Design of a Noise Eliminator System*

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A system of compression before recording and expansion upon playback has been developed to extend the dynamic range of a studio tape machine up to 110 dB. Accurate reconstruction of the signal independent of the tape recorder gain results from the use of a wideband cube root compression characteristic in conjunction with high- and low-frequency preemphasis. Three variations of the preemphasis-compression characteristic are optimized for 15, 7½ in/s, and the production of consumer tapes, records, and FM programs. While intended to be played through an expander, the consumer program material is also compatible with conventional equipment. System characteristics and design are discussed.

INTRODUCTION: The dynamic range capability of a studio tape machine can be greatly extended if the signal is suitably processed before and after recording. This paper will describe a new complementary signal processing system called the noise eliminator[1] which extends the dynamic range of a studio tape machine to 110 dB [1]. The system is also applicable to other noisy media such as FM, prerecorded tapes, records, or a microwave link. Consumer program material made through the record signal processor and played through a simplified version of the playback processor could produce a 35-dB or more improvement in dynamic range and at the same time be compatible with existing equipment.

Because tape machines use different amounts of high-frequency preemphasis for different tape speeds, an optimum system for raising the level of the recorded signal to overcome the tape noise must deliver less high-frequency content to the tape machine at the slower tape speeds. For studio mastering completely independent channels are desirable, but to produce compatible consumer program material stereo channels should have the same instantaneous gain so as to preserve the directional effects. The design of the noise eliminator system therefore includes three separate characteristics. Characteristic *A* is optimized for studio recording at 15 in/s and provides 110-dB dynamic range. Characteristic *B* is optimized for studio recording at 7½ in/s and provides a dynamic range of 102 dB. Characteristic *C* is optimized for stereo tape recording at 3¾ or 1⅞ in/s, for FM broadcasting, for records, and for background music service; and two or four channels are ganged together.

The system to be described utilizes high- and low-frequency preemphasis and a single wideband cube root compressor to produce the recorded signal, and a complementary expander and deemphasis for playback. In the single-band system the frequency response is constant and unaffected by inaccuracy in the tape machine. The choice

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[1] Patent pending.

of compression characteristic and the extreme dynamic range over which it applies eliminates the problem of expansion or compression occurring in the restored signal due to an error in the gain of the tape machine. As a result of the development of a two-quadrant multiplier/divider capable of controlling gain over a 60-dB range with 0.1-dB accuracy and the use of other precision components, a typical system accuracy of ±1 dB was achieved over the entire dynamic range along with frequency response flat within ± 0.2 dB from 20 Hz to 20 kHz.

It is to be noted that the noise eliminator system is a form of compandor, and it will not remove noise from an already noisy signal but will preserve its original dynamic range through the recording process.

DESIGN CONSIDERATIONS

The objective in the development of the noise eliminator system was to be able to record and reproduce live musical instruments and groups at their original acoustic levels with no audible tape noise whatsoever and without exceeding the 1% and preferably 0.5% distortion level of the tape. A number of experiments were made to determine what are the acceptable noise levels both during the absence and during the presence of music to meet this objective. Some experiments were also made to determine the peak signal requirement for the reproduction of a drum set.

These experiments revealed that the practice of recording with the standard vu meter occasionally reaching 0 to +3 vu usually results in peaks reaching far above the 1% distortion level of the tape which is typically at +6 vu for a studio machine equipped with a linearizer. Peaks from the tom-tom, cymbal, and cow bell reached as high as 24 dB above the reading of the standard vu meter when a capacitor microphone was placed at 4 in from the point of impact. Placing the microphone farther away reduced the peak to average level. In many positions the maximum peak occurred not at the initial impact but about 1 ms later as the vibrations propagated through the instrument. It was concluded that it is not always practical to preserve every single peak, but even so the vu meter readings on a 15-in/s studio machine cannot exceed −8 vu without causing excessive distortion on many types of program material. The resulting poor signal-to-noise ratio, however, prohibits this practice.

The listening tests revealed that for the noise to be negligible in the absence of program content it should be 90 dB or more below the 1% distortion level or below −84 vu. In the presence of a signal more noise is tolerable because the signal masks the noise in the ear. Audible tape noise, however, is predominantly high frequency, and it was found that noise 65 dB down could be plainly heard in the presence of a 500-Hz sine wave but could be just barely heard at frequencies above 3 kHz. Only by severely restricting the bandwidth to about one-half octave centered at 500 Hz could the tone be made to sound completely pure in headphones. For this reason a compandor system which divides the audio into separate frequency bands and controls the level of each independently, such as in the Dolby A-System [2], was strongly considered.

A multiband system, however, has the disadvantage that errors in the frequency response of the tape machine cause additional errors in the gains of the expanders used for playback with resultant further degradation of the frequency response. Inaccuracy in the gain of the tape machine also results in a frequency response error, but this problem can be solved by spreading the compression evenly over the entire dynamic range.

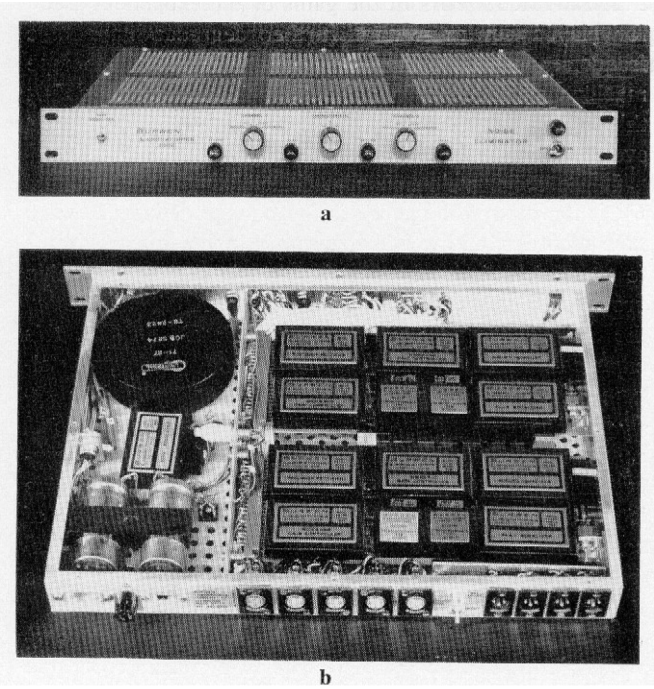
If the compression characteristic $E_T = E_{in}^{1/n}$, where E_{in} is the input voltage, E_T is the voltage delivered to the tape recorder, and n is any number in the range from 1 to a practical limit of 5, a gain error in the tape machine produces a constant system gain error. At the output of the expander, whose output voltage is $E_o = E_T^n$, this error can be eliminated by readjusting the playback volume. The gain error in dB is multiplied by n. For example, in a system where $n = 2$, a 1-dB change in level at the tape represents a 2-dB change in level at the system input or output, and a gain error in the tape machine is correspondingly doubled in dB. Note that the type of compression and expansion considered here is not instantaneous, but acts so slowly as to avoid distortion of a steady-state sine wave.

With today's technology in multipliers the most convenient compression characteristic is a square root type where $n = 2$ and a 90-dB input range is compressed to 45 dB at the tape. This type of system is perfectly quiet at no signal, but listening tests showed that when the signal level was small, there was too much noise in the presence of the signal. In order to overcome the noise it is necessary to hold the signal level on the tape as close to the maximum limit as practical at all times. On the other hand there must be some variation in the level on the tape which the expander can measure with sufficient accuracy to reconstruct the original levels. Otherwise a pilot tone would have to be used to operate the expander.

The next most convenient exponent to use is $n = 3$ where a 1-dB change in level at the tape corresponds to a 3-dB change in level at the system input or output. This system, which was finally adopted, compresses a 90-dB input dynamic range to 30 dB at the tape. Small signals are thus considerably above the tape noise level.

The problems of a multiband system could be further alleviated by combining the principle of a dynamic noise filter [3] with a single wideband compandor. The dynamic noise filter behaves as a low-level expander for the extreme high and low frequencies, and the expansion takes place in such a way as to produce an audible reduction in the noise but little or no effect on the program material. To make the system perfect, a complementary high- and low-frequency compression system could be added to the record system. Furthermore the signal-to-noise ratio could be improved by adding high-frequency preemphasis in the wideband compandor. This system appeared promising and some experiments were made. It was found that after adding preemphasis in the compandor, the added improvement due to the automatically variable filter was only 5 dB, and this improvement would be further diminished if a tape were used having greater high-frequency output and corresponding equalization to flat response during playback.

The headphone tests had indicated that it would take an impractical number of compandors operating in separate frequency bands to reduce the noise during the signal to complete inaudibility. A compromise was made and it was decided that if the system would perform well



a

b

Fig. 1. Two-channel noise eliminator.

in an *A–B* test on piano music in headphones, the noise during the signal would be low enough. Thus the single wideband cube root compandor system with 13-dB high-frequency preemphasis and 5.4-dB low-frequency preemphasis evolved.

In any such compandor system there is a problem of overshoot in the signal level, while the compressor is measuring the signal to determine the gain reduction required. A delay in the signal path ahead of the gain control element to avoid the overshoot does not solve the problem because the expander on the playback side will not be measuring the same signal and may not reproduce a transient at its original level. The overshoot must be recorded and reproduced faithfully by the tape recorder. Experiments made with percussive instruments indicated that distortion of the first millisecond of the signal was unnoticeable provided the high-frequency content was attenuated. In other words, high-frequency or slew rate limiting of the signal for 1 ms is permissible. This type of limiting set at three different levels for characteristics *A*, *B*, and *C* is used in the final design of the record signal processor.

The Dolby system attempts to solve the overshoot problem by adding a direct signal path to the compressed signal which is clipped during the overshoot. Large-signal transients pass directly, but there is still a range of medium signal levels for which the leading edge of a transient is distorted; so the problem is not really eliminated.

A secondary consideration in the development of the noise eliminator system was the production of compatible program material for consumers. Noise reduction in professional recording, however, was not to be sacrificed at all for this purpose, and so separate characteristics *A*, *B*, and *C*, two for professional use and one for the consumer material, were provided.

SYSTEM DIAGRAM

The basic system is illustrated in Figs. 1 and 2. Each channel consists of a plug-in circuit card containing seven epoxy modules. Channel 1 is used as the record processor and channel 2 in the same chassis as the play processor. The input signal passes through an active transformer which is a unity-gain differential input to single-ended output amplifier used to reject common mode line noise. The signal from the active transformer feeds the record equalizer which provides high- and low-frequency preemphasis. Next the signal passes through the gain control element which is connected as a divider. The divider provides a gain of 0 to 60 dB determined by the *Y*-input dc control voltage. This device, called the wideband gain controller, which is the heart of the system, can control the gain with an accuracy of 0.1 dB over this range, has low noise, and has a maximum harmonic distortion of 0.1%.

The output from the divider then feeds a high-frequency or slew-rate limiter in the record equalizer module. Limiting occurs about 2 dB below tape saturation at all frequencies. At 20 kHz the output capability is re-

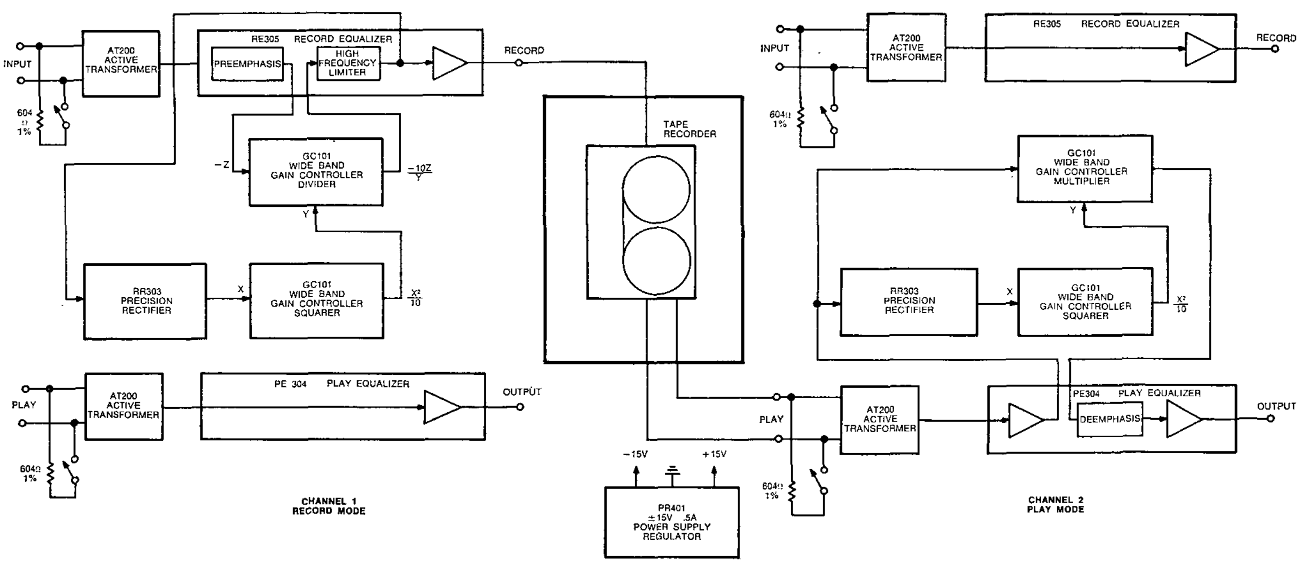


Fig. 2. System diagram.

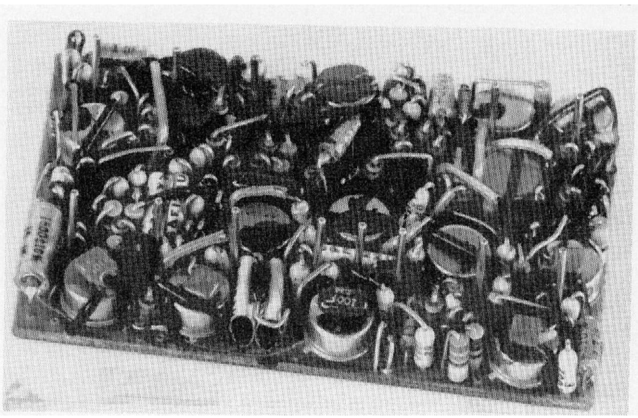
duced below this level to allow for the preemphasis in the tape machine. In characteristic A for 15 in/s the maximum output is down 5 dB, in characteristic B for 7½ in/s it is down 14 dB, and in characteristic C for consumer program material the maximum 20 kHz output is down 20 dB. After the limiter the output is adjusted to the proper level and fed to the record side of the tape recorder.

Control of the gain of the divider is accomplished by measuring the output of the high-frequency limiter in the precision rectifier module. The precision rectifier delivers a dc output voltage proportional to the audio input, and this is channeled to the X input of a squarer module. The output of the squarer is then proportional to the square of the record signal. This voltage becomes the control voltage for the divider. The record signal level is thereby controlled by a feedback loop which senses the actual record signal level and controls the gain such that the record voltage is proportional to the cube root of the input, and a 1-dB change in the record signal corresponds to a 3-dB change in the input signal. Because the preemphasis occurs ahead of the compressor, the tendency to limit at high frequencies is minimized. It was noted that the gain control element is able to vary its gain over a 60-dB range. This means that a 90-dB range of input signal will be compressed by 60 dB to produce a range of signal levels at the tape recorder of 30 dB. The control voltage is limited in the precision rectifier so that the divider gain cannot exceed 60 dB. Therefore extremely low-level signals are amplified in direct proportion to the input. At the maximum input signal the divider gain is not reduced to 0 dB, except for high-frequency signal inputs where the preemphasis network increases the signal and requires additional compression capability. At 400 Hz the compression range is 82 dB or from -66 to $+16$ vu in terms of the input level. The equivalent input noise level of the system is at -94 vu, making the total dynamic range 110 dB.

At the tape the $+16$-vu signal is compressed down to $+5$ vu and the -94-vu signal is brought up to $+50$ vu. The range of signal levels at the tape of -50 to $+5$ vu is well above the tape noise level and below the 1% distortion point. Note that referred to the output of the system, tape noise is reduced 8 dB by the preemphasis alone.

The tape recorder is assumed to have unity gain, and its output is connected to the active transformer in channel 2. After adjusting the signal level in the buffer amplifier portion of the play equalizer module, it is fed simultaneously to a multiplying element and a precision rectifier. The output of the precision rectifier is squared as in channel 1 and applied to the Y input of the multiplier. Because the precision rectifier output is proportional to the tape signal, the Y control voltage is proportional to the square of the tape signal, and the output of the multiplier is proportional to the cube of the tape signal. Thus the expansion characteristic is exactly complementary to the compression characteristic.

Following the multiplier the signal passes to a deemphasis network in the play equalizer which restores the signal to flat response. After adjusting the output level in a buffer amplifier, the reconstructed output closely matches the system input over the -94 to $+16$-vu dynamic range.



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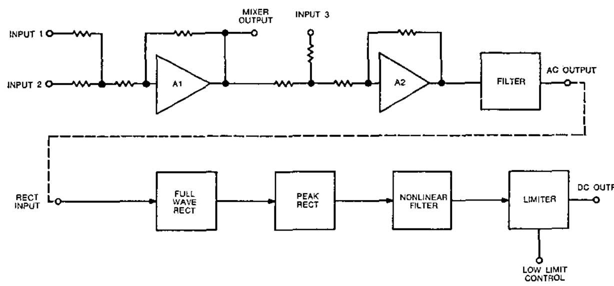

b

Fig. 3. **a.** Inside a precision rectifier module. **b.** System diagram.

Two additional blocks shown in channel 1 consist of an active transformer and the buffer amplifier in the play equalizer which can be used to monitor the compressed signal from the tape. For calibrating the tape machine an OFF mode is provided in which the input signal passes at unity gain through a buffer amplifier in the record equalizer to the record terminal, and the play signal passes at unity gain through a buffer amplifier in the play equalizer to the output terminal. Both manual and remote-control switching are provided to switch each channel to either the RECORD or the PLAY mode. The precision rectifier, the squarer, and the divider can be switched to connect into the play equalizer in channel 1 just as is shown in channel 2. Sharing these three modules between the RECORD and PLAY modes saves both equipment and space and improves the record-playback accuracy for a single channel.

The precision rectifier module is shown in more detail in Fig. 3. It was noted that in characteristic C, designed for the production of consumer material, stereo effects are preserved by making the instantaneous gains of two or four channels the same. Ganging of the channels is accomplished by mixing the high-frequency limiter signals from stereo channels together in the amplifiers $A1$ and $A2$ so that the dc control voltages in all channels are derived from the sum of all four with an appropriate adjustment in gain. The combined audio signals at the output of $A2$ then pass through a 10-kHz low-pass filter and on to the rectification circuits.

All the circuits in the precision rectifier and the other blocks of the system use operational amplifiers, nearly 100 for the two-channel system, to achieve high accuracy, independent of the semiconductor characteristics. The circuits use 1% components and internal trimming to eliminate the accumulation of tolerances.

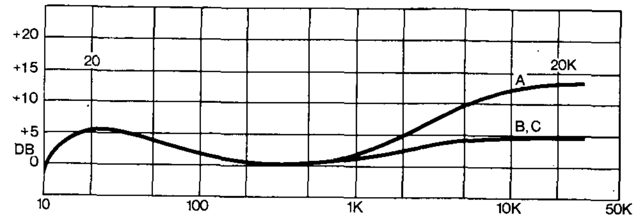

Fig. 4. Record preemphasis.

The ac output from the filter is then full-wave rectified and peak rectified using circuits which have threshold levels below 1 mV. Following the peak rectifier the signal passes through a multistage nonlinear filter and then to a limiter which limits the maximum output to +10 V and the minimum output to either 316 mV for characteristics *A* and *B* or 1 V for characteristic *C*. The combination of the peak rectifier and the nonlinear filter provide a dc output voltage which has a fast rise time constant of approximately 1 ms and a slow decay time. The decay time has been made as rapid as possible consistent with less than 0.05% distortion of a 20-Hz audio signal due to modulation. The final staggering of the time constants was made by listening to the compressed signal on various types of program material and percussive instruments and was chosen to provide a smoothly varying and consistently high-level signal at the tape as an audio transient died out.

In the initial design of the system the same 13-dB high-frequency preemphasis was used for three different tape speeds. Because of the additional preemphasis in the tape machine at the slow speeds, there was a need to reduce the signal level whenever it contained very much high-frequency content. This was accomplished by increasing the high-frequency gain in the peak rectifier module so as to depress the entire signal level. The system worked very well with a studio machine but poorly with a consumer tape machine because of inaccurate reproduction of the high-frequency transients. Also the compressed program material which was intended to be compatible suffered from reductions in the signal level, due to high-frequency instruments, which were completely out of proportion to the apparent loudness of these instruments. The system was accordingly changed to provide flat response with a 10-kHz cutoff in the precision rectifier, and the preemphasis was reduced for the slower tape speeds.

PERFORMANCE

The final preemphasis curves chosen for characteristics *A*, *B*, and *C* are shown in Fig. 4. Characteristic *A* for 15 in/s has 13-dB boost at 20 kHz, but characteristics *B* and *C* for 7½ and 3¾ in/s have only 4.4-dB boost. By using precision networks the overall response of the system, as shown in Fig. 5, through the record and play

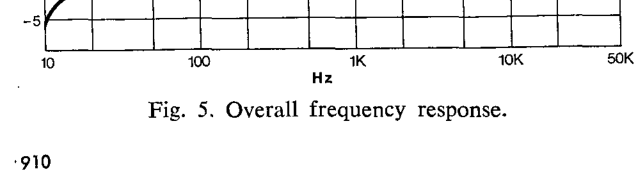

Fig. 5. Overall frequency response.

channels has been made flat to within 0.2 dB from 20 Hz to 20 kHz. The system places great stress on the bandwidth of the divider element in the record channel. Because the divider consists of a multiplier in the feedback path of an operational amplifier, the bandwidth necessarily varies with the divider gain. By attenuating the high frequencies in the operational amplifier at slightly less than 9 dB per octave, the bandwidth has been made to vary from 10 MHz at 0 dB gain to 70 kHz at 60 dB gain. The response is carefully controlled so that the contribution to the system attenuation at 20 kHz is only 0.1 dB for the smallest input signal.

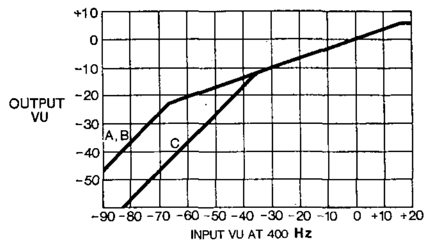

Fig. 6. Record compression.

Fig. 6 shows the record compression characteristic which is 3 dB/1 dB over an 82-dB input range for characteristics *A* and *B*. For characteristic *C* the maximum gain is limited so that the background noise of the compressed program material will not be excessive. This limit is adjustable by remote control to accommodate various types of material.

Due to the precision of the components the reconstruction of the output levels from the compressed signal is accurate within 1 dB over the entire dynamic range and is more accurate for a smaller dynamic range. Harmonic distortion is primarily due to the multiplier and the divider and is typically 0.1% at +16 vu system input and output at 400 Hz (Fig. 7).

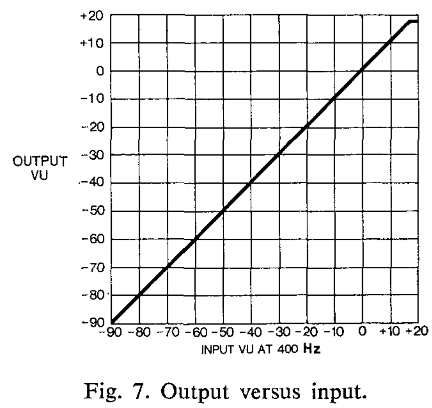

Fig. 7. Output versus input.

In addition to preserving the quality of the signal, the system is designed to provide low-frequency compensation for a deficiency that seems to occur in nearly all tape machines. The response is adjustable from 0 to +15 dB at 16 Hz as shown in Fig. 8. An adjustable network is also provided which can accommodate future high-resolution recording tapes which may have higher output at the high frequencies. The low-frequency compensation is included in the output buffer amplifier of the record equalizer and the high-frequency compensation in the input buffer amplifier of the play equalizer.

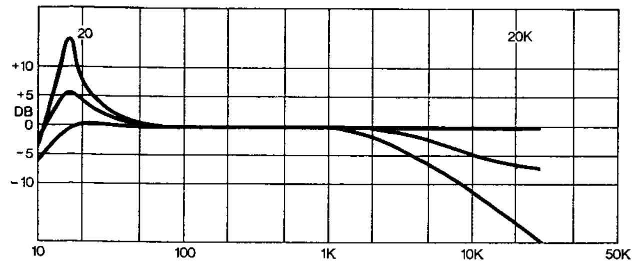

Fig. 8. Compensation for recorder and high-resolution tapes.

Experiments with percussive instruments indicate that the system characteristics provide low distortion reproduction with audibly negligible slew-rate limiting at the leading edge.

CONCLUSION

A precise compandor system has been developed which more than achieves the objective of recording and playback of live musical instruments without audible tape noise and with low distortion. It compresses a 110-dB input dynamic range to 55 dB on the tape, and it accurately reconstructs the signal upon playback. The high performance has been made possible partly by the development of a unique wideband low-noise two-quadrant multiplier/divider. The techniques used are applicable in simplified form to the development of new consumer equipment which offers the possibility of reducing the noise from FM, records, and tapes by 35 dB or more. In a consumer version circuits can be shared among the channels to save components. Program material produced for such equipment is playable with pleasing results on conventional equipment.

REFERENCES

[1] R. S. Burwen, "110-dB Dynamic Range for Tape," *Audio*, p. 49, (June 1971).

[2] R. M. Dolby, "An Audio Noise Reduction System," *J. Audio Eng. Soc.*, vol. 15, p. 383 (Oct. 1967).

[3] R. S. Burwen, "A Dynamic Noise Filter," *J. Audio Eng. Soc.*, vol. 19, pp. 115-120 (Feb. 1971).

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