DESIGN AND CONSTRUCTION OF AN ARTIFICIAL REVERBERATION PRODUCER

by

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ABSTRACT

The purpose of this work was to select design criteria, and to design and build a device to simulate reverberation by producing a succession of closely-spaced echoes such as those produced in an actual room or auditorium by reflections from the walls. Such a device can be used in studying the effects of different reverberation conditions upon speech and music.

The device which was constructed utilizes magnetic recording in the following manner. The original sound, to which reverberation is to be added, is recorded on loops of tape, and then artificial echoes of this sound are picked up from the tape by pickup heads a short time later. The sound is erased from the loops of tape at a point before the tape enters the recording head.

The device consists of two parts: four loops of tape traveling at 15 inches per second, and a high-speed loop of tape traveling at 69 inches per second. The high-speed loop was provided in order to decrease the time between the original sound and the first echo. Circuits were also provided to feed the first echo back to the recording head at reduced amplitude, thus producing a second echo, which is fed back and produces a third echo, and so on, ad infinitum.

This paper describes the design and construction of the entire mechanical part of the device, and the electronic and magnetic circuits

for the high-speed tape loop, including circuits for feeding the echoes back to the recording head. Tape-head frequency response characteristics at a tape speed of 69 inches per second were measured, and proper equalization circuits were devised and constructed to provide approximately flat overall frequency response. Alternating bias and erase current were used on the tape. Proper frequency and magnitude for bias and erase current were determined, and an oscillator was constructed to provide this current.

When music or speech is fed into the device, a reverberation effect is added, but there is a "flutter" sound in the reverberation, similar to the "flutter" echo obtained when sound is reflected between two hard walls.

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I. INTRODUCTION

A. Statement of the Problem

The problem was to select design criteria, and to design and to build a device which would produce a succession of closely-spaced echoes such as those produced in an actual room or auditorium by reflections from the walls. In this manner, reverberation would be simulated.

B. Importance of the Problem

The question of the reverberation conditions most desirable for rooms and auditoriums has always been very important, and yet is still undecided. If a room is too reverberant, that is, if sound persists in a room too long, then it is difficult to understand speech, because echoes of previous words interfere with later words, and music sounds "jumbled", because echoes of previous notes interfere with later notes, especially in rapidly-played passages. But if the walls of a room absorb too much of the sound instead of reflecting it, music is not as pleasing as with a little more reverberation. Speech is more pleasing and even more intelligible with the proper reverberation.

A device which would produce controllable echoes synthetically, that is, without requiring a certain type room or auditorium, would be very valuable for the study of the effects of different reverberation conditions upon music and speech. The device would have to be capable of producing echoes of any desirable spacing and amplitude within a certain range. It was the object in this work to design and build such a device.

C. History of the Problem

Several devices which artificially simulate reverberation to the human ear have been built, but they lack the qualities of flexibility and control required for scientific study of the effects of reverberation.

The author has found that the effect of a large cavern can be produced from an ordinary phonograph by simply mounting a second pick-up arm displaced about 60° around the disc from the usual pickup. This provides an echo delayed about 0.13 second, at a turntable speed of 78 rpm. It is necessary to have the needles of both pickups riding in the same groove on the record.

A synthetic reverberation device was developed by Begun and Wolfe using magnetic recording.[1] A recording head recorded the sound on a moving, endless loop of steel tape. Pick-up heads then picked the sound off the tape as it traveled along, thus providing the echoes. An erase head removed the sound at a point just ahead of the recording head. A disadvantage of the machine was that a click was produced each time the soldered joint passed a pickup head. Other disadvantages were that it was not possible to get the echoes spaced closer than about 0.1 second apart, and at the time this device was built, magnetic recording was not developed to the point where it could produce good fidelity.

Goldmark and Hendricks built an electro-optical synthetic reverberation device as follows:[2] The original sound was recorded on the

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[1] S. J. Begun, S. K. Wolfe, "On Synthetic Reverberation", Communications, v. 18, 8, August 1938.

[2] P. C. Goldmark, P. S. Hendricks, "Synthetic Reverberation", Proceedings I. R. S., p. 747, December 1939.

rim of a rotating phosphor-coated disc by means of a modulated light source directed through an optical system. Then photocells, placed around the disc, picked off echoes through another optical system. The logarithmical decay of the light images in the phosphor as they passed the photo-tubes gave the reverberation effect. A high-pressure mercury-vapor lamp was used as the modulating light source, and considerable difficulty was encountered in this device.

The principle of sending sound down a long tube and picking it up at the far end with a microphone has been used by W. R. Johnson[1] to produce synthetic reverberation. Four tubes, of lengths of 25 ft., 50 ft., 75 ft., and 100 ft., were used, thus spacing the echoes 0.023 second apart, since sound travels about 1100 feet per second. A scheme of feeding the picked-up echoes back to the driving unit and sending it down the tubes repeatedly in order to lengthen the reverberation time was used. Considerable difficulty was encountered because of the great attenuation of the high frequencies in being sent down the long tubes.

D. Basic Method Used

Magnetic recording was used in this investigation. Magnetic recording has, within the last few years, reached the stage of development where sound can be recorded and reproduced with greater fidelity by this means than by any other means. The sound, of which echoes are desired, is recorded on a loop of moving magnetic tape, and then picked off by pickup heads a short time later. Thus an artificial echo is produced.

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[1] G. W. Curran, "An Artificial Reverberation System", Audio Engineering, p. 13, May 1948.

The sound is erased from the loop of tape at a point before it enters the recording head.

II. DESIGN CRITERIA

In the original design, the machine consisted of four loops of tape, moving at a linear speed of 15 inches per second. Provision was made for each loop to have an erase head, a recording head, and a pick-up head, and all heads were adjustable along the tape in order to provide different time delays. The heads for these four tape loops, the horizontal bars along which the heads can be moved, and the drive mechanism for the tapes, are in the upper part of the machine, illustrated on pages 10, 11, and 12. Only one of the four loops of tape is on the machine in these pictures, in order to show better the details of the machine.

The four loops of tape were used to decrease the time between echoes, as explained below:

When a pickup head is placed against the recording head on one of the loops of tape, the center-to-center spacing of the two heads is about $1\frac{1}{2}$ inches. This means that the first echo cannot be had in less than 0.10 second after the original sound is recorded on the tape (with a tape speed of 15 inches per second). Using all four loops, the original sound can be fed to all four recording heads simultaneously. Suppose the spacing between the recording head and the pickup head was 2 inches on the first loop of tape, $2\frac{1}{2}$ inches on the second loop, 3 inches on the third loop, and $3\frac{1}{2}$ inches on the fourth loop. Then let the sound picked up by the pickup head on the first loop be used as the "original" sound. Then echoes would occur at time spacings corresponding to a distance of $\frac{1}{2}$ inch along the tape, or 0.0333 second. The time difference between the "original" sound and the first echo could be made as small as desirable, even down to zero, by making the spacing between the recording head and the pickup head the same value for two or more loops.

This method is satisfactory for decreasing the time between echoes in cases where the original sound can be discarded, as in making records or in radio transmitting, and the first picked-up sound used as the "original" sound. The delay of at least 0.10 second would cause no trouble in these cases. But anytime the actual original sound is heard, as when an audience is attending a live program, this method would not work, because of the 0.10 second delay between the original sound and the first echo. With this great a delay, the human ear will hear a distinct echo instead of the effect of a lingering sound.

To overcome this shortcoming, it was decided to add to the machine another loop of tape traveling at a much higher linear speed. A loop of tape driven directly by the rim of a 17-inch-diameter, 78 rpm turntable has a linear speed of 69 inches per second. With this tape speed, the time between the original sound and the first echo can be about 0.02 second, and this delay is small enough to simulate reverberation, and the original sound can still be used. Then, by feeding the first echo back to the recording head at reduced amplitude, a second echo can be obtained; the second echo will produce a third echo, and so on, providing an infinite series of decreasing echoes.

The four-loop part of the machine has advantages. Time between echoes can be decreased right down to zero (when the original sound can be discarded). This capability is extremely useful in some acoustical studies. Also, the four-loop part of the machine can be combined with the high-speed loop to produce various combinations of echoes superimposed on an exponentially decaying train of echoes, or each echo from the four-loop system can be given a train of reverberation following it, by the high-speed loop.

Because of the large amount of work involved, it was decided to divide the original program into two parts. A senior thesis investigation of building the electronic circuits for the magnetic heads for the four-loop part of the machine was undertaken by Daniel von Recklinghausen.

The author designed and constructed the entire mechanical part of the machine, both the four-loop part and the high-speed loop part. (The author supervised the part of the work which required the services of a skilled machinist). The author also designed and constructed the electronic and magnetic circuits for the magnetic heads on the high-speed tape loop, including circuits for feeding the echoes back to the recording head.

III. MECHANICAL DESIGN

Three magnetic mediums were considered:

(1) discs with magnetic coatings around the rims,

(2) a wide, multi-channel tape,

(3) several loops of standard $\frac{1}{4}$-inch tape.

The discs were not used because of the difficulties of getting a very smooth, concentric coating on the rims. The slightest irregularities in the rims would cause great wear if the heads were held in contact with the rims, and difficulties would be encountered in recording high audio frequencies if the heads were held a short distance away from the rims. A wide, multi-channel tape was not used because of the difficulty of maintaining intimate contact between the tape and many movable heads. The standard $\frac{1}{4}$-inch tape loops were chosen because of this tape's high quality and availability. It is necessary to form the loops by joining each end of a straight piece of tape; loops are not available from tape manufacturers. However, by making a careful, diagonal splice, virtually all of the noise caused as the splice passes the pickup head is removed. With a diagonal splice, even if it separates slightly, there is never a time when there is no magnetic material bridging the poles in the head, whereas with a transverse splice, parallel to the gap in the head, if the splice separates ever so slightly, then there is no magnetic material bridging the poles in the head when the splice passes the gap.

A 17-inch diameter phonograph turntable was available, and so it was used as the drive mechanism for the tapes. It was mounted in the simplest manner, on two vertical sheets of $\frac{1}{2}$-inch 24 ST Dural, and a horizontal plate of 3/8-inch Dural was used to support the head carriers and idler pulleys for the four loops as shown on pages 10, 11, and 12.

The drive capstan was made 3.66 inches in diameter to drive the tapes at 15 inches per second, with a turntable speed of 78 rpm. A block of Dural and the turntable were bolted together and put in a large lathe, and the block of Dural was turned down to form the capstan. Both the turntable and the capstan were revolved together in the lathe in order to make the capstan as nearly concentric with the turntable as possible. A small cylinder was left projecting at the top of the capstan. A steel sleeve was put tightly around this projection, and then a bronze bearing was mounted for this steel sleeve to revolve in. The steel-on-bronze makes a good bearing, and the bearing serves to keep the capstan in true alignment. See Fig. 1.

The idler wheels for the loops of tape were mounted on pivoted bars, as shown in Fig. 2. The adjusting screws, which allow for slightly different lengths in the tape loops, push against the pivoted bars with pieces of sponge rubber. Sponge rubber is used in order to be able to adjust the tension in the tape, without allowing the pivoted bars to vibrate, as springs might do.

The way in which the head carriers for the four loops are mounted on the horizontal bars can be seen on pages 10 and 11.

The mechanical design and construction for the high-speed tape loop was quite simple. The tape loop is driven directly by the rim of the turntable, at a speed of 69 inches per second. As is shown on pages 10 and 12, a small plate was bolted to the top of the turntable base, and

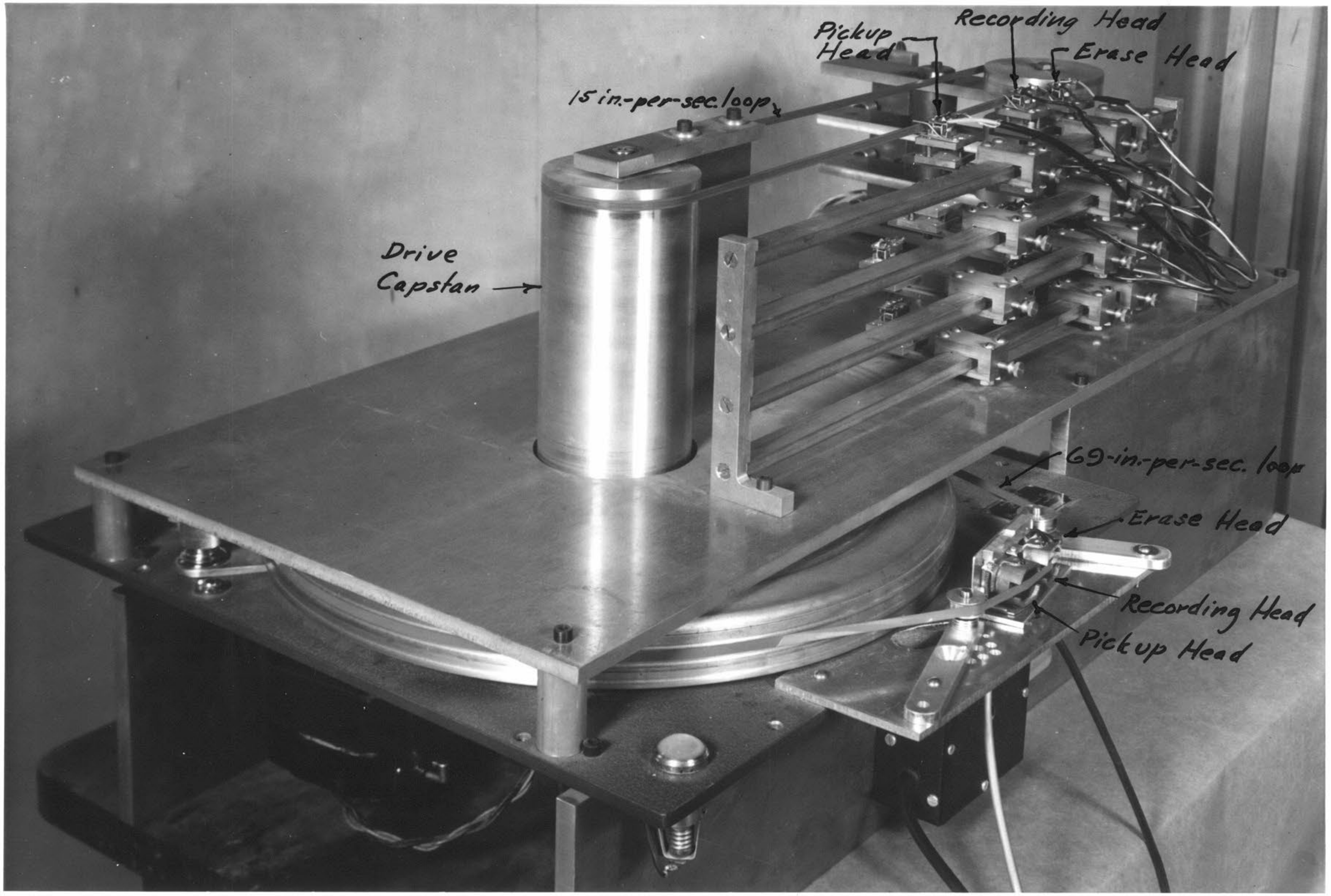

Pickup Head
Recording Head
Erase Head
15 in.-per-sec. loop
Drive Capstan
69-in.-per-sec. loop
Erase Head
Recording Head
Pickup Head

Fig. 1

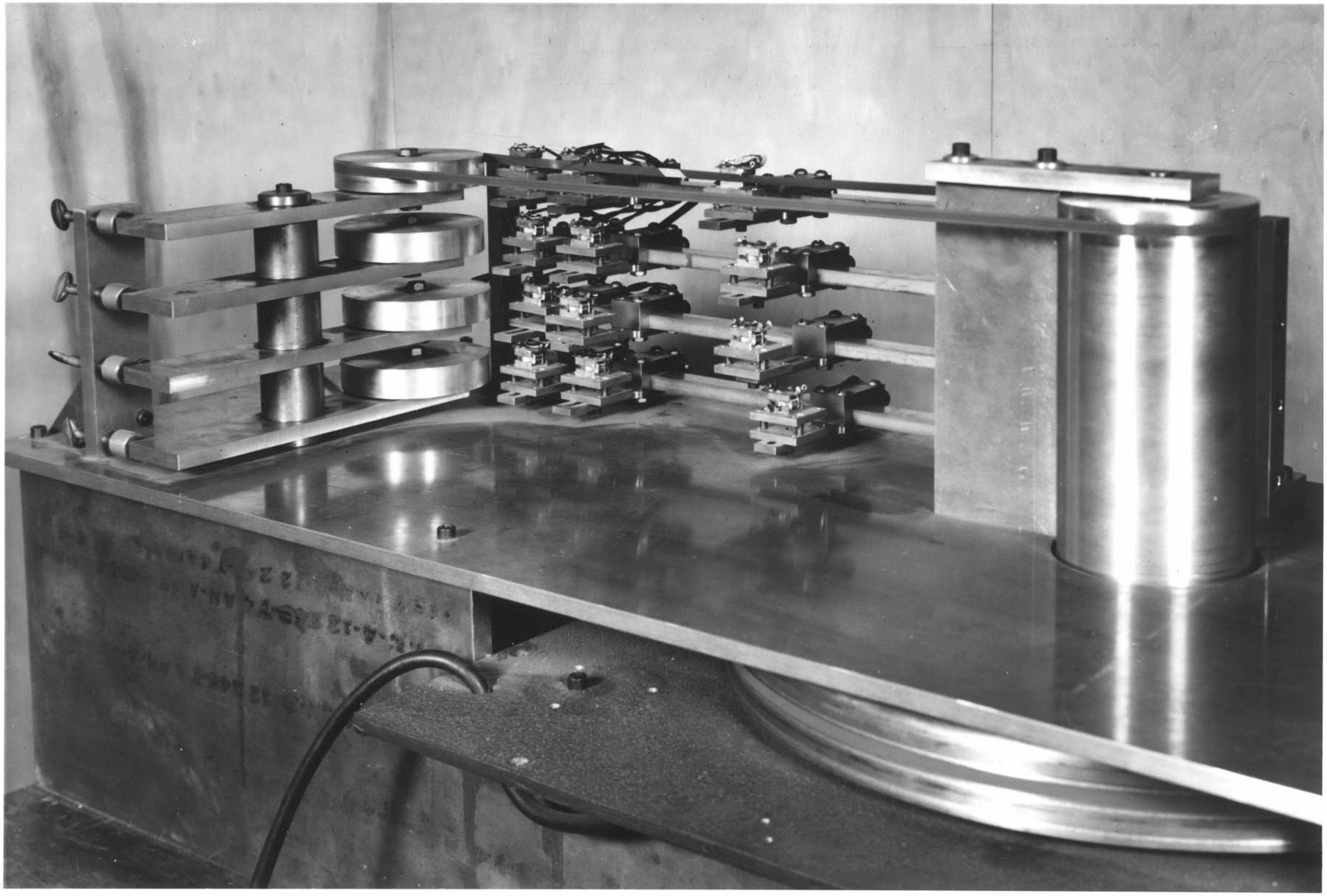


Fig. 2.

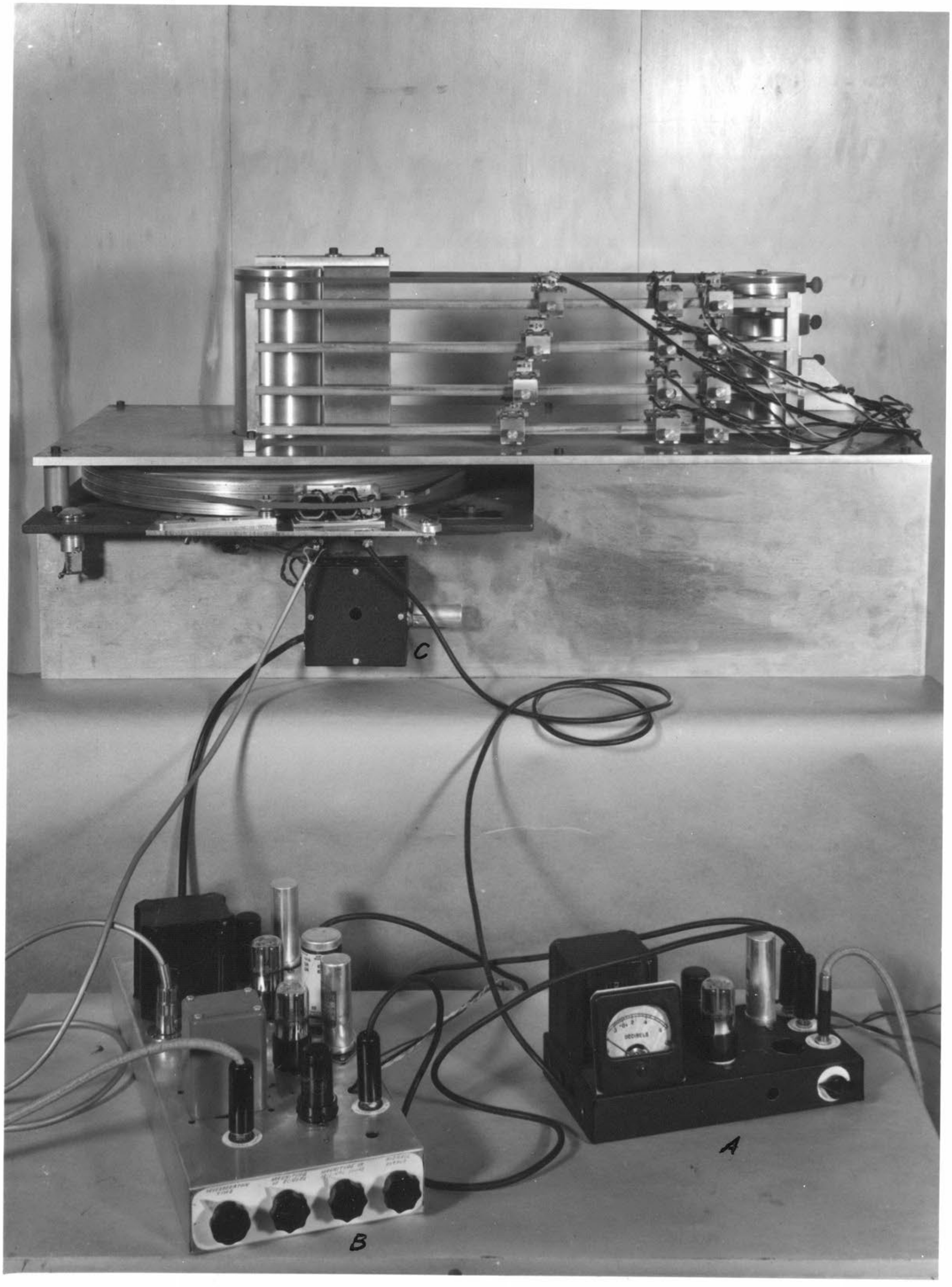


Fig. 3. ENTIRE SYSTEM FOR THE 69-IN.-PER-SEC. LOOP

A. Recording Amplifier
B. Playback Amplifier and Mixers
C. Bias-Erase Oscillator

the heads for this loop were mounted directly on this plate. A unit consisting of an erase head, a recording head, and a pickup head was obtained from Magnecord, Incorporated. The center-to-center spacing between the recording head and the pickup head is $1\frac{1}{4}$ inches. At this speed, this makes the time delay 1/55 second, or 18 milliseconds. Idler rollers were constructed and mounted as shown on page 10, in order to make the tape enter and leave the heads at the proper angle, that is, without wrapping too far around the heads.

Magnecord heads were used because they have the following characteristics: wide pole pieces and smooth gaps, which cause the tape to ride smoothly over them; capability of handling full audio range; recording and pickup head are well-shielded with mu metal; erase head does not operate too hot, even for the closed loop of tape, which has little time to lose heat before it passes through erase head again.

IV. ELECTRONIC CIRCUITS

A. Block Diagram and Basic Electronic System

The electronic circuits discussed in this paper serve to provide erase-bias current, feed the recording head, and amplify the output of the pickup head, for the 69-inches-per-second tape loop. In addition to serving these functions, the circuits also provide considerable

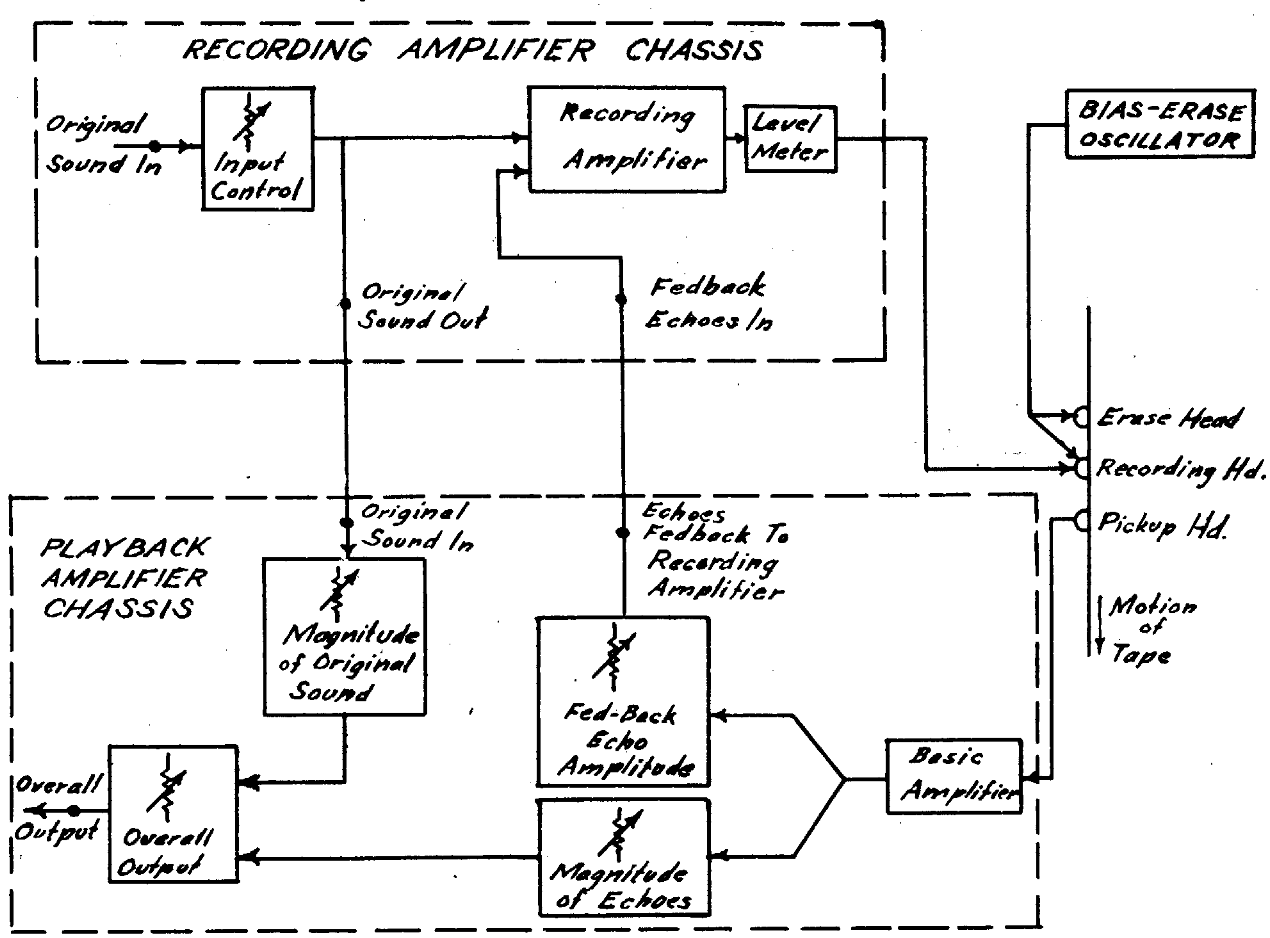


Fig. 4. BLOCK DIAGRAM OF SYSTEM

flexibility in combining various channels and feeding the sound from the pickup head back to the recording amplifier. Isolating stages have been used to prevent the various channels from interacting. The actual equipment is pictured on page 12.

Refer to the block diagram of the system in Fig. 4. The original sound has a path to the output bypassing the entire recording-playback system, with a separate level control "Magnitude of Original Sound". The original sound is also fed to the recording head, and picked up from the tape 18 milliseconds later. This picked-up sound is fed to the output through the control "Magnitude of Echoes", thus providing a single echo. This picked-up sound is also fed back to the recording amplifier, recorded again, picked up and fed back again, ad infinitum. The control, "Fed-Back Echo Amplitude", determines the drop in amplitude of a pulse echo each time the echo is fed back through the system. It was calibrated in db, by opening the feedback loop, feeding in a signal at "Fed-Back Echoes In", and measuring the output at "Echoes Fedback to Recording Amplifier". For example, since the time spacing between the recording head and pick up head is approximately 1/55 second, then if this control is set at -1 db, pulse echoes will decay with a slope of -55 db per second; and if the control "Magnitude of Echoes" is set 1 db below the control "Magnitude of Original Sound", then the first echo will be 1 db below the original sound, and the second echo will be 1 db below the first, this giving a reverberation time of 1.1 seconds (for 60 db decay). This condition is illustrated in Fig. 5. If the control "Fed-Back Echo Amplitude" is left set at -1 db, then the slope at which the echoes decay remains -55 db per second, but the whole train of echoes can be raised or lowered relative to the original sound by the control "Magnitude of Echoes", as illustrated in Fig. 6.

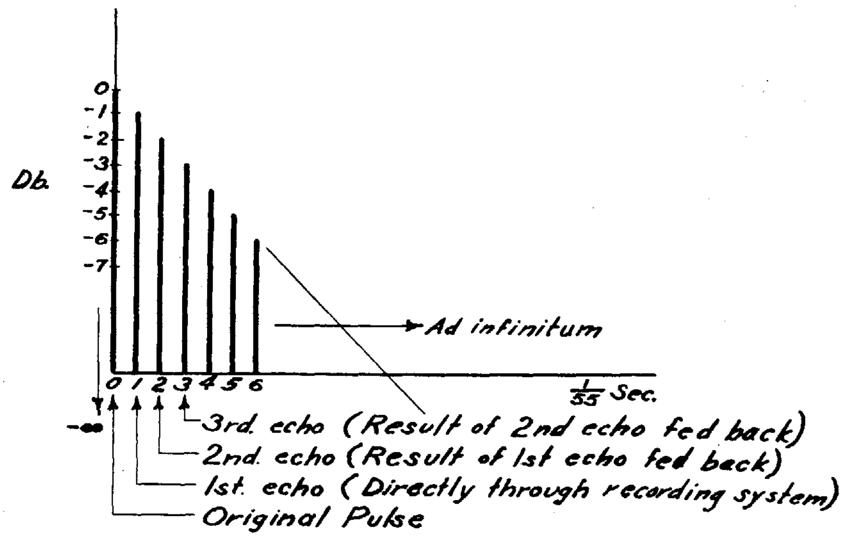


Fig. 5

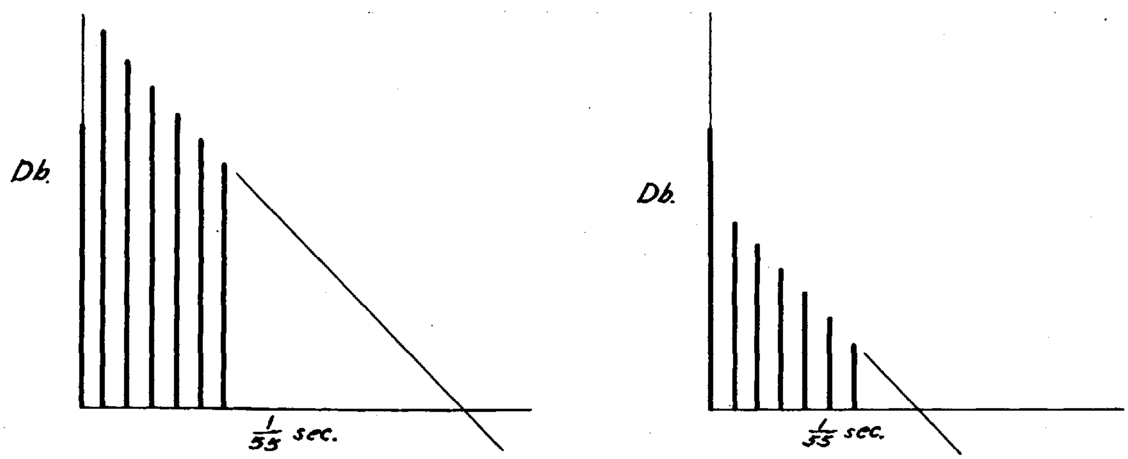


Fig. 6

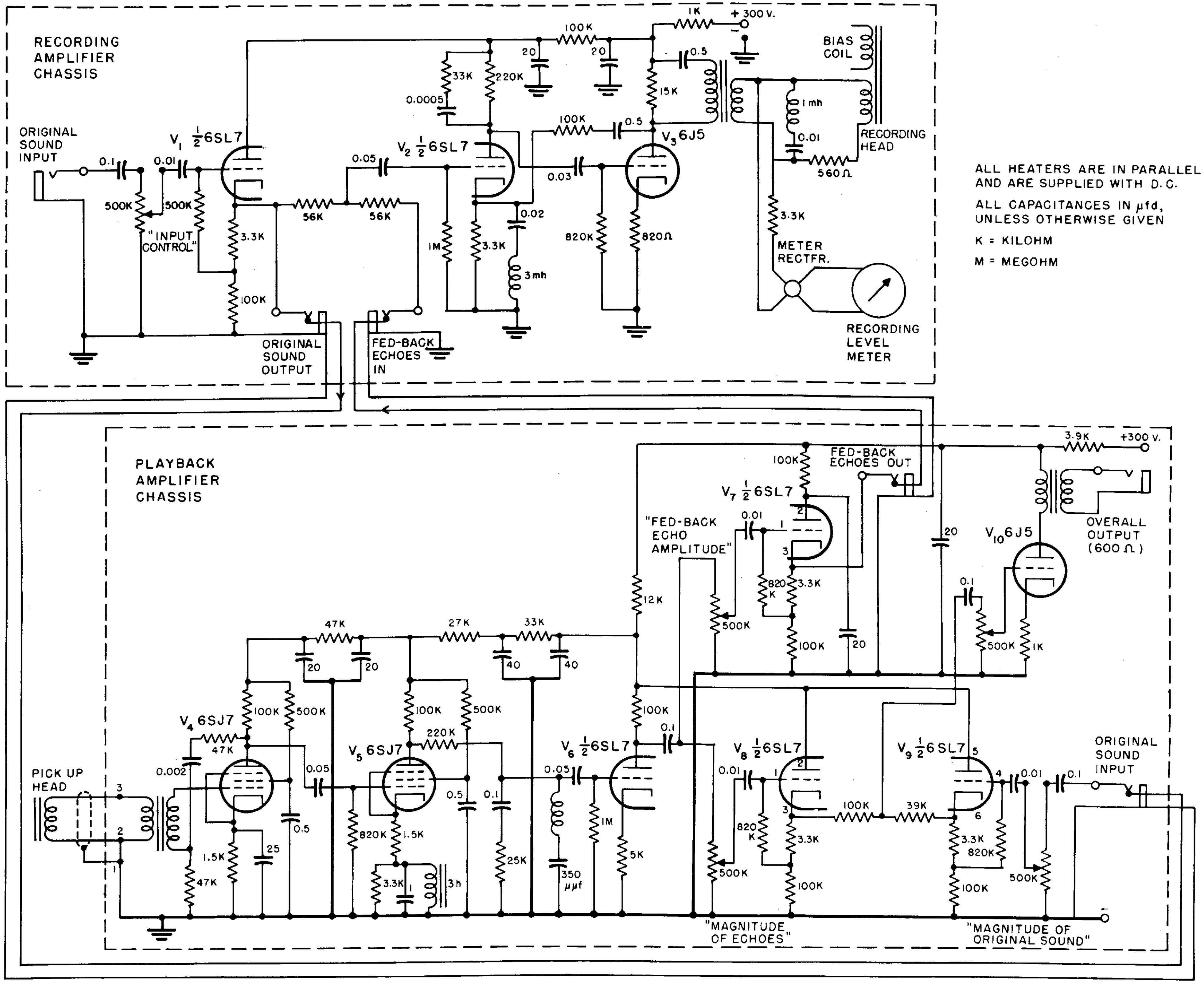
If the control, "Fed-Back Echo Amplitude", is turned to zero db, then the picked-up echo is fed back to the Recording Amplifier at the level which is required to produce its own amplitude, that is, the following echo will be of the same amplitude as the echo which produced it. Then the system will go into oscillation at a frequency at which the phase of the fed-back echo is in phase with the signal which produced it. If the frequency response of the recording system were perfectly flat, then there would be many frequencies at which the system could oscillate. However, in the actual case, the system will oscillate at the frequency at which the phase relation is proper and also the gain of the system is greatest.

Except for some changes to be discussed later, which were made in the basic recording and playback amplifiers, Fig. 7 shows the entire electronic circuit as originally designed to provide the controls and flexibility of the system in the block diagram described above. Following the input stage V_1 is a mixer circuit in which the echoes, fed back from V_7, are added to the original sound, and both the original sound and the fed-back echoes are then recorded. The outputs of V_8 and V_9 are added, providing the combining of the echoes with the original sound for the output of the entire system. Four cathode followers were used to feed these mixers for several reasons. First, this type mixer makes the interaction between the two channels being combined practically zero, since looking back into the cathode circuit of a cathode follower, the impedance is very small; in these cathode followers,

$$R_{internal} = \frac{R_K r_p}{(1+\mu)R_K + r_p}$$

$$= \frac{103.3 \times 44}{(71) \times 103.3 + 44}$$

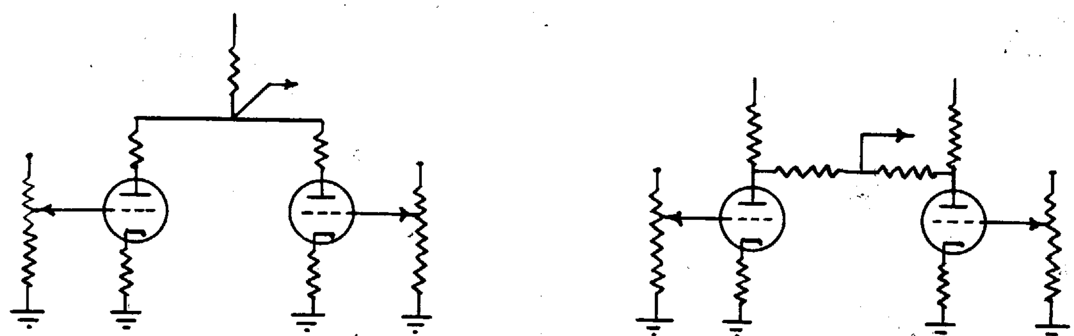
$$= 0.617\,k, \text{ or } 617\,\Omega.$$



SCHEMATIC DIAGRAM OF RECORDING AND PLAYBACK AMPLIFIERS

FIG. 7

In the type mixers shown in Fig. 8, the high-frequency response of each



(a) (b)

Fig. 8.

channel was found to be altered slightly by the level setting in the other channel. The second advantage of the cathode followers used is that V_1 and V_7 provide low impedance sources for feeding the cables connecting the two channels. The third and perhaps most important advantage of the cathode followers is the very low input capacitance; in these cathode followers,

$$C_{in} = C_{gp} + C_{gk}(1-K') \text{ where } K' \text{ is the gain.}$$

$$= 2.8 + 3.2(1-0.982)$$

$$= 2.86\ \mu\mu fd.$$

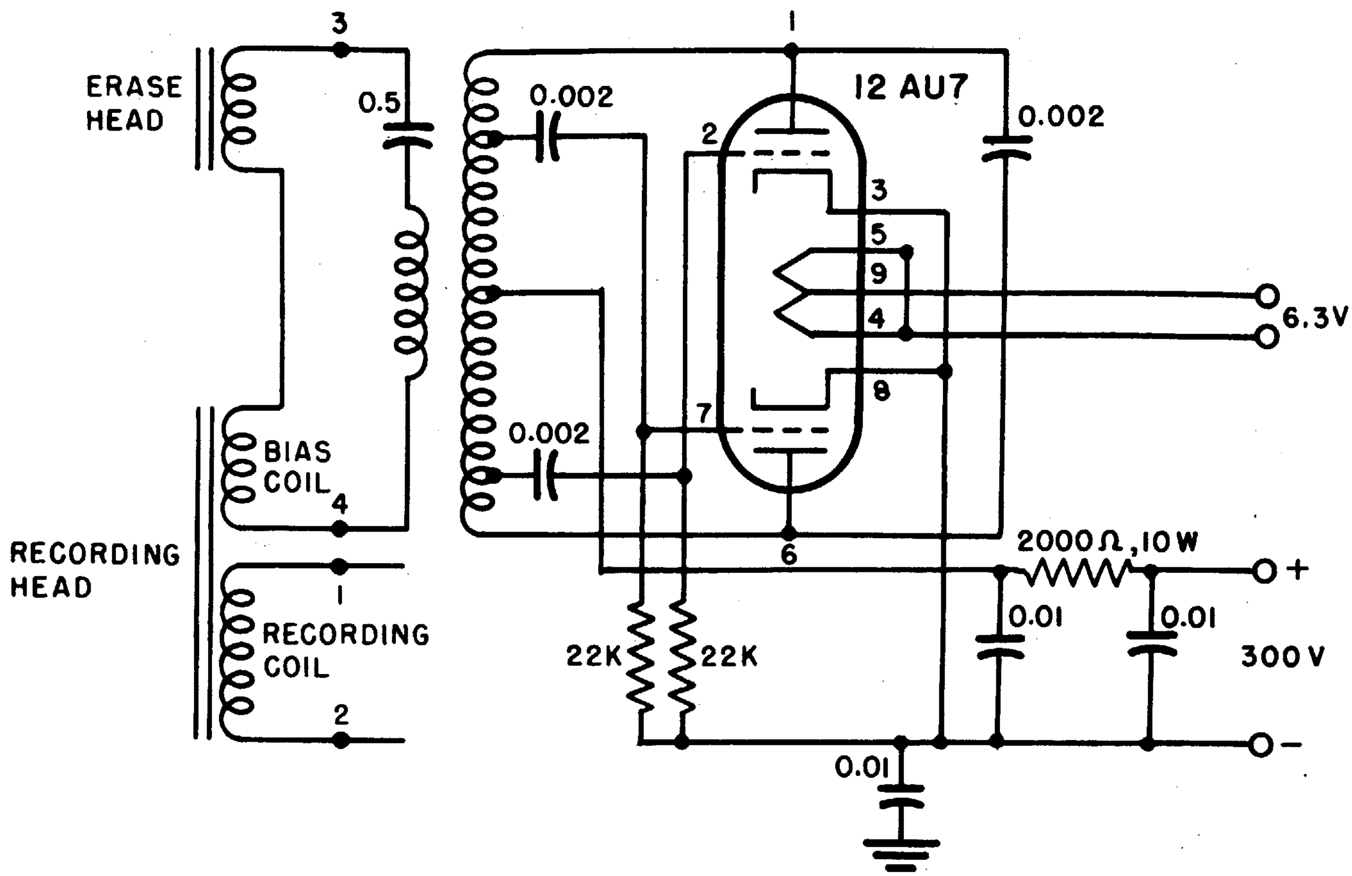
This very low capacitance prevents appreciable attentuation of the high frequencies when the potentiometer in the grid circuit is set around its mid-position, at which point there is effectively a high impedance source (approximately 125K since 500K potentiometers are used) feeding the grids. Thus the same shape of the frequency response curve is maintained regardless of the position of the potentiometers.

The unequal resistors joining the cathodes of V_8 and V_9 serve to make the controls "Magnitude of Echoes" and "Magnitude of Original Sound" contribute equal amounts to the overall output when they are in the same position.

B. Actual Bias-Erase, Recording, and Playback Circuits

1. Bias-Erase Circuit

It was found experimentally that at a tape speed of 69 inches per second, a bias-erase frequency of 54 kc gave satisfactory results: low distortion, and low enough noise on the tape. There is nothing critical about the frequency of 54 kc; the only requirement is that the frequency must be about this high to prevent the production of audible heterodynes with the highest audio frequency recorded, and 54 kc is the frequency which is recommended by the manufacturer of the heads. Fig. 9 shows the schematic diagram of the bias-erase oscillator used. Since the recording coil and the bias coil in the recording head must both produce magnetic flux in the same iron, there is tight coupling between the coils. Because of this coupling, it is necessary to insert the 0.5 microfared capacitor in series with the bias coil, so that the impedance (very low at audio frequencies) of the erase head and the coil in the oscillator will not effectively short out the bias coil at audio frequencies, and thereby greatly load up the recording circuit. On the actual machine, this oscillator is in



BIAS-ERASE OSCILLATOR

FIG. 9

the black box directly under the heads of the high-speed tape loop, (page 12) and receives its power on a cable from the power supply.

Curves were run to determine output at different frequencies, as a function of the bias —erase current. (Fig. 10). As these curves show, the output increases with the bias current. If the curves were taken further, a peak would be reached. However, the noise on the tape was also found to increase as the bias current was increased, and so there is no reason for increasing the bias current above the point where the tape noise is slightly greater than the noise in the playback amplifier, for above this point

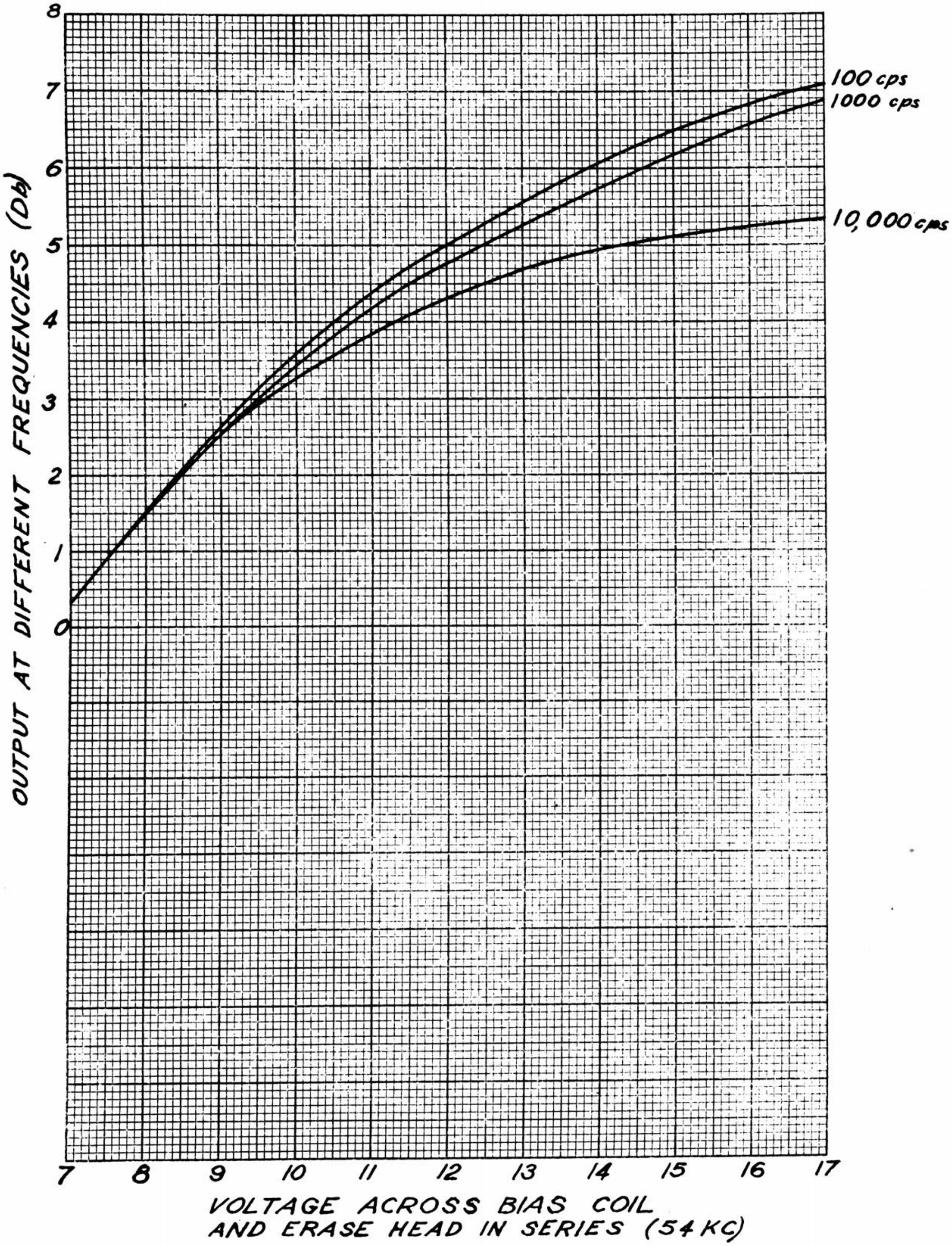

8
7
6
5
4
3
2
1
0
OUTPUT AT DIFFERENT FREQUENCIES (Db)
100 cps
1000 cps
10,000 cps
7
8
9
10
11
12
13
14
15
16
17
VOLTAGE ACROSS BIAS COIL
AND ERASE HEAD IN SERIES (54 KC)

Fig. 10

the signal-to-noise ratio is determined by the tape, and is not limited by the amplifier noise. A disadvantage of high bias current is that the high frequencies are not increased as much as the lower frequencies, as is shown by the 10,000 cps curve in Fig. 10. A voltage of 13 volts across the erase head and the bias coil in the recording head in series was used because it produces a current high enough to raise the tape noise above the playback amplifier noise, thus preventing the amplifier noise from limiting the signal-to-noise ratio, and still does not heat the heads and tape too much.

2. <u>Recording and Playback Amplifier</u>

a. <u>Frequency Response Characteristics Desirable</u>

It was decided to make the frequency response of the recording-playback system as flat as practical. Although no actual walls reflect all frequencies the same, additional filters can be added externally at any time to simulate any particular reflecting material, and a flat-response system provides a good reference point. Also, by having a flat-response system, all frequencies will be fed back with the same amplitude, and thus a greater total amount of audio power can be fed back before the "Fed-Back Echo Amplitude" control is turned high enough for the system to break into oscillation.

b. <u>Tape-Head Characteristics at Tape Speed of 69 Inches per Second</u>

Fig. 11 shows measurements of the tape-head frequency

Log Scale 20 to 20000 on long side × 10 to the half inch, 5th lines accented.
MADE IN U. S. A.

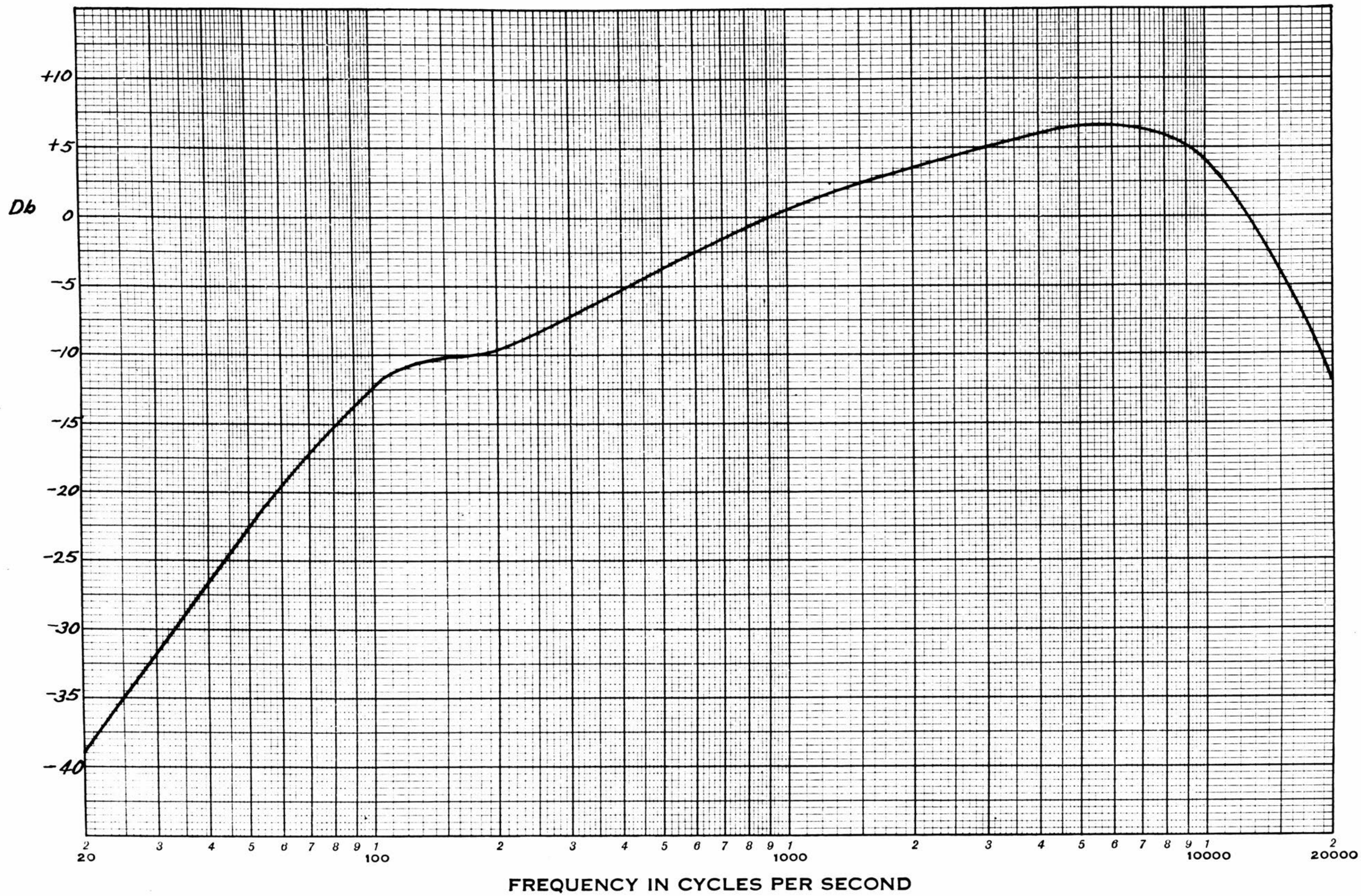


Fig. 11. TAPE-HEAD (MAGNECORDER) CHARACTERISTICS AT 69 IN. PER SEC.

response characteristics at a tape speed of 69 inches per second, using the Magnecorder heads and Minnesota Company tape. Similar data were obtained for other brands of tape. The curve of Fig. 11 was obtained by maintaining constant current in the recording head at all frequencies, and feeding the pickup head into a playback amplifier for which the frequency response curve was also run by inserting a small (0.1 ohm) resistor in series with the playback head with the tape not moving. Then by combining the response curve of the playback amplifier with the output curve, the tape-head characteristics curve was obtained. Thus this curve gives the response which would be obtained by maintaining constant current in the recording head, and using a playback amplifier whose frequency response is flat.

From about 200 cps to 3000 cps, this curve rises at the rate of 6 db per octave. This rise is due to the fact that the voltage induced in the playback head is proportional to the time rate of change of flux in it, and hence if the amplitude of the sinusoidal swings of flux is held constant and the frequency doubled, then the rate of change of flux is doubled, corresponding to 6 db. It was expected that this rise should continue well beyond 20,000 cps, up to a frequency at which the effective gap length in the heads (primarily the playback head) becomes comparable to the wavelength being recorded. At 13,800 cps, the wavelength recorded is 5 mils, with tape speed of 69 inches per second. The effective gap length of these heads is approximately 1 mil. However, the peak in the response curve occurs at about 7000 cps and then drops off at higher frequencies. But this response curve runs high enough for the audio range, with equalization, and so no further work was done to try to boost its high end. From about 100 cps down, the curve drops very rapidly at this high tape speed, because the wavelengths being recorded at the low frequencies are so long at this high tape speed that only a small portion of a cycle on the tape is in contact with the poles of the heads. There is a slight flattening out of the curve between 100 cps and 200 cps.

c. Requirements and Design of Amplifiers

As the above curve indicates, considerable boosting is required at the low frequency end. As a starting point for the amplifier, the recording amplifier was built to put the same current into the recording head at all frequencies, and the playback amplifier was given considerable boost at the low frequency end by means of the feedback loop in the grid circuit of V_4. The original circuit as worked out was as shown in the schematic in Fig. 7, (page 18), except that the following parts were not in the original circuit:

1. The series L-C circuit in the cathode circuit of V_2, and the R-C feedback link from the plate of V_3 to the cathode of V_2.
2. The R-C branch in parallel with the plate load resistor of V_2.

3. The parallel R-L-C circuit in the cathode of V_5. The 1.5K cathode resistor went straight to ground.

4. The R-C low-pass circuit, and the series L-C circuit in the plate circuit of V_5.

All cathode resistors except that of V_4 were left unbypassed. A meter was put across the transformer feeding the recording head, to indicate the proper recording level. Because of the coupling between the bias coil and the recording coil in the recording head, considerable voltage at the bias frequency of 54 kc was induced in the recording coil. This signal was many times greater than the recording signal. To prevent it from masking the recording voltage on the meter, the series-resonate filter consisting of the 1 mh coil and the 0.01 capacitor was put across the transformer. Both the recording and the playback amplifier had considerable reserve gain for subsequent equalization. The response curves of the original recording and playback amplifiers are shown in Figs. 12 and 13, and the response curve of the overall recording-playback system using these amplifiers is shown in Fig. 14. The response curve of the recording amplifier was run by feeding from an oscillator into the receptacle, "FedBack Echoes In", and measuring the current through the recording head by measuring the voltage across the 560 ohm resistor in series with the head. Approximately 1 milliampere was put through the head. The response curve of the playback amplifier was run by inserting 0.1 ohm in series with the playback head, and

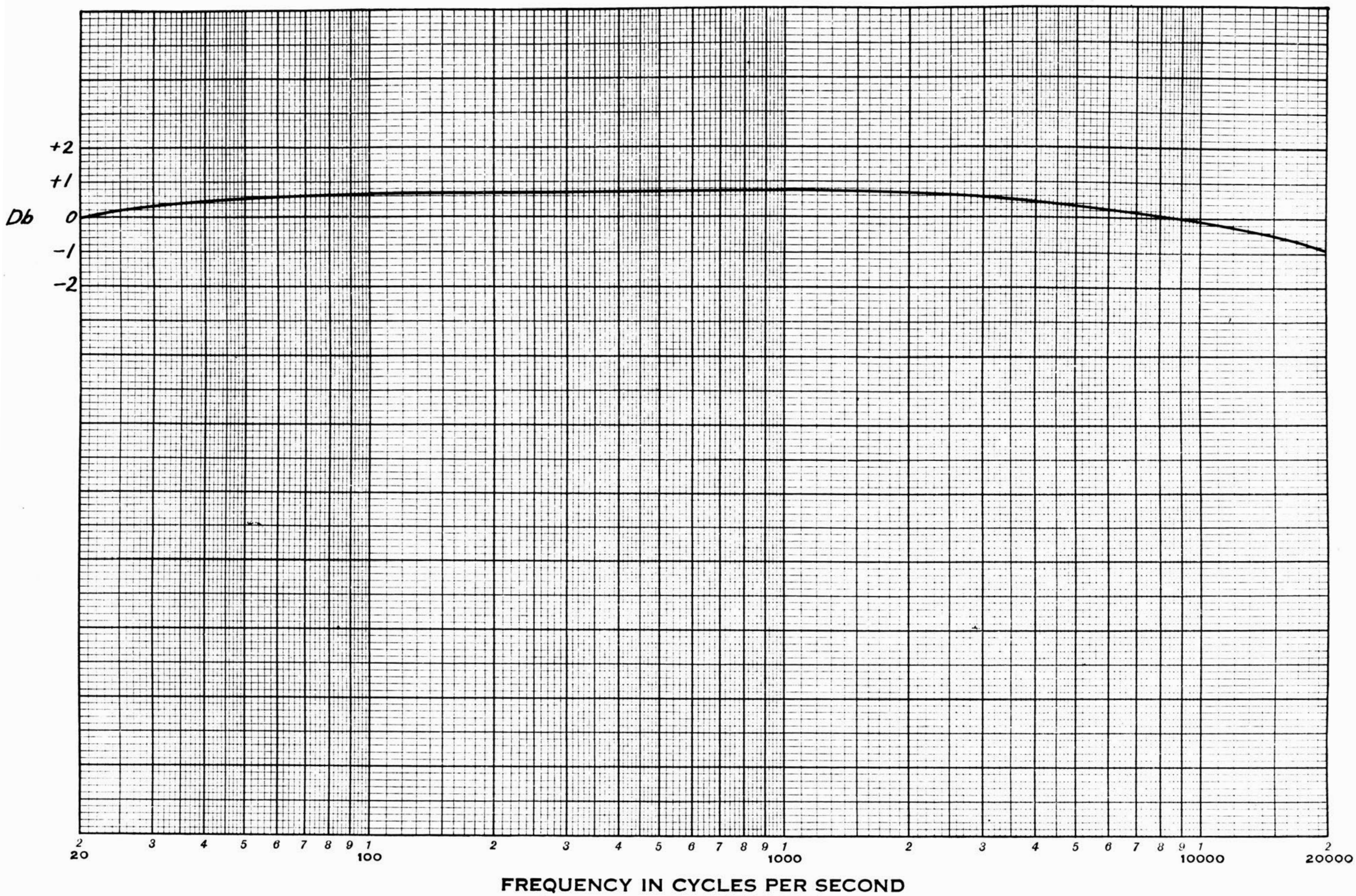


Fig. 12. ORIGINAL RECORDING AMPLIFIER RESPONSE

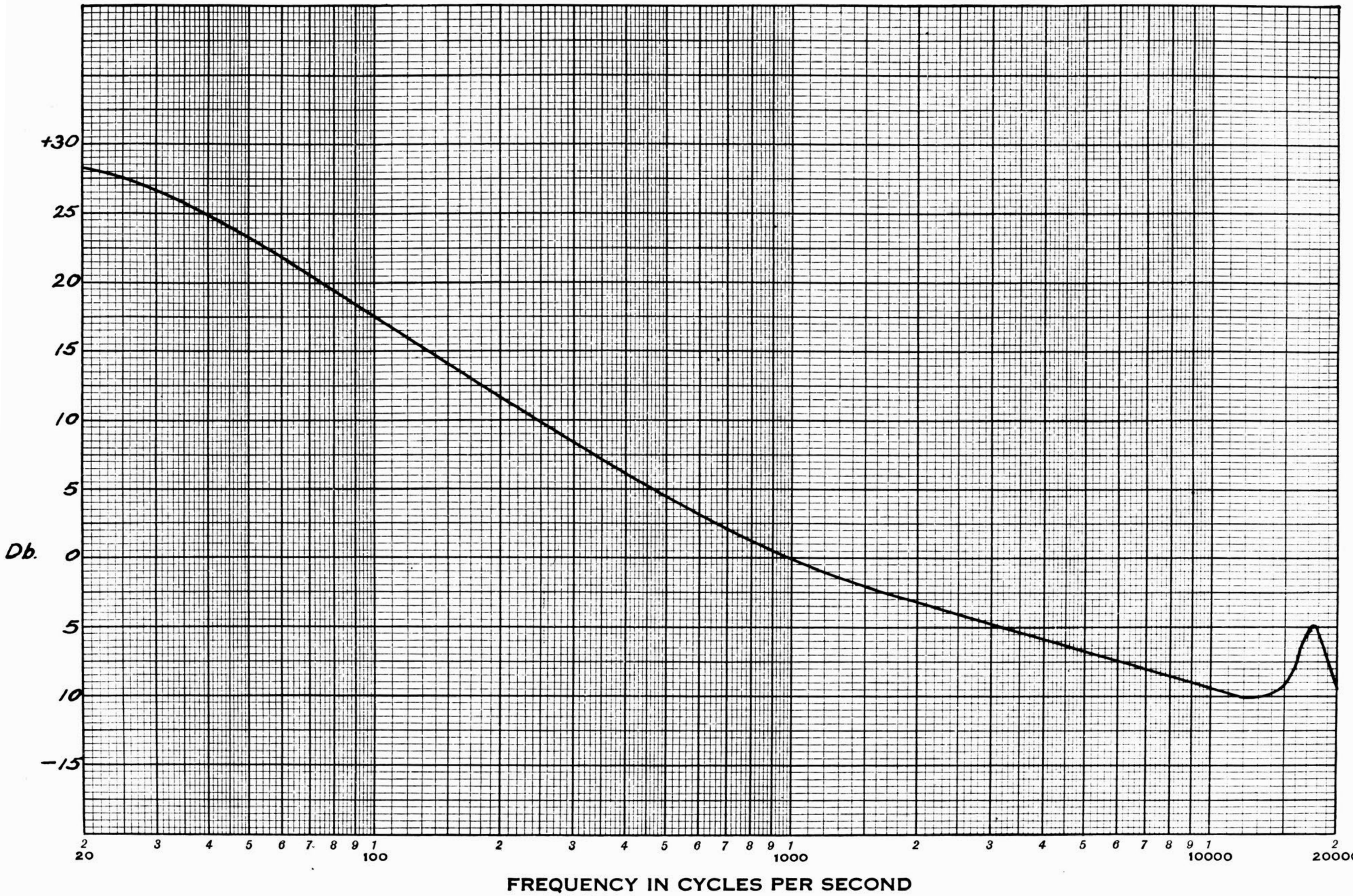


Fig. 13. ORIGINAL PLAYBACK AMPLIFIER RESPONSE

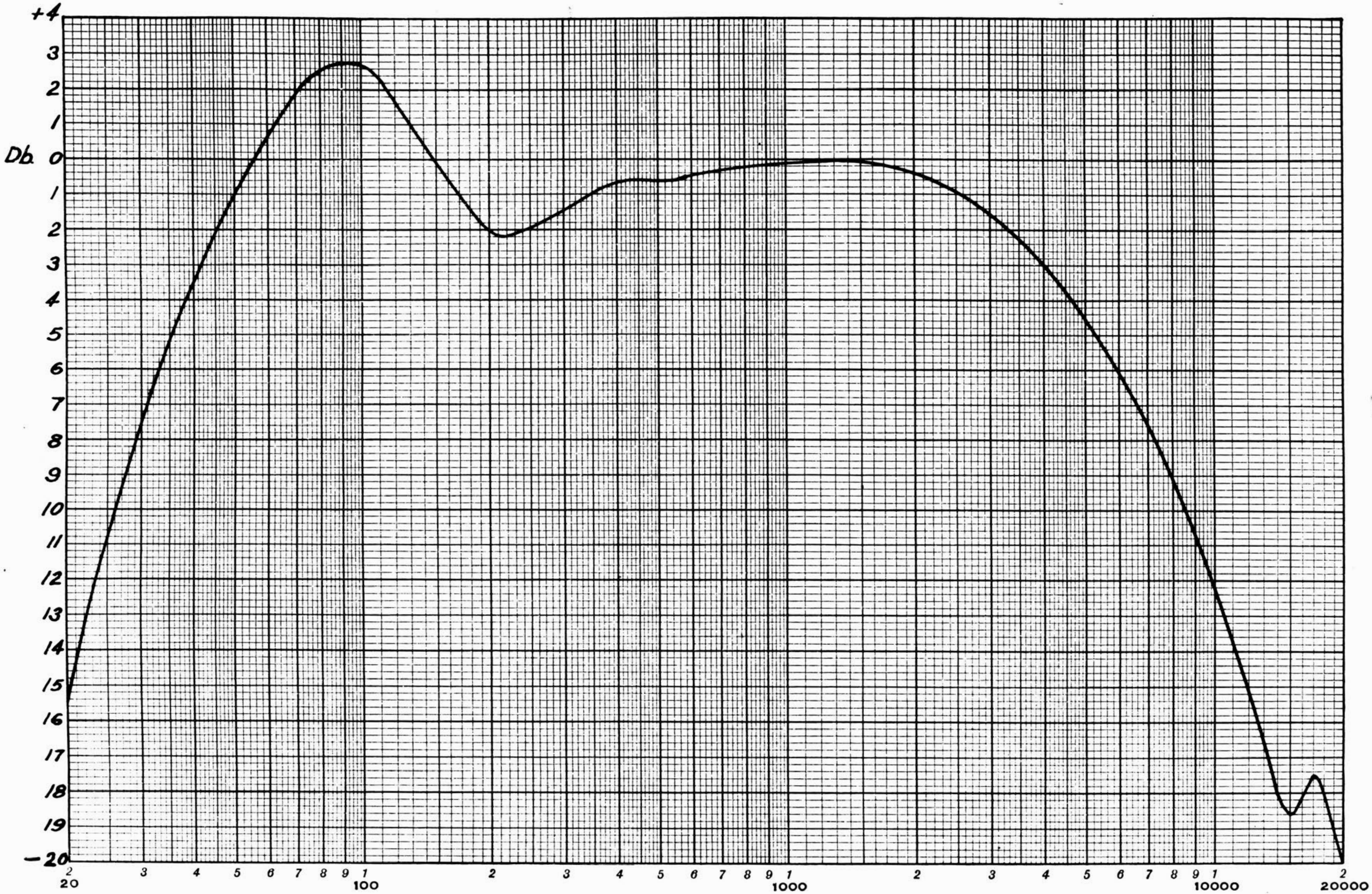


Fig. 14. ORIGINAL OVERALL RESPONSE

producing a voltage across this resistor of the magnitude which gave about the same output as the tape did at 1000 cps. The output was measured at the "Fed-Back Echoes Out" receptacle since the frequency response in the feedback loop is most critical. The frequency response would be practically the same measured at the Overall Output receptacle. The response curve of the overall recording-playback system was run by feeding into the recording amplifier at "Fed-Back Echoes In" and measuring the output at "Fed-Back Echoes Out" on the playback amplifier.

In the overall recording-playback response curve, (page 30) the peak at 100 cps is caused by the flattening out of the tape-head response curve (page 24) between 200 cps and 100 cps, and the steep bass boost curve in the playback amplifier. The drop below 100 cps is caused by the great drop in the tape-head response curve. Steps were taken to remove this 100-cps peak. First, a twin-T R-C network to attenuate 100 cps was inserted between V_4 and V_5, but this pulled down the response below 100 cps too greatly. Finally, the R-L-C parallel circuit was inserted in the cathode circuit of V_5. This parallel circuit resonates, that is, becomes a high impedance at 100 cps, thereby providing degenerative feedback and knocking down the 100-cps peak in the response curve. The fact that this circuit provides feedback makes it sufficiently sharp. In fact, the 3.3K was added to keep it from being too sharp. Then the response curve dropped down from about zero db at 100 cps, as the frequencies were lowered. Therefore, the R-C low-pass filter was put in the plate circuit of V_5. This circuit was designed as follows. See Fig. 15.

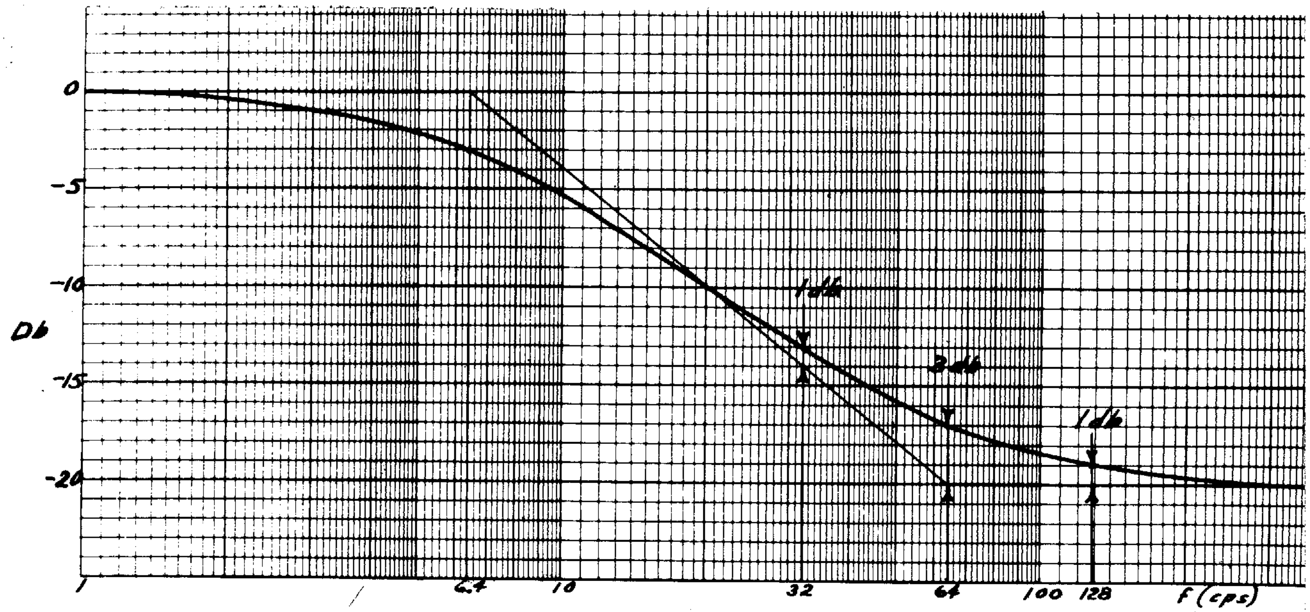


Fig. 15.

The "break" frequency, the frequency at which the asymptote to the network response curve begins to rise with a slope of -6 db per octave, is

$$f_1 = \frac{\omega}{2\pi} = \frac{1}{2\pi R_1 C}$$

$$= \frac{1}{2\pi \times 25 \times 10^3 \times 10^{-7}}$$

$$= 64 \text{ cps}$$

for the values of R_1 and C used. The network response curve itself is 3 db above the asymptotes at the "break" frequency, and 1 db above the asymptotes at twice and half the "break" frequency. The asymptote rises with -6 db per octave slope to the lower "break" frequency, which is

$$f_2 = f_1 \frac{R_1}{R_1 + R_2}$$
$$= 64 \cdot \frac{25}{25+220}$$
$$\doteq 6.4 \text{ cps}$$

for the value of R_2 used. These values of components are reasonable. Of course, the R-C circuit actually attenuates all the frequencies except the low frequencies, but the amplifier was designed with sufficient reserve gain for just such purposes. With the R-L-C circuit in the cathode of V_5 and the R-C circuit in the plate of V_5, the low frequency end of the response curve was as shown in Fig. 16.

The L-C circuit in the plate circuit of V_5 is series-resonant at 54 kc, and serves to reject the bias-erase signal which gets induced in the pickup head, from the tape and by direct coupling from the erase head and the bias coil on the recording head.

Next, steps were taken to raise the high-frequency response. In order to obtain a good signal-to-noise ratio, it was decided to boost the high frequencies in the recording amplifier rather than the playback amplifier. For this purpose, the feedback loop from the plate of V_3 to the cathode of V_2, along with the series-resonant L-C circuit in the cathode of V_2, was used. The feedback loop provides negative feedback except at the high frequencies near the resonant point of the

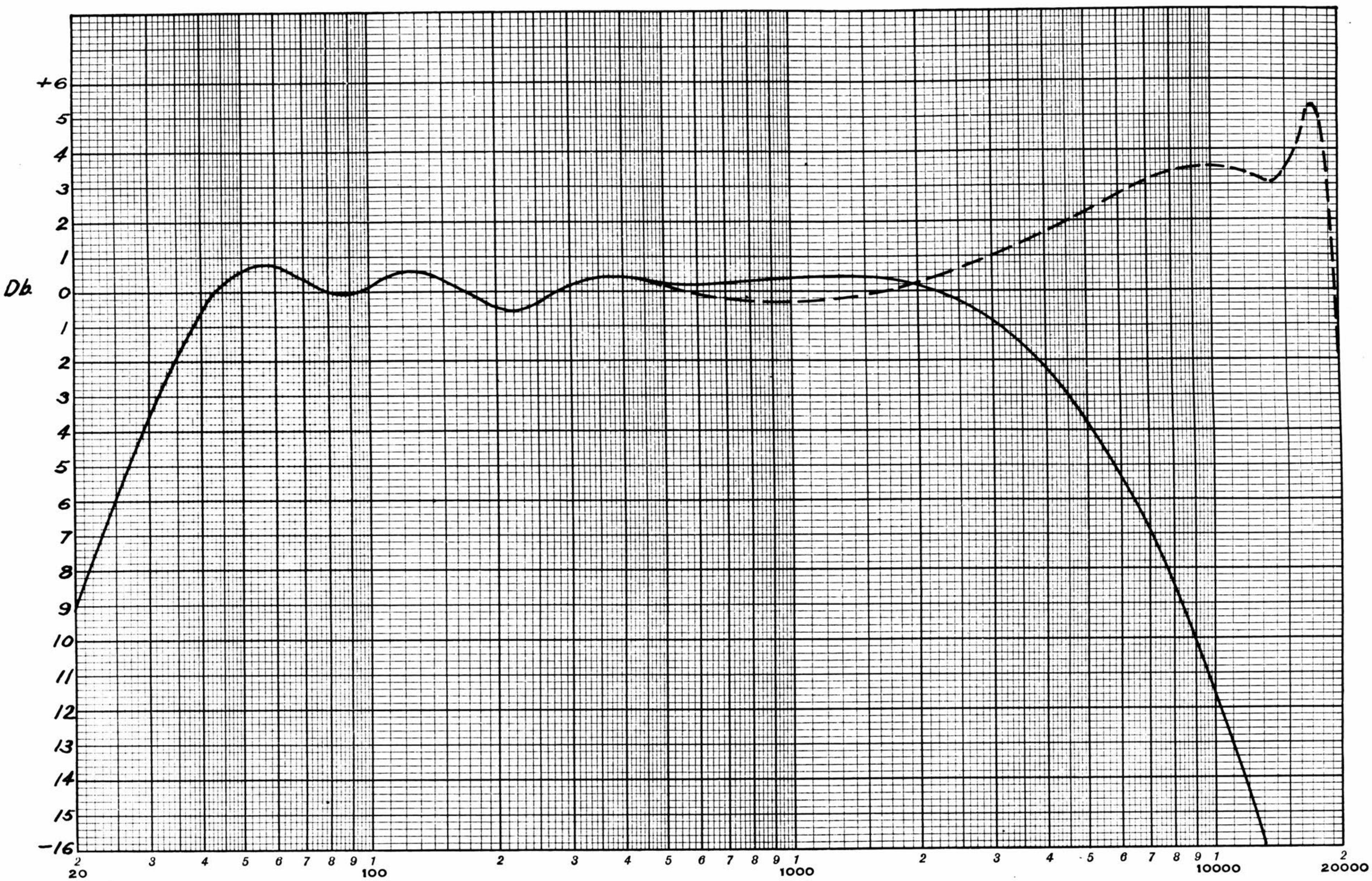


Fig. 16. STEPS IN FLATTENING OVERALL RESPONSE

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L-C circuit, where the L-C circuit shorts out the fed-back signal to ground. The L-C circuit resonates at the frequency

$$f = \frac{1}{2\pi\sqrt{LC}}$$

$$= \frac{1}{2\pi\sqrt{3\times10^{-3}\times2\times10^{-8}}}$$

$$= 20{,}500 \text{ cps.}$$

With this circuit inserted, the high frequencies rise as shown by the dashed curve in Fig. 16. To pull this curve down some, the R-C branch was put in parallel with the plate load resistor of V_2, thereby decreasing the gain of this stage at the high frequencies. The recording process took this boost in current at the high frequency end without appreciable distortion. The final response curve, flat within ± 1 db from 35 to 10,000 cps, is shown in Fig. 17. The frequencies above 10,000 cps could be boosted by by-passing the cathode resistors of V_5 and V_6 with small capacitors, but this would increase the noise in the playback amplifier.

Response curves of the recording amplifier and the playback amplifiers were then run again, and are shown in Fig. 18 and Fig. 19.

The wide use of negative feedback in the amplifiers contributes toward stability of gain, and thus helps to prevent the system from drifting into oscillation when the "Fed-Back Echo Amplitude" is set close to unity feedback. The use of

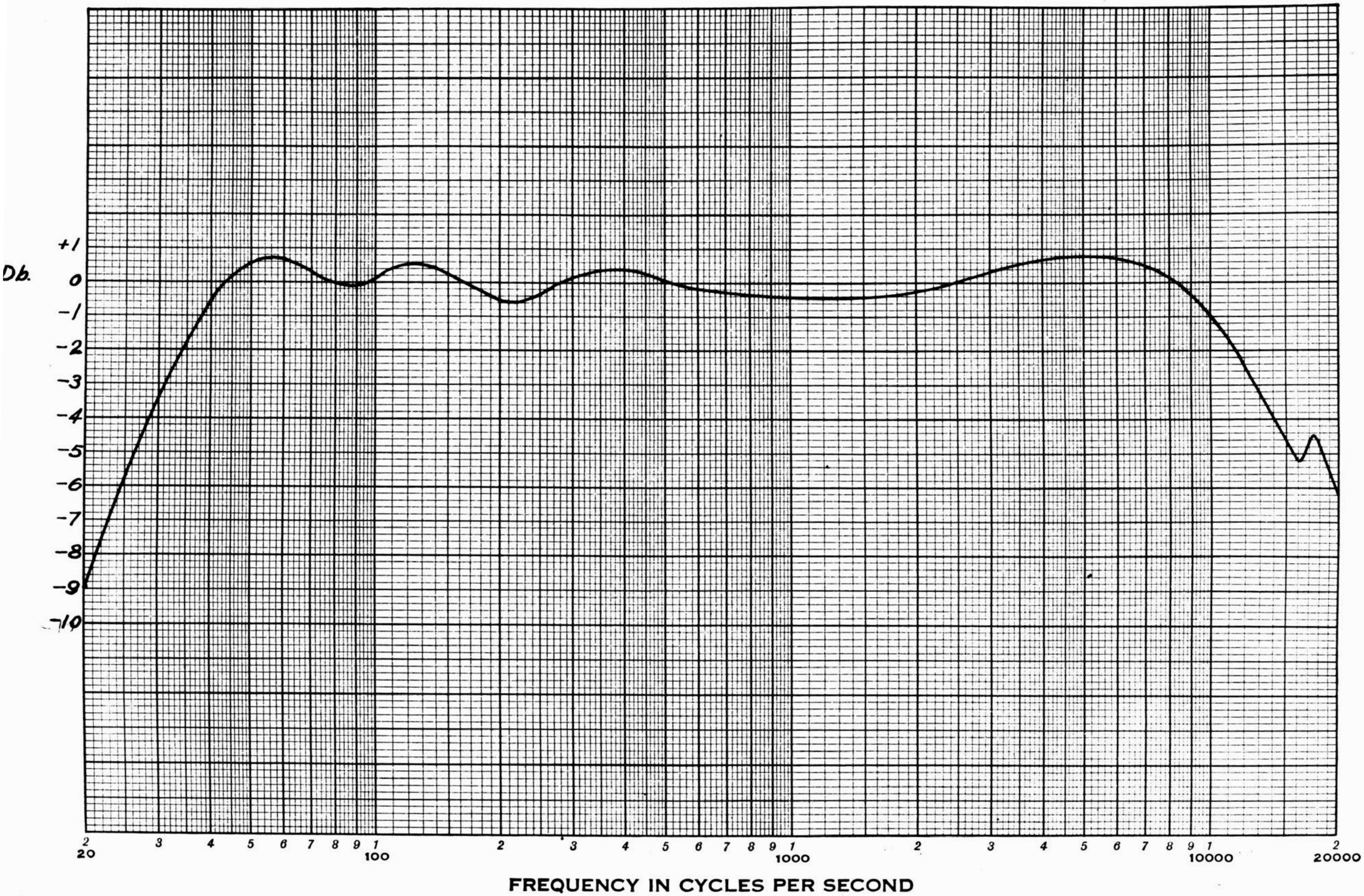

Fig.17. FINAL OVERALL RECORDING-PLAYBACK RESPONSE

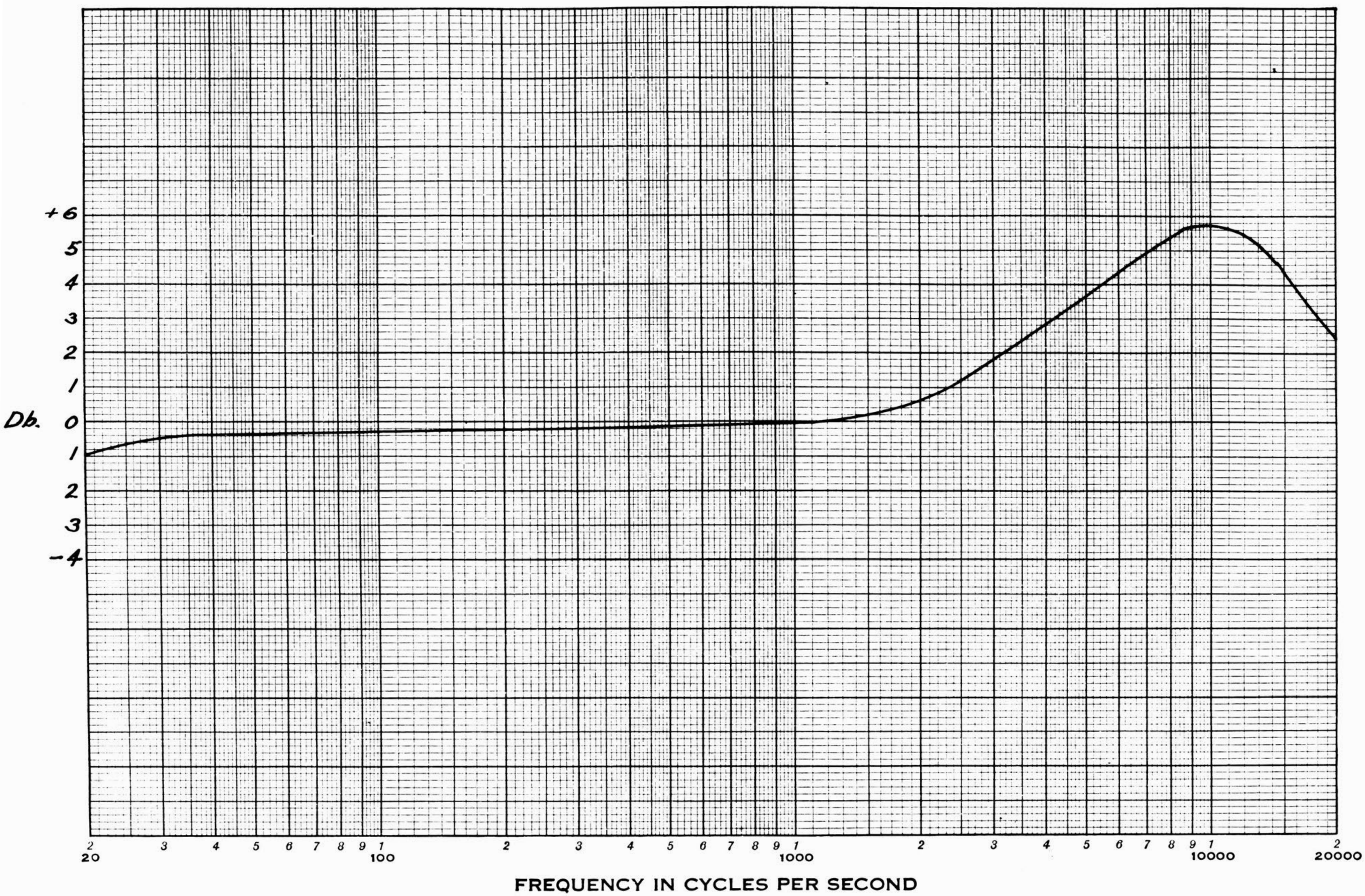

Fig. 18. FINAL RESPONSE OF RECORDING AMPLIFIER

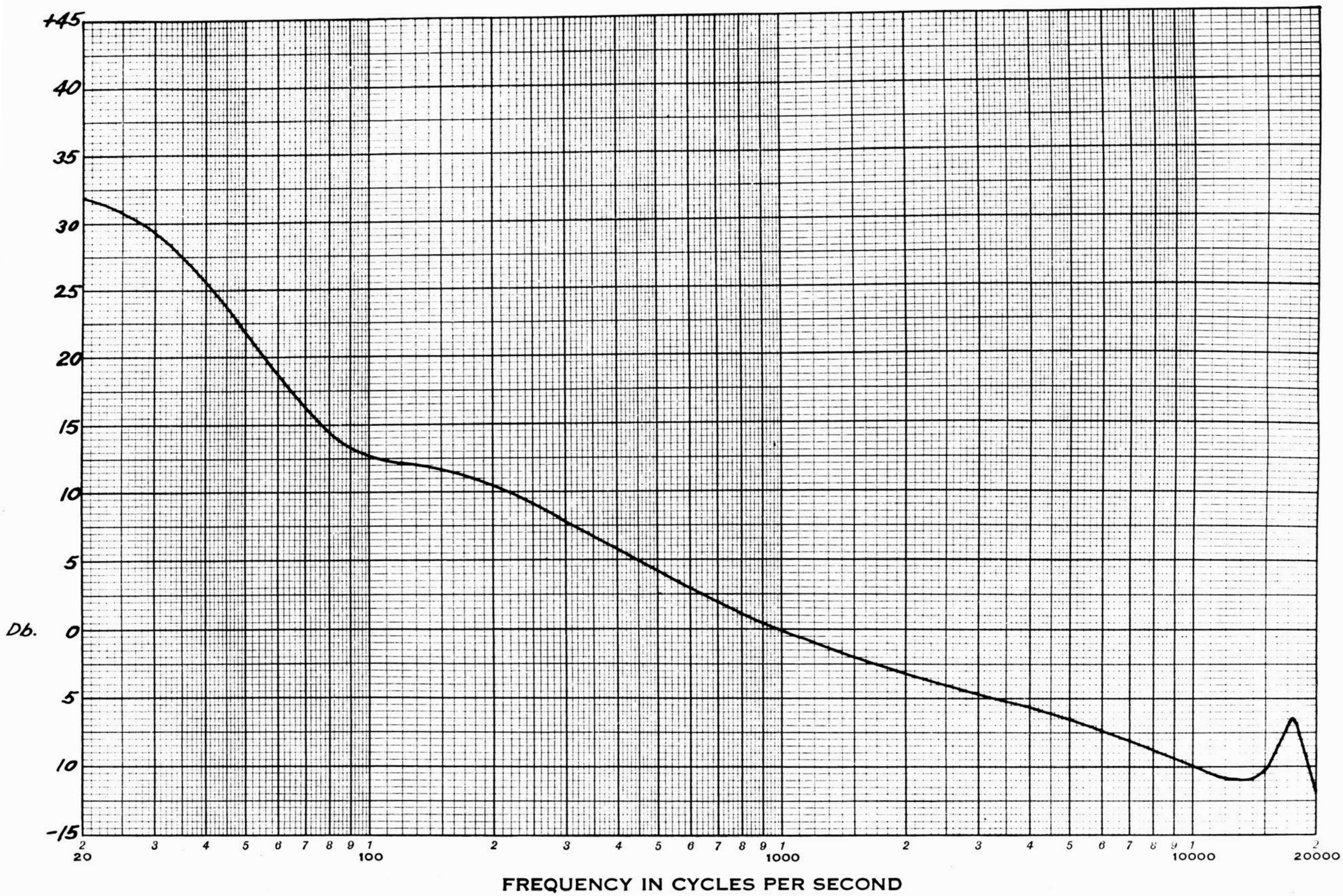

Fig. 19. FINAL RESPONSE OF PLAYBACK AMPLIFIER

a regulated power supply also helps in this respect.

3. Signal-to-Noise Ratio

It was feared that at the high tape speed, the inhomogeneity of the magnetic material on the tape would produce great noise, since the voltage induced in the pickup head is proportional to the rate of change of magnetic flux, and the rate of change of flux is proportional to tape speed for any given magnetic pattern on the tape. The high speed does not increase the output of the signal, because the recorded pattern due to the signal is spread out in proportion to the speed. However, a signal-to-noise ratio of 42 db was obtained; that is, the entire noise of the recording-playback system is 42 db below the output at which approximately 2% harmonic distortion occurs. The noise was measured in this test with a meter which covers the range from 10 cps to 150 kc, and hence measured noise outside the audio range. Since there may be a considerable portion of the noise above the audio range, due to the high tape speed raising the frequencies of the noise caused by inhomogeneity in the tape, then the signal-to-audible noise ratio may be even better than 42 db.

The use of a.c. bias on the tape kept the tape noise lower than if d.c. bias had been used. Direct current was used on the tube heaters in both the recording and playback amplifiers to reduce the noise in the amplifiers. As stated under the section on the bias-erase oscillator, the bias current was increased up to the point at which the tape noise was above the amplifier noise, in order to increase the signal output, thereby obtaining the highest signal-to-noise ratio. The fact that the high end of the audio frequency spectrum was boosted in the recording amplifier instead of the playback amplifier also contributed toward a high signal-to-noise ratio.

The input stage in the playback amplifier was shock-mounted in soft rubber in order to reduce noise from microphonics. Of course, in a signal-to-noise ratio test, microphonic noises are not included.

V. FURTHER DISCUSSION OF FEEDBACK PHENOMENA IN THE SYSTEM

An explanation of the operation of the system has been given in section IV, under part A, Block Diagram. Pulse echoes were used in this explanation. When pulse echoes are used (of duration less than 1/55 second) in the device, the original sound has ceased before the first echo is feedback, and so there is no combining of the fedback echo with the original sound (within the feedback part of the system). But in actual sounds, the original sound is still going on at the same time at which the picked-up sound is fed back. Due to the variation in phase shift of different frequencies through the entire recording-playback system, some frequencies are fed back directly in phase with the original sound, thereby reinforcing these frequencies, and other frequencies are fed back directly out of phase with the original sound, thereby decreasing these frequencies. At other frequencies, at which the phase shift lies between 0° and 180°, the reinforcing or decreasing will lie between the extremes. The spacing between the recording head and the pickup head accounts for a rapid change in phase shift as the frequency is changed. There is some change in phase shift through the recording and playback amplifier as the frequency is changed. But just due to the 1/55 second time lapse between the recording head, at 55 cps and every multiple of 55 cps, the voltage induced in the pickup head will be right in phase with the current in the recording head; and at $27\frac{1}{2}$ cps, and at $27\frac{1}{2}$ cps plus any multiple of 55 cps, the voltage induced in the pickup head will be directly out of phase with the current in the recording head. Thus every $27\frac{1}{2}$ cps, the phase shift of the induced voltage changes 180°.

A. Steady-State Frequency Response of the Feedback Part of the System

For a steady-state sinusoidal input signal the block diagram of the feedback part of the system is as shown in Fig. 20. In this case, the time delay between heads is important only in the way it produces phase shift. In Fig. 20, e_{out} is only the portion of the overall out-

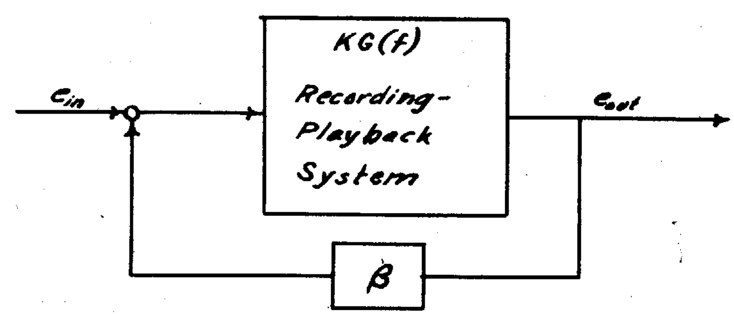

Fig. 20.

put which comes from the channel controlled by "Magnitude of Echoes". β introduces practically no phase shift since it is just the cathode follower V_7, and its magnitude is determined by the setting of the control "Fed-Back Echo Amplitude".

All the phase shift comes from KG(f), which is the transfer function of the recording-playback system. For a steady-state signal of a frequency at which the fed-back signal is directly in phase with the input signal, βKG(f) is real and positive, and the KG(f) must also be real and positive since β is always real and positive. If the control, "Fed-Back Echo Amplitude", is set at -6 db, then at a frequency at which the fed-back signal is in phase with the input, $\beta KG(f)$ is equal to $+\frac{1}{2}$. Then

$$\frac{e_{out}}{e_{in}} = \frac{KG(f)}{1-\beta KG(f)}$$

$$= \frac{KG(f)}{1-(+\frac{1}{2})}$$

$$= 2KG(f).$$

If the control, "Fed-Back Echo Amplitude", is set at -6 db, then at a frequency at which the fed-back signal is directly out of phase with the input, $\beta KG(f)$ is equal to $-\frac{1}{2}$.
Then

$$\frac{e_{out}}{e_{in}} = \frac{KG(f)}{1-(-\frac{1}{2})}$$

$$= \frac{2}{3}KG(f).$$

Therefore, as the audio frequency spectrum is swept through, approximately every $27\frac{1}{2}$ cps the response curve should change in level by 20 log $(2 \div 2/3)$, or 9.5 db, with the "Fed-Back Echo Amplitude" control set at -6 db. This statement depends upon KG(f) having constant magnitude, and it has been made approximately so, within ± 1 db, from 35 to 10,000 cps. The frequency response curve just discussed is shown in Fig. 21, as obtained from the actual equipment. It is plotted only from 100 cps to 1000 cps since above this range, the distance on the

half inch, 5th lines accented.
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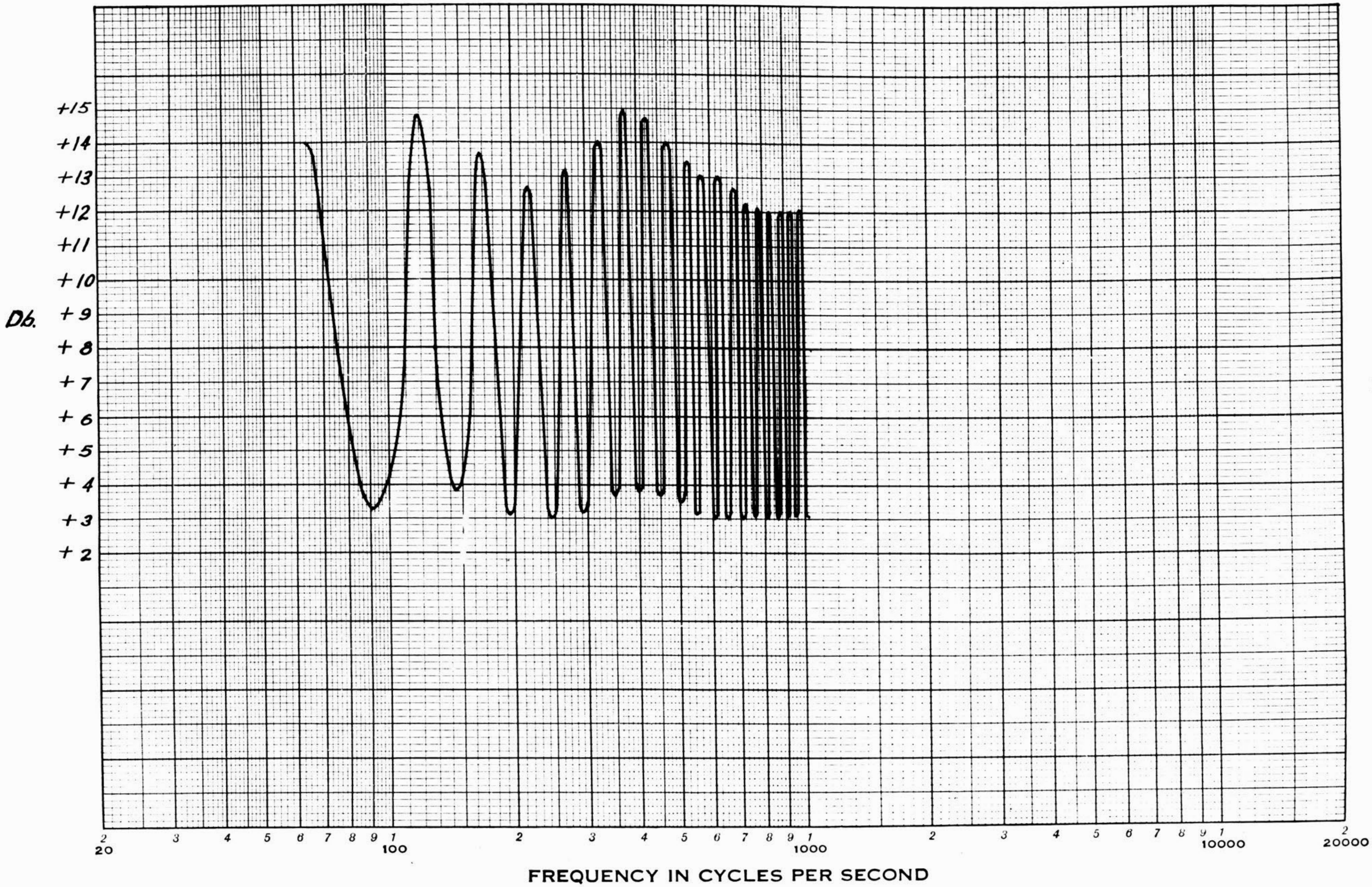

Fig. 21. OUTPUT OF ECHOES CHANNEL WITH FEEDBACK $(|\beta K G(f)| = \frac{1}{2})$

plot corresponding to $27\frac{1}{2}$ cps is very small. But the humps continue approximately every $27\frac{1}{2}$ cps. The irregularity of the humps is caused by the slight waves in the recording-playback frequency response curve, Fig. 17, p. 36. If β is increased by raising the control "Fed-Back Echo Amplitude" (toward 0 db), the humps will be more pronounced. And of course if $\beta KG(f)$ is made equal to 1 (by setting the control, "Fed-Back Echo Amplitude" at 0 db), the device will go into oscillation.

B. <u>Steady-State Frequency Response When the First Echo is Added to the Original Sound, (With No Feedback)</u>

For this case, the block diagram is shown in Fig. 22. The change

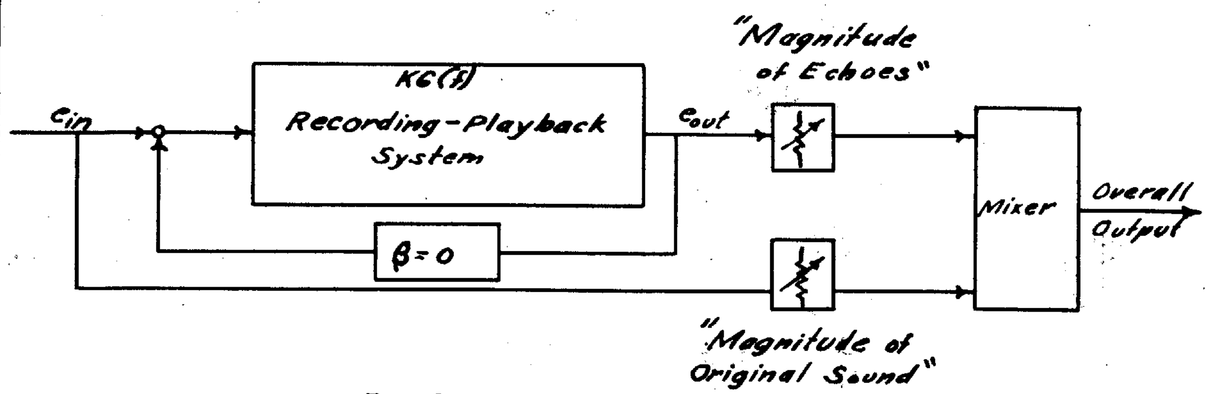

Fig. 22.

in phase shift through KG(f) as the frequency is changed is still the important factor. Frequencies at which the phase shift through the recording-playback system is zero will add directly to the input signal, and frequencies at which the phase shift is 180° will subtract directly from the input signal. If the "Magnitude of Echoes" is set at half (6 db lower) the value of "Magnitude of Original Sound", then at some frequencies (in phase) the overall output will be $1\frac{1}{2}$ times (or 3.52 db above) what it would be with just the original sound, and at other frequencies (out of phase), the overall output will be $\frac{1}{2}$ (or 6 db below) what it would be with just the original sound. The curve for this condition is shown in Fig. 23. Again here, the humps occur approximately every 55 cps, and the irregularity in the humps is caused by the waves in the recording-playback response curve, Fig. 17, p. 36.

If the output of the echoes channel with feedback (Fig. 21) is combined with the original input, a curve similar to Figs. 21 and 23 would be obtained, except that the humps would be even more exaggerated, because at the frequencies at which the fed-back signal is directly in phase with the input, the output of the echoes channel would be greatest, and at these same frequencies the output of the echoes channel would add directly to the input signal. This is the condition existing in the device when the control "Fed-Back Echo Amplitude" is turned up to give feedback, and both the controls "Magnitude of Echoes" and "Magnitude of Original Sound" are turned up.

Audio Frequency—Semi-Logarithmic
Log Scale 20 to 20000 on long side × 10 to the half inch, 5th lines accented.
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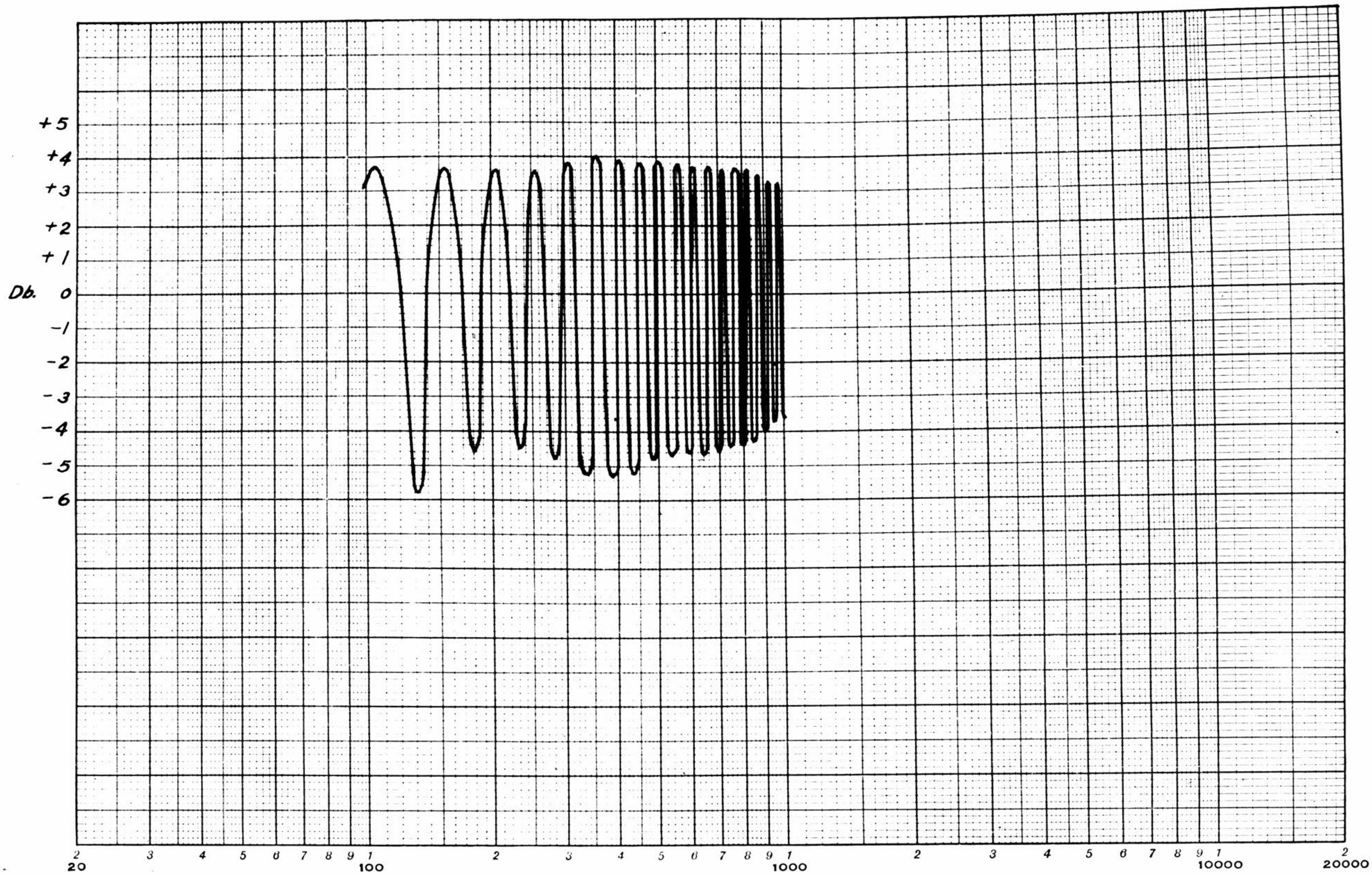

Fig. 23. ORIGINAL INPUT ADDED TO FIRST ECHO (NO FEEDBACK)

VI. LISTENING TEST AND SUGGESTIONS FOR FURTHER WORK

When music or speech is fed into the device, an effect similar to reverberation is produced. However, the reverberation sounds unnatural -- more like a "flutter" echo. Since music and speech contain many frequencies, some of these frequencies are reinforced and others are decreased by the device, as explained in the preceding section. As a certain frequency is being either reinforced or decreased, however, the steady-state condition is approached in steps occuring every 1/55 second due to the time delay in the recording-playback process. It is believed that this building up or decreasing of different frequencies, occurring in 1/55 second steps accounts for the "flutter" sound in the reverberation. The effect may be similar to that obtained in a one-dimension room, or between two hard parallel walls.

It may be that additional pickup heads spaced along the tape loop would improve the reverberation, since they would give the effect of echoes coming from more than one point in a room. Also, some frequencies being reinforced by one head would be decreased by other heads. One additional head was tried on the device, but no definite conclusions were reached.

The use of the machine for studies in acoustics has many possibilities. This is an unlimited problem in itself, beyond the scope of this thesis.

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