIONAL CONVENTION

March 20–23, 1961

Waldorf-Astoria Hotel and New York Coliseum, New York, N. Y.

Prospective authors are requested to submit all of the following information by the *DEADLINE DATE OF OCTOBER 21, 1960.*

1) 100-word abstract *in triplicate,* title of paper, name and address
2) 500-word summary *in triplicate,* title of paper, name and address
3) Indicate the technical field in which your paper falls:

Aeronautical & Navigational Electronics
Antennas & Propagation
Audio
Automatic Control
Bio-Medical Electronics
Broadcast & Television Receivers
Broadcasting
Circuit Theory
Communications Systems
Component Parts
Education
Electron Devices
Electronic Computers
Engineering Management
Engineering Writing & Speech
Human Factors in Electronics
Industrial Electronics
Information Theory
Instrumentation
Microwave Theory & Techniques
Military Electronics
Nuclear Science
Production Techniques
Radio Frequency Interference
Reliability & Quality Control
Space Electronics & Telemetry
Ultrasonics Engineering
Vehicular Communications

Note: Original papers only will be considered, not published or presented prior to the 1961 IRE International Convention; any necessary military or company clearance of paper is to be granted prior to submittal.

Address all material to:

Dr. Gordon K. Teal, Chairman
1961 Technical Program Committee
The Institute of Radio Engineers, Inc.
1 East 79th Street, New York 21, N. Y.

Perception of Stereophonic Effect as a Function of Frequency*

W. H. BEAUBIEN†, MEMBER, IRE, AND H. B. MOORE†

Summary—A literature study and listening tests have been conducted to contribute to an understanding of the stereophonic effect as a function of frequency. The literature study failed to reveal tests showing loss of stereophonic direction for any part of the audio spectrum and pointed to arrival time difference of the transient portion of sound waves as the significant contributor to stereophonic perception.

Tests employing actual program material with a specially developed Stereo Spectrum Selector showed the extreme lower frequencies to have equal or superior directional content to the higher frequencies. The perception loss at any frequency may be of a quantitative nature rather than strictly related to certain wavelengths. Test results and consideration for future developments suggest adoption of full frequency stereophonic systems.

Introduction

A BETTER understanding of the stereophonic effect as a function of frequency would help in the establishment of standards for radio and television stereophonic broadcasting, and would also be desirable as an effective aid in the efficient design of tape and phonograph home sound reproduction systems.

For example, many different limits for stereophonic perception at various upper and lower frequencies have been mentioned in the past two years. These suggested limits usually lead to practical system advantages such as reduced RF bandwidth requirements for broadcasting systems, and also certain economies and marketing niceties for tape or phonograph equipment.

Enthusiasm for these considerations, however, could tend to obscure the true nature of the stereophonic effect. More studies of stereophonic fundamentals, on the other hand, could help to prevent that occurrence, and could also provide for future advancement in the state of the art.

In order to contribute to this important area of acoustical technology, we have been conducting a study which is characterized by two parallel and continuous efforts. These are a literature search and a listening test program.

This is a progress report of the results to date from these efforts. It will be presented by referring first to a brief glance at the literature study and some related theoretical considerations. Following these will be a description of the special test equipment and the testing techniques employed. Tabulated data will then be given with resulting conclusions listed. Also, some speculative analysis of the results will be undertaken with the objective of stimulating interest for additional exploration of these concepts by other experimenters.

* Received by the PGA, February 25, 1960. Presented before the AES Annual Meeting, October 8, 1959, and reprinted with permission from the *Journal of the Audio Engineering Society*, April, 1960.

† General Electric Co., Utica, N. Y.

Literature Survey

The literature search uncovered 88 references dealing with work from as long ago as 1921 up to the present. Included were papers from foreign countries such as England, Italy, France, Holland, Germany, and Russia. Noticeably absent were tabulated test results with loudspeakers indicating lack of stereophonic perceptions over any particular frequency range.

One Russian scientist [61] described tests of an effect similar in mechanism to the binaural but which can be realized by stimulating the cutaneous (skin) receptors. A marked wooden stick was placed successively between the left and right hands, knees, and toes of blindfolded observers. The stick was struck and the observer was asked to identify the lateral point where it was struck by interpreting the sensations in the particular parts of the anatomy where the stick was placed. The accuracies achieved were impressive.

This purely physical directional perception may be further verified by knowledge of the ability for a deaf person to perceive the point of origin of a foot stomp on the floor; and may be the reason that some hi-fi fans like to play their systems at loud volumes—for body stimulation.

However, the great majority of the papers were not so dramatically in variance with anticipated theories and most of them involved work with earphones.

Trimble [81] found that directional perception by intensity through earphones was independent of frequency and that phase sensitivity dropped off about 1000–2000 cycles. Trimble also discovered that intensity was almost as effective when operating antagonistically to phase difference as when alone.

Since the linear distance between the ears is not enough to produce large intensity differences from loudspeaker sound sources, many experimenters have measured and cited the shadowing effect of the head at frequencies above 300 cycles as the cause of intensity derived effects.

Perception at lower frequencies is then thought to be based on phase difference. However, reports of actual stereophonic perception tests to evaluate these theories using loudspeakers rather than earphones were not found in the literature!

Snow [71] used loudspeakers to reproduce a human voice while investigating the effects of arrival time. His conclusions indicated that arrival time was important but might contribute confusion to a stereophonic system.

On the other hand, Clark, Dutton, and Vanderlyn [21] considered difference of arrival time of a wave front at the two ears as the most significant factor. They also believed that the brain uses some sort of nerve combination such that one nerve requires stimulation by two others simultaneously before it will respond. Computer engineers will recognize this hypothetical effect as being similar to the "logic and gate" so common to computer circuitry. A firing of any one of the localized "gates" theoretically constitutes measurement of arrival time difference of a pulse at the two ears. According to Clark, Dutton, and Vanderlyn, some evidence in support of their theory has been published. Experiments on cats are described in which clicks separated by a known time interval were supplied to the two ears independently and the right and left lobes of the brain observed. In accord with the "gate" theory, maximum response was obtained from the right lobe when the click to the left ear was advanced by a time corresponding to a sound coming from the left and vice versa.

Moir and Leslie [54] mention that the ear can ignore reflected sounds and state that an inhibiting effect has been found in researches into the neural mechanism where the sense organ once discharged is unable to fire again for a time interval up to approximately two milliseconds. A second longer but unexplained inhibiting effect is also mentioned.

Theoretical Considerations

This information provided by available literature helps us postulate a theoretical picture of the stereophonic effect.

The inhibiting effects mentioned by Moir and Leslie may be responsible for the Haas, or precedence effect. They also probably make it possible to perceive direction even in a live room with a large amount of reverberation. Perhaps the "gates" respond to first the direct sound wave reaching the ears and are not able to "reset" by the time the reverberations reach the ears or by the time standing waves have built up.

In addition, these inhibiting effects also tend to demonstrate the importance of the first part, or transient, of the wave front to reach the ears and suggest why sharply transient sounds such as switching "clicks" seem to be much more directionally perceptible than continuous waves.

For example, we have made preliminary tests using audio signal generators and intensity differences in loudspeakers to attempt directional perception vs frequency checks, and poor results at all frequencies were realized except when the observer listened for the click turning on the generator. Even when the click was eliminated and the signal strength raised gradually, the only consistent results came from listeners clever enough to register their first impression as to the sound source location. The majority, however, turned in ambiguous results at all frequencies.

We felt substance had been given to the theory that the arrival time of transient portions of program material was the significant factor in stereophonic perception. We then decided to run a series of tests with actual program material to determine the directional perception as a function of frequency.

Stereo Spectrum Selector

In order to evaluate the directional contribution of the lower and upper frequency components in actual stereophonic program material, it was necessary to develop original techniques and equipment to eliminate selectively these frequency components without otherwise affecting the acoustical reproduction characteristics of a system.

Stereophonic sound systems of today usually contain two separate channels of information. For purposes of this report these will be labeled L (left) and R (right). If the algebraic sum $(L+R)$ can be derived, it represents the monophonic portion of the program; and if the algebraic difference $(L-R)$ is produced, it represents the stereophonic directional content.

It is then possible to eliminate specific portions of the $L-R$ information by passing it through high pass (to test lows) or low pass (to test highs) filters. In order to eliminate system phase ambiguities resulting from the inherent phase shifts in these filters a compensating phase shift is imposed upon the $L+R$ information.

The sum of these two signals (L') fed into one amplifier and speaker and the difference (R') fed to another amplifier and speaker will constitute a stereophonic system with full range audio response but with selectively limited directional information.

Frequency components falling within the attenuation range of the filter setting will tend to appear in the middle of the two speakers regardless of their intended position in the original program material. By alternately switching between this limited stereo and full stereo, observers could be tested for their ability to correctly identify full stereo. The Stereophonic Spectrum Selector was developed along these lines. A block diagram of the testing system is shown in Fig. 1.

Much time and effort was spent in preparing the equipment to meet high performance goals. Particular troubles were encountered in developing the matrix circuits so that the full stereo would be normal in all respects and data to be presented will show that this goal was realized. A Rondine turntable was employed together with a General Electric reluctance cartridge, tone arm, pre-amplifiers, amplifiers, and speakers. The system response was basically adjusted for RIAA compensation.

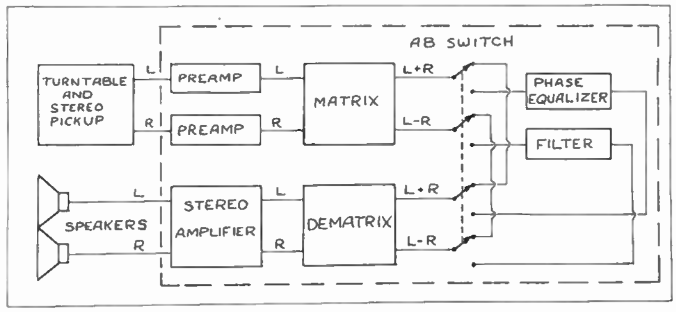


Fig. 1—Block diagram of stereo spectrum selector.

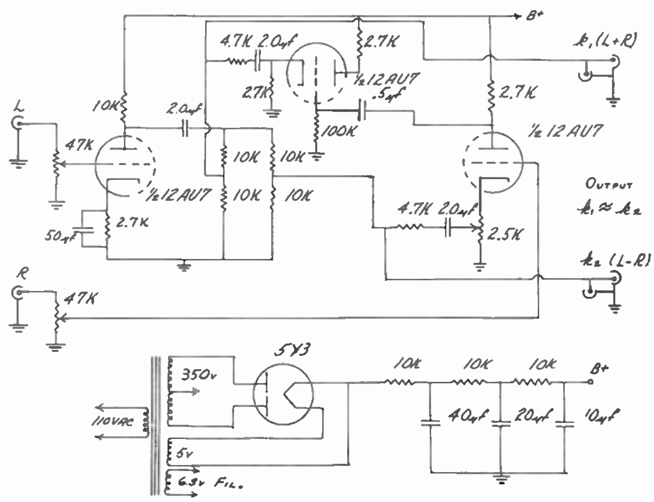


Fig. 2—Matrixor.

The circuit for matrixing (produces $L+R$ and $L-R$) is given in Fig. 2. The matrixor produced a flat response from 40 to 20,000 cycles with a gain less than one. The separation over this range was better than 30 db. The criteria used for adjusting over the band were as follows:

1) With input $L=K$, $R=O$, $L+R$ must be within ± 3 per cent of $L-R$.
2) With input $R=K$, $L=O$, $L+R$ must be within ± 3 per cent of $L-R$.
3) With input $L=R=K$, $L+R$ must be within ± 2 per cent of two times the nominal output of conditions 1 and 2 above, while $L-R$ must be better than 30 db down.

The second matrixor after the filter (restores L' and R') is identical to the first matrixor and adjusted the same. In use, the information going into its input and coming from its output is essentially the inverse of the first matrixor.

The amplifier was adjusted to provide $\frac{1}{4}$ watt of power to each speaker with a 1-kc, 7.5-millivolt test signal into each channel. Then the system response at the speaker terminals was set in accordance with the RIAA reproducing characteristic curve except during the testing of the low frequency information the bass was boosted 5 db at 100 cycles to compensate for speaker response falloff. The channel outputs tracked within ± 1 db.

The filter and equalizer were designed and constructed specifically for these tests. The filter circuits are given in Fig. 3 high pass (lows test) and Fig. 4 low pass (highs test). Adjustments $P3$ and $P4$ were brought to the front panel and calibrated in frequency at the 1.5 db down points so that the filter output would be 3 db down at calibrated frequency when both were set at the same frequency. The gain of nonfiltered information was set equal to unity by the internal cathode follower potentiometers. Filter response for the indicated knob settings is shown in Fig. 5 (lows) and Fig. 6 (highs).

The equalizer circuits are shown in Fig. 7 (lows) and Fig. 8 (highs). Adjustment $P6$ is a front panel knob calibrated in frequency. Gain of equalized information is unity over the entire band from 40 to 20,000 cycles.

To test the effectiveness of the equalizer to compensate for the phase shifts introduced by the filter, identical signals were fed into each unit and the outputs compared for phase correspondence. The results showed 0 phase angle between the signals within $\pm 2°$ as the frequencies were varied from 40 to 20,000 cycles.

Separation measurements were made for the system. An RCA constant velocity test record was used for frequencies from 15,000 to 1000 cycles and a London RIAA test record for 1000 cycles down. Each side of the record contained only one channel. Separation curves are shown in Fig. 9 using 0 db for reference throughout the spectrum.

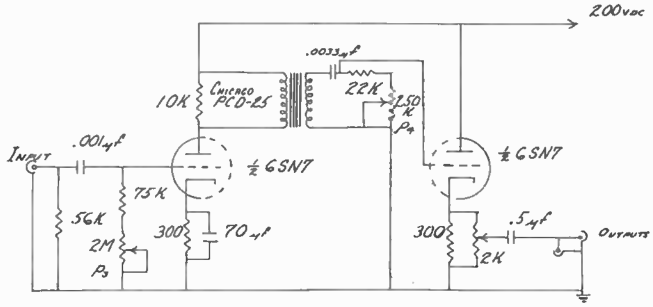


Fig. 3—High pass filter (test lows).

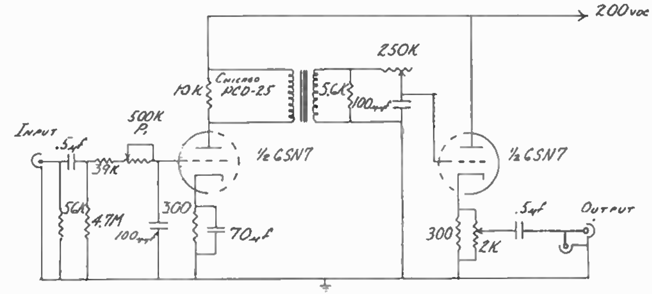


Fig. 4—Low pass filter (test highs).

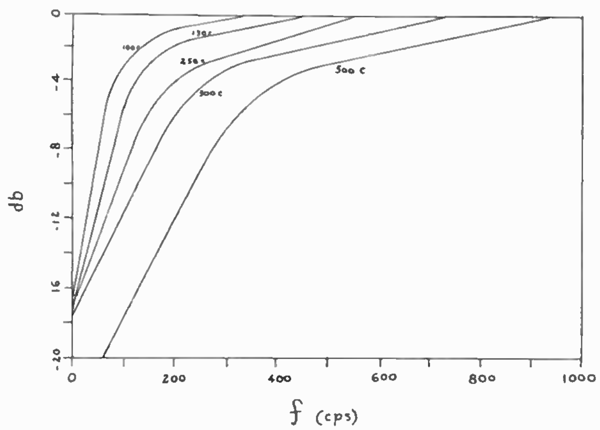


Fig. 5—Filter response (lows).

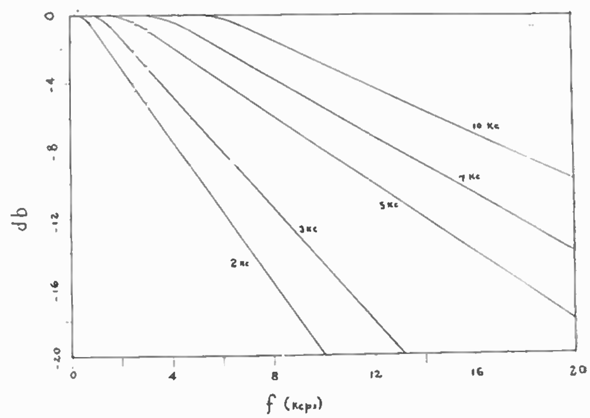


Fig. 6—Filter response (highs).

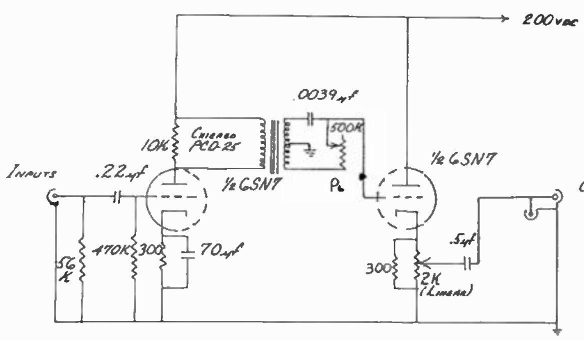


Fig. 7—High pass filter phase equalizer (test lows).

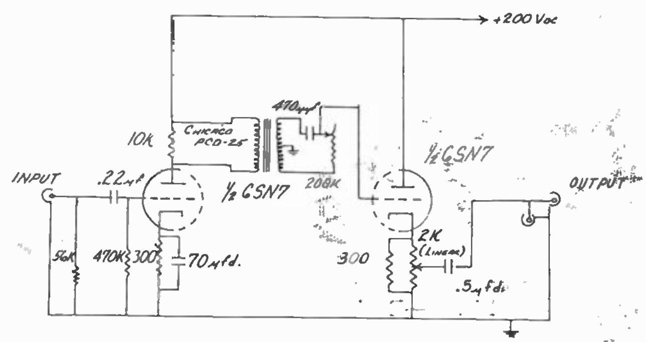


Fig. 8—Low pass filter phase equalizer (test highs).

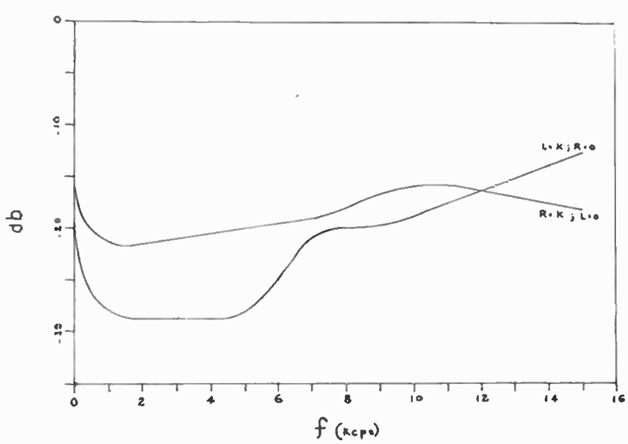


Fig. 9—System separation.

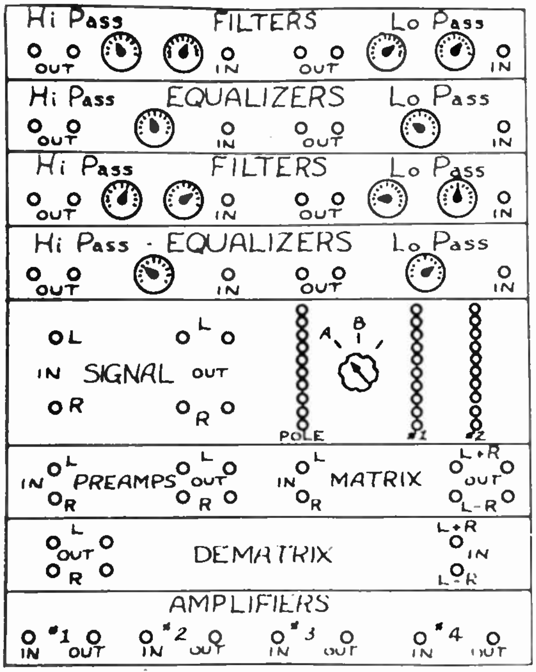


Fig. 10—Stereo spectrum selector panel.

Fig. 11—Equipment photograph.

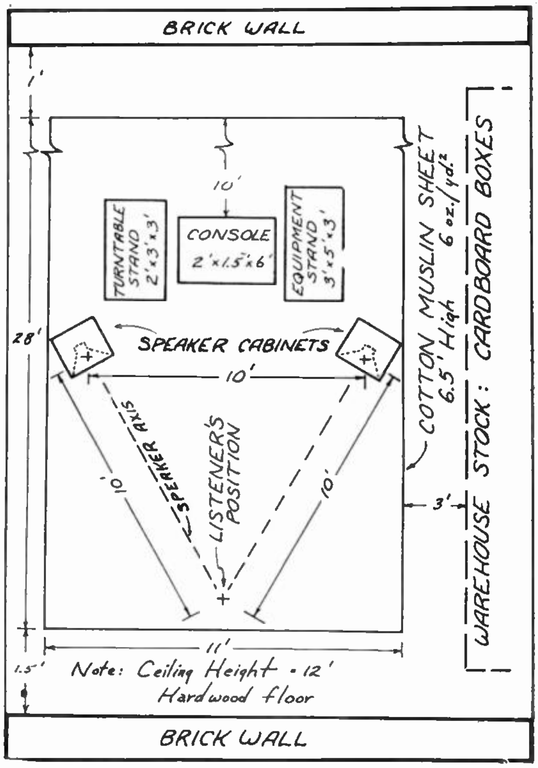


Fig. 12—Room and equipment arrangement.

All the above equipment excepting the turntable, tone arm, cartridge, and speakers is housed within a standard relay rack. The front panel arrangement is shown in Fig. 10. We call this unit the Stereo Spectrum Selector.

The photograph, Fig. 11, had the arrangement altered slightly for illustrative purposes.

Testing Techniques

The equipment in the test room was, however, actually arranged in accordance with Fig. 12.

The switch on the panel that changed from "full" stereo to "limited" stereo (often called an A, B switch) was operated by a technician in accordance with signals from the listener.

Technical and nontechnical listeners were used. Half of the nontechnical listeners were women. The need for concentration by the observers was considered important enough for us to bar kibitzers from the area.

Medical hearing tests (125 cycles to 12,000 cycles) were conducted on *most* of the listeners but no correlation existed between hearing ability and listening test results.

All listeners were educated as to what they were comparing and told that low (or high) frequency instruments would seem to be further to the outside under full stereo conditions. The testing began with a roll-off at a specific high or low frequency and the subject was required to make five different decisions prior to proceeding to the next lower (or higher) roll-off frequency.

No time limit was imposed and the listener was permitted complete freedom in switching between conditions A and B. The full stereo condition was changed between A and B in accordance with a master test sheet for each of the five decisions. (Sometimes A was "full" stereo, sometimes B was "full" stereo.) The tests involving low frequencies were conducted at a different time than those involving high frequencies.

The recordings that were used varied in accordance with listeners' desires. Selections were heard from the following recordings:

1) Capitol—Volume 1 "What's New on Capitol Stereo,"
2) RCA—"Gaite Parisienne,"
3) Counterpoint—"A Study in Stereo Sound,"
4) Angel—"Tchaikovsky Symphony No. 4 in F Minor, Opus 36,"
5) Capitol—"Shearing on Stage,"
6) Capitol—"Donnybrook with Donegan."

For the majority of the tests, selections from record 6) above were used because of good instrument localization. The music from the bass fiddle provided low frequency localization while the drums seemed to assist in high frequency stereo perception.

Test Data

The data of twenty-seven observers in the tests for low frequency perception is tabulated in Fig. 13. Circled items are incorrect decisions. f_0 is the frequency where the $L-R$ information is 3 db down.

Fig. 14 indicates the listener technical status, listener sex, and percentage of correct test replies. The average test score for specific low frequencies is given in the right hand column. The average test score for all low frequencies for each listener is given in the bottom row.

The data of twenty-seven observers in the tests for high frequency perception is tabulated in Fig. 15. Circled items are incorrect decisions. f_0 is again the frequency where the $L-R$ information is 3 db down.

Fig. 16 indicates listener technical status, listener sex, and percentage of correct test replies for the high frequencies. The average test score at a particular frequency is given in the right hand column. The average test score for all high frequencies for each listener is given in the bottom row.

Fig. 17 shows a plot for the average test scores for the low frequency tests, and also one for the average scores of the high frequency tests.

f_0	Test #	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21	22	23	24	25	26	27
500	1	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B
	2	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	B	A	A	A
	3	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A
	4	B	B	B	B	B	B	B	A	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B
	5	B	B	B	B	B	B	B	B	B	B	B	B	A	B	B	B	B	B	B	B	B	B	B	B	B	B	B
300	1	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A
	2	A	A	A	A	A	A	A	B	A	A	A	A	B	A	A	A	A	A	A	A	A	A	A	A	A	A	A
	3	B	B	B	B	B	B	B	B	B	B	B	B	A	B	B	B	B	B	B	B	B	B	B	B	B	B	B
	4	B	B	A	B	A	B	B	A	B	B	B	B	A	B	B	B	B	B	B	B	B	B	B	B	B	B	B
	5	A	A	A	A	A	A	A	B	A	A	A	A	B	A	A	A	A	A	A	A	A	B	A	A	A	A	A
250	1	B	B	A	B	A	B	B	A	B	B	B	B	A	B	B	B	B	B	B	B	B	A	B	B	B	B	B
	2	B	B	B	B	B	B	B	A	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B
	3	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B
	4	B	B	B	B	A	B	B	B	B	B	B	B	A	B	B	B	B	B	B	B	B	B	B	B	B	B	B
	5	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	B	A
150	1	A	A	A	A	A	A	A	A	A	A	A	A	B	B	A	A	A	A	A	A	A	A	A	A	A	A	A
	2	B	B	B	B	B	B	B	B	B	B	B	A	A	B	B	B	B	B	B	B	B	A	A	B	B	B	B
	3	A	B	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	B	A	A	A	A	B	A
	4	B	B	A	B	A	B	B	A	B	B	B	B	B	B	B	B	B	B	B	B	B	B	A	B	B	B	B
	5	B	B	B	B	B	B	B	A	A	B	A	B	B	B	B	B	B	A	B	B	A	B	A	B	B	B	B
100	1	B	B	B	B	B	B	B	A	A	B	B	B	A	B	B	B	B	B	B	B	B	B	A	B	B	B	B
	2	A	A	A	A	A	A	A	A	A	B	A	A	B	A	A	B	A	A	A	A	A	B	A	A	A	A	A
	3	B	B	A	A	B	B	A	A	A	B	B	B	A	B	B	A	B	B	B	B	A	B	B	B	B	B	B
	4	A	A	A	A	A	A	A	B	A	A	A	B	B	A	A	A	A	A	A	A	B	A	B	B	A	A	A
	5	A	A	A	A	A	A	A	B	A	A	A	A	B	A	A	A	A	A	B	A	A	A	B		A	A	A

Fig. 13—Test data (lows).

Code #	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21	22	23	24	25	26	27	Avg.
Sex	M	M	M	*M	F	M	*M	M	M	M	M	M	F	M	M	M	M	M	F	F	M	M	M	M	M	M	M	—
**T or NT	T	T	T	T	NT	T	T	T	T	NT	T	NT	NT	T	T	NT	T	T	NT	NT	T	T	T	T	T	T	NT	—
f_0 (CPS)																												
500	100	100	100	100	100	100	100	80	100	100	100	100	80	100	100	100	100	100	100	100	100	100	100	80	100	100	100	97.9
300	100	100	80	100	80	100	100	40	100	100	100	100	20	100	100	100	100	100	100	100	100	80	100	100	100	100	100	93.5
250	100	100	80	100	60	100	100	60	100	100	100	100	60	100	100	100	100	100	100	100	100	80	100	100	100	80	100	93.5
150	100	80	80	100	80	100	100	60	80	100	80	80	60	100	100	100	100	80	100	100	60	80	40	100	100	80	100	86.8
100	100	100	80	80	100	100	80	20	60	80	100	80	0	100	100	60	100	100	80	100	60	80	40	80	100	100	100	80.9
Average	100	96	84	96	84	100	96	52	88	96	96	92	44	100	100	92	100	96	96	100	84	84	76	92	100	92	100	

Average of Technical Listeners—91.4 Average of Non-Technical Listeners—88

* Substitute Listeners.
** T = Technical.
NT = Non-Technical.

Fig. 14—Test scores (lows).

f_0	Test #	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21	22	23	24	25	26	27
2,000	1	A	A	A	A	B	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A
	2	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A
	3	B	B	B	A	A	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B
	4	B	B	B	B	A	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	A	B	B	B
	5	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B
3,000	1	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	A	B	B	B	B	B	B	B	B	B	B	B
	2	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A
	3	B	B	B	B	A	B	B	B	B	B	B	B	B	B	B	B	A	B	B	B	B	B	B	B	B	B	B
	4	A	A	A	A	B	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A
	5	A	A	A	A	A	A	A	A	A	A	A	A	B	A	A	A	A	A	A	A	A	A	A	A	A	A	A
5,000	1	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	B	A	A	A
	2	B	B	B	B	A	B	B	B	B	B	B	B	B	B	B	B	A	B	B	B	B	B	B	A	B	B	B
	3	A	A	A	A	B	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A
	4	A	A	A	A	B	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A
	5	B	B	B	B	A	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B	B
7,000	1	B	B	B	B	B	B	B	A	B	B	B	B	B	B	B	B	A	B	B	B	B	B	B	B	B	B	B
	2	B	B	B	B	A	B	B	B	B	B	B	B	B	B	B	B	A	B	B	B	B	B	B	B	B	B	B
	3	A	A	A	A	A	A	B	A	A	A	A	A	B	A	A	A	B	A	A	A	A	A	A	A	A	A	A
	4.	B	B	B	B	A	B	A	B	B	B	B	B	B	B	B	B	B	B	B	B	A	B	B	B	B	B	B
	5	A	A	A	B	B	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	A	B	B
10,000	1	A	A	A	A	B	A	A	A	A	B	A	B	A	A	A	B	A	A	B	A	A	A	B	A	A	A	A
	2	B	B	A	A	B	A	A	B	A	A	A	B	A	A	B	A	B	B	A	A	A	A	A	B	B	A	A
	3	B	B	B	B	B	B	B	A	A	A	B	A	B	B	B	B	A	B	A	B	A	B	A	A	A	A	B
	4	A	A	A	A	A	A	B	B	B	A	B	B	A	A	B	B	A	A	A	B	B	A	A	A	A	B	A
	5	B	B	B	A	A	B	A	A	A	A	A	A	B	A	A	B	A	A	B	A	A	A	B	A	A	A	B

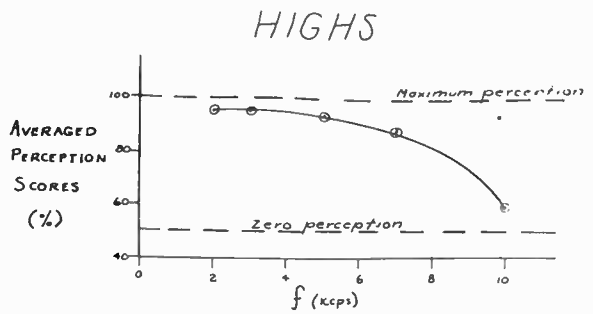
Fig. 15—Test data (highs).

Code #	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21	22	23	24	25	26	27	Avg.
Sex	M	M	M	F	F	M	M	M	M	M	M	M	F	M	M	M	M	M	F	F	M	M	M	M	M	M	M	—
**T or NT	T	T	T	NT	NT	T	NT	T	T	NT	T	NT	NT	T	T	NT	T	T	NT	NT	T	T	T	T	T	T	NT	—
f_0 (CPS)																												
2,000	100	100	100	80	40	100	100	100	100	100	100	100	100	100	100	100	100	100	100	100	100	100	100	80	100	100	100	96.4
3,000	100	100	100	100	60	100	100	100	100	100	100	100	80	100	100	80	80	100	100	100	100	100	100	100	100	100	100	96.4
5,000	100	100	100	100	20	100	100	100	100	100	100	100	100	100	100	100	80	100	100	100	100	100	100	60	100	100	100	95.0
7,000	100	100	100	80	40	100	60	80	100	100	100	100	80	100	100	100	40	100	100	100	80	100	100	100	100	80	80	89.5
10,000	80	80	100	80	40	100	60	20	40	40	60	0	100	80	40	60	40	60	60	60	40	80	60	40	40	40	100	59.2
Average	96	96	100	88	40	100	84	80	88	88	92	80	92	96	88	88	68	92	92	92	84	96	92	74	88	82	96	

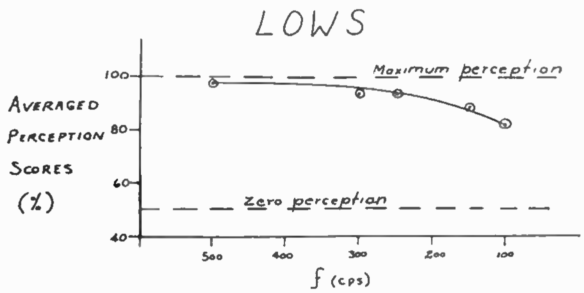
Average of Technical Listeners—89 Average of Non-Technical Listeners—84

** T = Technical.
NT = Non-Technical.

Fig. 16—Test scores (highs).



(a)



(b)

Fig. 17—Plot of test scores.

Conclusions

1) The high degree of directional perception with program material was in contrast to the ambiguities experienced in the preliminary tests with continuous waves.

2) Directional perception during the high frequency tests vanished at around 10 kilocycles.

3) Directional perception during the low frequency tests was 80 per cent, even at the lowest filter setting employed—100 cycles.

4) Nontechnical listeners did as well as technical listeners.

Speculative Analysis of Results

The improvement in stereophonic directional perception for actual program material compared with continuous waves probably results from the ability of the ears to measure arrival time difference of the transient characteristics inherent to sound waves present in program material.

This would obviously account for perception at the higher frequencies, but the situation at the lower frequencies may not be so easily understood. The high pass (test lows) filter used in these tests would not significantly alter steep transients. This leads to the interesting hypothesis that moderately sloped transients produced by lower pitched instruments represent an important contribution to their directional localization.

The very slight loss in perception with the low frequency filter adjusted to filter $L-R$ information below 100 cycles (*i.e.*, correct perception average score 80 per cent) may have been more the result of a quantity change in the stereo information than an effect characteristic of the wavelengths involved. To illustrate this point further, interesting results (similar or even lower perception scores) might come from eliminating an identical quantity of $L-R$ frequencies from another part of the spectrum—say 1500–1600 cycles or 5000 to 5100 cycles, etc.

The perception scores in the low frequency tests as well as the high frequency tests are more remarkable when it is realized that a setting of the filter for one frequency did not completely attenuate the other frequencies below (or above) the indicated value.

For example, a reference to Fig. 5 will show that the $L-R$ information at 50 cycles is down only 9 db when the setting, or 3-db point, is at 100 cycles (and a reference to Fig. 6 will show that the $L-R$ information at 14,000 cycles is down only 9 db when the setting is at 7000 cycles). A step function filter would most probably result in higher average percentage scores for perception of the lows (or highs)!

In addition, the program information at the extreme low and high frequencies was undoubtedly limited by present recording and reproduction abilities. Future advancements in either or both of these areas would most probably result in higher perception scores. These considerations and the test results provide strong evidence that all low frequencies are important for a good stereophonic reproduction system. And even though the tests show that frequencies around 10,000 cycles are not directionally perceptible when removed from the $L-R$ information, expectations that improved scores would result from sharper cutoff filters and future advancements in recording and reproduction techniques, make it desirable that premature conclusions be avoided relative to their importance in stereophonic sound reproduction.

Therefore, good stereophonic sound systems would seem to require reproduction of the stereophonic effect over the entire audible band.

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Listener Ratings of Stereophonic Systems*

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***Summary*—Subjective listening tests have been completed which indicate that stereo in any of the forms compared is preferred over monaural, but normal stereo from two full range speakers well physically separated is, in all probability, the most preferred.**

I. Introduction

As a result of recent tests reported in a paper,[1] presented at the 1959 Fall AES Convention which indicated that high and low frequency stereophonic directional information was perceptible, it was suspected that differences between stereophonic sound reproduction techniques would be discernible. Therefore, to learn which reproduction techniques would be most appreciated, a program of subjective testing was undertaken.

II. Background

The home phonograph industry has marketed a variety of different stereophonic playback systems resulting in the generation of some confusion. The respective manufacturers quite naturally praise the merits of their systems and each considers their approach distinctly superior. Eliminating the economic and sales niceties of each system, this study was undertaken to determine which system approach is most appreciated when observers have the opportunity to compare the stereophonic sound from one against another while all other equipment conditions remain equal. The systems tested include those which might be classified as normal two-speaker stereo, three-speaker mixed-lows stereo, three speaker phantom-center stereo, one piece console-type stereo, and spread-sound console stereo.

* Received by the PGA, May 3, 1960. This paper was published in the 1960 International Convention Record, pt. 7, pp. 64–72.
† General Electric Co., Utica, N. Y.
[1] W. H. Beaubien and H. B. Moore, "Perception of the stereophonic effect as a function of frequency," *J. Audio Engrg. Soc.*, vol. 8, pp. 76–86; April, 1960.

III. Procedure

Listeners were selected from draftsmen, engineers, technicians, and secretaries. Only one listener was permitted in the testing area at a time. Each test required 15 decisions because it included six systems compared two at a time.

The listeners were educated to the extent that the only significant changes which should be judged should concern the reproduced sound panorama. They were informed that loudness and response differences between systems had been eliminated as much as possible because any such differences should not enter into their decisions. Each listener was given the instructions as seen in Fig. 1 and asked to record his decisions in the table provided. In filling out the test sheet, the listener compared condition P to condition T until he selected a preference. For example, if P was selected in this case, he wrote P into the upper left box. Next, he compared T with U and recorded his preference in the next block down, etc. Complete freedom of switching systems was provided the listener by the use of a remote control multiple-push-button switch.

The General Electric speakers employed included one model of the high fidelity component type whose response curves were almost identical. The amplifiers were also similar and were General Electric 20 watt high fidelity components. Speakers and amplifiers shifted jobs from test to test. Program material was selected which contained information in the center, as well as on the left and right.

You are about to compare different techniques used to reproduce stereo. The only changes you should hear will concern the *reproduced sound panorama:* The loudness and response differences have been reduced by adjusting the systems so that these differences are not discernible to a trained ear. This has been done because loudness and response differences should not enter into your decisions.

It is up to you to determine your order of preference of these six reproduction techniques (compare only two at a time and indicate your preference independent of other decisions).

Place letter of stereo condition preferred in boxes.

	T	U	A	Q	W
P					
T					
U					
A					
Q					

Test No. ________

Name ______________________ Date ________

Fig. 1—Instructions and typical test sheet.

Fig. 2 shows a block diagram of the equipment and one arrangement used. The eight amplifiers used were needed to correct for response characteristics between systems. The switching circuitry is given in Fig. 3 and shows, for example, that normal stereo was heard from speakers 1 and 4 and came from the dual amplifier directly under the dual preamplifier. Single-cabinet-console stereo was derived from the same dual amplifier, but came from speakers 2 and 3. Then the mixed-lows test involved this same dual amplifier, but LC prototype high pass filters were introduced to knock down the lows from speakers 1 and 4 while the matrixor, another amplifier, and a low-pass filter supplied the mixed lows to speaker 2. Final measurements of system responses were documented after being set by trained ears.

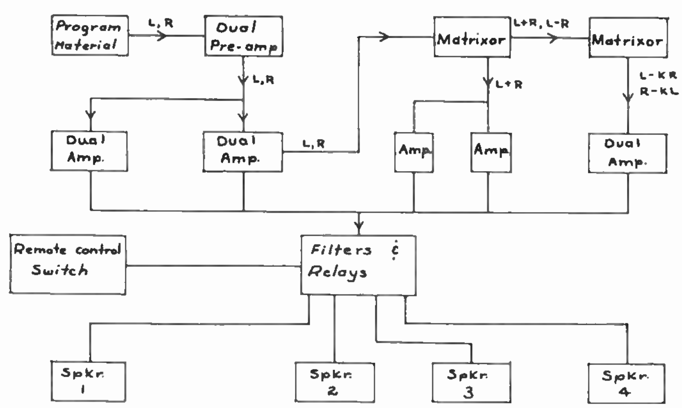


Fig. 2—Equipment arrangement block diagram.

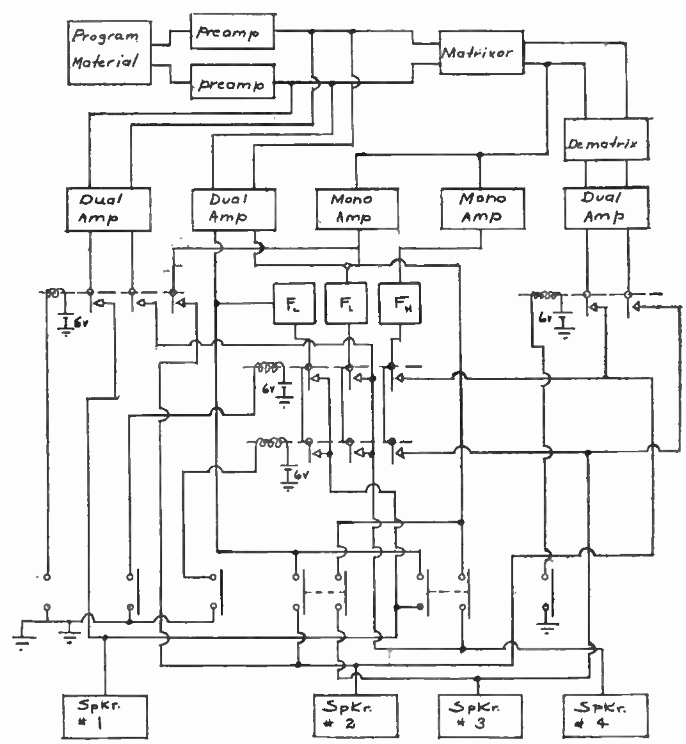
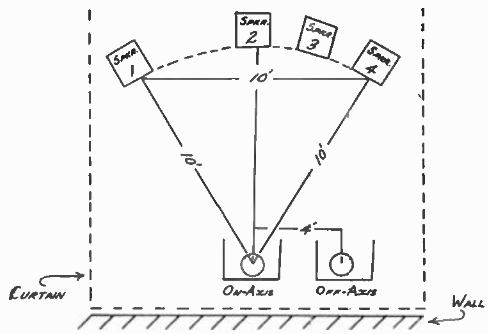


Fig. 3—Switching circuitry.

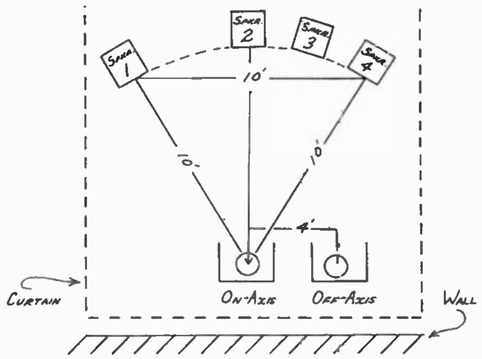
The initial series of tests comparing the systems listed in Fig. 4 were run using high fidelity component disk phonograph reproduction equipment.

In the remainder of the paper, these systems will be respectively referred to as: 1) Normal stereo, 2) Mixed-lows center, 3) Mixed-lows right, 4) Monaural, 5) Phantom center, 6) Console stereo.



System	*Speakers*
Normal Stereo	1 & 4
Highs Left and Right >300~ Mixed Lows in Center <600~	1, 2 & 4
Highs Left and Right >300~ Mixed Lows in Right Center <600~	1, 3 & 4
$L+R$ Monophonic	2
Normal Stereo & $L+R$ in Center (center level = −6 db)	1, 2 & 4
Console Stereo	2 & 3

Fig. 4—Disk tests. Listening area conditions: 1) Room was large. 2) Ceilings were high. 3) Curtains were light. 4) Wooden floor was uncovered.



System	*Speakers*
Normal Stereo	1 & 4
Highs Left and Right >300~ Mixed Lows in Center <600~	1, 2 & 4
Highs Left and Right >300~ Mixed Lows in Right Center <600~	1, 3 & 4
Normal Stereo & $L+R$ in Center (center level = side level)	1, 2 & 4
Console Stereo with $(L-R)$ increased 2.7 times	2 & 3
Console Stereo	2 & 3

Fig. 5—Tape tests. Listening area conditions: 1) Room was large. 2) Ceilings were high. 3) Curtains were light. 4) Uncovered wooden floor.

The record played was Capitol's "Donnybrook with Donegan," because it had a bass fiddle recorded on one channel, a drum on the other, and a piano near center. The playback was established such that the bass normally emanated from the left speaker. 12 listeners took the test sitting on axis. After changing the system identification letters, 12 listeners repeated the test the next day but sitting off axis to the right.

The last set of tests tabulated in Fig. 5 were conducted employing a broadcast quality tape deck in combination with the same high fidelity amplifiers and speakers. Significant deviations from Fig. 4 amount to:

1) Increasing the level of the center speaker for the phantom center system.
2) Substituting console stereo with the difference information increased 2.7 times to spread the stereo effect beyond the speakers in place of the monaural system. This will be referred to as *spread stereo console.*

The tape played was RCA's "Love in the Afternoon," which had a guitar recorded on one channel and an accordion on the other. The center was filled with information from the drums and bass instruments. Again, 12 listeners were tested on axis and 12 off axis.

IV. Results

The tabulated data collected from the 24 observers, who listened to the phonograph record, and from the 24 observers, who listened to the tape selections, are documented in Appendix I.

Table I is a summary of the disk tests. It was possible for any system to obtain 60 votes of preference from each listening position or a possible total of 120 votes.

Table II tabulates the votes of the tape tests. Since some of the differences between system total votes is small, it is important to determine whether the differences are significant or not.

TABLE I

Summary of Disk Results

	Votes On-Center	Votes Off-Center	Total Votes
Normal stereo	49	52	101
Phantom center	52	32	84
Mixed-lows center	28	48	76
Mixed-lows right	31	32	63
Stereo console	19	13	32
Monaural	1	3	4

TABLE II

Summary of Tape Results

	Votes On-Center	Votes Off-Center	Total Votes
Normal stereo	48	54	102
Mixed-lows right	45	41	86
Mixed-lows center	37	38	75
Phantom center	34	35	69
Spread-stereo console	11	6	17
Stereo console	5	6	11

Votes		(101) Normal Stereo	(84) Phantom Center	(76) Mixed-Lows Ctr.	(63) Mixed-Lows Rt.	(32) Stereo Console
(84)	Phantom Center	3.5 per cent				
(76)	Mixed-Lows Center	<0.1	23.5			
(63)	Mixed-Lows Right	<0.1	0.8	11.6		
(32)	Stereo Console	<0.1	<0.1	<0.1	<0.1	
(4)	Monaural	<0.1	<0.1	<0.1	<0.1	<0.1

$\boxed{\#}$ = Per cent probability of a random difference (Student's "t" test)
<0.1 per cent = Highly significant, 1–5 per cent = probably significant, >5 per cent = insignificant conclusion.

Fig. 6—Analysis of disk data.

Votes		(102) Normal Stereo	(86) Mixed-Lows Rt.	(75) Mixed-Lows Ctr.	(69) Phantom Center	(17) Spread-Stereo Console
(86)	Mixed-Lows Right	1.5 per cent				
(75)	Mixed-Lows Center	<0.1	14			
(69)	Phantom Center	<0.1	1.7	41		
(17)	Spread-Stereo Console	<0.1	<0.1	<0.1	<0.1	
(11)	Stereo Console	<0.1	<0.1	<0.1	<0.1	20

$\boxed{\#}$ = Per cent probability of a random difference (Student's "t" test).
<0.1 per cent = Highly significant, 1–5 per cent = probably significant, >5 per cent = insignificant conclusion.

Fig. 7—Analysis of tape data.

V. Discussion of Results

The testing approach is specifically designed for statistical analysis, where the probability of random ranking, random agreement, and random difference can be computed.[2] A sample set of calculations is included in Appendix II. The calculations show, via Snedecor's F test, that the ranking is accurate while application of the standard error test indicates the probability of random agreement to be less than 0.01 per cent.

Fig. 6 summarizes the significance of the disk test voting. For example, the probability that a random difference exists between the 101 votes assigned to normal stereo and the 84 votes assigned to stereo with a phantom-center speaker is 3.5 per cent. A 3.5 per cent probability means that a good chance exists of a preference for normal stereo as compared to the phantom-center stereo system. The less than 0.1 per cent figures under normal stereo indicates that a preference is very certain of existing for normal stereo over all other systems tabulated.

Similarly, Fig. 7 reviews the significance between the tape test voting results. This chart tells the same type of information as the previous one. For example, in this test, the 41 per cent probability that the difference is random between the 75 votes given the mixed-low-center system over the 69 votes accorded the phantom-center system means that if a preference exists, it was not demonstrated by this test. Fig. 8 presents the results in a plotted form for consolidated examination.

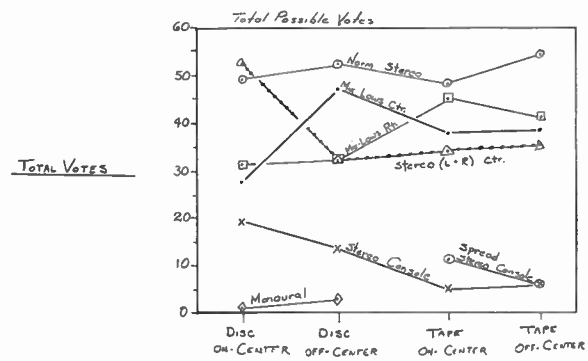


Fig. 8—Results.

The changes occurring between the disk tests on center and off center were probably a function of the extreme shifts in the sound panorama. For example, under the normal stereo condition, the bass fiddle was heard on the left. Then when listening to mixed-lows right, the bass fiddle was heard from the right and no left information existed. On the other hand, small panorama changes occurred during the tape tests, since the lows normally emanated from center, and extreme left and right information didn't disappear.

The change in popularity of the stereo system employing a phantom-center speaker may be related to the level of center information. The lower level used in the disk tests appears slightly preferred over the level used in the tape tests. This conjecture is compatible with the final results, since the lower the level of information from the center speaker, the closer this system approximates normal 2 speaker stereo.

[2] F. A. Olson and K. Schjonneberg, "Listening Test Methods and Evaluation," General Electric Co., Utica, N. Y., Publ. No. TIS60HRR2; March, 1960.

VI. Conclusions

1) Console stereo was significantly preferred to single-speaker $L+R$ sound.
2) Two-speaker normal stereo averaged an over-all high preference compared to other tested systems.
3) Of the forms of stereo tested, all which presented a wider angle spread were preferred to console-type stereo.
4) "Off axis" listening of normal two-speaker stereo supported its preference as well as, if not more than, "on axis" listening.

Subjective listening tests have been completed which indicate that stereo, in any of the forms compared, is preferred over monaural, but normal stereo from two full-range speakers well physically separated is, in all probability, the most preferred.

Explanation of Sample Calculation Sheet

A. To fill in the squares, the number of people that preferred one system over each of the other ones must be known. This number is put into the top half of the square. Since the *rows* represent favorable votes and the columns represent unfavorable votes, it can be seen that in the sample calculation, 10 people preferred "P" over "T" and 2 people preferred "T" over "P". The total, of course, must equal our total number of judges ($m=12$).

The bottom half of the square for each comparison is filled in with the $jC2$ of the favorable votes in that box. If j equals the number of favorable votes, then

$$jC2 = \frac{j(j-1)}{2}$$

and for the case of "P" vs. "T", our $j=10$ and

$$jC2 = \frac{j(j-1)}{2} = \frac{10(9)}{2} = 45$$

which is placed in the appropriate square. All the comparison squares are filled out in this manner.

B. On the right hand side on top.
1) $R=\sum j$ for that row (where j again equals the favorable votes, or for P row, $R=31$).
2) R^2 is then computed.
3) $P=R/N$, where $R=\sum j$ and $N=m(n-1)$. n is the number of systems being used in this comparison. N here $=12(6-1)=60$.
4) $\sum jC2$ is the sum of the individual $jC2$ for each *row*.
5) We then are interested in $\sum R^2=$ sum of R^2 column and $J=$ sum of $\sum jC2$ column.

C. With the above data, we can now compute the probabilities on the bottom of the calculation sheet.

1) *Probability of Random Ranking (Snedecor's F Test)*:

$$S_{\max} = \frac{m^2(n^3-n)}{12} = \frac{(144)(216-6)}{12}$$

$$= \frac{(144)(210)}{12} = 2520$$

from m and n again we can find k

$$k = \frac{3(n-1)}{n+1} = \frac{3(5)}{7} = 2.15$$

and $kS_{\max} = (2.15)(2520) = 5400$.
We know $\sum R^2 = 7212$.
Thus, $S=\sum R^2 - kS$ max $= 7212 = 5400 = 1812$
$S-1$ then $= 1811$.
Q is found then from the equation

$$Q = \frac{S_{\max}}{S-1} - 1 = \frac{2522}{1811} - 1 = 1.39 - 1 = 0.39.$$

"Snedecor's F" is given as $\dfrac{m-1}{Q} = \dfrac{11}{.39} = 28.$

Entering the Snedecor's F chart,[3] we find that for 12 judges and 6 systems and for a 1 per cent probability of random ranking, our Snedecor's F should be equal to 3.41. Our $F=28$ which indicates much less than 1 per cent probability of randomness.

2) *Probability of Random Agreement (Standard Error Test)*:

$$\eta = \frac{m(m-1)(n)(n-1)}{2(m-2)^2} = \frac{(12)(1)(6)(5)}{2(100)}$$

$$= \frac{(66)(30)}{100} = 19.8$$

$$\beta = \eta(m-3) = 19.8(9) = 178,$$

$$J = \sum(\sum jC^2) = 803,$$

$$Z = \frac{4J}{m-2} - \beta = \frac{(4)(803)}{10} - 178 = 320 - 178$$

$$= 142.$$

Our standard error then $= \sqrt{2Z} - \sqrt{2\eta - 1} = \sqrt{284} - \sqrt{39.6-1} = 16.9 - 6.2 = 10.7$.

On the table under this test, we can see that our probability of random agreement is way below 0.01 per cent.

[3] H. Arkin and R. R. Colton, "Tables for Statisticians," Barnes and Noble, Inc., New York, N. Y., p. 117; 1953.

3) *Probability of Random Difference (Student's t Test):*
Since our agreement was good and our ranking accurate, we can enter the the sequence of preference into the chart under this test with its appropriate p in the adjacent column. We are interested in the significance of this preference, and we can test the significance of any two systems using the following method.

a) Comparing A and W we then call $p1=0.87$ and $p2=0.82$ (the p's for A and W).
b) Find $\bar{p}=(p1+p2)/2=0.845$.
c) Find $\Delta p=p1-p2=0.05$.

D. We then find the value of the Student's t from the equation

$$t = \Delta p \sqrt{\frac{N}{2\bar{p}(1-\bar{p})}},$$

So

$$t_{AW} = 0.05\sqrt{\frac{60}{1.69(0.155)}} = 0.05\sqrt{\frac{60}{0.262}}$$
$$= 0.05\sqrt{229} = 0.05(15) = 7.5.$$

E. Entering this value along with the degrees of freedom $\phi=N-2=58°$ into the Student's t distribution curve,[4] we interpolate a probability of 44 per cent.

F. A reference level of 5 per cent probability is usually accepted as significant, and it can be seen that the 44 per cent probability between A and W is too large; and, therefore, if a preference exists between A and W, it was not demonstrated by this test.

[4] *Ibid.*, p. 116.

Appendix I

Data

Disk Tests On-Center

P = mixed-lows right, T = console stereo, W = normal stereo, U = mixed-lows center, Q = phantom center, A = monaural.

Listener \ Systems Compared	PT	PW	PU	PQ	PA	TW	TU	TQ	TA	WU	WQ	WA	UQ	UA	QA
1	T	W	P	Q	P	T	T	T	T	U	Q	W	Q	U	Q
2	P	W	P	Q	P	W	U	Q	T	W	Q	W	Q	U	Q
3	P	W	U	Q	P	W	U	Q	T	W	W	W	Q	U	Q
4	P	W	P	Q	P	W	U	Q	T	W	Q	W	Q	U	Q
5	P	W	U	Q	P	W	U	Q	T	W	Q	W	U	U	Q
6	T	W	U	Q	A	T	U	Q	T	W	Q	W	Q	U	Q
7	P	P	P	Q	P	W	U	Q	T	W	W	W	Q	U	Q
8	P	W	P	Q	P	W	U	Q	T	W	W	W	Q	U	Q
9	P	W	P	Q	P	W	U	Q	T	W	Q	W	Q	U	Q
10	P	W	P	Q	P	W	U	Q	T	W	W	W	Q	U	Q
11	P	W	P	Q	P	W	T	Q	T	U	W	W	Q	U	Q
12	P	W	P	Q	P	W	U	Q	T	W	W	W	Q	U	Q

Cont'd on next page

Off-Center

U = phantom center, T = stereo console, A = normal stereo, P = mixed-lows center, W = mixed-lows right, Q = monaural.

LISTENER \ SYSTEMS COMPARED	UT	UA	UP	UW	UQ	TA	TP	TW	TQ	AP	AW	AQ	PW	PQ	WQ
1	T	A	P	W	U	A	P	W	T	A	A	A	P	P	W
2A	U	A	P	U	U	A	P	W	T	A	A	A	P	P	W
3A	U	A	P	W	U	A	P	W	Q	P	W	A	P	P	W
4	U	A	P	U	U	A	P	W	T	P	A	A	P	P	W
5	U	A	P	W	U	A	P	W	T	P	A	A	P	P	W
6	U	A	P	W	Q	T	P	W	T	P	A	Q	P	P	W
7A	U	A	U	U	U	A	P	W	T	A	A	A	P	P	W
8	U	U	U	U	U	A	P	W	T	A	A	A	W	P	W
9A	U	A	P	U	U	A	P	W	T	A	A	A	P	P	W
10A	U	A	P	W	U	A	P	W	T	A	A	A	P	P	W
11	U	A	P	W	U	A	P	W	T	A	A	A	P	P	W
12	U	A	U	U	U	A	P	W	T	A	A	A	P	P	W

Tape Tests On-Center

Q = phantom center, U = console stereo, A = normal stereo, T = mixed-lows center, P = mixed-lows right, W = spread-stereo console.

LISTENER \ SYSTEMS COMPARED	QU	QA	QT	QP	QW	UA	UT	UP	UW	AT	AP	AW	TP	TW	PW
1	Q	A	Q	P	Q	A	T	P	W	A	P	A	P	T	P
2	Q	A	T	P	Q	A	T	P	W	A	A	A	P	T	P
3	Q	A	T	P	Q	A	T	P	W	T	A	A	P	T	P
4	Q	A	T	Q	W	A	T	P	U	A	A	A	T	W	W
5	Q	A	T	P	Q	A	T	P	U	A	P	A	P	T	P
6	Q	Q	Q	Q	Q	A	T	P	U	A	P	A	P	T	P
7	Q	A	Q	P	Q	A	T	P	W	A	A	A	T	T	P
8	Q	A	T	P	Q	A	T	P	W	A	P	A	P	T	P
9A	Q	A	T	P	W	A	T	P	W	T	P	A	T	T	P
10A	Q	A	T	P	Q	A	T	P	W	A	A	A	P	T	P
2A	Q	Q	Q	Q	Q	A	T	P	U	A	A	A	P	T	P
3A	Q	Q	Q	Q	Q	A	T	P	U	T	P	A	T	T	P

Off-Center

W = mixed lows center, P = normal stereo, T = stereo console, A = phantom center, U = spread-stereo console, Q = mixed-lows right.

LISTENER \ SYSTEMS COMPARED	WP	WT	WA	WU	WQ	PT	PA	PU	PQ	TA	TU	TQ	AU	AQ	UQ
1	W	W	W	W	W	P	P	P	Q	A	T	Q	A	Q	Q
2	P	W	W	W	Q	P	P	P	P	A	U	Q	A	Q	Q
3A	P	W	A	W	Q	P	P	P	P	A	T	Q	A	A	Q
4	P	W	W	W	Q	P	P	P	P	A	T	Q	A	Q	Q
5	P	W	A	W	Q	P	P	P	P	A	T	Q	A	A	Q
6A	P	W	A	W	W	P	A	P	P	A	U	Q	A	A	Q
7	P	W	W	W	W	P	P	P	P	A	T	Q	A	Q	Q
8	P	W	A	W	W	P	P	P	Q	A	U	Q	A	A	Q
9A	P	W	W	W	Q	P	P	P	P	A	T	Q	A	A	Q
10A	P	W	W	W	Q	P	P	P	P	A	U	Q	A	Q	Q
2A	W	W	W	W	W	P	P	P	Q	A	U	Q	A	Q	Q
12	P	W	A	W	Q	P	P	P	P	A	U	Q	A	Q	Q

Appendix II

Sample Calculation

Date: 3/4/60 Title: Disk (On-Center Test) Engr.: Peter R. Ransom

	P	T	W	U	A	Q	C	G	R	R^2	P	j^{c2}
P		10	1	9	0	11			31	961	0.518	136
		45	0	36	0	55						
T	2		2	2	1	12			19	361	0.317	69
	1		1	1	0	66						
W	11	10		10	6	12			49	2401	0.82	226
	55	45		45	15	66						
U	3	10	2		1	12			28	784	0.467	115
	3	45	1		0	66						
A	12	11	6	11		12			52	2704	0.87	257
	66	55	15	55		66						
Q	1	0	0	0	0				1	1	0.017	0
	0	0	0	0	0							
C										7212		803
G												

Number of judges $m = 12$
Number of sets $n = 6$

$$j^{c2} = \frac{j(j-1)}{2} \qquad \sum R^2 = 7212 \qquad J = \sum j^{c2} = 803$$

Probability of random difference

$$p = \frac{R}{N}; \quad N = m(n-1) = 60$$

$$\bar{p} = \frac{p1 + p2}{2}; \quad \Delta p = p1 - p2$$

$$t = \Delta p \sqrt{\frac{N}{2\bar{p}(1-\bar{p})}}$$

Prob. = Curve 1 or 2[4]

Set	p	$\bar{p}$	Δp	t	Prob
A	0.87				
		0.845	0.05	0.75	44
W	0.82				
		0.67	0.3	3.5	0.1
P	0.52				
		0.49	0.05	0.55	53
U	0.47				
		0.39	0.15	1.67	10
T	0.32				
		0.168	0.303	4.39	<0.1
Q	0.017				

Probability of a random agreement

$$\eta = \frac{m(m-1)n(n-1)}{2(m-2)^2} = 19.8$$

$$\beta = \eta(m-3) = 178$$

$$J = 803$$

$$Z = \frac{4J}{m-2} - \beta = 142$$

$$\text{Std. Error} = \sqrt{2Z} - \sqrt{2\eta - 1} = 10.7$$

Standard Error	Probability of random agreement
1	31.7 per cent
2	4.6 per cent
3	0.3 per cent
4	0.01 per cent

10.7 = No Randomness

Probability of random ranking

$$S_{\max} = 2520$$

$$S_{\max} + 2 = 2522$$

$$kS_{\max} = 5400$$

$$\sum R^2 = 7212$$

$$S = \sum R^2 - kS_{\max} = 7212 - 5400 = 1812$$

$$S - 1 = 1811$$

$$Q = \left(\frac{S_{\max}}{S-1}\right) - 1 = 0.39$$

$$F = \frac{m-1}{Q} = 28.2$$

Snedecors F = 28.2
5 per cent F = 2.41
1 per cent F = 3.41

$$k = \frac{3(n-1)}{n+1}$$

28.2 = Accurate Ranking

A 1 7/8-IPS Magnetic Recording System for Stereophonic Music*

P. C. GOLDMARK†, FELLOW, IRE, C. D. MEE†, MEMBER, IRE, J. D. GOODELL†, AND W. P. GUCKENBURG†

Summary—The primary aim of this work has been to develop a stereophonic system for recorded music, using a small, inexpensive and practical cartridge with magnetic tape as the information carrier. In order to make the cost of the recorded cartridge comparable with a disk containing an equivalent amount of music, it was realized that basic developments in magnetic recording such as efficiency of recording and reproducing techniques were necessary to meet the packing density requirements.

This paper describes developments which have led to a recorded cartridge one fifth of the volume of a disk and capable of playing more than one hour of stereophonic sound uninterrupted. In order to obtain the desired signal-to-noise ratio and frequency response to 15 kc, radical improvements have been made to tape, recording system and playback head.

In conjunction with this work, a fully automated tape machine has been developed. The machine is equipped with a changer-type mechanism and accommodates a number of cartridges which are played and rewound completely automatically, one after another, furnishing music for several hours. The machine requires no manual threading and has a rewind cycle of less than twenty seconds for an hour-long tape.

Introduction

As PART of a long range development program in the field of magnetic recording which CBS Laboratories undertook on behalf of Minnesota Mining and Manufacturing Company, prerecorded tape systems for the home have been under study over a period of several years.

In order that prerecorded tape can take an important place in the field of home entertainment, one must take into account a great many requirements, some of which are not easily met. For instance:

1) The tape must be contained in a compact cartridge in such a way that no part of the tape is exposed.

2) The amount of tape must be small and the cost of the cartridge low in order that the price of the final product can approach that of the record.

3) The sound should be stereophonic with provision for three tracks for maximum flexibility. More about this later.

4) A complete musical composition should be played without interruptions, that is, without reversing the cartridge or tape.

5) The quality of sound should be at least as good as the best of existing prerecorded media.

6) The durability of the tape and cartridge must be high enough so that after several hundred plays, the sound remains unchanged.

7) It should be possible to place a number of cartridges on a tape machine equipped with a changer-type mechanism so that one can provide music for several hours.

The outcome of these studies and subsequent developments which we believe will satisfy the above conditions and requirements will be discussed.

General

It was clear from the outset that one was dealing with a system rather than just a few components. Thus intensive development work over a period of several years progressed simultaneously in such areas as methods of signal recording, magnetic transducers and playback heads, design of cartridges and tape transport mechanisms. CBS Laboratories' system work, in close cooperation with Minnesota Mining and Manufacturing Company, also included the development of a new tape with characteristics that provided optimum matching into the over-all performance.

Late last fall, the new prerecorded system was in a sufficiently advanced stage to be demonstrated to most members of this industry.

Some of the important features and parameters of the new tape cartridge system are as follows:

1) Tape speed is $1\frac{7}{8}$ inches per second. The width of the tape is 150 mils, the thickness 1 mil, and there is provision for three tracks. Each track is 40 mils wide.

2) The cartridge is approximately $3\frac{1}{2}$ inches square and $\frac{5}{16}$ inch thick. The cartridge contains sufficient tape to play continuously for 64 minutes, and thus will carry more than 98 per cent of music compositions without interruptions. The space occupied by the cartridge in its container is approximately 4 cubic inches as compared with an LP record in its envelope with approximately 20 cubic inches.

3) The tape machine which will be demonstrated today can accommodate five cartridges and play them automatically one after the other. A cartridge can be rejected during any part of its play, similar to a record changer. The production versions of this machine now under development by Zenith will provide fast forward and reverse speeds. The same instrument will also serve as a home recorder using the new cartridges with blank tape.

* Received by the PGA, May 26, 1960. Presented at the IRE International Convention, New York, N. Y.; March 23, 1960. Abstract published in 1960 IRE INTERNATIONAL CONVENTION RECORD, pt. 7, p. 116.

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Earlier reference was made to a third track which is located in the center of the 150-mil wide tape.

Extended studies have been undertaken in CBS Laboratories to determine the optimum acoustic conditions desired by the listener in the average home while playing prerecorded music. Conventional stereophonic music, as now recorded, provides only a portion of the sounds that are perceived by the listener sitting in a concert hall. A large percentage of the total acoustic energy which reaches the listener's ears is reverberated and delayed sound which is considerably depleted of its original stereophonic character. Experiments in CBS Laboratories have shown that in a space simulating the average living room, a much more exciting and realistic sound can be produced giving an illusion of "being there." Thus, it is intended to record on the third track as an optional feature on the new prerecorded tape system, the stereophonic sum signal delayed and reverberated to an optimum degree.

The new medium will provide maximum flexibility and a new dimension in sound. The reproducing instruments can be manufactured for two or for three tracks.

Later some of the electrical and magnetic characteristics of the new system will be discussed. The data and curves shown are already based on the newly developed tape and represent the over-all behavior of the entire system.

Following the section dealing with the magnetic aspects of the new system, some of the mechanical problems and their solutions as encountered will be described.

Magnetic and Electrical Characteristics

Component Developments Required

In order to achieve adequate signal-to-noise ratio, frequency response and dynamic range at a tape speed of $1\frac{7}{8}$ ips, significant developments of most components used in magnetic recording are required. For instance, due to the shorter wavelengths encountered, developments have been aimed at reducing the wavelength dependent losses.

Wavelength Dependent Losses Reduced

Fig. 1 lists the important losses which have been minimized in the present system.

Losses in Reproduction: 1) There is an exponential reduction of the replay head flux with decreasing recorded wavelength due to the finite separation between the surface of the tape and the replay head pole pieces. At 15 kc and $1\frac{7}{8}$ ips, this loss is almost 0.5 db per microinch separation.

2) Another important loss is associated with the azimuth alignment between the replay head gap and the line of constant recorded magnetization across the track width. For a conventional 90-mil wide track, a loss of 6 db occurs at 15 kc and $1\frac{7}{8}$ ips for a misalignment angle of 3 minutes.

3) The proportion of replay head flux shunted by the gap will increase when using the narrow gaps necessary to resolve the shortest wavelengths recorded at a tape speed of $1\frac{7}{8}$ ips. In order to maintain a high efficiency it is necessary to compensate for a reduction in gap length by a corresponding reduction in gap depth.

WAVELENGTH DEPENDENT LOSSES

Reproduction

1. **Separation of head and tape surface.**
2. **Azimuth alignment of head and tape.**
3. **Replay head efficiency.**

Recording

1. **Tape thickness loss.**
2. **Recording field configuration loss.**
3. **Non uniformity of tape particles.**

Fig. 1.

Losses in Recording: 1) A separation loss of the type described for reproduction occurs during recording due to the finite coating thickness. Those particles remote from the tape surface will thereby give an attenuated contribution to the tape surface flux and so will contribute less to the replay head flux.

2) The magnetization of a recorded tape will not be uniform throughout the coating thickness since it depends on the rate of extinction and the direction of the recording field when the critical value for recording is reached after the tape has passed the recording gap. In addition to this, a further loss can occur due to change in phase of the recorded signal through the coating thickness caused by the vertical curvature of the effective recording plane of the recording head field.

3) For high resolution of the effective recording plane, a sharp cutoff of the recording field must be accompanied by a high uniformity in the magnetization characteristics of the individual particles of the tape. Elimination of particles with low critical fields for switching will also reduce self-demagnetization effects.

The separation loss has one advantage in slow speed tapes for audio since, due to the shorter wavelengths involved, print-through is correspondingly reduced allowing new thin tape backing materials to be used with safety.

New Developments in Magnetic Recording Components

Although the major loss component, called separation loss, is inherent in presently known magnetic recording systems, it has been possible by improvements of tape and heads to achieve performance characteristics approaching those presently obtained from $7\frac{1}{2}$ ips machines. Such performance is achieved with a track width of 40 mils. Having a narrow track reduces the alignment problem.

It has been found that a conventional laminated ring type playback head can be constructed to be responsive up to 15 kc with a 1.5-mv output from a tape having $\frac{1}{3}$ mil coating thickness. A subassembly of the 2-track version of such a head is shown in Fig. 2. The replay head coils fit over the projecting laminations. Since the recorded wavelength at 15 kc is only $\frac{1}{8}$ mil, it is necessary to form an effective magnetic gap of $\frac{1}{16}$ mil (or 1.5 microns). It was found that a 1-micron thick spacer gives satisfactory head resolution in prolonged use. By manufacturing the multitrack head in two halves, automatic colinearity of the gaps is assured and in practice the 10-kc sensitivity of the tracks differ by less than 4 db.

Similar mechanical refinement is necessary, of course, in the recording head. Fig. 3 shows a plot of the field distributions at 0.1-mil spacing for various gaps. It is seen that the field decrement increases somewhat with gaps which are large compared to the spacing. Thus, a long gap might be thought advantageous, especially since the vertical field decrement is also reduced. In practice, however, the expected improvement does not occur, probably due to the relatively greater vertical component of the effective recording field. Considerable development has been carried out to improve the recording field configuration for the very short wavelengths involved in this system. This will be reported on later.

Significant advances have been made in the recording media by Minnesota Mining and Manufacturing Company leading to considerable reduction of the separation loss effects. Firstly, a tape lacquer formulation has been developed which is relatively soft, giving good head-to-tape contact. Particle rub-off on guides and heads has virtually been eliminated and the consequent amplitude variations considerably reduced at the shortest wavelengths. In addition, CBS Laboratories developed a higher output and lower noise tape as a result of changes in the magnetic material itself. Previous work has concluded that a reduction of effective particle size results in lower tape noise. The improvement achieved is shown in Fig. 4, where the weighted noise response for the existing tape is compared with the new tape using optimum bias for each. A 4-db lower noise level is obtained in the midfrequency range. Higher over-all output is also obtained from the new material. It is found that the short wavelength efficiency is par-

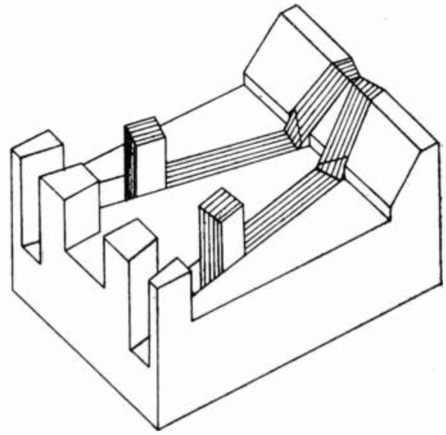

Fig. 2—Two-track replay head subassembly.

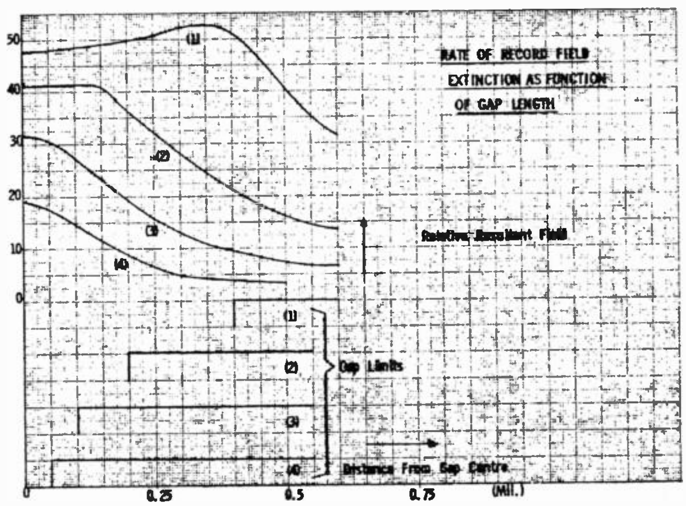

Fig. 3.

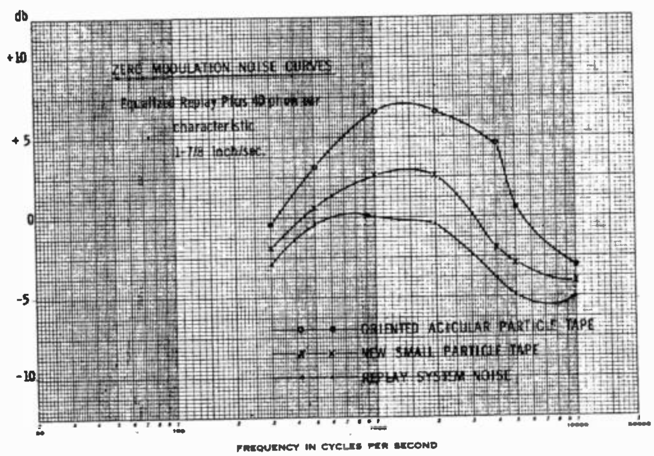

Fig. 4.

ticularly improved. One reason for this is that a deliberate attempt was made to reduce the spread of critical fields required for magnetization change in the individual particles. For acicular particles, better control of the size and shape is required, and for effectively spherical or cubic particles, it is necessary that the acicularity be kept low enough to make the crystal anisotropy dominant in all particles. Fig. 5 shows the improvement resulting from recording with the new tape using one of the high efficiency recording heads compared to that obtained with conventional $1\frac{7}{8}$ ips recording.

Equalization Techniques, Performance of System

The recording equalization adopted for the new $1\frac{7}{8}$ ips record-replay system is shown in Fig. 6. This curve was derived by performing many listening tests on a variety of program material. It is the optimum characteristic which meets the requirement to load the tape optimally at all frequencies without overload danger. Using this in conjunction with the replay equalization (Fig. 7), a flat response is obtained from 30 cps to 15,000 cps at -18 db relative to a level giving 3 per cent distortion at 1 kc. Under these conditions, the ratio of the maximum signal level at 1 kc to the zero modulation system noise is 54 db. The 10-kc signal response at this maximum signal level is -12 db relative to that at low frequencies. Typical equivalent signal-to-noise ratio for professional $7\frac{1}{2}$ ips half-track systems is 54 db with a corresponding 10 kc signal response at -6 db. Thus the new system with its own recording and replay characteristic approaches the 7.5 ips performance available today and has been found to be entirely adequate for all types of musical programs.

Mechanical Design Problems and Solutions

One of the central problems in prerecorded tape systems is the design of the tape packaging. Obviously, it is necessary to satisfy requirements of convenience as well as provide adequate protection for the tape. Naturally, high quality performance with respect to music reproduction is a prerequisite.

In order to popularize prerecorded tape it is essential to eliminate the process of manual threading between the reels. This requirement is dictated by the need for avoiding manual threading and also by the requirement to make the cartridge compatible with a practical automatic changer mechanism.

On first examination the notion of threading the tape permanently between two side-by-side reels contained in the cartridge is attractive. However, every practical design incorporating both the supply and take-up reels in the cartridge requires that sections of the tape be exposed through openings in the cartridge walls with consequent dangers of damage. Even in a single cartridge player there are many difficulties involved in

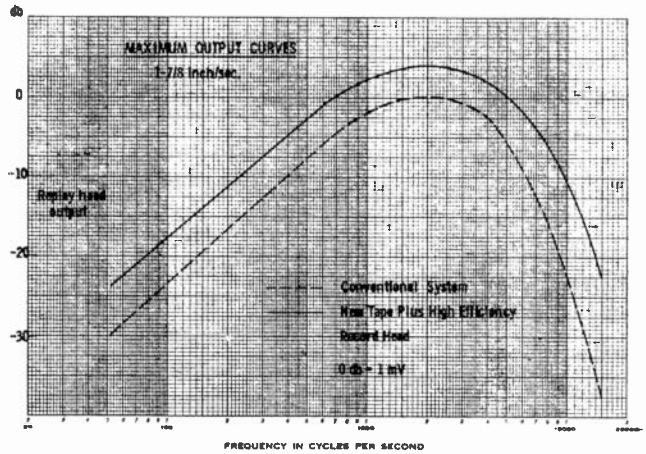

Fig. 5.

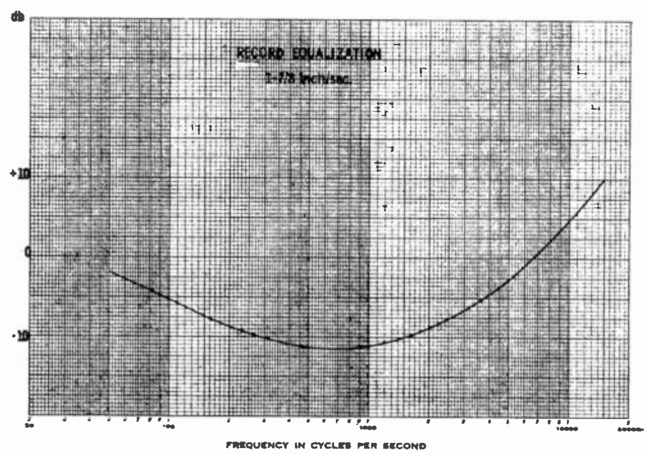

Fig. 6.

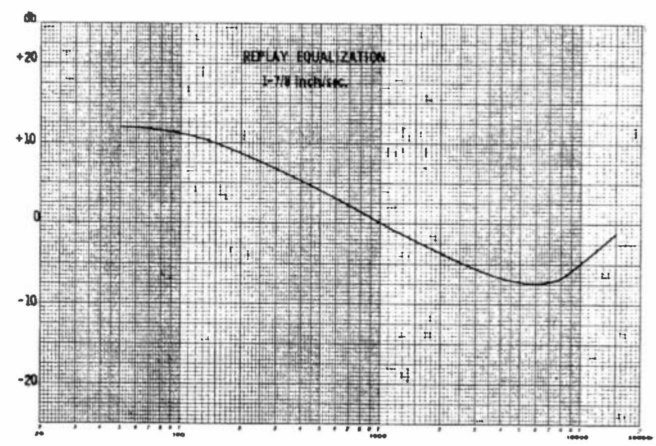

Fig. 7.

coupling the tape of a dual reel cartridge to the drive system and the heads, but when the design of an automated changer is considered, these problems increase rapidly in number and magnitude.

A basic consideration in any type of cartridge is the need for relatively high speed transport in so-called "search" operations. If flanges are used on the reels inside the cartridge, the bulk is considerably increased and many problems of stability are encountered. Thus, high speed winding without flanges requires some method of maintaining a separation between the tape and the cartridge walls.

The three-dimensional geometry of the reeled tape, the driving spindle in the transport mechanism, the walls of the cartridge and other components call for strictly orthogonal relationships or some automatic dynamic adjustment and an accurate system of tape guidance. Otherwise, the cumulative errors in repetitive reeling of the tape, even on the same machine, will lead to telescoping or angular displacement of the tape reel with respect to the cartridge walls. In brief laboratory experiments these problems may not be evident, but in long term field use the increasing friction produces instabilities in the tape speed and eventually may completely block the reel from rotating.

The problem of smooth reeling without any flanges was solved by introducing a novel guiding member in the cartridge with adequate compliance to insure a smooth rewind cycle. This arrangement allows a tape with an hour of playing time to be rewound in twenty seconds. (Five-second rewind has been achieved in the laboratory.)

Threading of the tape is accomplished by means of a leader permanently attached to the take-up reel in the mechanism. The end of the rewind cycle leaves the permanent leader in the threading path of the machine.

A very simple and economic solution was used for the design of the coupling between the reeled tape and the permanent leader. This consists of a "U"-shaped device attached to the end of the tape in the cartridge and so shaped that it seals off the only opening in the cartridge when the tape is fully rewound. The permanent leader terminates in a dumbbell-shaped element that readily mates with the "U"-shaped clip. The dumbbell attached to the permanent leader can slip through the "U"-shaped clip in a vertical direction with only a light detenting restraint but provides an absolute coupling in terms of horizontal pull when the two members are engaged (Fig. 8).

In order to eliminate variations in back tension with dynamic changes in effective reel diameter, a felt pad is spring-loaded against the surface of the tape as it leaves the cartridge and the supply reel is operated in free running bearings. This provides excellent tensioning characteristics and at the same time maintains the cartridge complexity cost at a minimum. Fig. 9 shows the tape deck and the felt pad.

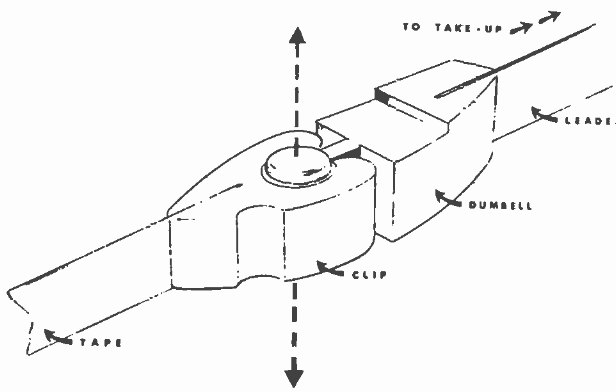

Fig. 8—Cartridge coupling members.

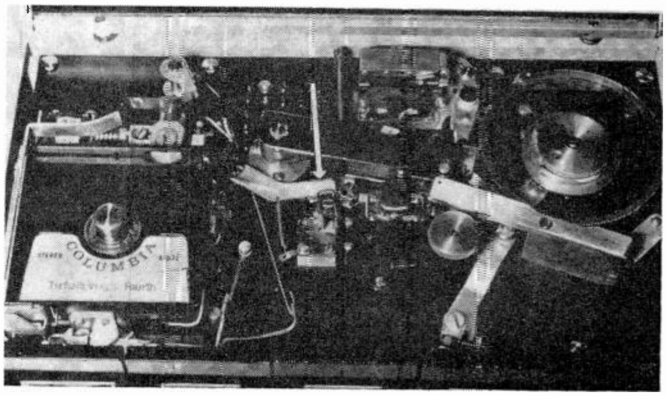

Fig. 9—Tape deck showing felt pad.

Some kind of braking mechanism is essential to avoid partial unreeling and fouling of the tape within the cartridge under normal conditions of handling. The brake must be positive, reliable and simple to assemble. The device selected consists of a linkage mounted in the cartridge hub and spring-loaded in a ratcheting relationship with teeth molded in the cartridge wall. When the cartridge is placed on the machine the spindle releases the brake automatically. The brake is shown in Fig. 10.

The facility for driving the cartridge hub during the rewind cycle must be designed so as to permit random rotary orientations of the spindle with respect to the cartridge hub in the loading process. This is accomplished by means of radial slots around the inner periphery of the hub and a spring-loaded two-toothed drive in the spindle (Fig. 11).

The cartridges are designed with mating surfaces that couple them (see Fig. 12) together in a stable vertical stack. This feature contributes considerably to the ease with which they may be handled and loaded in a changer mechanism. The patterns are unsymmetrical so that the cartridges must be correctly oriented or they cannot be fitted together. Other details of the mechanism make it impossible to load the cartridge in any way that results in improper operation.

The resulting cartridge design is compact, inexpensive and dependable. Actually, of course, the cartridge de-

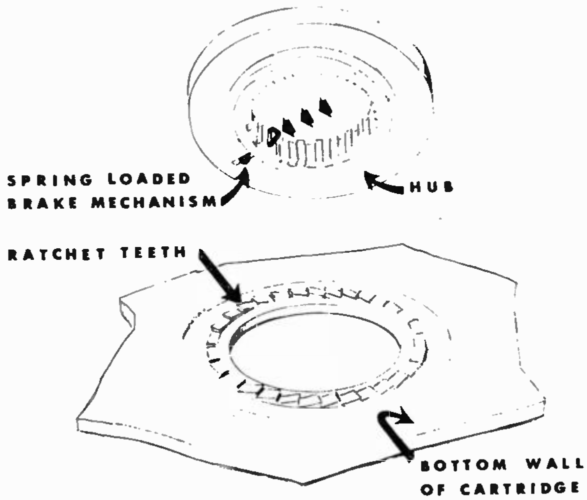

Fig. 10—Cartridge brake mechanism.

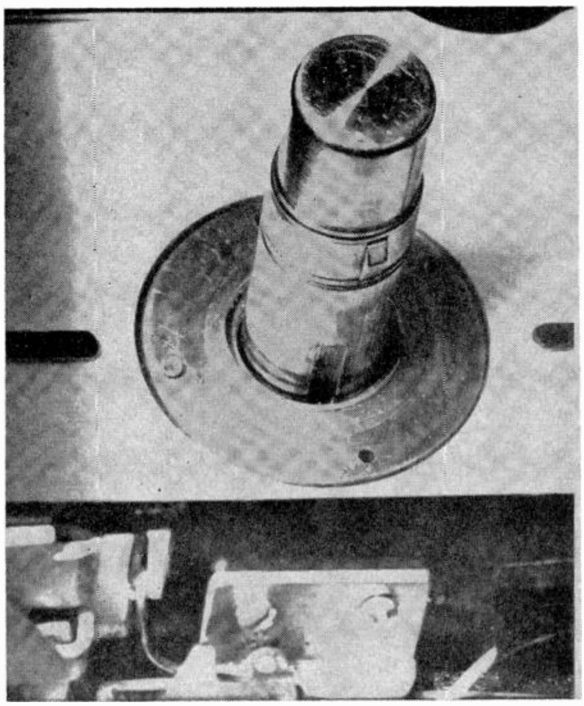

Fig. 11—Cartridge spindle.

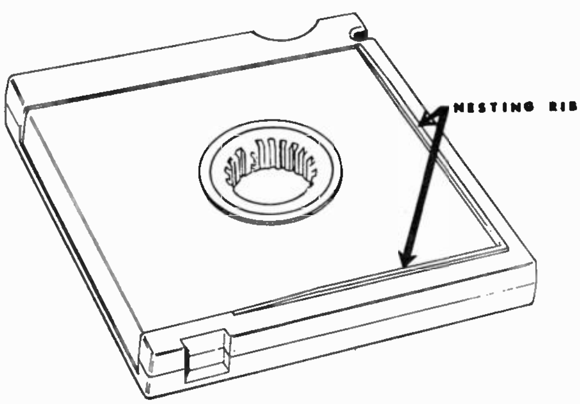

Fig. 12—Cartridge nesting ribs.

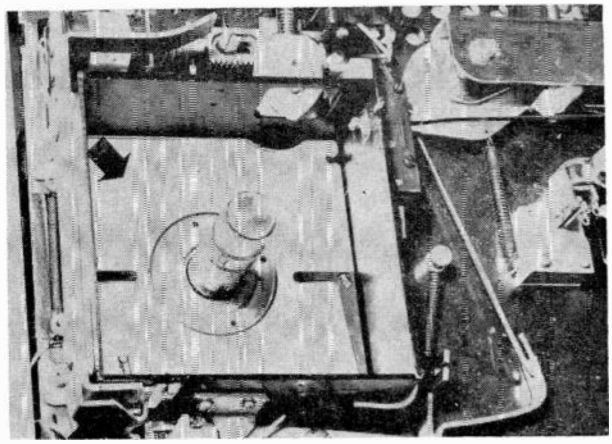

Fig. 13—Cartridge well.

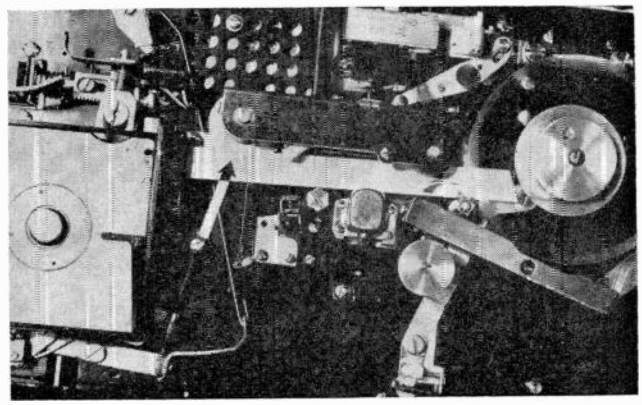

Fig. 14—Straight line path for tape.

sign was carried on in conjunction with the development of mechanisms capable of handling it in a fully automated changer so as to eliminate any mutually exclusive features. The actual changing mechanism consists simply of a spring-loaded platform in a well (Fig. 13), with which the supply spindle is coaxial, and an appropriate escapement. The latter is an essentially conventional device.

There are two escapement levers that operate in tandem on opposite sides of the cartridge well. One of the escapement levers is placed close to the corner from which the tape is fed to maintain accurate positioning between the clip terminal and the threading path.

The path for the tape is a straight line from the cartridge to the supply reel during the threading operation. When the tape has been pulled from the cartridge and starts to wind on the supply reel, the pressure pad that supplies the back tension and the pressure roller are automatically brought into position (Fig. 14).

The take-up reel is operated with a conventional slipping clutch drive.

The successive cycles of operation are programmed by a multiposition rotary switch and several mechanical interlocks. The slipping clutches, brakes, speed changing idlers and the like are operated from the three-dimensional surfaces of a single complex cam (Fig. 15). All pressure-roller, pressure-pad and escapement operations may be programmed via suitable cam designs. It is necessary to provide a number of mechanical and a few electrical interlocks to prevent improper manual interference with machine operations, but these are relatively simple and straightforward in design.

The straight line character of the tape path does not require intermediate idlers and consequently the guidance problems are minimized. However, as in all such drives, it is important to maintain the pressure roller axis parallel to the axis of the capstan. This is accomplished by introducing sufficient compliance in the mounting of the pressure roller so that it is self-adjusting within small limits. The spring loading provides a simple adjustment for correcting major pressure differentials across the idler surface (Fig. 16).

Obviously, there must be some means for sensing the end of the tape and various other portions of the operating cycle. In this machine these results are obtained by means of a simple analog computing linkage that cannot be disclosed in detail at this time. However, the method is independent of the length of the tape in a given cartridge and has displayed a very high degree of reliability.

The authors wish to express their appreciation for the advice and assistance given during the course of this work by B. B. Bauer, A. A. Goldberg, J. C. Jeschke, H. R. Sherman, E. L. Torick, and J. C. Wistrand, of CBS Laboratories, and Barbara Ivins, formerly with CBS Laboratories. We also wish to acknowledge the wholehearted cooperation of Dr. W. W. Wetzel and his associates of the Magnetic Products Division, Minnesota Mining and Manufacturing Company.

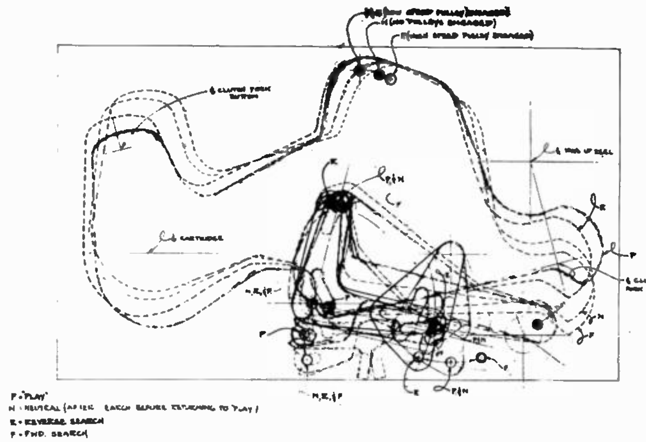

Fig. 15—Programming cam.

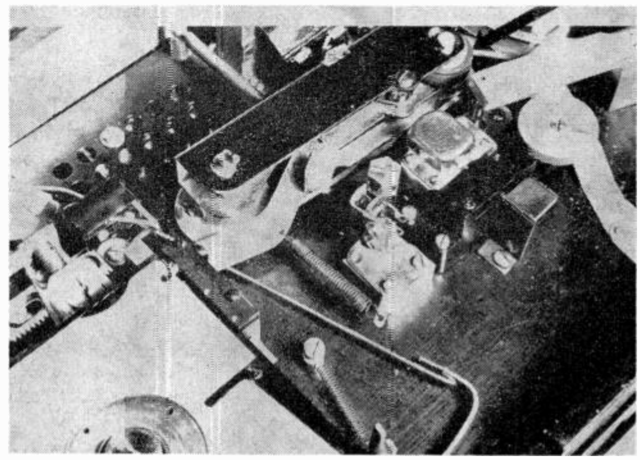

Fig. 16—Pressure roller.

Signal Mutuality in Stereo Systems*

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Summary—Four stereo systems are compared:

1) 3 microphones, 3 independent transmission means or "tracks," and 3 speaker output or "channels," designated 3-3-3.
2) 3 microphones, 2 sound tracks and 3 outputs using a bridging or "derived" center channel, designated 3-2-3.
3) 2 microphones, 2 tracks, 3 outputs with derived center, designated 2-2-3.
4) A stereo microphone pair in a single housing with stereo separation derived by directional response of the 2 microphones, using 2 sound tracks and 3 play back speakers with derived center, designated SD-2-3 (SD for "stereo-directional" applied to the microphone).

Each of the 4 systems is shown to contain mixtures of all signals in each channel.

Crosstalk may be defined as the inadvertent transfer of a signal from one channel to another. Signal mutuality is the natural consequence of one microphone in a stereo array detecting signals pertinent to other microphones.

The magnitude of differences between the 4 types of stereo studied are found to be small—of the order of less than 4 decibels.

Delay effects are similar in the first 3 types, but where a single microphone location is used and dependence is on directional pattern for stereo separation, the delay effects are different. A separate study of the combined effects of sound delay and quality was made to corroborate the suspected delay effects of the so-called "stereo microphone."

Introduction

THE CONCEPT of 3-channel stereo derived from 2 sound tracks is predicated on the principle that if 2 microphones are properly placed relative to each other and to the sound source, their combined output would be that of a microphone between them, and that this microphone that wasn't there can be recovered by recombination.[1,2]

Geometry tests[3] show that 2-track 3-channel stereo is capable of yielding results closely approximating observations of live sound in the original geometric array. Introduction of a third channel is recognized as making possible a wider array and also increasing the listening area.

Recent Experience

Whatever the span or stage width, raising the level of the center channel narrows the apparent stage width. Lowering the center-channel level has the opposite effect. These effects are as would be predicted. In the limits, the center channel which is too loud predominates and focuses all events in the center, or one which is absent vacates the center leaving a void between the flanks. In the properly adjusted condition, the derived center channel will retain the dimensional integrity of a soloist at one side of the podium in front of a symphony orchestra, or an intimate group like a string quartet.

While a center-channel adjustment is indicated, it is by no means critical. Center-channel levels of -3 db to -10 db have been experienced because of room or hall acoustics. It appears that meeting these values plus or minus 2 decibels suffices to satisfy the listener that the sound curtain is solid, continuous and natural.

Again, geometry tests using the 2-track derived third channel indicate a reasonable latitude of at least plus or minus one decibel for the level of the center channel to retain a reasonable similitude of the original sound geometry.

Steinberg and Snow[4] noted the focusing effect of the center channel, and also noted that the 3-channel system exhibited the important advantage that the virtual shift of apparent sound for side observing locations was smaller. This has been corroborated in the recent experiments leading to the thought that the term wide-stage stereo should be worded to include the concept of wide audience stereo.

Another observed fact is that the geometric integrity of both small and large sound sources is retained; a soloist or string quartet remains compact and a large orchestra is extended over the same wide-stage 3-channel speaker configuration. Thus, in one example, the listener subtended an angle of 90° between flanking speakers, yet a string quartet was depicted as grouped in the stage center.

This writer's early derived third-channel systems were compared to "phantom" systems, but the derived

* Received by the PGA, March 31, 1960; revised manuscript received June 10, 1960.

Presented in part at the following meetings: AES Convention, New York, N. Y., October 8, 1959 (as part of a joint paper with Robert C. Avedon of Electro-Voice); IRE Section Holloman Air Force Base, New Mexico, February 8, 1960; IRE Section Tulsa, Okla., February 18, 1960; IRE Section Shreveport, La., April 5, 1960; Joint meeting of the Chicago IRE and the Chicago Acoustical and Audio Group, Chicago, Ill., May 13, 1960. (Terminology used herein was evolved prior to Magnetic Recording Industries Association's definitions and so is not consistent therewith. See "Terminology for stereo with two signals and a derived center output," this issue, p. 183.)

† Klipsch and Associates, Inc., Hope, Ark.

[1] P. W. Klipsch, "Stereophonic sound with two tracks, three channels by means of a phantom circuit (2PH3)," *J. Audio Engrg. Soc.*, vol. 6, pp. 118–123; April, 1958.

[2] P. W. Klipsch, "Terminology for stereo with two signals and a derived center output," IRE Trans. on Audio, vol. AU-8, p. 183; September–October, 1960.

[3] P. W. Klipsch, "Wide-stage stereo," IRE Trans. on Audio, vol. AU-7, pp. 93–96; July-August, 1959.

[4] J. C. Steinberg and W. B. Snow, "Physical factors," Symp. on Auditory Perspective, *Elect. Engrg.*, vol. 53, pp. 9–32, 214–219; January, 1934.

system is analogous rather than similar to the "phantom" circuit. Thus, where I designed the derived system as "2PH3," short for "2-track phantom-derived 3-channel," it seems appropriate to drop the "phantom" and designate the system simply as "2-3 stereo," short for "2-track-derived-3-channel stereo." To include the microphones, 2 or 3, the system could be designated "2-2-3 stereo" or "3-2-3 stereo." In the case of the "stereo microphone," a pair of cardioid-directional microphones achieving separation by directionality, the designation could be "SD-2-3 stereo" short for "stereo-direction microphone, 2-track, derived-3-channel stereo." Where only 2 digits are used, the implication is to the sound tracks and playback channels. Consistent with the above, "3-3-3 stereo" would designate 3 microphones, 3 electrically-independent sound tracks, and 3 output channels.

Actually 3-3-3 stereo, 3-2-3 and 2-2-3 stereo differ only slightly. Each microphone overhears program material pertaining to other microphones so that 3 "independent" channels are not really independent.

Caused perhaps by witnessing improperly conducted experiments or demonstrations, skepticism has arisen in some quarters as to the validity of the derived center channel, the view being that it is some sort of bootstrap operation or a violation of some physical principle. It is not the purpose here to show that the derived third channel is the identical equal of the system using 3 independent transmission channels, but to show a remarkable similarity. That it works has already been demonstrated and accepted. This paper is largely by way of explaining why it works.

Analogy

In wire communications, it is possible to have 3 communication channels with 4 conductors.[5] This may be done by the phantom circuit of Fig. 1, or the common return circuit of Fig. 2. Both represent 3 independent circuits, in that each excludes signals from the others. Each requires 4 conductors.

In a 2-track tape machine or a 2-groove wall disk recording, only 2 independent channels are available, and a third channel cannot be independent but must contain signals pertaining to at least one other circuit. Perhaps a "phantom" head could be devised for a tape machine, but for the present discussion it will be assumed that a 2-head stereo tape machine is capable of transmitting only 2 tracks. Tape and disk speeds are considered such as to exclude an additional track by carrier frequency. In other words, practical stereo is currently limited to 2 sound tracks.

The most practical evaluation of 2-track stereo with the derived third channel takes the form of a geometric study in which sounds are generated and detected in a contrived stereo array, and observers attempt to locate sounds by listening to the original sound and to the sound reproduced over a stereo speaker array. This was done by Steinberg and Snow,[4] and by the present author.[3] Comparison of the live and reproduced listening tests showed the same order of magnitude of errors for each, indicating that the 2-2-3 system is highly effective and approaches live listening in accuracy.

[5] P. W. Klipsch and R. C. Avedon, "Signal mutuality and crosstalk in two and three track three channel stereo systems," presented at Audio Engrg. Soc. Convention, New York, N. Y.; October 8, 1959.

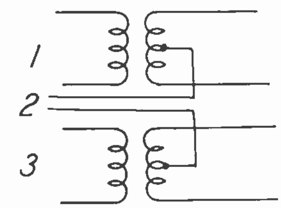

Fig. 1—Phantom circuit; requires 4 conductors.

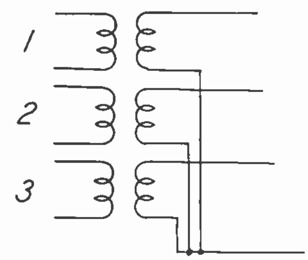

Fig. 2—Grounded return circuit for transmission of 3 signals; requires 4 conductors.

Combinations

Consider the basic problem of transmitting 3 sound sources over 2 sound tracks and recovering the output for reproduction over 3 speakers. If 3 sources are A, B, C, one track could contain $A+B$, the other $A-B$ and their sum for the center speaker $B+(A+C)/2$. Many other arrangements may be written down, all of which lack the degree of independence afforded by 3 separate sound tracks.

But then, how independent are 3 "independent" sound tracks, when, for example, a center microphone "hears" all three sources, A, B and C?

Mutual Signals in Each Sound Track

Crosstalk may be defined as the inadvertent transfer of a signal from one channel to another. Signal mutuality is here defined as the natural consequence of one microphone in a stereo array picking up signals pertinent to other microphones. Thus, in Fig. 3, where A, B and C are sound sources and a, b and c are microphones, each microphone hears each sound source. This is not crosstalk, but signal mutuality, and is a perfectly natural and proper phenomenon. But it has the same qualitative effect as does crosstalk in that it dictates that the channels are not independent.

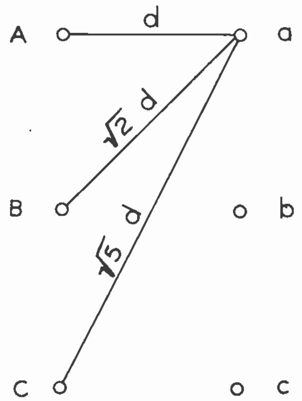

Fig. 3—Sound sources A, B and C are spaced from microphones a, b and c as shown. Each microphone "hears" each sound source, the intensity being inversely proportional to the distance squared, or the amplitude inversely to distance.

Analysis

Now suppose the sound source A produces a voltage output from microphone a of unit volts. The sound intensity is inversely proportional to the square of the distance (true only for anechoic conditions, but assumed here for analysis), or expressed in pressure, amplitude or volts, inversely with the distance. Thus, source B produced $1/\sqrt{2}$ or 0.71 volt at microphone a, and source C produced 0.45 volt at a.

After suitable amplification, equal for each sound track and reduced to unity reference, the 3 outputs are:

$$A + 0.7B + 0.45C$$
$$0.71A + B + 0.71C$$
$$0.45A + 0.7B + C.$$

The third channel is simply the mirror image of the first.

Thus, each of the 3 "independent channels" contains a mixture of all 3 signals. Although the "3 independent channel" system implies complete independence, the "3-3-3" actually contains considerable signal mutuality.

The above is illustrated in Fig. 4(a). The third-track values, being the mirror image of the first, are not reiterated.

The 3-2-3 system can be illustrated as in Fig. 4(b). The microphone outputs are the same as before. They are added or combined along the paths indicated, then "volume controlled" to a unity reference, then added or combined again, and the center channel only "volume controlled" again by the bracketed factor of 0.5. The flanking channel output with A referred to unity has 3 db more B signal than the 3 independent sound track system, and 3.6 db more C signal. The center channel, to reduce B to a unity basis, requires a "volume control" or multiplying factor of 0.5 after which it contains 1.5 decibels more A and C components than in the 3-3-3 system.

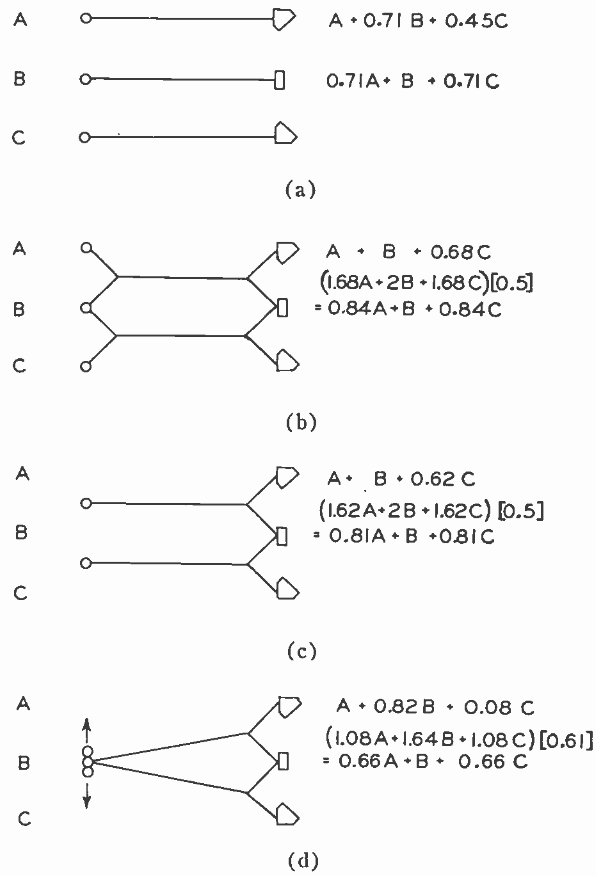

Fig. 4—(a) 3-channel stereo with 3 microphones, 3 independent transmission tracks, and 3 speaker output channels. Designated 3-3-3 stereo. Each speaker output contains a component from each microphone. The quantities are voltages or amplitudes, not intensities, so 0.71 represents minus 3 db. (b) 2-2-3 stereo with 3 microphones, 2 sound tracks, 3 output channels. Each output again contains signals from all microphones. Note the similarity between the individual channels of this system with the corresponding channels of the 3-3-3 stereo system. (c) 2-2-3 system with 2 microphones. This appears to be a closer approximation to the 3-3-3 system than does the 3-2-3 arrangement. (d) SD-2-3 stereo system using a stereo-directional microphone, 2 tracks and 3 channels. Of all the 2-track systems, this appears to approach most closely the 3-3-3 system.

It should be remembered that this analysis, assuming anechoic conditions and ignoring delay effects will not ordinarily be met with in practice. Room reverberation will reduce the separation or increase the mixing, "crosstalk" and signal mutuality still further.

The 2-2-3 system may be illustrated as in Fig. 4(c). Using the same techniques, one gets about the same channel outputs as in the 3-2-3 system; actually, the approximation to the 3-3-3 system is better by a fraction of a decibel.

In the quantitities expressed in (b) and (c) of Fig. 4, the figure in brackets or the multiplying factor of 0.5 (voltage basis) or −6 db applied to the center output channel is a figure typically met with in practice.

Delay effects seem to be applicable to the same degree in the 3 systems.

The Stereo Microphone

Students of the delay effect have noticed that the sound which arrives first from a pair of sources is influential in determining direction. Steinberg and Snow[4] pointed out that "quality," or the ratio of direct to reverberant sound (a function of distance), influences the sense of direction. Deatherage and Hirsh[6] showed that an intensity change can be used to compensate for delay, within limits.

W. B. Snow[7] showed that the delay or time difference effects are important in sound localization.

There seems to be consistent evidence to indicate that sound intensity differences may be used to compensate for arrival time differences, but it also appears that this can be true only up to some limit. Intuitively, this should be true for extreme cases where, for example, a click is separated into two distinct sounds. De Boer[8] and Haas[9] have shown relations between sound intensity and time delay. Though de Boer's work dates back to 1940, and Haas to 1951, the delay-intensity effect seems to have been dubbed the "Haas effect"; perhaps it should be renamed the de Boer effect.

In the Stereo microphone, obviously the delay values are different than in the previous stereo arrangements. Fig. 5 shows the theoretical and actual polar pattern (measured at 440 cycles). Using the cardioid for computation, the 2 sound-track voltage values are computed from the polar sensitivity, taking account of the distance. The SD-2-3 system is shown in Fig. 4(d). The upper sound track yields

$$0.61A + 0.5B + 0.05C,$$

which "volume controlled" to unit value of A is

$$A + 0.82B + 0.08C.$$

Of course, the lower track is the mirror image of this.

Again adding, the 3 outputs are as shown in Fig. 4(d), with a multiplying factor of 0.61 to bring the center channel to unit value of the B signal.

So far, the stereo microphone analysis has ignored the time-delay effects. And here the computed values for the center channel fail to be corroborated in recording-playback experiments, some 9 to 12 db instead of 6 db being required to bring this channel into balance.

This discrepancy is attributed to delay effects. Logic indicates that proximity of the B source to *both* microphones produces the computed monophonic component, and the delay from flanking sources reduces their importance more than the mere loss in intensity.

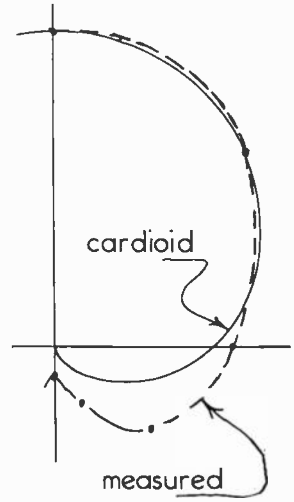

Fig. 5—Cardioid and measured polar response at 440 cycles of Telefunken SM-I microphone.

Experimental data are still limited for the SD-2-3 system. Two experimental recordings[10] indicate an excess monophonic component requiring a reduction of the center channel output. One[11] recording produced less monophonic component but still some 3 db more than "normal," that is, a center channel adjusted for "conventional" recordings needed about 3 db added loss to play the stereo microphone recording in proper focus. The other was a test recording which was found to require 6 db extra loss in the derived output channel.

Delay

To derive an independent and corroborative measure of the combined effect of delay and quality, we tried an experiment. Note that distance affects sound intensity, delay and quality. Intensity effects alone were considered in the foregoing analysis. The difference between the stereo microphone and the other arrays should be capable of explanation in terms of combined delay and quality.

Two speakers were set up so as to subtend an angle at the observer of approximately 45°. One speaker was left fixed at 7 feet distance. The other was moved from its reference distance of 7 feet in 2-foot increments from 5 feet to 21 feet. The power to the fixed speaker was left constant. The power to the movable speaker was varied in an attempt to make the virtual sound source appear to be between the speakers. A voice recitation was recorded on a tape loop and re-recorded to form a series of identical repetitions.

In Fig. 6, curve 1 shows the power-level change necessary to produce a sensation of the sound coming

[6] B. H. Deatherage and I. J. Hirsh, "Auditory localization of clicks," *J. Acoust. Soc. Amer.*, vol. 31, pp. 486–492; April, 1959.

[7] W. B. Snow, "Effect of arrival time on stereophonic localization," *J. Acoust. Soc. Am.*, vol. 26, pp. 1071–1074; November, 1954.

[8] K. de Boer, "Stereophonic sound production," *Philips Tech. Rev.*, vol. 5, pp. 107–114; April, 1940.

[9] Von Hermut Haas, "Über den Einfluss eines einfachechos auf die Horsamkeit von Sprache," *Acoustica*, vol. 1, pp. 49–58; 1951.

[10] The microphone was a Telefunken model SM-2 condenser microphone loaned by Audio Fidelity Professional Products, Inc., New York, N. Y.

[11] Using the Telefunken SM-1 with the microphone pointed "end-on" toward the stage center and 180° configuration.

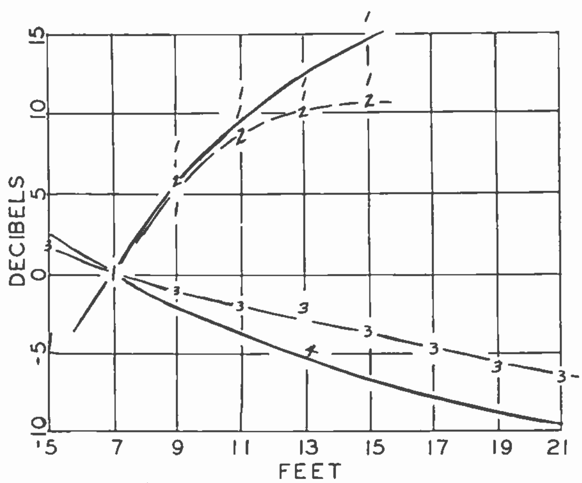

Fig. 6—Two speakers, one fixed, the other movable, were placed to subtend approximately 45° at an observer location. As the distance of one speaker was varied the observer adjusted the volume to cause the sound to come virtually from midway between them. Fixed speaker 7 feet from observer; movable speaker at distances shown on abscissa scale. Ordinates, db differences.

Curve 1 shows the power level fed the movable speaker to overcome its distance, delay and quality effects.

Curve 2 at each location, the fixed speaker was cut off and the power required to produce constant intensity at the observer was measured with a sound level meter.

Curve 3 is the sound level produced by the movable speaker when fed the power necessary for "focus."

Curve 4 computed inverse square law intensity at observer.

from between the speakers. Beyond about 8 feet difference, the sound failed to coalesce and was heard as coming from two sound sources.[12] The experiment was conducted in a room 16×25 feet with an 11-foot ceiling. The room was of about "average" reverberation characteristics. It has been judged to be a good listening room for sound reproduction, but not perfect for recording.

At each location of the movable speaker, with the power necessary to coalesce the sound between the speakers, the fixed speaker was turned off and the sound intensity measured at the observer's location. This was plotted as curve 2.

At each location, a constant power was fed to the movable speaker and the sound level measured at the observer. This is plotted as curve 3. A certain word in the recitation was found to be usable as a reference point at which to read the sound-level meter.

Finally, the inverse-square-law intensity was computed and plotted as curve 4.

The difference between curves 3 and 4 is due to non-anechoic conditions. At a distance difference of 6 feet, the volume loss resulting from distance was only 3 db actual, $5\frac{1}{2}$ db calculated. The difference in signal necessary to center the virtual sound source between the speakers was 12 db. The volume increase at the observer's location was 10 db.

There appears to be a 12-db power increase necessary to overcome a 6-foot delay; the sound intensity increase was 10 db. These values are suggested as being of the magnitude observed of extra attenuation required for the center channel with the SD-2-3 compared to the other microphone applications, for one recording experiment.

It was suggested that such a microphone would permit dispensing with the middle channel, but this was wishful thinking: the fallacy should be obvious without experiment. With only 2 channels, the stereo micophone recordings were just as void in the middle as were conventionally made 2-track recordings.

The difference in center-channel level is neither good nor bad. Excellent playback results were obtained. The fact that one more control becomes necessary is also neither good nor bad. If it eventually is judged bad, the monophonic component could be reduced by means introduced between the microphone and the recording.

The effort has been to explain a lack of time delay effect in terms of a center-channel level change necessary to compensate.

It should be pointed out that Fig. 6 contributes to the delay-intensity relation; comparison with Snow's Fig. 3 summarizing de Boer's and Haas' data, shows at least an agreement in order of magnitude if differing in slope and curvature.[7] One point of difference is that the intensity compensation fails in this writer's experiments for time delays exceeding about 8 msec, *i.e.*, the sounds fail to coalesce beyond about 8 msec. One point of similarity might be that our curve 1 plots alongside the de Boer curve but with about twice as much delay for a given intensity change.

Conclusion

Failure to sense the sound as coalescing at distance differences more than 6 or 8 feet may be due to the size and shape of the room, or due to the small base distance of only 7 feet. A larger room would have permitted a larger base distance. It is surmised that as the base distance increases, the distance difference could also increase, and that as the base distance is increased and the ratio of distance difference to base distance decreased, the amount of sound intensity change necessary to compensate for a given delay will decrease. Thus, it is surmised that the size of the room and the small base distance gave an exaggerated observed volume necessary to compensate a given delay. But one of the experimental recordings with the prominent center-channel response was performed in this same room so the effects remain comparable.

All the stereo systems shown have been proven to be workable and to add considerably to listening satisfaction compared to 2-channel stereo. If fidelity of tone is defined as accuracy of tone, as measured by frequency response, distortion, spatial coverage, and listener judgment of accuracy of tonal reproduction, then fidelity of geometry could be defined as the accuracy with which observers place virtual sound sources compared to the actual relative locations of the sources. All

[12] Limitations expressed and implied earlier and in various references are corroborated here.

4 of the stereo systems shown have been shown to yield a geometry superior to 2-channel systems. The analysis of this paper shows relatively little differences between them. The 3-3-3 or so-called 3-independent-channel system is still considered the reference standard even though it is shown to be not completely independent. Geoemetry tests with a 2-2-3 system compared favorably with observations of direct sound. To the listener devoted to enjoying music, the derived systems have afforded superior pleasure.

Final Summary

1) All the stereo systems shown are subject to signal mutuality in that each channel contains signals pertaining to the other channels. Thus, the so-called 3-independent-channel system is not really independent. The various 2-track derived-third-channel systems differ by only a slight amount from the so-called 3-independent-channel system, both in analysis and in experiment.
2) The stereo microphone as essentially a 2-track system affords the effect of a center microphone. Although delay effects influence the balance and focus, compensation amounts simply to lowering the center-channel playback level in some cases. While this stereo technique can stand a lot more study, it seems at present to offer attractive features. As with other microphone techniques, recordings so made are highly adaptable to the derived center channel, which is always necessary where wide stage and wide audience areas are demanded.
3) The fallacy of trying to achieve a center channel without the center speaker should be obvious with or without the present analysis.

   A glance at Fig. 4 should make it evident that a center channel of zero output could not approach a 3-3-3 response geometry regardless of microphone type or placement.
4) All the stereo recording systems are amenable to using a derived center channel and the expense is small for a large gain in stereo geometry.
5) As between the various 3-channel systems, there is not enough comparative data to indicate a preference. From the earlier work, it would be possible to theorize that systems which include the delay effect would offer better separation. But the reduced delay effect in the stereo microphone system may be compensated by a volume reduction in the center output channel. At least, within as yet undetermined limits, sound intensity may be used to compensate for the delay effects for stereo localization purposes. It may later be found that effects other than localization derive from compensation of delay by intensity.
6) This writer's best recordings were done with 2 microphones prior to his own experiments with the 2-3 stereo derivation, but in all cases the derived third channel [Fig. 4(c)] was found effective; even where a 2-loft pipe organ was recorded, the center channel portrayed its honesty by remaining unnoticed, and a soloist with symphony orchestra standing 6 feet to the left of the podium was heard in the same spatial relation on playback. Opportunities with the stereo microphone have been too limited for us to assign definite values. Naturally there is a tendency to want this system to be successful, as it represents a simplification of recording technique and less pounds and packages to transport and put in place.
7) One advantage is apparent in the stereo microphone in the absence of the double-Doppler effect, wherein a rapidly moving object producing a pitched sound (as the whistle of a rapidly moving locomotive) appears as 2 separate sounds in conventional spaced-microphone recording and as a natural movement of sound when the stereo microphone is used.[13]

Appendix I

Oscillographic examination was made of a tape recorded with the stereo microphone. This examination consisted of putting one sound track on the vertical plates and the other on the horizontal plates. In order to adjust gain and check phase shifts, we first used a test tape, a monophonic signal produces a line at 45° slope. Most stereo signals produce a random pattern, but a monophonic component produces an elliptic pattern with a definite slope of its major axis, tending to 45° for sum and 135° for difference monophonic component, with variations from these angles depending on a prominence of the signal of one track over the other.[14]

The particular tape examined indicated a strong monophonic component with major-to-minor axis ratios of 2:1 for ensembles and 3:1 for solos.

The observed excess signal put out by the center speaker thus appears to be due to a higher monophonic amplitide than expressed by the quantities indicated in Fig. 4(d). In addition, the lack of delay effects in the stereo microphone exaggerates the monophonic effect.

It is tentatively the opinion that, when one is playing recordings made with the stereo microphone, the only way to control the center output is by means of a volume control for the center speaker.

Appendix II

In Fig. 4, the individual outputs were arrived at by simple addition. There is a possibility that addition on a power basis would yield results more consistent with the observed effects.

[13] P. W. Klipsch, "The double Doppler effect in stereophonic recording and playback of a rapidly moving object," *IRE Trans. on Audio*, vol. PGAU-8, p. 105; May-June, 1960.

[14] B. B. Bauer, and G. W. Sioles, "Stereophonic display patterns," *J. Audio Engrg. Soc.*, vol. 8, pp. 126–129; April, 1960.

Stereophonic Localization: An Analysis of Listener Reactions to Current Techniques*

JOHN M. EARGLE†

Summary—Playback localization plots, similar to the technique employed by Steinberg and Snow in the Bell Laboratory experiments of 1933, afford perhaps the only means of evaluating quantitatively the performance of representative stereophonic systems. In the present tests, which deal with two-channel systems comprising a center bridged loudspeaker in addition to the two flanking loudspeakers, it is seen that the performance of wide-angle loudspeaker arrays can be optimized to give accurate localization over a large listening area. As an aid to the evaluation of test data, brief discussions of loudness and time-delay effects—the essential factors providing localization—are included.

Introduction

THIS paper deals with stereophonic localization tests similar to those originally performed by Steinberg and Snow in 1933 as part of the Bell Telephone Laboratory symposium on auditory perspective [9]. In these tests, sounds precisely located on a sound stage are recorded and then played back over a loudspeaker array. Listeners are then asked to plot on a set of charts provided to them the apparent locations of the reproduced sounds. In this manner it is possible to evaluate different microphone and loudspeaker configurations as to their ability to recreate the original geometry of the sound stage.

Although their approach is very much like that of the Steinberg-Snow tests, the present tests represent a considerable narrowing of scope. Whereas Steinberg and Snow tested a wide variety of techniques including the use of bridged center microphones and speakers as well as independent three channel systems, the present tests deal only with two channel transmissions with a bridged center loudspeaker in the playback configuration. One further change should be mentioned. The loudspeaker configurations used in the present tests were such as to subtend an angle in excess of 80° with the listener as compared with an angle of about 35° in the Steinberg-Snow tests. This increase of the angular width of the virtual stage created by the loudspeakers thus makes possible an evaluation of the trend toward wider loudspeaker placement for stereophonic reproduction.

It is fully realized that the ability of a system to reproduce accurate geometry is not the sole criterion for judging its worth. In something as subjective as the enjoyment of music, which is the ultimate purpose of these systems, the proof is in the listening and not in the testing. However, if quantitative comparisons of systems are to be made, then stereo geometry tests as outlined above appear to be the only means at hand.

Physical Factors Involved in Stereophonic Localization

There are two kinds of geometrical separation between events perceived on the virtual sound stage produced by the loudspeaker array: depth (longitudinal) separation and angular (lateral) separation. Depth separation, unlike lateral separation, is not unique to multichannel systems, for it can be reproduced by a monophonic (single-channel) system. Consider what happens when a sound source recedes from an observer. Three effects can be noticed: the over-all intensity becomes less, the timbre of the sound becomes less distinct (due to the more rapid attenuation of high frequencies relative to middle and lower ones), and finally, the ratio of direct to reverberant sound decreases. Any one of these effects alone is sufficient to convey the image of a receding sound source.[1] The role played by the ratio of direct to reverberant sound might not be immediately apparent. When an observer is close to a sound source, most of what is heard comes directly from the sound source itself; relatively little comes from room reflections and echoes. When the sound source is more distant, the amount of direct sound is reduced considerably while the amount of room reflections and echoes remains fairly constant. This change in ratio is always characteristic of a receding source (except under anechoic or free-field conditions, and the effect, even when artificially produced by, say, an echo chamber, can create the illusion of a receding sound source even when the over-all intensity of sound remains unchanged.

Lateral, or angular, separation is a phenomenon which demands two or more channels. It depends not only upon the relative intensities of the loudspeakers but also upon the variations in acoustic time delay caused by differences in path length between the observer and the several loudspeakers as well as the differences in path length between microphones and sound sources.

Consider a listener located on the meridian plane between two identical loudspeakers. If the listener is then provided with a dual potentiometer by means of which he can raise the level of one loudspeaker and simultaneously lower the other, the listener can cause the

* Received by the PGA, May 31, 1960.
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[1] See [9], p. 14.

apparent source of sound to move back and forth along a continuum between the two loudspeakers, with localization tending toward the louder of the two.

Assume another set of conditions in which the amplitudes of the loudspeakers are fixed and the time delay between them made variable. By controlling the time delay the listener can again cause the apparent source of sound to shift along a continuum between speakers, with localization tending toward the leading loudspeaker.

Extending the notion, the listener can be provided with means for varying simultaneously both time delay and relative intensity. It then becomes possible to make the factors aid or oppose each other. That is, the apparent source can be shifted by raising the level of one loudspeaker relative to the other, and then restored it to its original position in the center by introducing a slight time delay in the louder of the two speakers.

Many studies have been made of the phenomenon just mentioned—the neutralization of shifts due to loudness differences by a time-delay shift of the opposite sense. Deatherage and Hirsch, using binaural headphones, have shown that the displacing effect of a 2-msec delay can be compensated for by raising the intensity of the lagging earphone by about 30 db [1]. Using loudspeakers, Snow has shown that the displacing effect of a 3-msec delay can be compensated for by an intensity imbalance of about 8 db. Although the data for tests of this sort will vary according to actual test conditions, it is generally observed by Snow that arrival time differences up to about 3 msec require progressively greater loudness imbalance in order to have the apparent source remain in the center. For time delays greater than 3 msec, the displacing effect appears to remain constant and can be compensated for by about 8 db of imbalance [8].

Testing Conditions and Procedure

The stereophonic system used in the tests employed two microphones, two storage-transmission channels, and three loudspeakers, with the third loudspeaker bridged across the output of the two channels. This system may be termed a "2-2-3" array, making use of a convenient three-digit nomenclature which fixes in order the number of microphones, channels, and loudspeakers in a stereophonic system. For example, a "3-2-2" array would be one employing three microphones (the center microphone bridged into both channels), two channels, and two loudspeakers. A "3-3-3" array would of course be comprised of three channels each with its independent microphone and loudspeaker.

The tests were carried out in a room approximately 16 by 25 feet with acoustical characteristics similar to those of an average living room of similar dimensions. A moderate amount of carpeting and drapery helped to minimize standing waves.

Altec M-20 microphone systems (omnidirectional) were connected into the two channels of an Ampex 601-2 stereophonic recorder-reproducer. The levels of the channels were precisely balanced by placing both microphones in the same sound field and by setting the gains of the two record amplifiers for equal deflection on the VU meters. Each microphone was then placed at its proper recording location. Figs. 1(a) and 2(a) show the sound stage arrangements for Tests 1 and 2, respectively.

Recordings were made with a reader going from station to station making the same announcement at each. The playback configuration included two Klipschorn loudspeaker systems for the flanking positions and a Klipsch Model-H for the bridged center position. The signal for the bridged center position was not taken from a third power amplifier but rather from a series arrangement of the flanking loudspeaker amplifiers. The circuit, shown in Fig. 3, has the advantage of being only two thirds as costly as the usual three-amplifier arrangement. At the same time the cross-talk between flanking channels is low enough for stereophonic purposes, and a calculation of load impedances will show negligible mismatch if amplifiers of high quality are used. Note that the bridged speaker receives an additive mixture of both channels, thus preserving normal polarities. In addition, the bridged loudspeaker is provided with a calibrated L-pad so that its level may be precisely adjusted relative to the flanking speakers.

A variety of listening tests were made with listeners both on and off the axis of the middle speaker. In these tests the listeners were asked to locate on charts the apparent locations of the reader as he went from station to station.

The reproduced plots are not averaged from a number of listener tests. Rather, they are selected as being the most representative individual plots for each test.

Evaluation of Test Data

The data for Test 1 show the localization plots for listeners both on and off axis for three different level settings of the middle loudspeaker. At Fig. 1(b) the level is too high, thus causing a drastic narrowing of the virtual stage. At Fig. 1(c) the level has been dropped an additional 3 db, and the virtual stage begins to take on more natural proportions. A further drop of 3 db gives almost ideal balance. Reducing the level below this point resulted in a splitting apart of the sound continuum. Center events tended to pull toward one side or the other, and the objectionable "hole-in-the-middle" became apparent. The optimum level of the bridged loudspeaker was arrived at empirically by simple trial and error, and although the level was noticed to vary with room geometry, it was found to be fairly critical. That is, there is a range, perhaps 2 db wide, outside of which the geometry suffers. These observations hold only for the reproduction of spoken announcements. It will be seen in a later test that for musical reproduction there appears to be a much wider operating range for the bridged center loudspeaker.

It is interesting to note the effect of listener location upon stage width. For observers on the axis of the center loudspeaker the virtual stage is at its maximum [Fig. 1(d)]. For the listener off axis [Fig. 1(g)], the stage has been shifted to the left with an accompanying reduction in angular width—a natural consequence of shifting arrival times and intensities between loudspeakers and listeners as the listener moves about the room.

The data for Test 2 is shown in Fig. 2. This test is like the first except that the array of stations is one row deeper, thus calling upon the system to reproduce a greater degree of depth separation. Examination of the plots for Test 2 shows that the depth of the virtual stage is much less than the depth of the sound stage. In both plots the front-row stations (1, 9, 5, and 12) appear well forward on the stage; this is to be expected since the ratio of direct to reverberant sound is highest for these stations. In contrast, the plots tend to overlap in depth for the stations on the second and third rows. Lateral separation for all rows appears to be accurate, but the back rows doubtlessly suffer in depth separation because the difference in the direct to reverberant sound ratios for the two rows is not large. We might well mention at this point a limitation perhaps of the sense of hearing rather than of the system of reproduction. The acuity of the ears in detecting small lateral displacements has been alluded to earlier, but unfortunately they do not enjoy a comparable accuracy in depth separation. Depth separation with any degree of exactitude is difficult even when there is no intervening electroacoustical system. Where facilities allow, it is interesting to have the listeners plot the stations of a "live" reader behind a suitable curtain (one which is optically opaque but acoustically transparent). This may well serve as the final measure of accuracy in the experiments for certainly one cannot expect the stereophonic system to exhibit a greater degree of accuracy than is possible with live plots! Steinberg and Snow included live plots in their tests of 1933, and it was observed that the depth separation thus perceived was no

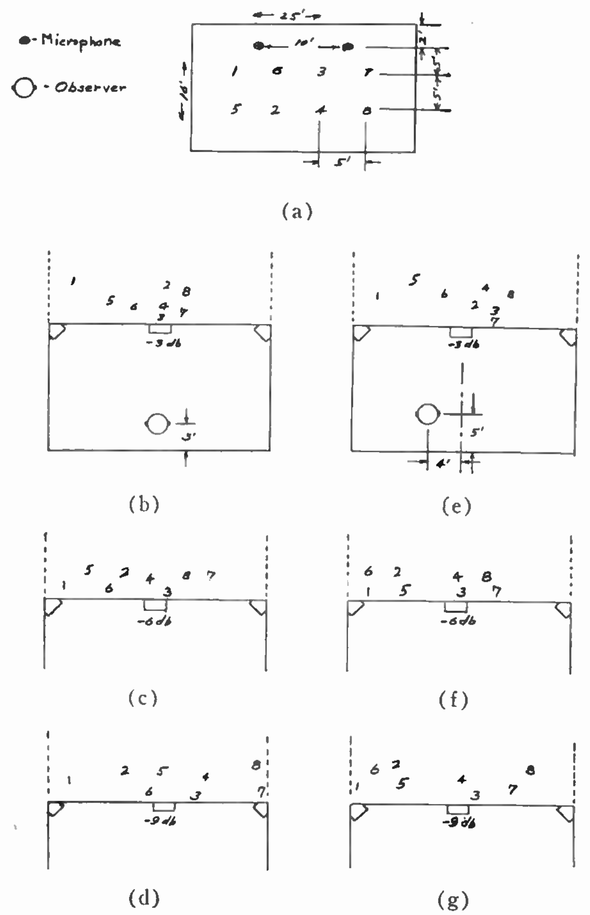

Fig. 1—(a) Sound stage for Test 1. (b)-(d) Listener on-axis. (e)-(g) Listener off-axis.

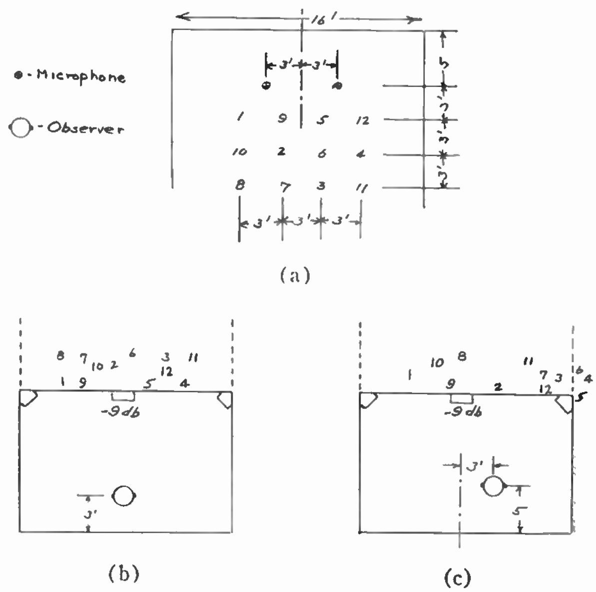

Fig. 2—(a) Sound stage for Test 2. (b) Listener on-axis. (c) Listener off-axis.

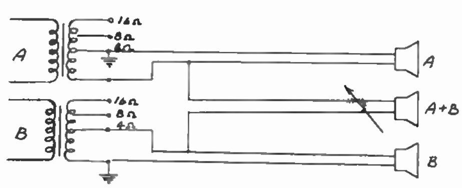

Fig. 3.

better than that reproduced over most of the systems tested.[2] Similar observations have been made more recently by Klipsch [6].

Test 3 makes use of a previously recorded tape of a four-piece jazz ensemble. The observers were asked to plot the apparent sources of the instruments in the usual manner. A musical instrument, such as a piano, does not emanate from a near point source as does the human voice. Accordingly, the observers plotted the virtual sources as areas and not as points on their charts. No specific directionality could be assigned to the string bass owing to the ear's relative inability to locate low-frequency sources. Fig. 4(a) shows the original recording geometry for the ensemble. Fig. 4(b) shows the virtual stage perceived by an observer on axis while Fig. 4(c) and 4(d) show the plots of observers considerably off axis. Note that at these extreme listening positions adequate geometry is preserved.

Conclusions

Perhaps the most immediate conclusion that can be drawn from the tests is that the balancing conditions required for successful geometry testing with vocal announcements are far more stringent than those required for ordinary stereo listening to music. A slight imbalance between channels during a geometry test can cause a large shift in apparent positions, whereas a change of perhaps 3 db will not appreciably alter the musical geometry. There are two reasons for this. First, musical sounds are not points but areas, and a shift of an area would be harder to detect than a corresponding shift of a point source. In addition, there is the attitude of the listener; when music is played he easily focuses his attentions on musical values while the technique of the playback tends to go unnoticed.

[2] See [9], p. 13.

The virtual stage is widest when the observer is located on the axis of the middle loudspeaker; it diminishes in width as the listener moves off axis. Considering Test 1, it is seen that the angular width of the stage, when final balance has been secured, is of the order of 90°. For the off-axis observer under the same balance conditions [Fig. 1(g)], the angular width is closer to 70°. For the test with the jazz ensemble the angular width diminishes more slowly as the observer moves off axis with the result that even at the extreme positions the angular width is still great enough for the listener to appreciate. For general stereophonic listening the room in which these tests were performed has proved itself quite satisfactory if the listener remains in that third of the room opposite the speakers. Any other position would necessitate rebalancing.

For a given listening position, the virtual stage width becomes a function of the intensity of the bridged speaker. The data for Test 1 point this out vividly. At one extreme the array becomes in effect a monophonic system with all localization taking place in the middle. At the other extreme the sound continuum is broken, and the listener becomes aware of two divergent sources. This is generally considered objectionable, but it may not be at all out of character for certain kinds of antiphonal music. Many examples of early choral and organ music could be pointed to here.

Thus, it becomes possible for a large array of speakers to reproduce the intimacy of, say, a string quartet, where the usual listening angle is rather small, as well as the broad expanse of a symphonic ensemble. This then is a distinct advantage of a "2-2-3" wide-speaker array over a "2-2-2" system with closer placement. The three-speaker array can adapt itself to a variety of musical demands, whereas the close two-speaker array

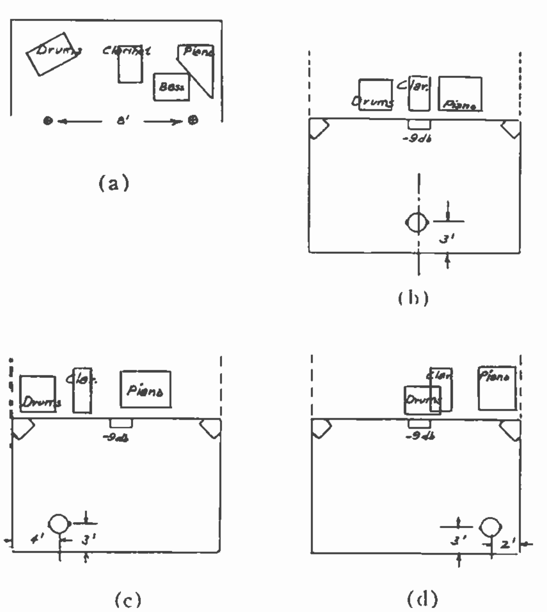

Fig. 4—(a) Recording studio. (b) Listener on-axis. (c) and (d) Listener off-axis.

can never give the illusion of angular width in excess of the physical angle subtended by the speakers at the listener.

In its most flexible form the "2-2-3" system would include three controls for the output circuitry: an over-all gain which would raise and lower both channels simultaneously by the same amount; a balance control (one that raises one channel while simultaneously lowering the other), to compensate for off-axis positions; and finally, a bridged center speaker level control to facilitate widening or narrowing the virtual stage according to the demands of the music.

Acknowledgment

The author wishes to acknowledge the generosity of Paul W. Klipsch in allowing extended use of the facilities of Klipsch and Associates, Inc., Hope, Ark., in performing the test herein reported.

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Compatible Cartridges for Magnetic Tapes*

MARVIN CAMRAS†, FELLOW, IRE

***Summary*—Magnetic tape may be offered in cartridge form at a price competitive with phonograph disks. Cartridges of different sizes are designed either for high quality or for maximum tape economy. All of these will operate on present-day machines, as well as on automatic designs. A cartridge changer allows records to be played in sequence. The erase feature offers interesting possibilities for sale of pure music separate from the sale of cartridges.**

Introduction

An inconvenience of present-day tape recorders is the threading operation. To overcome this objection, cartridges of many different kinds have been proposed. While solving the threading problem, they usually introduce difficulties in operation or in cost. The cartridge about to be described is relatively free of such difficulties.

In arriving at the design of Fig. 1, we considered that a good cartridge should fit readily into one hand. It should seal the record from dust, protect against accidental erasure, and should have a minimum of parts, insuring reliability and low cost.

* Received by the PGA, April 15, 1960. Based on a paper presented at the IRE International Convention, New York, N. Y.; March 23, 1960.
† Armour Research Foundation of Illinois Institute of Technology, Chicago, Ill.

The new cartridge fulfills these requirements admirably. It is a flat round package which inserts directly, without orientation, into the slot of an automatic player, or into the hopper of an automatic changer.

As shown in Figs. 1 and 2, the cartridge is formed of a spool with a central opening that fits present recorders. On the inner edge of each flange a bead is molded, which holds a wide mylar end-leader securely in place, sealing the inside to protect the tape from dust. The $\frac{1}{4}$ inch wide recording tape passes through the flanges with ample clearance after the leader has been unwound. A leader at the inner hub actuates the automatic reverse or rewind operation.

The label on the upper side of the cartridge, and milled beveled rim, reinforce sight with sense of touch, so that the cartridge will be inserted right side up. But if one should try the wrong way, it would not fit, because the opening is keyed at the top to pass the beveled but not the square rim, as in Fig. 3(a).

Protection against accidental erasure is provided by a safety groove of Fig. 3(b). If this groove G is present, then the feeler F enters it when the machine is switched to recording position, allowing normal erasing or recording. If the groove is *not* present, the feeler is blocked, and the machine is prevented from erasing or recording. Unrecorded cartridges are molded with this groove, but recorded cartridges are not. A removable insert H is provided, so that protection may be added after a cartridge has been recorded; or the groove may covered with pressure sensitive tape.

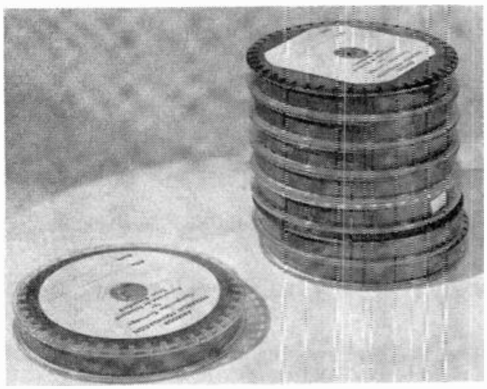


Fig. 1—Compatible tape cartridge.

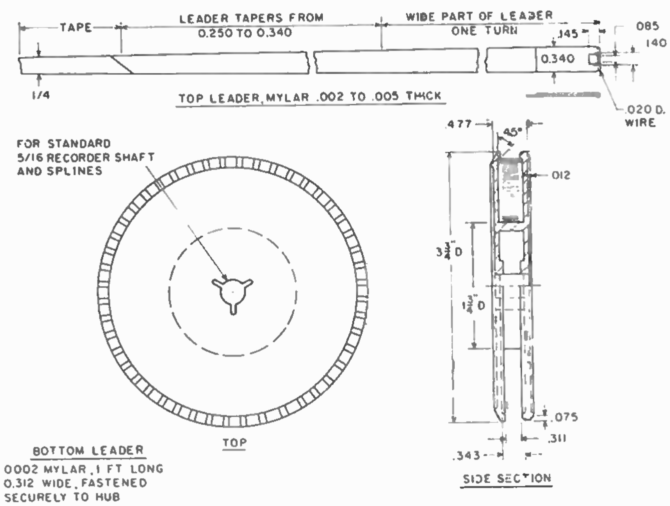


Fig. 2—Dimensions of compatible tape cartridge.

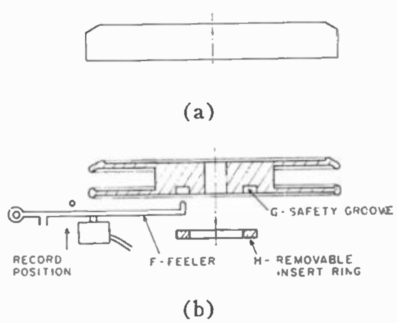


Fig. 3—Protective features in cartridge. (a) Opening shaped to prevent wrong insertion. (b) Cartridge section showing safety groove.

Machine Design

Automatic machines take countless forms. The handiest of these has a slot for insertion of the cartridge, as in Fig. 4. Operating sequence is explained in Fig. 5. When all the tape has gone through, the machine reverses automatically to play the second set of tracks, or rewinds rapidly, after which the cartridge is released. This type of machine is especially suitable for automobiles, where a cartridge may be picked out of a drawer and loaded with one hand, without taking one's eyes off the road (Fig. 6).

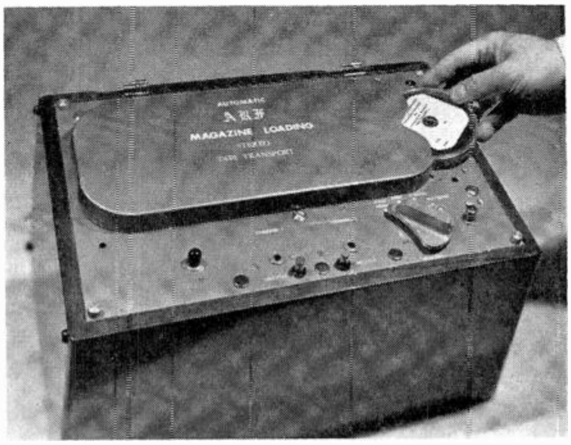


Fig. 4—Automatic cartridge machine.

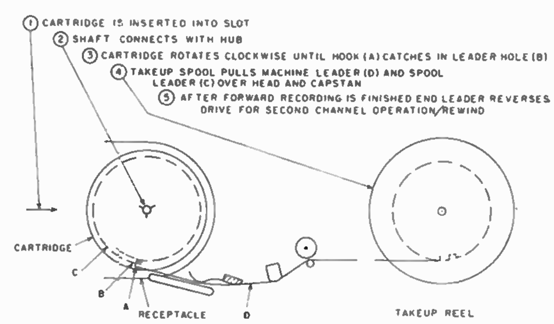


Fig. 5—Operation of tape cartridge on a machine.

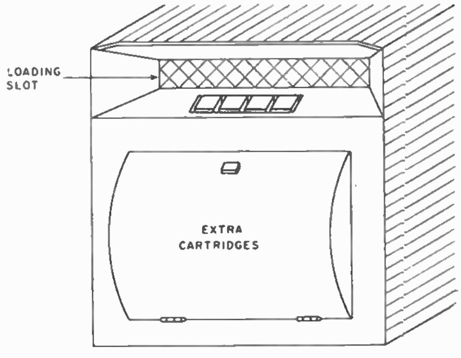


Fig. 6—Machine with front loading feature.

Automatic Changers

A very simple automatic changer may be added to the cartridge machine design. As shown in Fig. 7, the cover plate on the top is replaced with a plate that has a cylindrical holder for a stack of cartridges. When the machine is ready to change cartridges, a pusher moves the old cartridge to the left where it drops into a receptacle; at the same time it pushes the bottom spool from the stack into playing position, and then returns for the next cycle.

Deluxe music systems would use an arrangement as in Fig. 8. Here the cartridges are contained in individual compartments of a magazine (Fig. 9). In response to a pushbutton selector, the desired compartment is moved into place, and its cartridge is shifted into play position. After the playing cycle is finished, the cartridge is moved back to its compartment, and a new selection is made automatically.

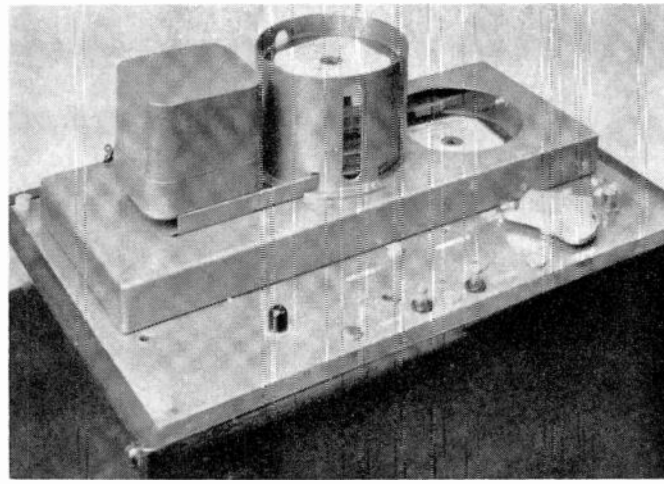


Fig. 7—Simple cartridge changer.

Semi-Automatic Operation

Inexpensive tape recorders can use semi-automatic loading as in Fig. 10. Here adapters were added to a standard tape recorder. For automatic threading the cartridge is placed on the left hand shaft and turned slightly until it engages the prethreaded leader on the machine. The recorder can then be operated in a normal manner. A special brake is provided on the right hand spool for smooth stopping after high speed rewind. Modifications of this kind can be built into many home recorders of present day design.

Fig. 8—Deluxe cartridge changer.

Compatible Operation

Next we come to the most elementary use of all, namely, operation on an ordinary spool type recorder. Millions of these have been built, and it is safe to say that such recorders will be used for a long time in the future. A connoisseur will not mind the threading if he owns a professional machine that gets the most of a recording. He will be interested in the library of programs that become available, as long as they fit his machine.

Actually, hand threading a cartridge is less involved than operating a manual disk phonograph. Placing the cartridge on the left hand shaft in Fig. 11 is comparable to placing a phonograph record on a spindle. Inserting the leader into the slot of the takeup spool is comparable to setting the tone arm on a phonograph. Nothing further is involved; there is no jacket to remove or to replace as with a disk.

The takeup spool in Fig. 11 has a ball detent in its hub. The ball detent snaps into the hole at the end of the leader, holding it strongly enough so that it will start in the takeup direction. At the end of rewinding, the leader pulls off of the right hand spool, and is tucked under the bead of the cartridge, so the cartridge is all wrapped up and ready to remove as a self supporting sealed package.

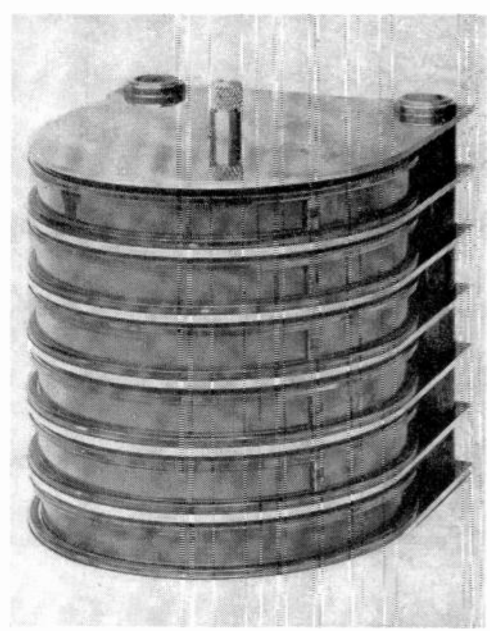


Fig. 9—Cartridges in a magazine retainer.

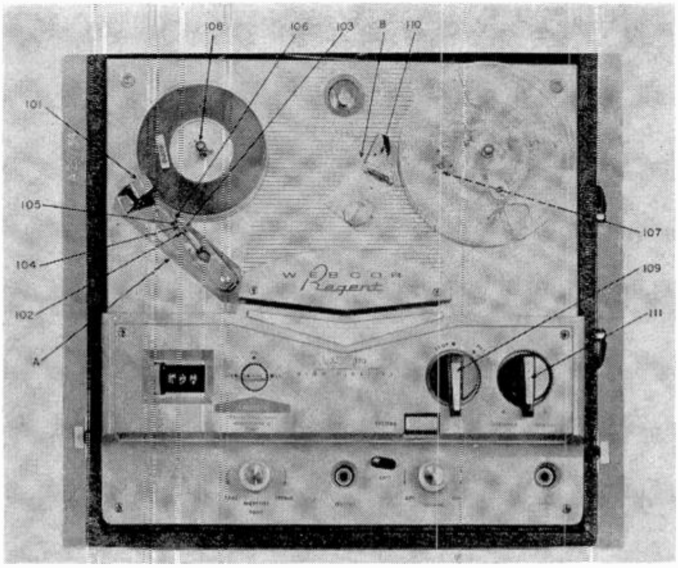


Fig. 10—Semi-automatic machine.

Fig. 11—Compatible use on a reel-type recorder.

Cartridge Size

What is the best cartridge size for general purposes? We can calculate this as soon as we are given specifications for tape speed, playing time, number of tracks, tape thickness and width, single or two way operation, etc.

But such factors depend on sound quality, cost, and other features convenient or attractive to the consumer. Science is partially helpful, in deciding these, but only when mixed with an ample quantity of opinion, intuition, foresight, speculation, and faith.

The proposed cartridge dimensions of Fig. 2 are conservative, and lead to the specifications of Table I. As seen from these specifications, playing time is consistent with stereo phonograph records, and with present tape libraries. The popular 32 minute interval is obtained even with the thickest grade of tape. Yet we have 64 minutes available for extra long programs. Or

TABLE I

Specifications for a Compatible Cartridge

Playing time	—48 minutes nominal for stereo (64 minutes maximum capacity, with 32 minutes the most likely in actual use). These times are doubled for monophonic.
Tape speed	—3.75 inches per second
Number of tracks	—4 (or more)
Tape width	—0.246 inch
Tape thickness	—One mil backing preferred for 48 minutes or less. The $1\frac{1}{2}$ mil tape may be used for 32 minutes, and the $\frac{1}{2}$ mil for 64 minutes.
Operation	—Two-way preferred
Over-all diameter	—3.75 inches

we can supply 32 minutes at $7\frac{1}{2}$ inches per second for extra high fidelity.

We should note that, under different circumstances, larger or smaller sizes of this cartridge design may be optimum. These can be interchangeable, just as we have different sizes of phonograph records. The cartridge is also universal in regards to running speed, tape width, and one-way or two-way operation.

Cost

A wise decision on tape speed must balance cost against performance. As speed is reduced, the price does not drop in direct proportion, because many costs remain fixed for a given piece of music, for example, the artist's royalities, packaging, advertising, and handling. One estimate indicates that doubling the speed of Table I would increase the retail price 17 per cent, and cutting the speed in half would lower the retail price 8 per cent. In all instances, the actual selling price of a cartridge is competitive to that of the equivalent long-playing disk record.

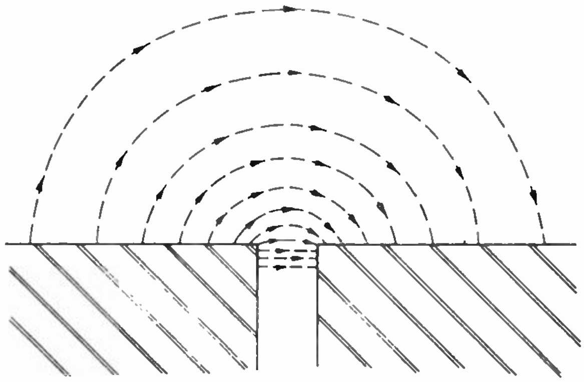
Frequency Response and Tape Speed

For a long time a rule of thumb in tape recording was "two thousand cycles per inch." At $7\frac{1}{2}$ inches per second we could expect about 15,000 cycles as the upper response limit. Next we tried for $3\frac{3}{4}$ inches per second, and were able to achieve a 15 kc at this speed also. However, the adjustments were quite critical, and difficult to maintain, and the dynamic range at higher frequencies was quite limited. Although the $3\frac{3}{4}$ speed has been a standard in home recorders for at least a decade, and although steady improvements were made over this period, most recorders still claim only a modest 7500 cycle response at this speed.

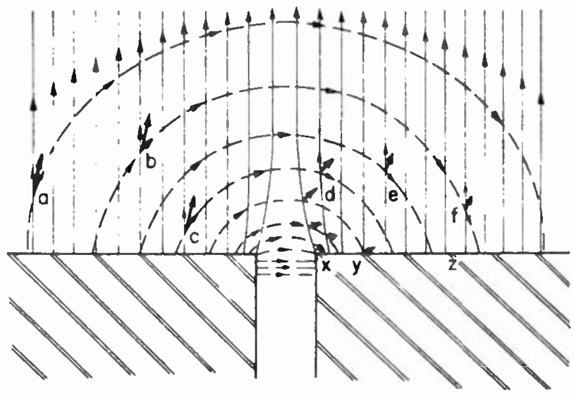
To obtain 15 kc response with good dynamic range at $3\frac{3}{4}$ ips or lower speeds, special care must be taken. Head gaps are cut down to about a micron, extra smooth tapes are indicated, and high frequency post-equalization is used.

In addition, special techniques may be necessary, as for example the cross field (X-field) head,[1] which

[1] M. Camras, "A new magnetic recording head," *J. SMPTE*, vol. 58, pp. 61–66; January, 1952.



(a)



(b)

Fig. 12—Flux paths of an X-field head. (a) Magnetic field produced by gap. (b) Superposed cross field adds vectorially to the gap field.

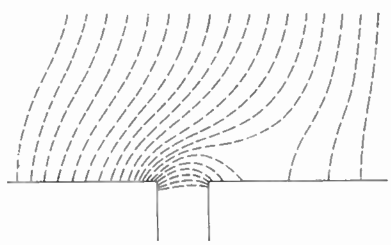


Fig. 13—Resultant of gap-field and X-field.

operates as shown in Fig. 12. To the usual semicircular field that surrounds the gap of a recording head, we add a vertical field. Vectorial addition of the components gives the resultant of Fig. 13. Near the trailing edge of the gap the field dies down more rapidly than in the simple head, giving better resolution, and allowing shorter wavelengths to be recorded. In the vertical direction the field is quite uniform throughout the depth of the recording layer; this means that the same bias adjustment is optimum for both long and short wavelengths. We do not have to compromise the high frequency response to insure distortionless recording of low and medium frequencies. Such results are shown in Table II. As described in the original paper, various modifications may be used. For example, bias alone may be supplied for the cross field, and signal alone for the gap field.

TABLE II

Comparison of Results, With and Without the X-Field, for the Same Experimental Head

Connection	Bias Required for Undistorted 100 Cycle Output	Bias for Maximum 10-kc Output	Loss at 10 kc Due to Setting Bias for Undistorted Low Frequency Response
Standard	1000 ma	600 ma	7 db
X-field	650 ma	560 ma	0 db
X-field	1000 ma (purposely over-biased)	560 ma	1.5 db

A similar development is the outside coil head of Fig. 14. A very small gap is defined by the lower polepieces which contact the tape surface. The energizing field is supplied by a coil and core on the side of tape opposite the recording gap. Here again the field is more uniform through the recording layer in the tape. Because of skin effect, high frequencies do not penetrate far into the top surface of the recording polepieces. Thus the short gap can be generous in depth, to allow for wear. This has been a real problem with conventional heads where depth can be only a few thousandths of an inch with short gaps.

With the aid of these tools we have been recording 20,000 cycles per inch. Under laboratory conditions we reached 40,000 cycles per inch, and are confident that we can obtain 100,000. A density of 40,000 corresponds to 15 kc response at a tape speed of only $\frac{3}{8}$ inch per second. Therefore, from a standpoint of frequency response, we can run at unbelievably slow speeds.

At this point it is appropriate to look at a graph (Fig. 15) that we used in 1951,[1] to show the trend of speeds for (audio) magnetic recording. At the time it seemed logical to extrapolate, and the dotted line, a, indicated that in about four more years we would be down to practically nothing. The curve has been revised with dashed lines to bring it up to date, and these show that we oversimplified the situation. Our 1951 curve branches into several parts. Branch b is performance that can be achieved in the laboratory. Here we have gone down to about $\frac{1}{4}$ inch per second, corresponding to a recording density of 40,000 to 60,000 cycles per inch.

Branches c, c', c'' are performance in commercial tape recorders. We seem to have leveled off at a speed centered around $3\frac{3}{4}$ inches per second for general purpose home recording, ranging up to $7\frac{1}{2}$ ips for high quality, and down to $1\frac{7}{8}$ ips for lesser requirements.

Since we obtain such excellent results experimentally, it might be feasible to standardize on a lower tape speed. We must pay for the lower speed with improved precision, less output, more amplification, and a lower signal-to-noise ratio. For a recorded density of

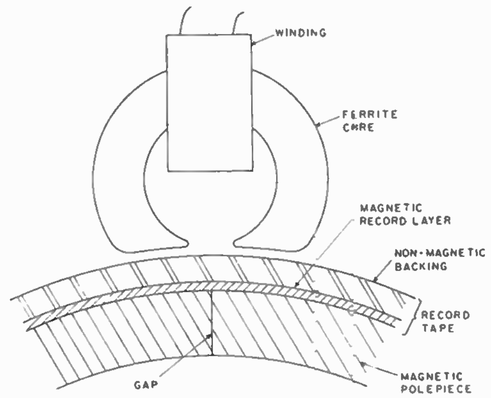


Fig. 14—The outside-coil head.

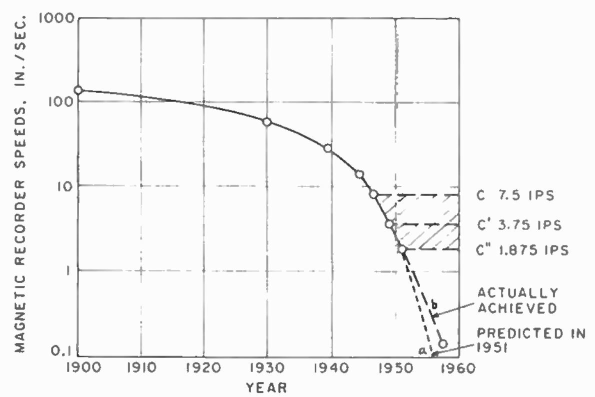


Fig. 15—Trends of magnetic recorder speeds.

10,000 cycles per inch the pickup gap must be 50 microinches or smaller. Azimuth must remain aligned within 2 minutes of arc. The tape must be super smooth and must not separate from the head by as much as a wavelength of light. Dynamic range at high frequencies is poor. The microgap playback heads are notoriously bad when one tries to record with them.

The above problems can eventually be solved or lived with. It is true that in the past, the public has accepted poorer standards of performance than those advertised. But we must consider another difficulty, even more important, and yet often overlooked, and that is the inherent unsteadiness of slow mechanical drives.

Wow and Flutter

As tape speed is reduced flywheel effectiveness goes down as the square of the speed. Other factors almost negligible at higher speeds become extremely serious. A few of these are shown in Fig. 16.

In Fig. 16(a) the length of tape, s, between capstan and head, fluctuates with the slightest irregularities in tension, especially with weaker and thinner tapes. Violin string vibrations, or squealing may be excited in this span, or in other spans coupled to s. As noted from the equation, flutter due to this effect is doubled whenever the tape velocity is cut in half.

Fig. 16(b) shows a precision capstan bearing greatly

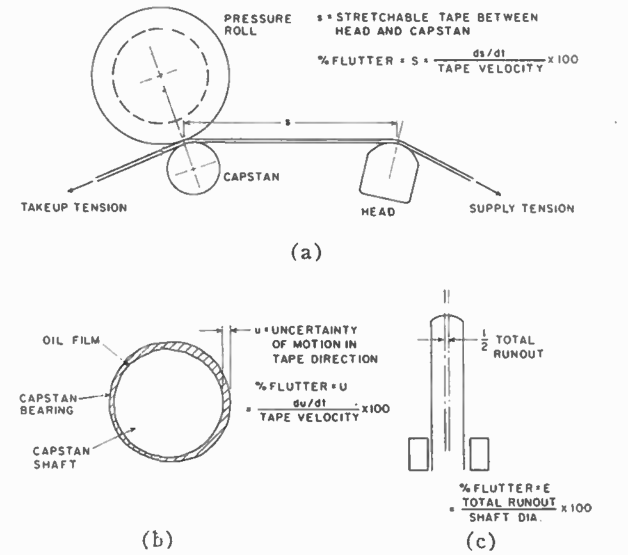


Fig. 16—Conditions contributing to wow and flutter. (a) Tape drive. (b) Journal fit. (c) Capstan eccentricity.

magnified. Neither the bearing nor the shaft are precisely round when measured within millionths of an inch. Conditions change constantly with wear, loading, bearing temperature, viscosity of the oil film etc.; but clearances u in the order of 0.0001 inch, are quite a trick to maintain in low cost mass production. Again we note that the flutter is doubled when we cut the speed in half.

Fig. 16(c), the capstan eccentricity, is always present.

The total cumulative irregularities resulting in wow and flutter may be summed up by

$$F = \sqrt{S_r^2 + U_r^2 + E_r^2 + \cdots S_p^2 + U_p^2 + E_p^2 + \cdots}$$

where the subscript r denotes recording, and the subscript p denotes playback. In addition to the irregularities considered above, we should add those due to the pressure rolls, belts, pulleys, motors, etc. When there is more than one recording and playback, each separate operation adds to the flutter.

Conclusions Regarding Performance

As the speed of a record medium is reduced, a point of diminishing returns is reached, where the loss in quality is not worth the savings in price. The optimum point falls rapidly at first, but reaches stability as the art matures.

We can draw on wisdom and experience of two older arts, motion picture photography, and phonograph records. In motion picture photography, 35-mm film is the standard of comparison. Sixteen millimeter is considered substandard but good enough for industrial and amateur use. Eight millimeter is marginal. Yet, in the laboratory we can demonstrate resolution that would make smaller sizes possible. It appears however, that for 8-mm film, we have reached an economic as well as a technical point of diminishing returns. At the 8-mm level, handling and selling costs are already such a large percentage of the retail price, that saving in raw film cost is minor.

In phonograph records the point of diminishing returns appears to be $33\frac{1}{3}$ rpm. It is quite interesting that from time to time we hear of "breakthroughs" where excellent quality is demonstrated at 8 rpm or less, on a four inch diameter disk record which plays for an incredibly long time. But up to now, even $16\frac{2}{3}$ rpm in commercial record changers leaves so much to be desired in quality that it is no threat to the $33\frac{1}{3}$ rpm standard. Again, the music and other fixed costs are such a large percentage of the cost of a record that the saving in material is minor.

The art of magnetic recording is changing so rapidly that it is difficult to agree on an optimum speed. However, one should consider that in commercial production, especially of low cost home machines, there ought to be an ample margin of safety to allow for wear and improper maintenance.

Enhanced Stereo With Multichannel Recording

Before stereodisks became available, tape recording had a virtual monopoly on stereophonic sound. At present stereodisks can provide two channels; but where three or more independent tracks are required, tape recording is still unique.

In the opinion of many listeners, two channel stereo is not entirely satisfactory. One problem is how to handle a soloist, as for example a singer in the center of the stage. If he is picked up by stereo microphones far to the right and left of him, and played back similarly through loudspeakers, much reverberation is added, and intimacy is lost. A third channel solves this problem. Another aid to realism is the recreation of room effects, giving a listener the sensation of being in the original recital hall. This also requires one or more additional channels.

For enhanced stereo three or four channels are none too many. One way to provide them is to run the tape in one direction only, using either three or all four of the standard tracks at the same time. Another possibility is the six track system of Fig. 17. We can use them three at a time, two way, or even all six at once.

Many aspects have not yet been explored fully. Pronents of enhanced stereo say that listeners prefer such systems even if the frequency response is restricted. But it is not known whether the enhanced stereo will increase one's tolerance for distortion, wow, flutter, or noise. A practical consideration is the objection by many stereo owners to loudspeakers housed in separate cabinets. Three or more speakers will be more of a problem than two.

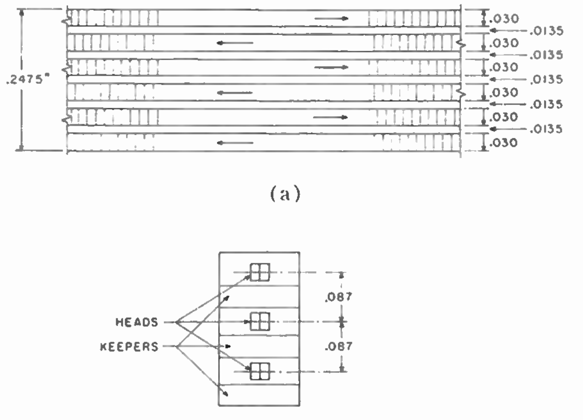


Fig. 17—Six track system on a $\frac{1}{4}$ inch tape. (a) Six track two way system. (b) Head spacing for six tracks.

Regardless of the number of tracks eventually chosen, the tape system is flexible enough to accommodate them, and in the meantime does not penalize the owners of less elaborate installations.

Self-Service Recording

The fact that magnetic records can be erased and rerecorded makes possible an entirely different form of merchandising. A dealer could stock cartridges of blank tape only. He would subscribe to a central service which would play master recordings to him over a telephone or coaxial line. He could dial for any master in a library of thousands of selections. At rerecording stations in his store, he could insert cartridges of blank tape, and in a few minutes they would have the desired recordings.

Many variations are possible. In smaller stores a self contained unit could receive both the cartridge and a coin in the slot; in return for which it would record any of a large number of masters. The customer could erase cartridges he no longer wanted, and rerecord his tape with new music.

Conclusions

A simple cartridge has been demonstrated for handling and storing tape records. It is compatible with present day equipment and with automatic machines. It is adaptable to future designs; yet is presently competitive with LP disks both in price and performance.

Acknowledgment

The author is indebted to A. P. Hultgren of American Molded Products, Chicago, Ill., for supplying the cartridge reels. L. Thunberg, S. Galus, C. Christensen, and P. Padva of Armour Research Foundation, Chicago, Ill., contributed to the design and construction of the cartridge machines.

Correspondence

Recorded Tapes*

Repeated perusal of recent articles concerning proposed new tape cartridge systems leaves me anxious. The CBS-3M system seems to have the greatest popular appeal; but as it will take longer than any other to reach production, and will involve a further hiatus in tape production, I like it least. Like many other audiophiles, I recently rebuilt my system for 4-track, 7½-ips tapes. The repertoire has only recently grown to the point where I can hope to obtain a few tapes to suit my left-handed tastes; and the CBS system would require another large-scale conversion—preceded by a drought of tape recordings of any kind, to judge by the effect of the ill-fated RCA Victor cartridge.

Not that I am against progress. But I have about as much use for a tape changer as a dog has for wheels, and as for reverberation, fine; but not at the expense of a third channel. And I shudder to think what will happen when wear affects the necessarily closer tolerances inside that sealed box.

It seems to me admirably demonstrated in a recent article by Camras[1] that economy is no longer a primary problem with standard tape. Now it is time for someone to get rid of the one remaining bogey: hiss. Even the 7½-ips tapes played with a 90-microinch head exhibit altogether too high a hiss level, and the published data on the CBS system doesn't promise much. Surely some of this astronomical packing density obtained by Goldmark, Mee, Goodell, Guckenburg, Brophy, *et al.*, could be traded for greatly improved signal-to-noise ratio. I know there are a number like me who would welcome a high-quality, compatible cartridge. I only hope it isn't too late to avoid another War of the Speeds.

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* Received by the PGA, June 10, 1960.

[1] M. Camras, "A compatible tape cartridge," IRE TRANS. ON AUDIO, vol. AU-8, pp. 62–67; March–April, 1960.

Terminology for Stereo with Two Signals and a Derived Center Output*

If laymen and technologists are to understand each other, they must use the same "dictionary." At the expense of being didactic, the following is submitted as a definition and terminology for the stereo system involving two signal channels and three speakers, the center reproducing a mixture of the two signals. It is proposed to call such a system a "2 channel with bridged center stereo," to be abbreviated "2–3 stereo." This differs from previous suggested terminology in view of the accepted definitions and terminology of stereo adopted by the Magnetic Recording Industries Association (MRIA).

Definitions agreed upon by MRIA and presented to the National Better Business Bureau comprise Standard Terminology as follows:

1) A *channel* is a single complete electronic transmission path for sound; it must include one or more loudspeakers. It may have a recorder and reproducer interposed as a time storage device. In a multichannel system, the number of channels is equal to the number of main transmission paths.

2) A *track* is a path which contains reproducible information left on a medium by recording means energized from a single channel.

a) A *recording channel* includes the means by which sound is prepared for storage on a single track.

b) A *playback channel* includes the means by which the recorded sound on a single track is reproduced.

The above definitions demand a new nomenclature for the stereo system involving two signals and three speakers with the center speaker deriving its output from a combination of the two signals.

Steinberg and Snow [1] used the expression "Bridging a third loudspeaker across the 2-channel system . . . . "

My own first paper [2] on this system called the center speaker a "phantom channel." There is an analogy to a phantom circuit but the principles are different, so this writer has abandoned the term "phantom."

The new definitions deny the center or bridged output as a "channel," leaving substantially a vacuum of terminology.

It is therefore proposed to name this system "2 channels with bridged center stereo;" being abbreviated "2-3 stereo" in contrast with my own former designation "2PH3 stereo" as short for "2-track (phantom derived) 3-channel stereo."

If more detail is desired the number of microphones may be indicated by a first number; thus "2-2-3 stereo" indicates 2 microphones, 2 channels, 3 speakers, "3-3-3" would be 3 electrically independent channels, "2 SD-2-3" would indicate a "stereo-directional microphone, 2 electrical channels, 3 speakers," etc.

It is not the purpose of this brief note to extoll the merits of the "2-3 stereo" system, but for those interested the appended bibliography affords a history of the art.

Note: W. A. Stocklin, Editor of *Electronics World*, after reading the above, has commented that some standard nomenclature should be used to identify a bridged or phantom third channel; but he did not believe that the suggestion "2-3 stereo" is the answer. He had no better suggestions to make. Other comments on this subject will be welcomed.

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* Received by the PGA, July 5, 1960.

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[5] P. W. Klipsch, "Circuits for 3-channel stereo playback derived from 2 sound tracks," IRE TRANS. ON AUDIO, vol. AU-7, pp. 161–165; November–December, 1959. Describes methods of "deriving" or "bridging" the center speaker across 2 channels.
[6] P. W. Klipsch and R. C. Avedon, "Signal Mutuality in Stereo Systems," presented at AES Convention, New York, N. Y., October 8, 1959, this issue, p. 166. This paper shows that 3 "independent" channels are not really independent due to signal mutuality and that a 2-2-3 system exhibits almost the same degree of independence.
[7] P. W. Klipsch, "Stereo geometry tests," (MS 51) submitted to *J. Audio Engrg. Soc.* (This paper compares localization of sounds rising with 3 microphones, 3 channels, 3 speakers, "3-3-3 stereo," with 2 microphones, 2 channels and 3 speakers, "2-2-3 stereo," and indicates approximately the same standard error for the two systems. For live, 3-3-3, 2-2-3 and 2-2-2 stereo the standard errors were respectively 0.042, 0.123, 0.137 and 0.30, where the numbers represent angular error divided by angle subtended by the array.

Contributors

William H. Beaubien (M'58) was born on December 31, 1923 in Toledo, Ohio. After serving in the U. S. Air Force as a B-29 navigator in World War II, he graduated from the University of Toledo in 1948 with the B.S. degree in engineering, magna cum laude.

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M. Camras

Mr. Camras contributed developments which are used in modern magnetic tape and wire recorders, including high frequency bias, improved recording heads, wire and tape materials, magnetic sound for motion pictures, multitrack tape machines, and stereophonic sound reproduction. He is a Fellow of the AIEE, and the Acoustical Society of America; and a member of SMPTE, AAAS, Eta Kappa Nu, Tau Beta Pi, and Sigma Xi. Offices held include PGA Chairman and Editor of these TRANSACTIONS and President of IIT Chapter Sigma Xi. He received the John Scott Medal in 1955, and the IRE PGA achievement award in 1958.

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After a two-year tour of duty with the United States Army, he joined Klipsch and Associates, Inc. Hope, Ark., as staff musical director in 1956. Since 1958 he has been a student at the University of Texas, Austin, Tex., pursuing the B.S. degree in electrical engineering. Presently he is general manager of the Austin Recording Company, Austin, Texas.

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P. C. Goldmark

Dr. Goldmark is a Fellow of AIEE, the Society of Motion Picture and Television Engineers, the Audio Engineering Society, and the British Television Society. In 1945 he was awarded the Television Broadcasters Association medal for his color television pioneering work, and in 1946 the IRE awarded him the Morris Liebmann Memorial Prize for electronic research. In 1960 he was given the PGA Achievement Award. He is a visiting professor for medical electronics at the University of Pennsylvania Medical School.

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John D. Goodell was born in Omaha, Neb., on September 20, 1909.

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From 1936 to 1945 he worked in the Central Research Laboratory of Siemens and Halske in Berlin, being engaged in high frequency measurement apparatus, and research work on relays and contact spark extinguishers. After the war, he joined AEG, in Berlin, where he performed pioneering work on audio magnetic tape recorders. He then assumed a research post with the Heinrich Hertz Institute in Berlin, in 1950, where he continued research work in the theory and components of magnetic recordings. In 1957, he joined CBS Laboratories, Stamford, Conn., to develop new techniques in magnetic recording.

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Paul W. Klipsch (A'34–M'44–SM'45), for photograph and biography, please see page 106 of the May–June issue of these TRANSACTIONS.

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Harwood B. Moore was born on March 9, 1928 in Providence, R. I. He received the B.S. degree in electrical engineering from the University of Buffalo, N. Y., in 1953, and continued graduate studies at Northeastern University, Boston, Mass., from 1954 to 1955.

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