NOISE REDUCTION SYSTEMS

There is a lot which appears on records and tapes that shouldn't be there. How do you go about getting rid of it, or at least reducing it? William King investigates.

OVER THE PAST FEW decades, the standard of reproduction of audio equipment has increased at an astounding rate; so much so that the public is beginning to demand programme material of very high technical quality.

One of the major problems in fulfilling this need is being able to recreate the full dynamic range of a live performance, of say, an orchestral concert. The heart of the problem is noise, and this is most likely to be worst in the audio link between the studio or concert and the listener at home, whether it be a disc, tape or FM radio link.

For most people a quality tape system with a good signal to noise ratio is far too expensive, and of those people who have FM tuners claiming a signal to noise ratio of 70dB, how many live close enough to a transmitter to have the required 1mV or so available at the front end?

Royal noise

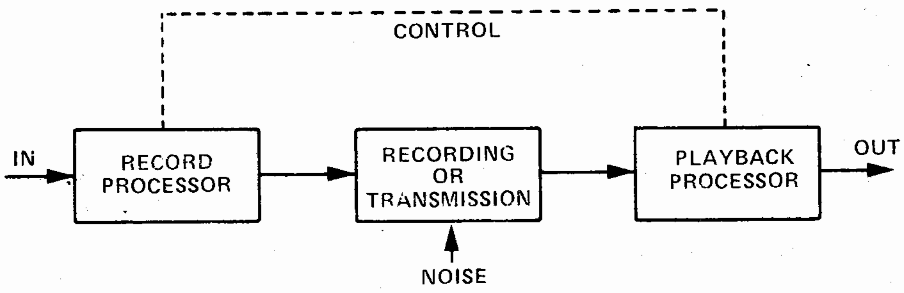
In the early days of sound broadcasting and recording, when the noise problem was much worse than today, attempts were made to reduce its effect by turning up the gain manually for quiet passages of music, thus in effect compressing the dynamic range so that even quiet passages would be above the noise floor. The BBC today still use this technique, in orchestral concerts relayed from the Royal Albert Hall.

The level of quiet passages can be raised by as much as 16dB, thus allowing listeners in poor reception areas who are listening to AM transmissions to be able to receive the entire programme without quiet passages being drowned in noise.

The average usable dynamic range of the domestic listening environment allows a usable dynamic range of a mere 65dB. This is about the same as the noise figure introduced between the studio and the listener at home. What use then are noise reduction systems?

Once a master tape has been finished copies will be made, and maybe later even copies made of the copies — but with each successive copying the tape noise will have increased by 6dB (roughly doubled). At home, the now popular 'compact cassette' system, due to its slow tape speed and narrow track width, has a noise figure of —50dB which is noticeable in any listening environment.

A block diagram of a complementary noise reduction system, showing the need for the overall control between input and output.



Being uncomplementary

Noise reduction systems are then a real necessity in the studio and can be justified in the home, and there are at present a number of systems in use, varying in complexity and effectiveness.

The broadest categories into which noise reduction systems fall, are complementary (two pass) units and non-complementary (single pass) types.

Complementary systems, such as the dolby and dbx systems, consists of two units, a processor and a deprocessor. Before being recorded, or transmitted, the audio signal is passed through the processor and on reproduction through the deprocessor.

Above the floor

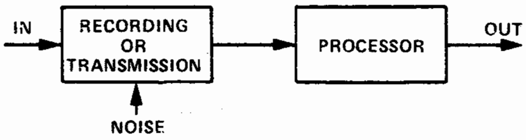
The simplest type of system is the compressor/expander type. In the dbx 122, the processor consists of an audio compressor, which will produce a change in output level of 1dB for a change in input level of 2dB. Thus, an orchestral concert with a dynamic range of 80dB when processed will have a dynamic range of only 40dB, giving the unit a compression ratio of 2:1.

The compressed signal is then recorded on a tape recorder as normal, and now, even the quietest passages (at —40db) will still be well above the noise floor of the tape.

On replay, the recorded signal is played back through the deprocessor, an expander, which restores the original dynamic range of 80dB.

When using a system like this with a tape recorder, the tape recorder must be accurately set up beforehand. If the machine has a frequency response accurate to within ±3dB, when the processed signal is replayed and expanded the frequency response will drop to within only ±6dB. Any defects in the tape will also be more noticeable — a drop out will sound twice as bad when expanded.

Noise reduction with this sort of system can be as much as 40dB, and because it works over the entire audio frequency range, will provide 'wideband' noise reduction. Unlike the dolby system, because the compression ratio is independent of relative levels, the dbx system requires no complicated setting up or calibration tapes.



Basic functioning of a non-complementary noise reduction system.

Dolby et al

Perhaps the widest known of all noise reduction systems is the dolby system. There are two versions of this system available, dolby 'A' for use in recording studios, and dolby 'B' a simpler version designed for consumer use.

The dolby A system also works on the principle of compression and expansion, but does not have a fixed compression ratio. The incoming audio signal is filtered into four bands, below 80Hz, 80 to 3kHz, above 3kHz, and above 5kHz.

As can be seen from the diagram, identical filter and compressor networks are used in the processor and deprocessor; — the only difference being the use of an adder in the processor and a subtractor in the deprocessor.

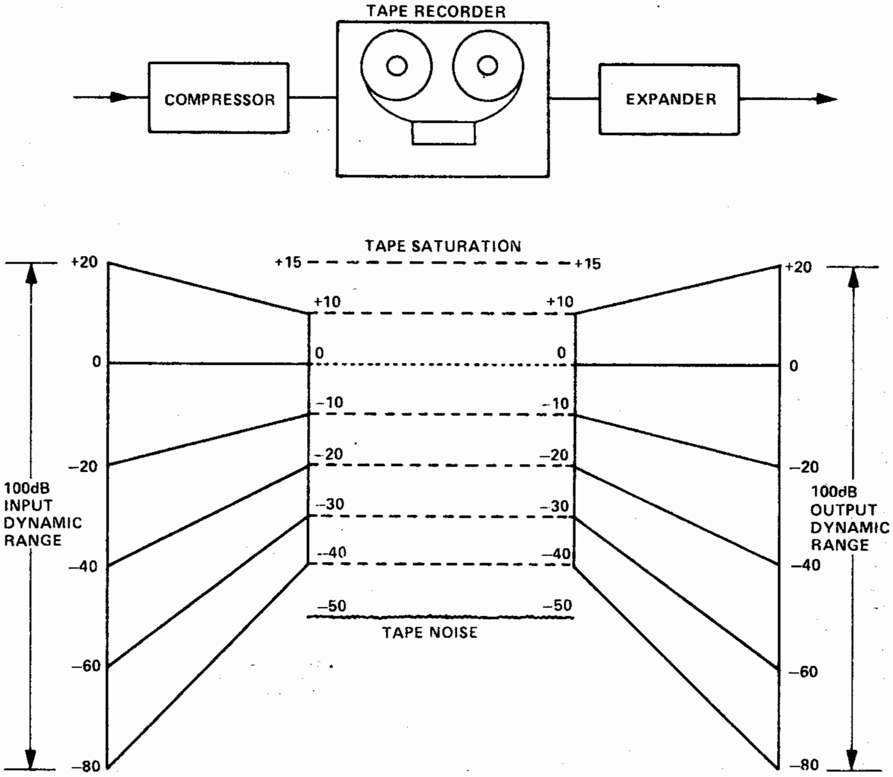
The dolby A system processor works only on high and medium level signals. If individual frequency components less than —40dB down on the operating level, they will pass through their particular band side chain without undergoing any form of compression. As signals increase in level, so does the amount by which they are compressed, varying from 0 for low level signals to up to 15dB compression for high level signals.

Banding together

Being a multi-frequency band system, the system minimises some of the problems associated with wideband systems such as the dbx, one particular advantage being that each band can have operating characteristics (such as attack and delay times) optimised reducing gain overhand and modulation products caused by too slow or too rapid gain changes.

The fact that the compression and

A compander circuit expander-compressor showing how a 2:1 compressor ratio functions in practise to give, in theory, a 100dB S/N ratio from a tape with a 50dB noise 'floor'.



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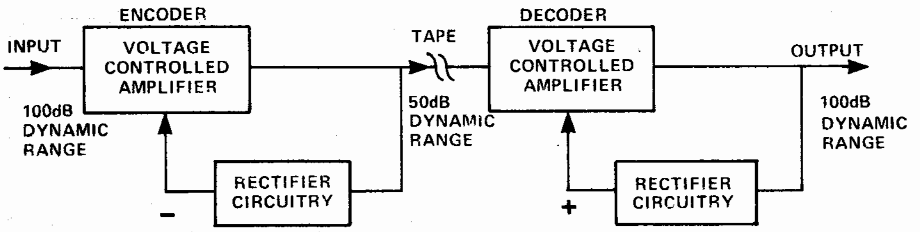


The dbx 122 unit is a refinement of the basic compander design. The addition of the rectification circuitry can be seen from the block diagram.

expansion ratios vary with the audio signal level means that in order to function properly, a tape machine using the dolby system will have to be accurately calibrated with level tapes, before use. Due to the difference in sensitivity and output level of different tapes it also is necessary to recalibrate the system when a different type of tape is used. From dolby A, the typical noise reduction figure which can be expected is 10dB up to 5kHz, rising to 15dB at 15kHz.

The simpler version, dolby B, is intended for consumer applications, and fitted as standard to many makes of cassette deck divides the audio signal into two audio bands. The system is based on the assumption that high frequency hiss will be far more noticeable than low frequency noise, and so does not process the low frequency content of an audio signal.

The compressor in the processor boosts low level high frequency signals by amounts of up to 10dB depending on the input level, giving on replay and deprocessing a reduction in noise of up to 10dB in the high frequency range. This increases the dynamic range of a cassette system from around 50dB to a max of over 60dB.

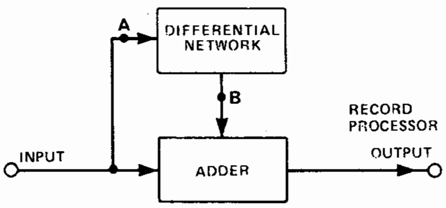
To B or A?

Dolby B, uses a non linear dynamic compression ratio and so must be carefully adjusted for the tape recorder and tape with which it is used; very few external user serviceable controls are found most on most cassette decks fitted with the dolby B system, for fear that people might incorrectly calibrate the device and produce worse results than if it were not used at all.

One of the drawbacks of complementary systems is that they will only reduce noise induced between the compressor and the expander. Complementary systems also offer no help with noisy tapes and discs recorded without being first processed, and of course, the various systems available are all incompatible so the engineer with a nice dolby system and a dbx tape is still at square 1!

A fillip to NRS

Single pass non-complementary systems offer some hope for non-encoded material. Philips DNL (dynamic noise limiter) has been around for a few years now, and was specifically designed to reduce noise in unprocessed cassette recordings (but is now also fitted in addition to the dolby system on some Philips reel-to-reel tape decks).



Top Simplified diagram of a Dolby A deprocessor circuit.

Right: The processor circuit to match that shown above.

Bottom: Diagram of the Dolby A differential networks.

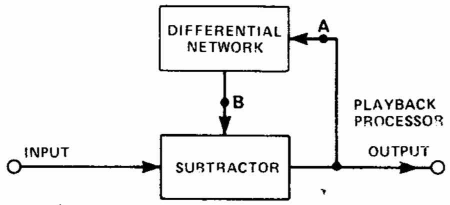
The DNL consists of a dynamic low-pass filter; noise is most noticeable in quiet passages of music (such as piano music), when most of the higher frequency signal is noise, and it is at this time that the DNL operates.

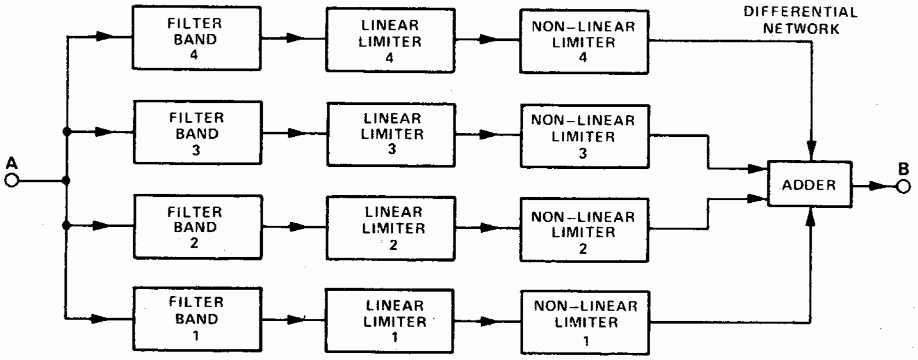
When the signal's high frequency component is strong, and sufficient to mask the high frequency noise, the cut off frequency of the filter increases and allows all the high frequency signal to pass unattenuated. When the level of the high frequency component is low, it is assumed that the noise will be dominant, and the cut off frequency of the filter is reduced, attenuating the noise, and, unfortunately some of the wanted signal.

Burwens DNF (dynamic noise filter) works on a similar principle but is much more flexible. The system senses the high frequency content of the signal. If there is a lot of high frequency energy present, the filter cut off frequency rises to 30kHz, and all the signal is passed, the high signal level masking the noise. When the signal reduces in level, so does the cut off frequency which falls to 500Hz when no signal at all is present, giving a very substantial noise reduction on blank gaps between pieces of music on tape.

Correlations!

Phase linears Auto correlator is an example of a very sophisticated non-complementary systems. It consists of a series of bandpass filters

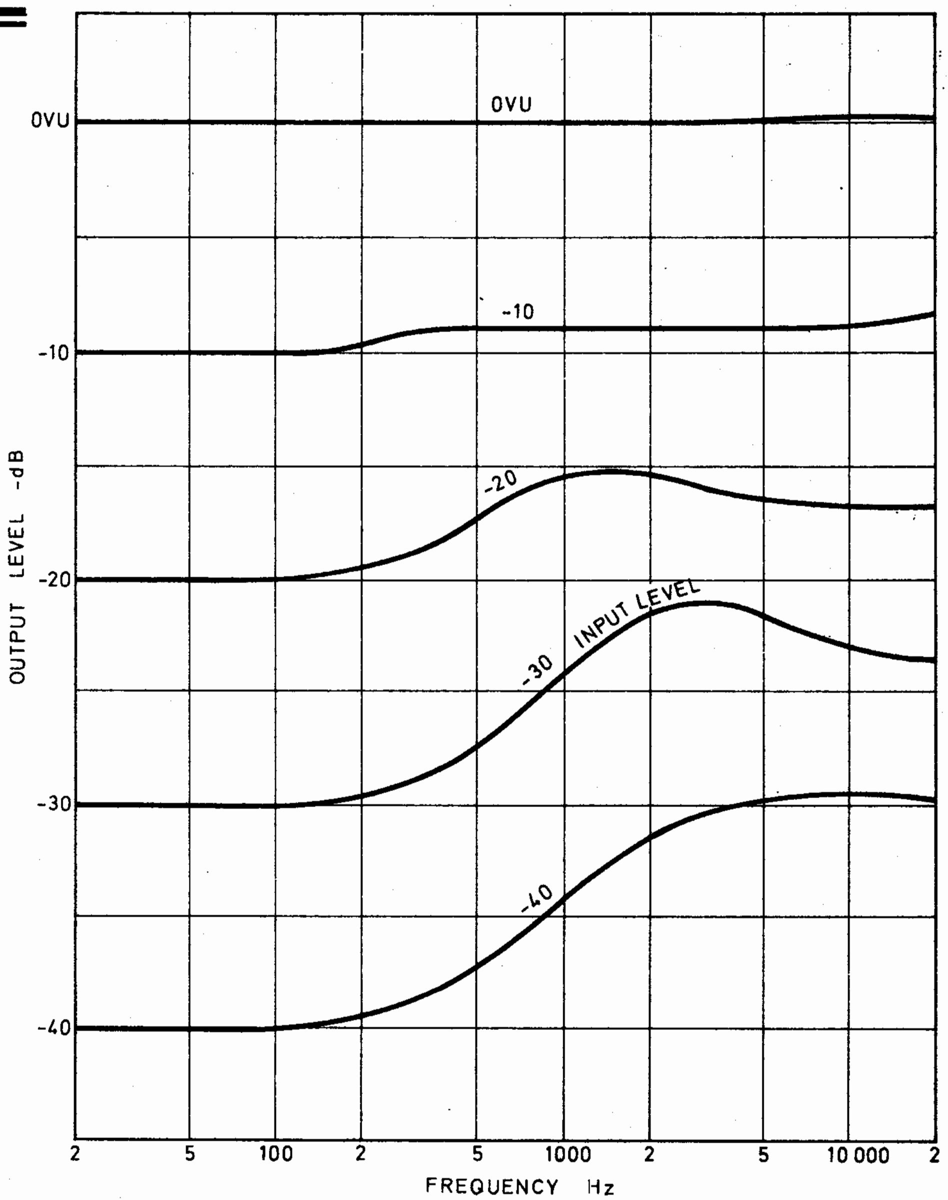




that can be opened, to allow signal in the frequency range of the filter through, or closed down to remove noise. The filters are controlled by the Auto correlation circuitry. Music contains mathematically related tones and is highly coherent (or correlated) in nature, whilse noise tends to be random and has a low correlation coefficient. What the Auto correlator does is to calculate the correlation coefficient, and use it to determine whether a signal is noise or music. If a particular signal is determined to be musical in nature, the appropriate filters are opened up to allow it, and its harmonics and overtones through.

If the correlation is low, the signal is treated as noise and the appropriate filters are activated preventing noise to pass through.

These three non-complementary systems, because of the way they operate will inevitably have an effect on the wanted audio signal, but if carefully set up can give a substantial reduction in noise with only a minor effect on the program content, and it must not be forgotten, that while two-pass systems only reduce noise induced between the processor and deprocessor, a non complementary one pass system can effectively reduce noise generated anywhere in the audio chain, from the studio to device itself.



Variable compression ratio of Dolby B system; note how frequencies below 100Hz are unaffected.

Snip the crackle and pop!

One recently developed type of noise reduction system is the SAE model 500 Impulse Noise Reduction System has been designed to remove unwanted clicks and pops caused by scratches in discs.

The SAE device recognises clicks and pops by their fast rise and fall times. If a signal has a fast rise and fall time, it is thus assumed to be an unwanted 'click', the click is removed and to prevent a period of silence disturbing the continuity, a small section of the preceding music is inserted in its place. This feat is accomplished using an analogue delay unit providing a delay of a few milliseconds, as the pop 'click' transient is about to enter the delay, the output of the device switches from the input of the delay to the output. Then, just before the click is about to leave the delay, the output of the unit again switches back to the input of the delay, thus removing the click. The typical click only has a duration of about a milliseconds, and so the switching to the delay goes unnoticed, but the click vanishes.

Studious future

What the future holds in store for "noise" is anyones guess; within a decade we may see digital tape-recorders as the standard in studios — using digital recording techniques will, if the sampling rate is high enough, eliminate noise induced in tape systems (or in broadcasts if we ever see pulse code modulation for broadcast stations) altogether, and will allow tapes to be copied any number of times without any extra noise being introduced. Already, today it is possible to buy add on units for video recorders to enable audio signals to be digitally recorded onto tape!

For the moment, however, for the engineers in recording studios using multi-track machines where track width is small, and for the hifi enthusiast at home with his cassette system, noise reductions systems are a necessity, and will remain so for some time. **ETI**

Simplified diagram of the workings of the SAE Impulse Noise Reducer. The 'switch' is the actual unit, which transfers the output connection from A to B when a click is present to eliminate the sound.