Dolby B-Type Noise Reduction System

Robert Berkovitz and Kenneth Gundry*

AN UNSATISFACTORY signal-to-noise ratio has remained the major obstacle to attaining an adequate level of performance from consumer media for music reproduction. This is especially true of the music-cassette, because of its slow tape speed and narrow track width, but it is also true of stereo FM broadcasting and the phonograph record. Although hopes were raised in recent years that further development of magnetic tape would eliminate its inherent noise as a problem, these hopes have been frustrated by the relatively modest gains achieved, and by studies which indicate that the available signal-to-noise ratio of present-day tapes is very near the maximum value imposed by theory.

It is therefore not surprising that numerous attempts have been made to devise methods of noise reduction satisfactory for professional and consumer use. However, almost all of the methods proposed have had unacceptable drawbacks.

The effectiveness of single-ended (non-complementary) systems, for example, which are designed to be used only during playback, extends only as far as the listener's willingness to sacrifice musical information. In principle, all playback-only systems depend upon the idea that the signal and the objectionable noise occupy separate domains; if this is correct, then the problem of noise reduction is one of defining the boundary between the domains, in terms of frequency and/or level, and designing a circuit to suppress everything on the "noise" side of the boundary. However, if the noise spectrum of ferric oxide cassette tape is taken as an example (see Fig. 1), it is seen that the noise, when passed through a standard DIN weighting network simulating the ear's sensitivity, remains considerable in the 1-4 kHz range. Since this range includes many of the lower harmonics and upper fundamental tones in music, it is not possible to suppress it, even at low listening levels, without serious loss of information. On the other hand, the noise within this range is so disturbing that if it is not reduced by such a circuit, the amount of subjective improvement obtained is minimal.

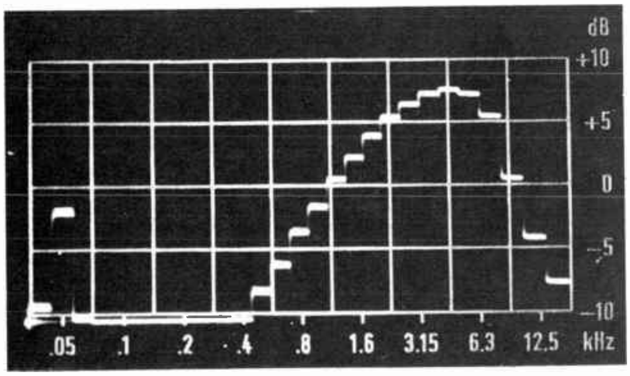

Fig. 1—DIN-weighted noise spectrum of low-noise ferric oxide cassette tape displayed on screen of real-time analyzer. Vertical scale is relative only; DIN noise measurement of sample shown was —47 dB with respect to 200 nWb/m. A substantial amount of the noise present falls well within the range of musical fundamental tones.

Complementary methods, i.e., those which require some signal processing or encoding during both recording and playback, offer greater promise, but can also present difficulties when put into practice. Pre- and de-emphasis schemes, for example, in which high frequencies are increased during recording and decreased by the same amount during playback, are only of limited value. Even in FM broadcasting, where such standardized pre-emphasis has been employed for many years, the usefulness of its continued application is in doubt. The primary problem is that modern microphones and recording equipment now routinely reproduce high frequencies at amplitudes so high that they were considered unlikely when current FM standards were set. Broadcasters are now forced to use limiters to prevent overmodulation, if they also wish to maintain reasonable levels at middle and low frequencies. In magnetic tape recording, pre-emphasis is difficult to use because tape saturation occurs at lower levels at high frequencies. Since high-frequency signals already present problems in cassette recording because of their short wavelength, added pre-emphasis would complicate a task which is already difficult.

The compandor type of noise reduction system, in which the dynamic range of the signal is compressed during recording and expanded again during playback, offers more promise. However, a simple compandor, even if precise in its action, also presents problems. In recordings and broadcasting, one of the most serious drawbacks of the compandor is the danger of signal overshoot, which can result in distortion or overmodulation of a transmitter.

An even more serious problem of compandors, from the listener's point of view, is noise modulation. When a conventional full-band compandor is used, low-level passages are recorded at a level higher than normal. They are then played back at reduced level, restoring correct signal dynamics and reducing noise at the same time (see Fig. 2). There can be no noise reduction effect during high-level passages, because this would require increasing the level of such passages during recording, resulting in overload. The simple compandor therefore requires that one assume that noise is not objectionable when the signal level is high. However, this is not always the case. A high-level bass drum beat, for example, does not mask high-frequency tape hiss; as a result the drum and other instruments introduce noise modulation during playback—each note is accompanied by a "swish" as the noise level rises for the duration of the note. While it is not audible with all types of program material, noise modulation limits the usefulness of the compandor considerably.

The extreme diversity of available source material and the high quality of present-day master recordings are the factors which really determine the conditions to be met by a satisfactory noise reduction system for home use. It must be remembered that many home listeners own playback equipment with very low distortion and wide frequency range,

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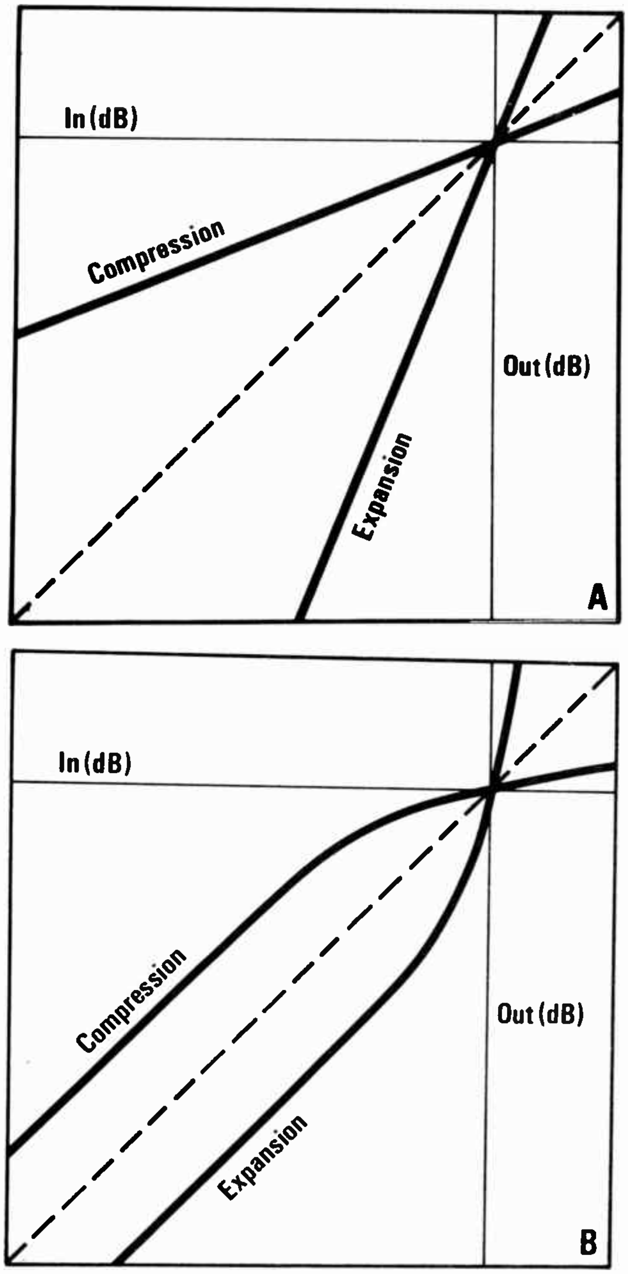

Fig. 2—Transfer characteristics of two conventional compandors (solid lines). A, constant-slope type; B, high-level type. Since compression and expansion are functions of signal amplitude only, in a single frequency band, such compandors fail to suppress noise whenever natural masking fails (see text).

disclosing audible effects which might have passed unnoticed in earlier times. Therefore, it is especially important that the program be recovered accurately after noise reduction, without addition of any audible sound. For the listener's sake accuracy of recovery and effectiveness of the system should not require adjustment of system parameters to match various kinds of program material. At the same time, the size and cost of the system should introduce no obstacle to its use. Furthermore, as a practical matter for the industry, it is clear that the system should require no modification of present professional practice in master recording, duplicating, or broadcasting.

The Dolby B-Type Noise Reduction System

The Dolby B-Type circuit is a specialized form of compandor which avoids the usual deficiencies of compandors. The operational principle of the B-Type system is complementary low-level compression and expansion in a frequency range which varies in bandwidth as the signal changes.

Most objectionable noise encountered in home listening is at middle and high frequencies, from about 500 Hz to the upper limit of audibility. In the interest of circuit economy, the action of the B-Type circuit has therefore been limited to this range. A feedback control circuit adjusts system parameters automatically as a function of signal level and spectrum, so that the system's action complements the psychoacoustic masking of noise which occurs naturally in the course of the program. A block diagram of a Dolby type of noise reduction system is shown in Fig. 3. The circuits used for encoding (during recording or transmission) and decoding (during playback or reception) are quite similar and can be considered as the same circuit, switched to operate in either mode.

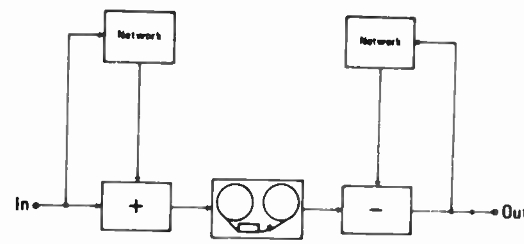

Fig. 3—Block diagram of Dolby type noise reduction circuit as used in typical record-reproduce chain.

The compression and expansion characteristics of the Dolby B-System are fixed and are referred to Dolby Level, a specific internationally standardized reference level. In the case of cassette tape, Dolby Level is a flux of 200 nWb/m; in FM broadcasting, Dolby Level is ± 37.5 kHz deviation.

Figure 4 is a block diagram of a switchable (encode-decode) B-Type circuit. There are two paths which the input signal follows: a *main path* (at the lower part of the figure) in which no change other than linear amplification occurs, and a *secondary path*, a variable filter through which only low-level, high frequency components of the input signal are allowed to pass. To encode the signal, the output of the secondary path is combined with signal in the main path *additively;* this boosts low-level, high frequency portions of the signal. Decoding is accomplished by feeding the secondary path from the circuit output, which is opposite in phase to the input (note phase inverter in Fig. 4); the secondary path is then part of an a.c. negative feedback loop which reduces output, i.e., the output of the secondary path is combined with the main path *subtractively.* In the decode mode, therefore, the circuit reduces the level of precisely the same information which was increased in level during encoding.

As Fig. 4 indicates, the action of the B-Type circuit is controlled by the output of the filter in the secondary path. Above a fixed threshold level, the bandpass of the filter, in turn, is modified by the d.c. feedback loop.

At very low levels, i.e., below the threshold, which at high frequencies is about 40 dB below Dolby Level, the output of the filter is not sufficient to generate d.c. feedback; consequently, the output of the secondary path is simply proportional to signal level within the filter pass band. The output of the circuit is then essentially as shown in Fig. 5.

As signal level rises above the threshold level, the rectified filter output is returned to the FET gate where it is applied as negative feedback, raising the filter cutoff frequency so

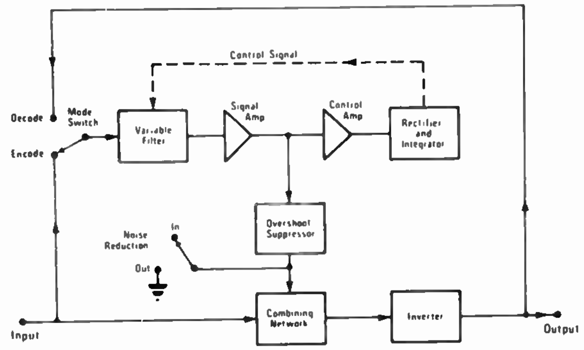

Fig. 4—Block diagram of Dolby B-Type noise reduction circuit. The configuration shown can be switched to encode or decode the signal.

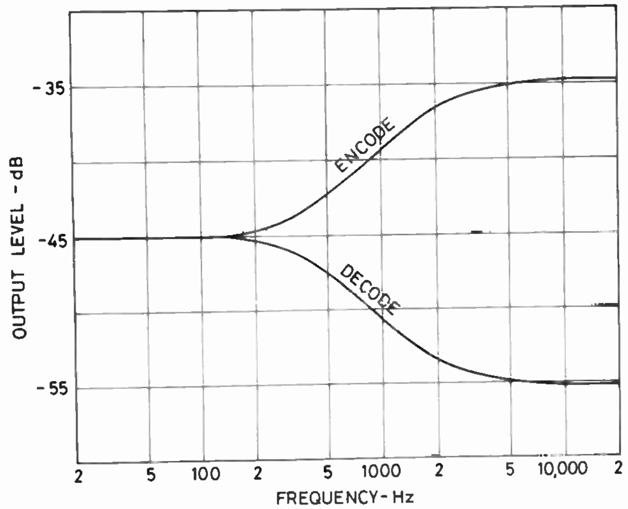

Fig. 5—Output of B-Type encoder and decoder circuits under low-level input signal conditions. The two operations are symmetrical and the result is an overall frequency response which is level.

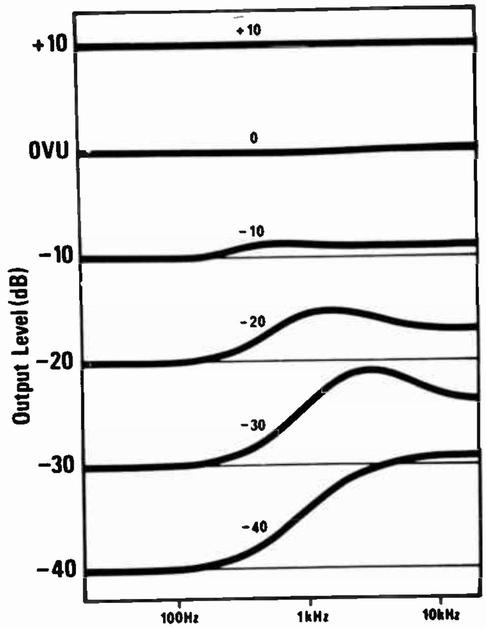

Fig. 6—Characteristics of encoding processor at several levels. The gradual reduction in boost with increasing level avoids possible tape overload.

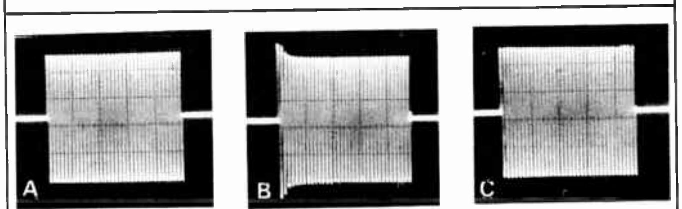

Fig. 7—Effect of the B-Type circuit on a tone burst; frequency = 3 kHz; burst duration, 12 milliseconds; low level = 40 dB; high level = +6 dB; (A) Input to system; (B) Encoded, (C) Encoded and Decoded.

that the output of the secondary path, while still increasing, no longer does so in proportion to the change in signal level. As signal level becomes even larger, the increasing d.c. feedback generated restricts the filter bandwidth further, and near Dolby Level the output of the secondary path remains relatively constant. The net effect is that the secondary path has no audible effect on output at low frequencies, and increasing effect with increasing frequency and decreasing level to about 40 dB below Dolby Level. At high levels, the effect of the extra signal is so small as to have no significance; at low levels, in the spectral region in which noise reduction is required, the increase during encoding is as much as 10 dB, and is of considerable importance.

The manner in which the secondary path changes from constant-gain to constant-output is determined by the adjustment of gain within the feedback loop. In addition, the exact variation in filter bandpass with changing level is set optimally by making the control amplifier frequency-dependent. The overall frequency response of a B-Type encoder circuit for different input levels is shown in Fig. 6.

A compandor operating over a wide frequency range must be designed to take into account the problem of noise modulation discussed above. If some high-level passages in the program differ sufficiently in frequency content from the noise components present, the latter will remain audible during the program in many cases. However, these passages cannot be increased in level when encoded, because of the danger of overmodulation. Under these conditions, compression may be applied intermittently, and high-frequency noise modulated audibly by mid-frequency components of the signal. The B-Type circuit overcomes this problem because it continues to function when a high-level signal occurs within its operating range; instead the feedback control shifts the range upward in frequency. This avoids the danger of overmodulation, but retains full noise reduction at frequencies higher than those masked by the signal.

The attack time of the B-Type circuit is dependent on the amount and rapidity of the signal change, due to the nonlinear design of the integrator, varying from about 100 milliseconds to as little as 1 millisecond. The recovery time of the rectifier-integrator is shorter than that of the human hearing system, about 100 milliseconds.

All compressors exhibit overshoot, including the B-Type circuit. However, the dual-path approach used makes it possible to reduce the amplitude of overshoots significantly. Overshoot, which can occur only in the secondary path (where it can be suppressed without affecting the main signal) is comparatively small, and essentially disappears when the signal is decoded again. When signal levels are low, or when changes in signal level take place slowly, there is no overshoot problem; when signal changes are large and rapid,

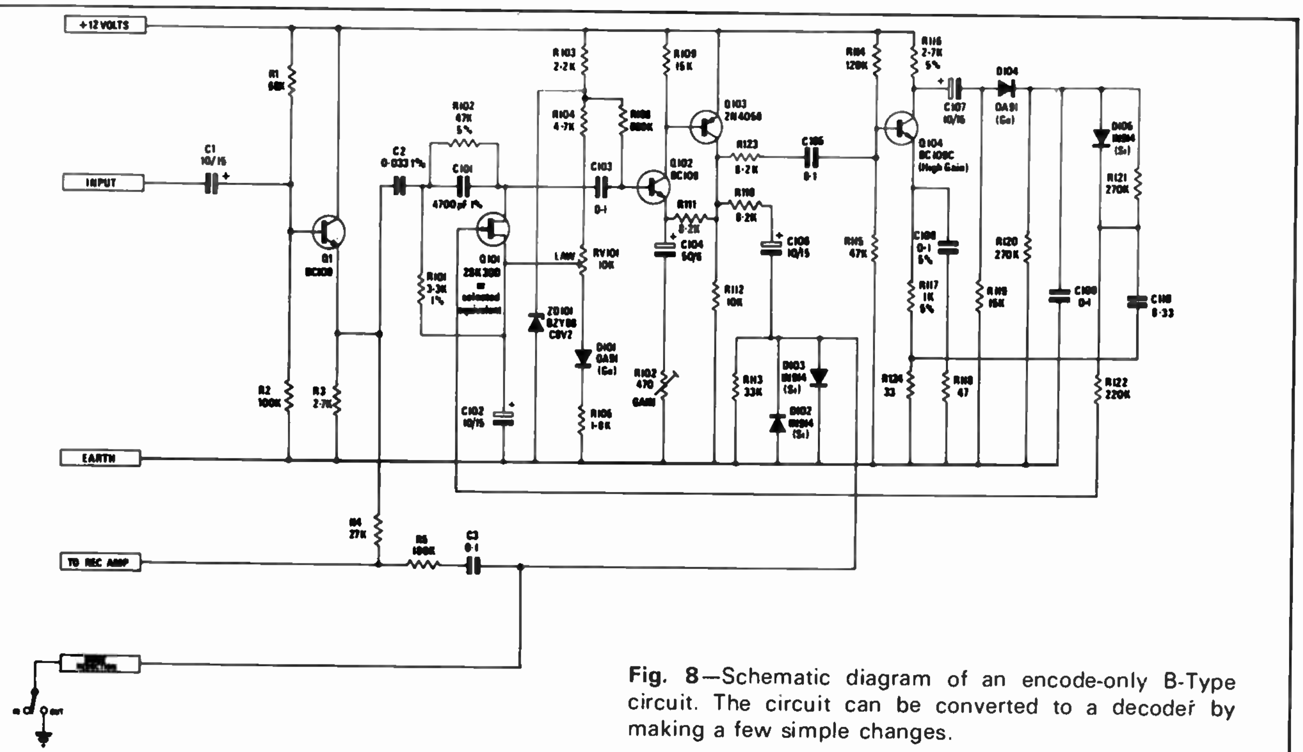

Fig. 8—Schematic diagram of an encode-only B-Type circuit. The circuit can be converted to a decoder by making a few simple changes.

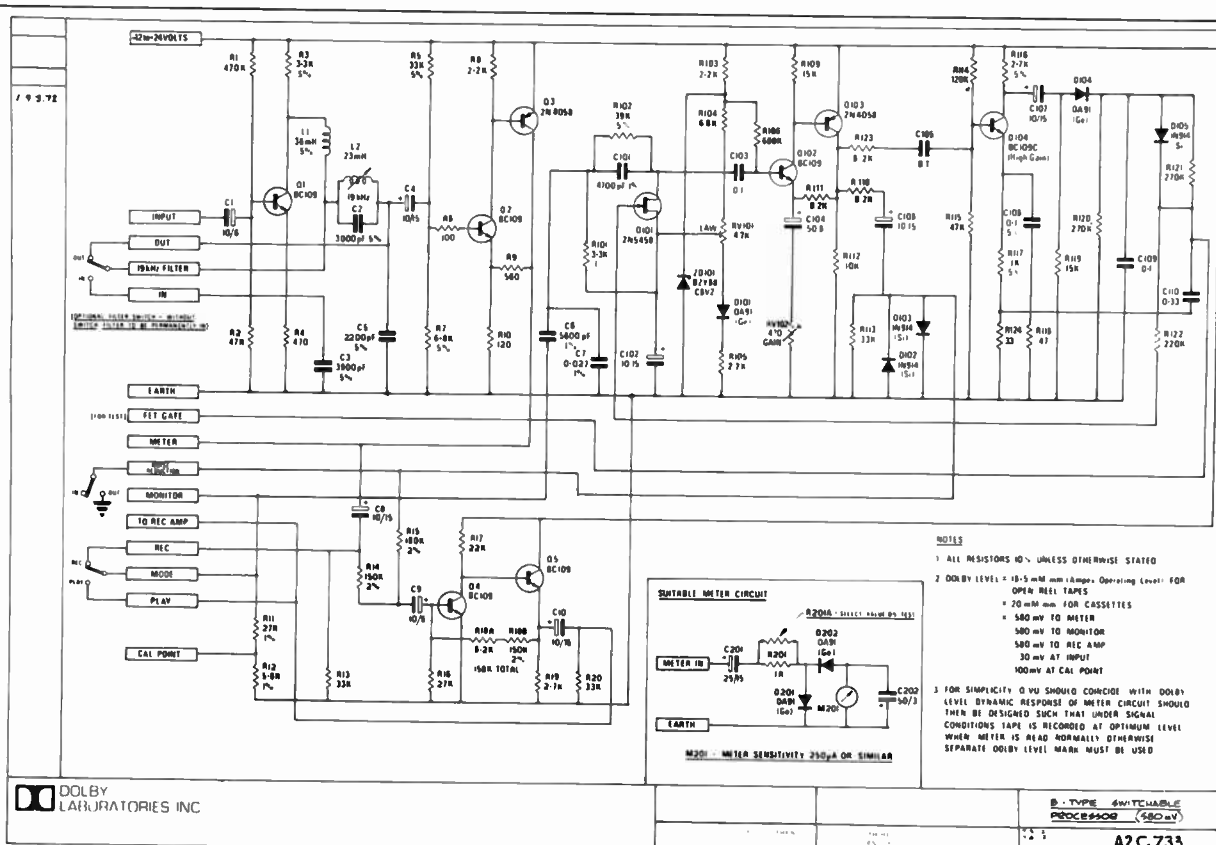

Fig. 9—Schematic diagram of complete switchable encode/decode circuit for use in tape recorder, including HF filtering, meter drive, monitor output and 26 dB of gain. Only a few more parts need be added to make complete record/play electronics for one channel of the recorder.

diodes in the overshoot suppressor stage limit the peaks of the overshoot. Since this takes place in the secondary path, the result of the suppressor action is to limit overshoot to a relatively small fraction of the full-level main path signal. Further, by restricting overshoot suppression to the secondary path, it is possible to avoid introducing audible distortion to the encoded signal. Because a complementary action takes place during decoding, the small remaining overshoot in the encoded signal is eliminated, and as with other effects produced during encoding, the original signal is restored. Figure 7 shows the result of encoding and decoding a short burst of 3 kHz, which changes in level from −40 dB to +6 dB.

Figure 8 is a typical schematic diagram of an encode-only B-Type circuit; the circuit for decoding-only is similar. As can be seen, only five transistors plus an FET are required; the parts cost of the circuit is approximately $2.40.

Figure 9 is the schematic diagram of a B-Type processor which has been designed to integrate noise reduction with other tape recorder electronics requirements as much as possible. The resulting circuit provides 26 dB of gain, whether or not noise reduction is in use, bias and multiplex filtering, and meter and monitor amplifiers. In fact, the only additional electronics needed to complete the recorder are a bias oscillator, recording amplifier (one transistor) and a microphone and head amplifier (two transistors). With the active elements used in the record/play switchable processor shown (eight transistors and one FET), the total used in the recorder, for two channels, is 22 transistors and two FET's. The cost to a manufacturer of the components shown in Fig. 9 is about $3.20, excluding the bias and multiplex filter components, which are, of course, necessary in the circuits of any properly designed tuner and recorder.

Dolby Laboratories and Signetics have collaborated in the development of an integrated-circuit version of the B-Type circuit. The IC is expected to offer manufacturers economy of assembly, elimination of adjustments, and somewhat smaller space requirement than the discrete-component version.

The characteristics of Dolby B-Type noise reduction can be summarized as follows:

1. Program recovery characteristics, with regard to frequency response, phase response, transients, and signal dynamics, are theoretically perfect; in practice, this ideal is attainable to any desired accuracy. Distortion in practical B-Type circuitry is considerably lower than that of the tape recorders or tuners with which it is used. Any type of program material can be encoded and decoded without audible loss.

2. The circuit is simple, inexpensive, and small in size, either in discrete-component or IC form.

3. The circuit is easy to manufacture and use because of the absence of critical components or adjustments. The circuit can be quickly and easily calibrated during manufacture, after which further calibration is not required. In use, only a simple level adjustment is necessary if tape of significantly different sensitivity is substituted for that formerly used.

4. No modification of broadcasting or duplicating practice is required to incorporate B-Type encoding. The use of the noise reduction system often makes worthwhile other improvements, however, such as extension of frequency response and dynamic range, or reduction of distortion by use of lower modulation levels, or some combination of these.

Effects Upon Noise Spectra

Figure 10 is a multiple exposure of the screen of a 1/3-octave real-time analyzer, allowing a direct comparison of the noise spectra at the output of a high-quality cassette recorder when different kinds of tape were used with and without the Dolby B-Type noise reduction circuit. Curve 1 is that produced by C90 ferric oxide tape; curve 2 is that of C90 chromium dioxide tape; curve 3 is produced by the same tape used for curve 1, but the B-Type circuit is switched "in," and curve 4 represents the noise spectrum of the chromium dioxide tape with the circuit in. The tapes shown were biased before the measurements were made; no changes in gain or other control settings were made during the tests, other than to set equalization differently for the chromium dioxide tape from (70 microsecond). In fact, most of the improvement in noise level obtained when chromium dioxide tape is used appears to be due to the change in equalization; if this change is not made, there is little advantage in chromium dioxide tape from a noise point of view. On the other hand, the combination of chromium dioxide tape, 70 microsecond equalization, and B-Type noise reduction results in an excellent noise figure, 57 dB below Dolby Level in the example in the photograph (DIN 45405).

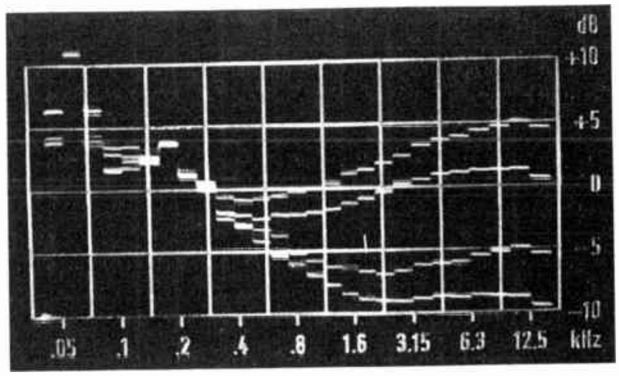

Fig. 10—Noise spectra at output of cassette recorder during playback of four kinds of tape. All tape was biased and control settings were the same for the four tests. (1) Ferric oxide C90 low-noise, −45.7 dB DIN 45405; (2) Chromium dioxide C90, −48 dB; (3) Same as (1) with B-Type noise reduction, −55 dB; (4) Same as (2) with B-Type noise reduction, −57.1 dB. All noise measurements referred to Dolby Level = 200 nWb/m.

The advantages of B-Type noise reduction are also obtained when the system is used for FM broadcast transmission and reception, i.e., the improvement in signal-to-noise ratio obtained by use of the B-Type circuit is approximately the same as that produced by a 10 dB increase in field strength. The significance of this improvement can be appreciated when it is realized that such an increase would usually require an increase in transmitter power by a factor of ten. Considerable experimentation and broadcast experience in the USA have demonstrated, as one would expect, that the area in which listening is satisfactory is greatly extended by use of the B-Type noise reduction system. Several American classical music FM stations are already broadcasting full-time using Dolby B-encoding.

Compatibility

When any improvement is made in a system as widely used as the compact cassette system, it is highly important that the new development should be fully compatible with existing equipment. Improved cassettes must be playable on any machine which can play old-type cassettes, and fortunately this is true of Dolby B-Type cassettes. Such cassettes are subjectively compatible (i.e., generally pleasing to the listener) when played without decoding circuitry, to a great extent because of the unique approach taken in the B-Type circuit.

Because most low-cost cassette machines are deficient in high-frequency response, the increase in low-level high frequency content in a B-Type cassette is usually welcomed by listeners with such equipment. Cassette recorders of higher quality, or the associated equipment with which they are used, contain tone controls which permit the balance to be adjusted to suit the taste of the listener. It is quite likely that many of the millions of B-Type encoded cassettes which have been made commercially are owned and played by listeners who are unaware of the special nature of the program material they hear. In any case, the subjective difference between encoded and other cassettes is sufficiently unobtrusive that none of the recording companies offering "Dolbyized" cassettes have found it necessary to offer old-type cassettes as an alternative.

It is worth noting that almost all pre-recorded cassettes are already compressed, for only in this way can the audibility of low-level passages be preserved in programs of wide dynamic range. B-Type cassettes differ mainly in that the listener now is able to remove the compression by pushing a button on his cassette machine restoring program dynamics and reducing noise. This is only possible because B-Type compression is standardized, while other types of compression vary considerably.

Commercial Use

Within a few years of its introduction, the Dolby B-Type noise reduction system has been licensed to most major manufacturers of consumer tape recorders. At the present time there are more than 40 licensees manufacturing over 100 different B-Type products. Licensee payments for use of the circuit are on a sliding scale, based on quantity, from a maximum of 50¢ (U.S.) to 10¢ per circuit. Royalty charges are typically 60¢ per stereo unit for a major manufacturer.

In addition, most of the pre-recorded cassettes now made in the United States, the United Kingdom and Japan are "Dolbyized," and many of the largest recording companies issue their cassette output in this form, among them Ampex and CBS in the United States, Decca and RCA in England, and CBS-Sony, Nippon Columbia, King, and Apollon in Japan. Pre-recorded open-reel tapes and 8-track cartridges are also becoming available. In the United States, a number of FM stations have already started to broadcast regularly in B-Type encoded form, and this procedure is under study in other countries as well. There is no royalty payable for encoding cassettes or other tape recordings, or broadcasts.

Conclusions

The reduction of background noise by the Dolby B-Type noise reduction system has contributed importantly to the improvement in quality of home tape recording and playback. It has helped to make the extension of frequency response, the reduction of wow and flutter, and other improvements worthwhile, particularly in cassettes. The unique characteristics of the B-Type system permit excellent noise reduction without program losses, noise modulation and other drawbacks which have afflicted earlier attempts to solve the noise problem. The simplicity and economy of the B-Type circuit facilitate its use in consumer products at all price levels. Æ

Noise Reduction Techniques

H. W. Hellyer*

LET'S TAKE A LOOK at one or two ventures into noise exclusion that have been at least a bit more ambitious than a mere clipping of playback peaks. One such system is Panasonic's NFD device. NFD, quite simply, mutes the line output unless the signals (on playback) are above a predetermined level and below a set frequency. This reduces hiss when the signal level is low. That is, you get what you want when you most want it.

In the RS 735US, there was a two-transistor, nine-diode circuit that gave very good results indeed. Figure 1 shows the basic configuration. Signal-to-noise ratio, when I tested it, with this noise filter employed, was as good as 66.5 dB. At 1 kHz, the improvement was a mere ¾ dB, but although at rated output level the NFD only made 1 dB difference to the S/N ratio, when the level of signal was down around the danger level, approaching what would have been obtrusive hiss, the circuit effectively blanked signal, and its action did not, as with so many compandor systems, provide an aural switchback.

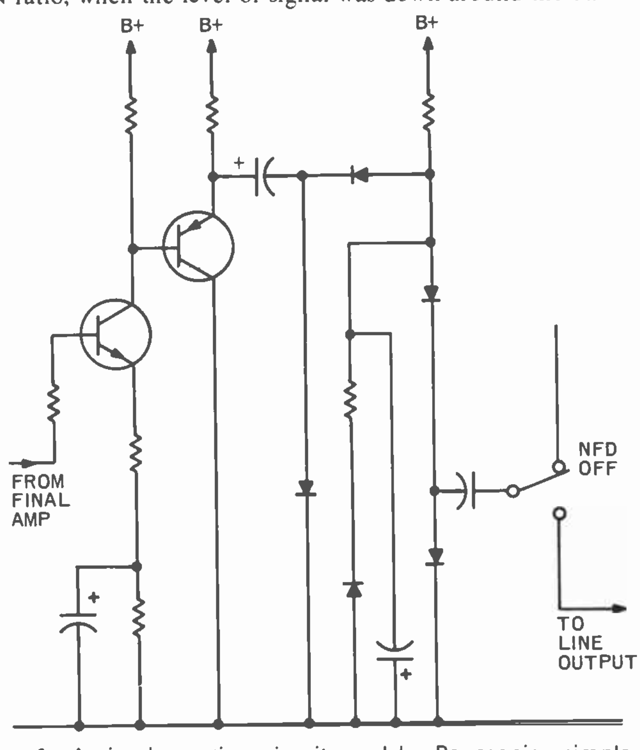

Fig. 1—A simple muting circuit used by Panasonic—simple, but effective, sensing the signal level and "killing" the line output when the signal drops dangerously near the noise level. The circuit shown is for one channel. The same two-transistor network is employed for the other channel, and this "commoning" can lead to problems.

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Taking the replay system a step farther, Philips has the DNL innovation, which should make much cassette work with other folk's tapes a really feasible possibility.

DNL means Dynamic Noise Limiter, and Philips (Norelco to you) argues thus . . .

"When music is played softly, it is made up almost entirely of pure tones in the middle and low frequency ranges with hardly any harmonics. This is mainly because very few musical instruments produce tones whose fundamental frequencies are much higher than 4.5 kHz. Tape hiss, however, is made up of sounds in the higher frequencies so that it is during the soft passages and silent intervals that it becomes most noticeable.

"When music is played loudly, it not only contains the lower and middle frequency pure tones, but also a great deal of harmonics, which give character to the sound. It is in the loud passages that noise suppression is unnecessary as the high frequency harmonics hide the tape hiss. Any filter action would make the music sound dull and unnatural.

"Therefore, if tape noise or hiss is to be suppressed, it must be completely eliminated in periods of no music signal, reduced during the soft passages of music, and left unsuppressed during the loud passages."

Thus, the oracle—begging one or two questions, like: "Pure tones—all instruments played softly?" and "What happens to the soft tones of one instrument when another plays loudly?" and "How soon after the loud noise ends does the suppression take place?"

The Dynamic Noise Limiter acts on replay, the argument being that it therefore allows complete compatibility, giving the benefit of noise suppression even to those poor, deprived owners of untailored cassettes. It is, effectively, a steep, low-pass filter which acts when there are no high signal frequencies.

Philips has been rather clever about it, allowing high frequency signals that exceed a predetermined level to bypass the filter: so there are two signal chains. Fig. 2 shows the block diagram. From the splitter, the signal takes two paths, one path merely inverts the phase without affecting the linearity while the other passes it through the tailoring process.

This process chops off the lower and middle frequencies, leaving only those above 4 kHz (approximately—you can't do these chopping actions abruptly without introducing almost ineradicable distortions, whatever the advertising copywriters say). This remaining high frequency band is now monitored so that the quieter parts of higher frequency are boosted. Hence the variable attenuator—it is both level and frequency-conscious.

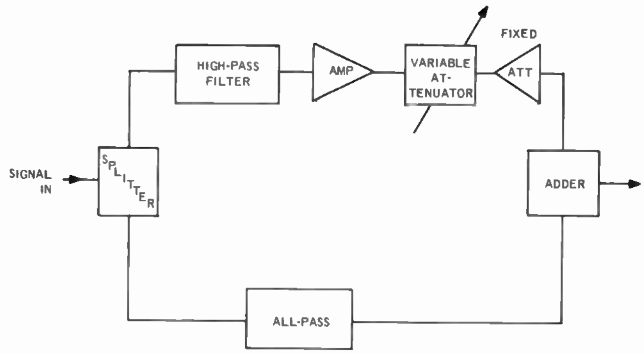

Fig. 2—Block diagram of the Philips (Norelco) Dynamic Noise Filter. The surprisingly effective though unsophisticated system acts on playback only and has the effect of an 18 dB/octave filter when the signal is low. A S/N ratio improvement of around 10 dB at 6 kHz and 20 dB at 10 kHz has been measured (unweighted). The high-pass filter takes effect above 4 kHz.

Fig. 3—The DNL circuit, four transistors, six diodes, and a handful of common components, can easily be made up into a neat set-side box—no bigger than a double pack of 20 cancer-sticks.

Adding together the processed and unprocessed chains should now, theoretically, give a signal whose low-level high frequencies have a quietened effect, while middle and low frequencies are unaltered and where the higher volume high frequencies are given their full, required weight. In theory, once again, the result should be a true replica of the original, but without the hiss.

And, I must admit, despite some initial misgivings because Philips demonstrated this device to us a year or so ago in an hotel room whose air-conditioning added some 30 dB to the ambient noise, the subjective effect is a cleaner sound, whatever the condition of the recording.

But I still feel that the answer is not to use a circuit that gives, as Philips claim, a 10 dB improvement of S/N ratio at 6 kHz and a 20 dB improvement at 10 kHz on replay, but to improve the overall record/replay process in such a way as to retain its original sound structure, not "tailor" it. Again, if you must have slow-speed, narrow-track recording, then you have to engineer out the hiss, not allow it to happen and *then* try to beat it.

So we come to Dolby and the now-famous stretching process that Dr. Ray Dolby pioneered. The original "A" process aimed at beating the "breathing" that compansion procedures forced disc users to suffer and cost more than some recording companies could afford. It begins its work during recording, splitting the audio path into a direct and a rectifier chain. But the expensive "A" system did this in four bites, carving up the frequency spectrum to give differential gain depending on signal level within the frequency bands. These are: below 80 Hz, from 80 to 3,000 Hz, above 3,000 and again above 9,000 Hz.

Both hiss and hum are present in the recording process, and while hum can be relegated to one low portion of the audio spectrum, hiss is a very different problem. It obtrudes into the very region where our ears happen to be most sensitive. It has measurable components that extend way upwards into what some engineer colleagues of mine call the "annoyance passband." Any crude way of militating against hiss will mutilate the upper frequencies which we need to preserve to get the clash and tingle of a full musical experience.

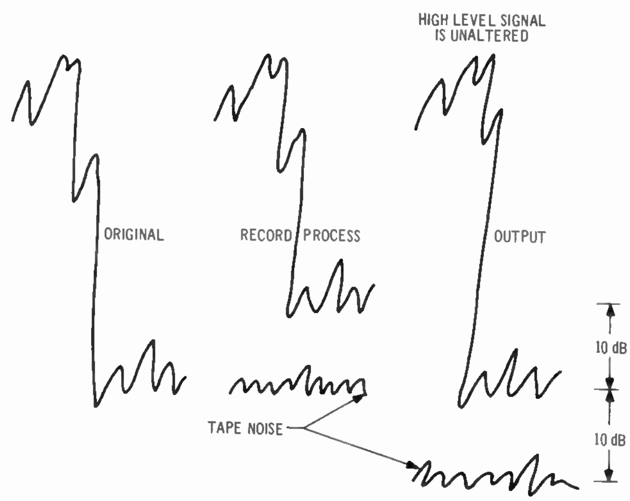

Fig. 4—One way of explaining the Dolby system: The original signal has its lower, levels down around the system noise. Processing during record gains some 10 dB of S/N ratio. Replay retains this, raising the lower levels of signal that much above the noise.

Again, the procedure is to let the noise remain when the music is loud enough to mask it. Masking—as a technical term—is a peculiar business. It depends as much on relative frequencies as on loudness, and has some strange anomalies to do with time difference and phase factors. Subject for a later discourse, maybe. At present, please take my word for it that the phenomenon happens, and by letting the main, high level signals straight through the system, Dr. Dolby follows the method we have roughly outlined already.

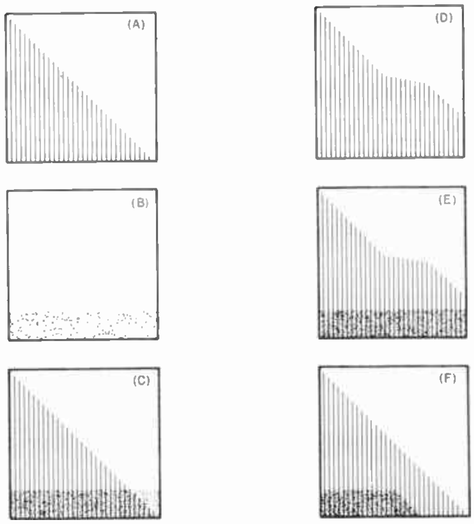

Fig. 5—An alternative explanation, as depicted by Dolby: A, music is made of sounds of different loudness with intervals of silence; B, noise of some kind is inescapable; C, when a tape recording is made and replayed the noise interferes with the low level signals, spoiling the program; D, the Dolby system boosts the lower signals during recording; E, those lower signals are still above the annoying noise during replay, as shown in F, the composite picture of the reconstituted sound with noise "reduced" by the carefully engineered boost and stretch system.

Fig. 6—Noise reduction units can be added quite easily to existing equipment. This Advent 100A has been enthusiastically received, despite the $250.00 price tag. My own special interest is harmonic distortion, and I was interested to note that the 100A was under 0.4% to 0 dB and less than 0.2% at lower levels. Output noise, –60 dB; noise reduction around 10 dB above 4 kHz, about 3 dB at 6 kHz. This is a stereo unit and well worth considering for slow-speed recording.

The subtlety lies in the treatment of the low-level signals, where noise is obtrusive. Dolby calls this the differential component, and this is, of course, relatively small—and hence more difficult to handle. It has to be remembered that the noise reduction system does not eradicate noise; it boosts weak signals to improve the signal-to-noise ratio, that's all.

That's all! Pause for hollow laughter! Arguable decisions are the threshold limit, below which noise-plus-signal will be processed, attack time, the response of filter circuitry to the information that a signal in need of treatment is coming along, the amount and nature of compression, and the way of ensuring a mirror image expansion and an avoidance of overshoot (which would process signals that did not need such treatment).

If the distortion has a duration of less than a millisecond, it will defeat the human ear. This is a smaller fraction than normal signal transients and our aural loudness-growth characteristic cannot distinguish the short-lived distortion.

The Dolby "B" system came into being when Ray Dolby was asked to dream up a modified noise reduction device for use with domestic equipment. The only feasible way to keep such a system within our budget was to forgo the technical requirement of four passbands and operate over the whole audio spectrum, this time making the sensor part of the apparatus listen for frequency as well as loudness, on a kind of sliding scale.

The system comes into action at about 600 Hz, with a maximum 3 dB effect. (O.K., so the ads say it extends above 2 kHz, but the sliding scale method means it really begins lower down). At 1.2 kHz it has a maximum 6 dB effect, has 9 dB at 2.4 kHz and reaches the advertised 10 dB above 4 kHz. The compression comes in about 45 dB below what has become known as the "Dolby level." This can be defined as a flux level on tape of 200 nanowebers per meter. Call this 0 VU.

In more technical terms, the differential chain splits into the rectifier path and into the linear path to the mixer for readdition to the main signal. The rectifier path contains boost circuits giving a 6 dB per octave flip to the higher frequencies. Then the output is rectified. This rectified signal effectively alters the dynamic resistance of an FET at the input end of the chain, and so gives a boost at low dynamic levels and practically no boost at high levels. By the simple device of driving the FET via a small coupling capacitor, Dr. Dolby achieved both a drop in gain with an increase in dynamic level and a change of the turnover frequency of the "threshold" as the level changes. The sliding scale, in fact.

At low levels the capacitor lets the FET see the full signal and gives a good 10 dB boost above 2 kHz. Increase the input level and the frequency above which this full boost is given begins to rise. Turn up the wick still more and the treble boost in the rectifier chain stops the over-saturation of the tape. To reinterpret, that means the tape is driven to its full limit when need be, at high dynamic levels (of original signal), but is allowed up to a 10 dB boost at lower signal levels. The replay mode is reciprocal.

The entire processed chain is inserted in a feedback loop around the main chain to subtract instead of add. The elegance of the system is that the same basic circuitry, and, indeed, a mirror-image printed circuit board makes production costs tumble and the add-on Dolby units now available should be within any enthusiast's purse. (Dolby IC chips are also coming soon.—Ed.)

Fig. 7—Slim, elegant, technically precise, one section of the Dolby A system as used by professional recording bodies throughout the world. Having had the chance to "rip one to bits," I can vouch for its engineering excellence.

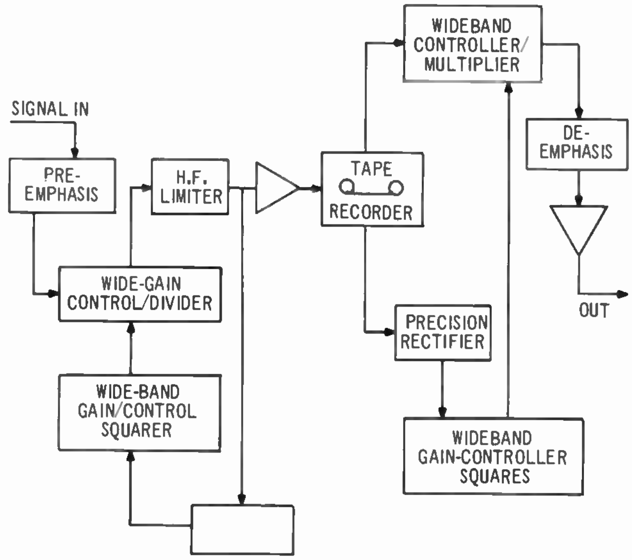

Fig. 8—Block diagram of the Burwen system, with refinements like active transformers and direct play equalizers omitted. The heart of the system is the rectifier module, monitoring the gain of two channels simultaneously in the "domestic" system. Operational amplifiers are used widely in this system with very high accuracy as a result.

My own tests with those available in the U.K. have confirmed that signal-processing of cassette-recorded music, speech, and sound effects have done wonders to guard against hiss and have not made detectable any audible worsening of the prime signal.

After Dolby, what? Well, according to Richard Burwen, quite a lot. In the December, 1971 issue of the Journal of the Audio Engineering Society, I came across the Design of a Noise Eliminator System which gave me much brain-searching and is at present exercising the pundits in those polite tomahawkeries of the erudite correspondence columns. (See also AUDIO, June, 1971.)

To begin with, the title of Richard S. Burwen's paper hits a sore point. The only way you eliminate noise, truly, is not to cause it. After the die is cast, all you can do is guard against it—which we have seen three different systems doing in the preceding notes.

Mr. Burwen took the critics by the ears at the 41st convention in New York on October 5, 1971. In February of that year, a paper of his entitled "A Dynamic Noise Filter" had aroused comment. He is more concerned with studio tape machines, just as Dr. Dolby was, and there seems little hope, at present, of such an elegant "domestic" solution to the noise reduction problem with a plain man's Burwen. But anyone who has been in the audio field as long as us (well, me) knows better than to say that something, anything, cannot be done.

So let's conclude with a brief look at Mr. Burwen's solution.

He set himself some pretty high parameters. His system was not, he told us, to exceed the present 1%, and preferably 0.5%, distortion level of good taping. He wants to record live music "with no audible noise whatsoever." So his first experiments were to determine peak recording levels.

Recording to +3 VU, a normal process, when 0 VU is the standard set limit and peaks above this as much as +6 VU are occasionally tolerated because of their short duration, meant that distortion on tape went over that critical 1%. He concluded—first point, and first stumbling block for his critics—that it is not always advisable to retain every peak.

Listening tests revealed that for noise to be negligible in the absence of program material, it had to be 90 dB or more below the 1% distortion level, i.e., better than -84 VU. Then he found that noise 65 dB down was audible with a 500 Hz sine wave but masked by frequencies above 3 kHz. You could reduce the bandwidth to about a half-octave centered on 500 Hz and get a pure tone—so the solution seemed to be split the waveband, per Dolby A.

But the multiband system, according to Richard Burwen, has the disadvantage of frequency response errors in the tape machine causing errors in the expansion process. The solution was to use the whole band but compress the 90 dB expected input to 30 dB at the tape. He then combined the principles of his dynamic noise filter (see June, 1972 issue of this magazine) with a single wideband compandor.

The dynamic noise filter acts as a low level expander at top and bottom of the frequency spectrum—again, something like we've seen before. Adding a high and low-frequency compression system seemed to be the answer, and high frequency pre-emphasis was intended to improve the S/N ratio. Some hellish problems raised themselves at this point, and Mr. Burwen went back to the drawing board. He finally produced three systems, A, B and C. Characteristic A is optimized for studio recording at 15 ips. It has a dynamic range of 110 dB and this is the one you'll see hailed in the ads! System B operates more modestly to give a 102 dB dynamic range at 7½ ips, and C is the one that may eventually interest us at 3¾ or 1⅞ ips for FM broadcasting, records or background music. If you want it in the words of the master: "The system . . . utilizes high and low frequency pre-emphasis and a single wideband cube root compressor to produce the recorded signal, and a complementary expandor and pre-emphasis for playback."

The important point slipped in later is that in the single-band system the frequency response is constant and is not affected by inaccuracy in the tape machine. Again, we shall leave the pundits to argue.

The high performance of the Burwen circuitry has been made possible by the low-noise two-quadrant multiplier/divider. Bettering Dolby by one magnitude in claim and applicable also to FM systems, it seems to offer possibilities, and we must wait and see what the outcome may be.

For my part, in this noise-polluted environment, I welcome any device that can help rid us of clamour. But noise is what you make it, and the tick of an obtrusive clock, as many an amateur recordist has found, can be as bothersome as a traction engine. The subjective results, applied to cassette, have been enormous—praise to the noise-breakers! Æ