Dolby® SR

Dolby® spectral recording

What It Is and What It Does

Dolby Spectral Recording
Dolby spectral recording (Dolby SR) is a new professional studio mastering system that yields recordings with exceptional purity of sound. Several important technical advancements are introduced in the new system. One is a substantial extension of available headroom, which allows the use of a uniformly high maximum recording level at all audio frequencies. Another is the practical elimination of the influence of noise and non-linearity on the reproduced sound. These advancements are achieved by new circuit functions, adaptive to the signal spectrum, and by the consistent application of minimum processing to the signal: *the principle of least treatment.*

Recording with Dolby SR is extremely simple. It can be used with any modern professional recorder; installation is rapid, operation is simple and reliable, and routine maintenance is unnecessary. Its unique Auto Compare feature allows immediate, dependable validation of recording system performance without additional instruments or delays in the studio work schedule. Plug-in module construction and compatibility with existing Dolby mainframes provide convenience and economy for studios that use Dolby SR.

The principle of least treatment

The ear is the final destination of all audio signals, and the most sensitive instrument for their analysis. An ideal audio device or system would impose no audible limitation on the signal passing through it. The design of Dolby SR has been carried out with close attention to the properties of human hearing, especially the need to prevent any audible artifacts of signal processing.

At the lowest signal levels, or in the absence of a signal, Dolby SR applies a fixed gain/frequency characteristic that reduces noise and other low-level disturbances by as much as 25 dB. Only when the level of part of the signal spectrum increases significantly does the circuit adaptively change its own spectral characteristics. When this happens, Dolby SR changes gain only at frequencies where change is needed, and only by the amount required. Adherence to this principle, the principle of least treatment, is critical to maintaining the extreme purity of sound audible in Dolby SR recordings.

Listening is the most demanding test

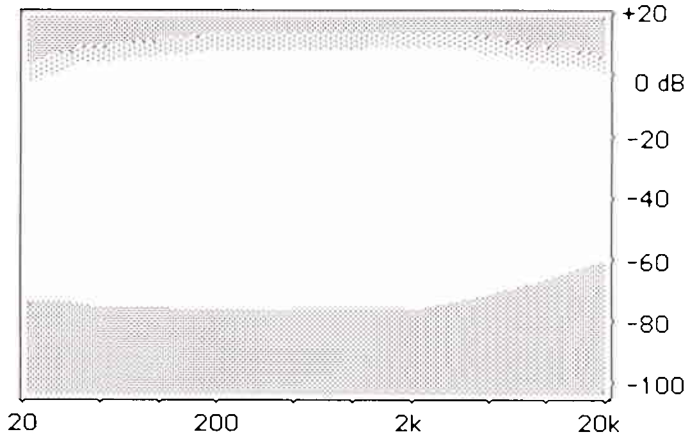
Laboratory measurements using test signals show that Dolby SR recordings contain very little noise, distortion and other impurities. However, meters cannot tell us how good Dolby SR tapes sound, because equipment does not respond to a recorded signal in as complicated a way as the ear and brain. The most important and reliable test of any signal processor is a careful comparison of line-in and line-out signals while a live recording is made in a quiet studio. We urge engineers, producers and recording artists to carry out such tests with Dolby SR, and to compare their own Dolby SR recordings to those based on any other technology.

Why Dolby SR is needed

All recordings and communication systems have definite dynamic range limits. However, a simple measurement of maximum level and noise level does not reliably indicate how recordings made with such a system will sound. Such a test says nothing about noise that appears only in the presence of a signal, or about system behavior when the signal is at the overload level. Analog tape, for example, saturates gently; digital recordings, on the other hand, clip fully if maximum level is reached even for less than a millisecond. Because Dolby SR increases recording headroom considerably, there is less risk of under- or over-recording. The engineer's working space is increased, and there is greater freedom for creative effort. Dolby SR provides effective protection during original recording, during mixdown, when equalization or specialized signal processing requires the lowest possible noise level, and when multi-generation copies are needed.

The analog tape recording window

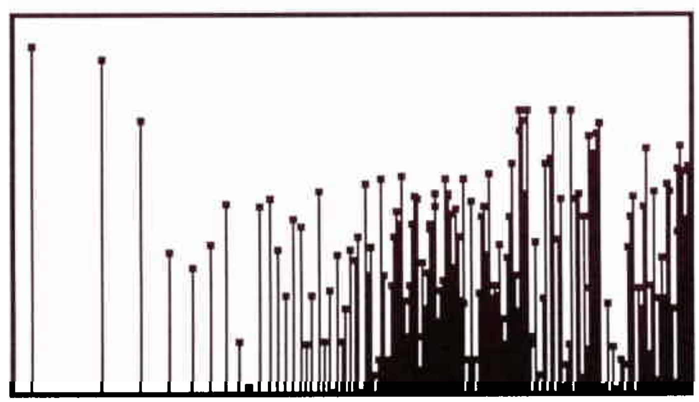
1



The limits of unassisted analog recording, using a standard professional recorder and tape at 15 ips, are sketched in Figure 1. The limit at high signal levels, actually a gradual overload, is at the top of the clear area. The noise level is the bottom of the clear area. Both the overload level and the noise vary with frequency. The central, open part of the sketch can be thought of as a window through which the signal must fit if it is to be recorded.

Recording a signal

2



A music program constantly changes in level and frequency content. A moment in such a program might have the spectrum shown in Figure 2. The signal level varies with frequency. The gain setting that would give the best recording at middle frequencies would cause the high frequencies to overload. One objective of Dolby spectral recording is to achieve as nearly optimum a level as is possible at all audio frequencies.

Finding the right level

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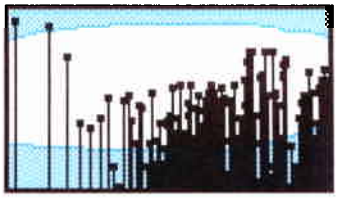

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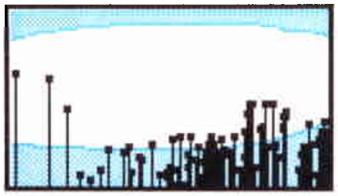
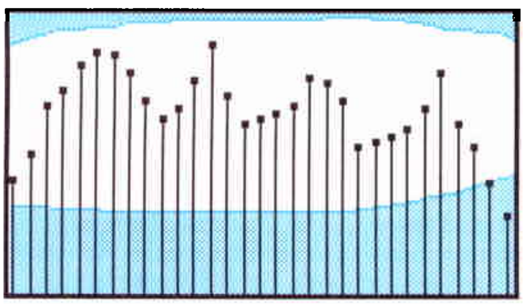
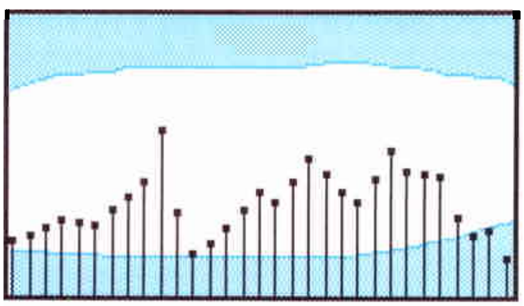


Figures 3 and 4 show what happens when the recording gain is adjusted; the signal spectrum moves up or down in the window. Even when the recording gain is set at the highest safe value, as in Figure 3, much of the capacity of the recording system, that is, the *spectral* space above the tops of signal components and inside the window, is unused. Dolby SR makes use of this capacity.

The simple compander



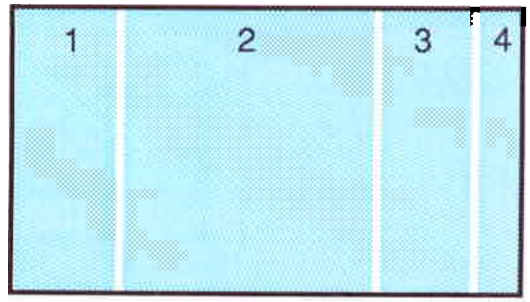
5a



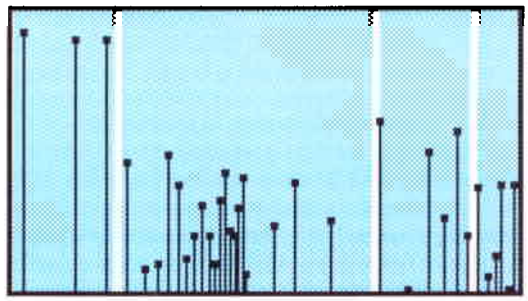
5b

The simple or broadband compander (compressor and expander) was first used in attempts to increase the dynamic range of recording systems. During recording, such a compander increases gain when the overall signal level is low; some companders also reduce gain when signal level is high. During playback, the compander's action is reversed; high-level signals, regardless of their frequency, cause all frequencies to be played back at a high level (Figure 5a), while low signal level, or no signal, causes playback gain to drop (Figure 5b). The *measured* dynamic range of a recording system may seem to be increased greatly. However, a simple compander meets its dynamic range specifications only when no signal is recorded. When a real program is recorded, the compander is often at rest when it is needed most. When it works, on the other hand, critical listeners can hear artifacts, like "pumping" modulation of the background noise or signal, as well as limited transient performance.

Dolby A-type noise reduction



6a



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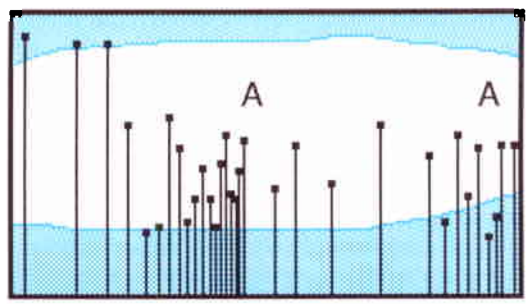

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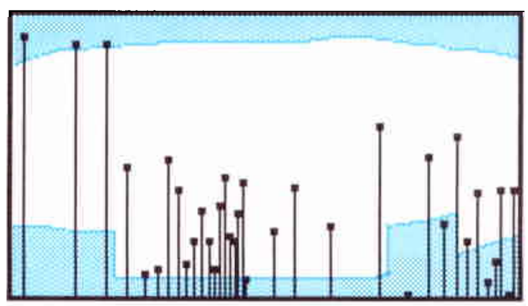

6d

No professional signal processor has ever been as widely used as Dolby A-type noise reduction. This is still true now, twenty years after its introduction. Dolby A-type processing substantially improves the efficiency of magnetic and optical recording media and audio communication systems. Dolby A-type signal processing relies on compression and expansion, but only at low signal levels, and separately in four frequency bands (Figure 6a). The signal components in each band (Figure 6b) are integrated; if this level is below a fixed threshold, it is boosted during recording (locations marked "A" in Figure 6c), and attenuated during playback (Figure 6d).

The boost used in Dolby A-type noise reduction is 10 dB across most of the audio band, increasing to 15 dB at very high frequencies. To improve recording further, it is not enough simply to increase these figures; the boost must conform more closely to the signal spectrum than is possible in a four-band system.

The auditory window

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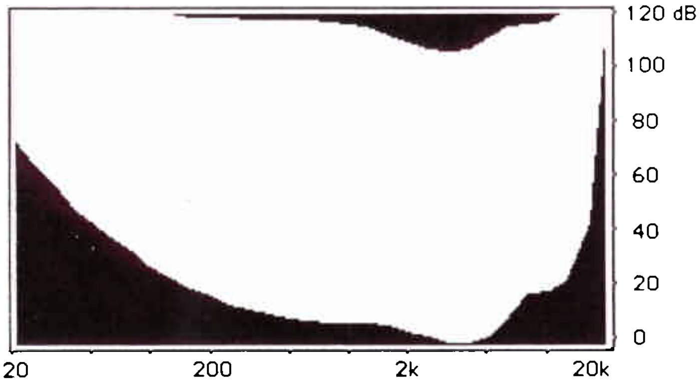

One way to define ideal sound reproduction is to show the limits of the human hearing system as a window, as we did for analog tape recording in Figure 1. Such an auditory window is sketched in Figure 7. The top of the graph corresponds to 120 dB, a continuous sound pressure level that is unbearably loud; at 5 kHz, sound changes to pain at about 110 dB. It is safest, of course, to leave a margin between such a level and the highest continuous sound level one aims at reproducing.

The boundary of the window near the top of the plot is 6 dB below the threshold of pain at each frequency.

The bottom of the window is the *threshold of hearing*; sounds at this level are just audible to a listener with sensitive hearing. The level of the background noise in a *very* quiet recording studio may be 10 to 15 dB. A recording system with a window like that shown in the figure could be played back without audible noise or overload, even if the highest-level signals were literally at deafening levels.

Playback of an analog recording

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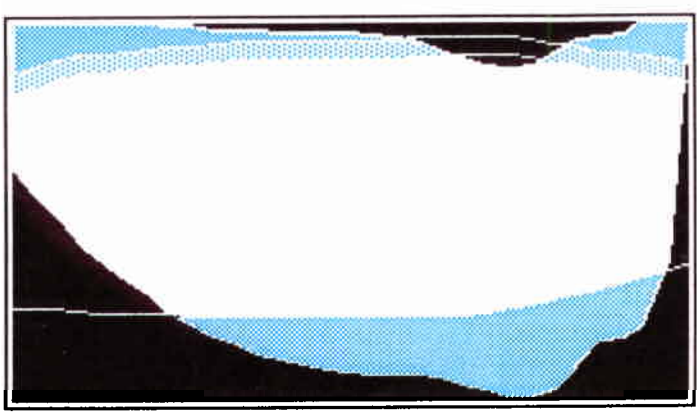

We can learn more about tape recording by superimposing the analog tape window on top of the auditory window. Sliding the analog window up or down corresponds to playing a tape at higher or lower level. In Figure 8 we have set the playback gain so that the maximum level signal that can be put on the tape lines up with a continuous sound pressure level of 110 dB. We will stay with that setting as we look at various recording system windows.

Several interesting facts are visible in this figure. One is that the noise of the tape will be audible only in a restricted range of middle frequencies, that is, where the auditory threshold is lowest. This is because noise or distortion components at higher and lower frequencies, even if only slightly below the threshold, are totally inaudible. Another observation is that if the audible noise in the mid-range frequency band could be reduced by 20-25 dB, no noise would be heard at all.

Disturbances produced by the presence of a signal

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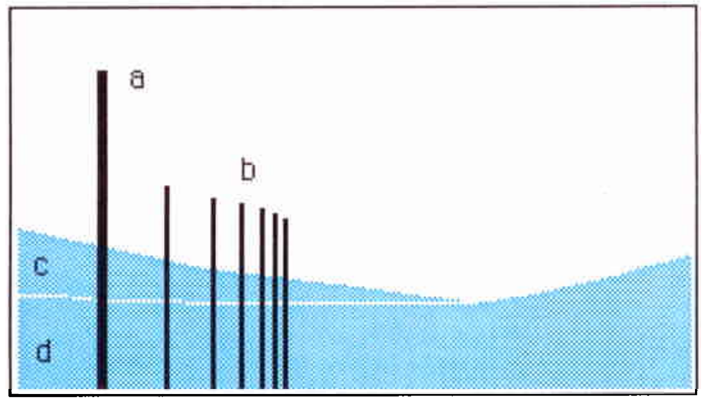

When no signal is present, the only low-level defect that can be measured is tape hiss. However, in the presence of a signal, the analog tape recording window closes further as other artifacts are added to the signal, layer by layer. Figure 9 shows several components of noise and non-linearity that can appear in the presence of a signal. The signal is shown as the vertical bar at (a), and is at a level that causes 3% harmonic distortion. These harmonics are shown in correct scale at (b). Modulation noise, which appears only when a signal is present, is spread over a wide range in the spectrum (c). The bottom layer of noise, tape hiss (d), is caused by statistical fluctuations in magnetic domain orientation in the tape coating.

Available dynamic range

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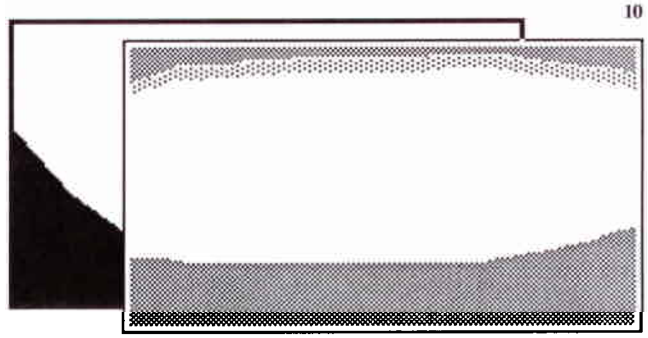

Another interesting fact is shown in Figure 10. Analog recording might be thought to be deficient in dynamic range at very low frequencies because saturation occurs at lower tape flux levels in that part of the spectrum. However, the opening in the analog window at low frequencies is actually *larger* than the opening in the auditory window. The same is true at high frequencies. If signal components at different frequencies were simply recorded at different gain settings, the effective dynamic range of analog recording could be extended considerably.

What Dolby Spectral Recording Does

Dolby SR at low levels

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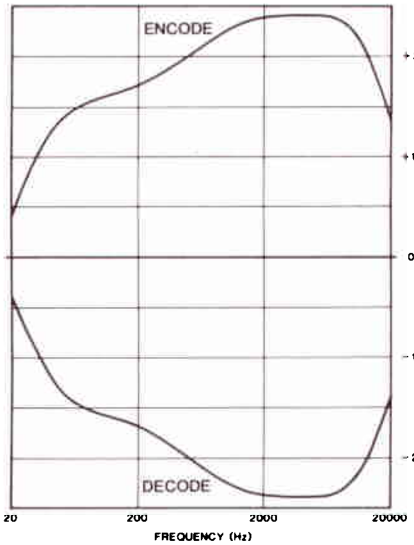
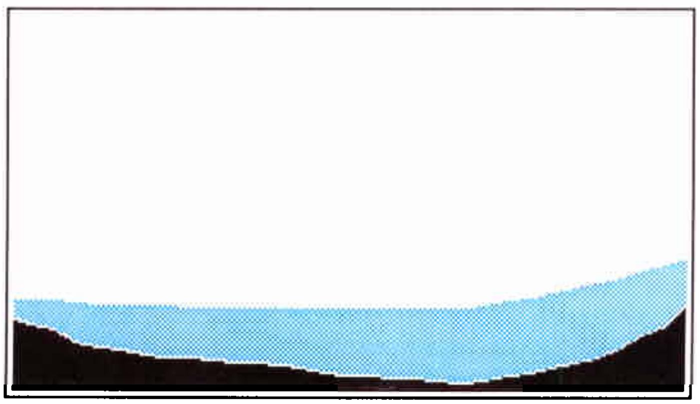


11b



The simplest way to suppress noise and other low-level recording defects is to use as high a recording gain as possible. From what we have already seen, an even better idea would be to use the optimum gain in each part of the spectrum. This is what Dolby SR does at very low signal levels. The result is a form of fixed equalization that does not change as long as the signal level stays below a certain threshold. When the recording is played back, the same equalization is applied in reverse, and any background noise is lowered by the same amount (Figure 11a). The upper curve in Figure 11b is the noise level of a typical professional tape at 15 ips. The lower curve shows the change in this level that results from the use of Dolby SR fixed low-level equalization. Since there are no dynamic changes taking place – the equalization is fixed – no dynamic side effects, audible or inaudible, are possible.

Dynamic action of Dolby SR at moderate levels

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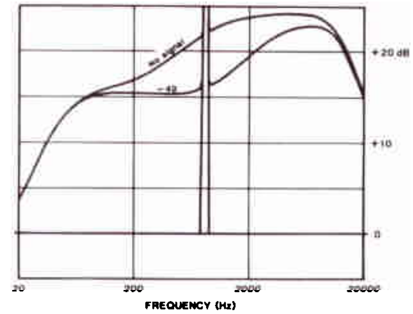

If the signal increases in any part of the spectrum, some adjustment of gain must be made to ensure that overload will not take place. This is done in a very gradual way by Dolby SR, so gradual that there is no danger of producing audible modulation of the signal or any other audible effect.

The principal mechanism of Dolby SR is a group of ten fixed- and sliding-band filters with gentle slopes. Those with fixed bandwidth are electronically controlled to vary their gain; those with fixed gain can be adjusted to cover different frequency ranges. By selecting and combining from the group, the Dolby SR control circuit can create an infinite number of filters through which the signal must pass before it is recorded. During playback, filters are automatically created that are the exact opposite of those used during recording.

Filter selection and adjustment is controlled by a continuous analysis of the signal spectrum and a process called *action substitution.* Action substitution determines which of the two types of filters will predominate and how each must be adjusted to produce the optimum composite filter (Figure 12). Even when the signal level increases substantially, the system is designed to deviate as little as possible from the fixed characteristic shown in Figure 11, in accordance with the principle of least treatment.

Dolby SR action at high levels

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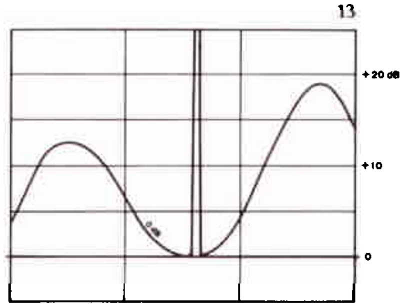

When a high-level component appears in the signal spectrum, Dolby SR assumes the kind of characteristic shown in Figure 13. In this example, a single tone at 800 Hz and at a level of 0 dB has been applied to the system input. Dolby SR reduces recording gain, but only at and near the frequency of the tone, and only by the amount needed to prevent overload. Above and below this part of the spectrum, the Dolby SR curve returns to the fixed, low-level characteristic. This action has results that are especially impressive when listening to a Dolby SR recording *without* decoding. Although bright in sound, the program does not appear to have been subjected to any dynamic processing. Since the only other system action consists of fixed equalization at very low levels, it is not surprising that during decoded playback, no trace of processing can be heard, except for remarkable clarity of reproduction.

Masking and Dolby SR

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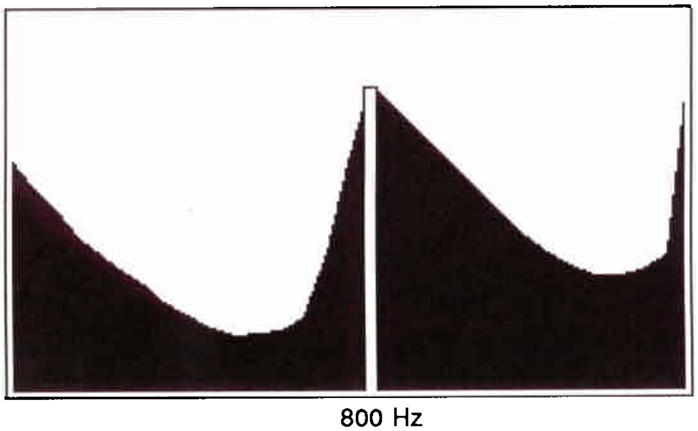

The changes that take place in the Dolby SR circuit are adaptive; that is, the system filter always adjusts itself to maintain the highest practical gain at every frequency as the signal *spectrum* changes. The human ear and brain also respond to these changes in the signal spectrum; one such response is a form of signal processing known as *masking*, one of the most extensively studied aspects of hearing.

Masking is the concealment of a low-level sound by a sound higher in level. A similar effect takes place in vision, when the daylight sky makes the stars disappear.

Most audio signal processing systems operate quite independently of the behavior of the ear and brain, and always take the same action that they would if nobody were listening to the output of the system. The extraordinary purity of sound of Dolby SR recordings is due in large measure to an elegant cooperation of adaptive signal processing and auditory masking.

In masking, a high level signal component raises the auditory threshold above and below the signal frequency. Sounds lower in level and near it in frequency disappear completely in this psychoacoustic "shadow" (Figure 14). In the Dolby SR circuit, feedback of the signal characteristics determines how each filter in the Dolby SR circuit must change to most closely envelop the masking "shadow." This is the way that Dolby SR applies as much gain as possible everywhere in the spectrum. The only region of the spectrum that is not boosted in gain is the region that is controlled by masking, where audible low-level information does not exist. Dolby SR electronic signal processing is silently traded for auditory signal processing in that part of the spectrum. It could fairly be said that although most of the Dolby SR system is on the circuit board, some of it is in the human brain.

Immunizing the system to errors

Another feature of Dolby SR is *spectral skewing*, which reduces the level of the incoming signal at extremely low and extremely high frequencies. Spectral skewing desensitizes Dolby SR to minor aberrations in tape-to-head contact and azimuth alignment, which might cause fluctuations in high-frequency response, and to head bumps or low-frequency variations in alignment tape levels. Although these effects are often inaudible, they can disturb the operation of other signal processing systems.

Recording complex signals at high levels

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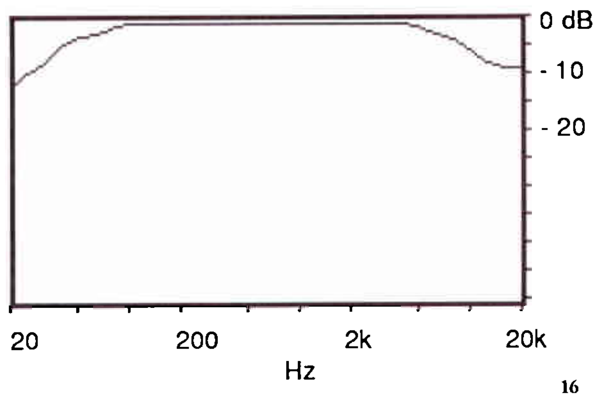

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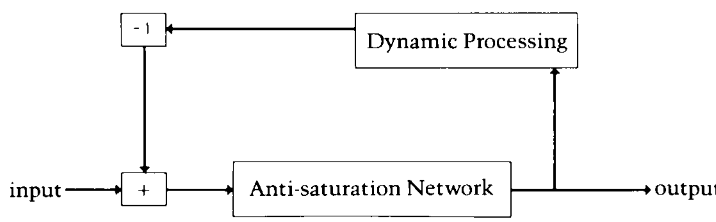

Headroom is as important as any other property of a recording system. Analog tape recording, as engineers know, has a gradual or "soft" clipping characteristic. In digital recording, there is no saturation region at all; there is simply clipping, in which the same digital "word" is recorded over and over as long as the signal remains above the limit. The Dolby SR circuit contains a feature designed specifically to deal with extremely high levels at low and high frequencies. The anti-saturation characteristic is sketched in Figure 15, and the circuit configuration shown schematically in Figure 16. Low-level signals pass through the side chain for processing; as signal level increases, an increasing proportion of the signal follows the direct path, which applies no dynamic signal processing. By introducing attenuation of high and low frequencies in this path, a significant increase in headroom is provided, further maintaining signal purity, with negligible effect on low-level operation.

Comparing Dolby spectral recording to other methods
We can compare the *static* performance of different recording systems by superimposing their windows and the auditory window (Figure 7). Any limitation that might be audible will appear as an obstruction that reduces the size of the opening in the auditory window. It is important to remember that this method of comparison does not show audible *dynamic* effects, such as modulation noise of analog systems; nor does it show low-level non-linearities, non-monotonicity, or effects of d.c. asymmetry, all of which may occur to varying degrees in digital recording systems. These effects all close the corresponding system windows from the bottom when a signal is present. In the figures that follow, for unassisted analog tape and Dolby SR, zero level is 320 nWb/m; for digital recording, zero level is 10 dB below the absolute clipping level. In all three examples, the relative vertical positions of the windows have been chosen so that the maximum recorded level will be presented at an acoustic level of 110 dB during playback.

Analog recording (no signal processing) [Figures 17,18]
Unassisted analog tape recording shows characteristic limitations in available headroom at low and high frequencies and a substantial level of noise at mid-frequencies.

Digital recording [Figures 19,20]
A typical digital recording provides performance that is better than unassisted analog tape in several obvious ways. The main drawbacks are the hard clipping barrier of digital recording and the disadvantageous spectral distribution of noise. Although the measured dynamic range of a digital recording system may exceed 90 dB, the noise level is not uniform with frequency. The noise level is extremely low at very low frequencies, much lower than the noise of analog tape, which is already more than adequate at low frequencies. However, digital system noise crosses the auditory threshold precisely in the spectral region where the ear is most sensitive. The usable improvement in noise level, especially in the presence of a signal, is not as great as theory predicts. Because the noise generated by a digital system is not random, and is therefore especially noticeable to the ear, it is normally masked by the addition of "dither" noise, elevating the final noise level.

Dolby SR [Figures 21,22]
This data was obtained using standard tape and a widely used professional recorder operating at 15 ips. The noise at the very bottom of the window could not be heard in a recording studio or control room unless the playback gain were increased considerably; under those conditions, maximum peak levels would approach or surpass the auditory threshold of pain. Played back at very high levels for test purposes, the audible noise floor of a Dolby SR recording is normally the noise of the microphone amplifiers, console electronics, or electronic instrument amplifiers. In a studio, with playback gain set as shown in the figure, and no signal present, the background noise of a Dolby SR recording is below the threshold of human hearing, and cannot be operationally improved.


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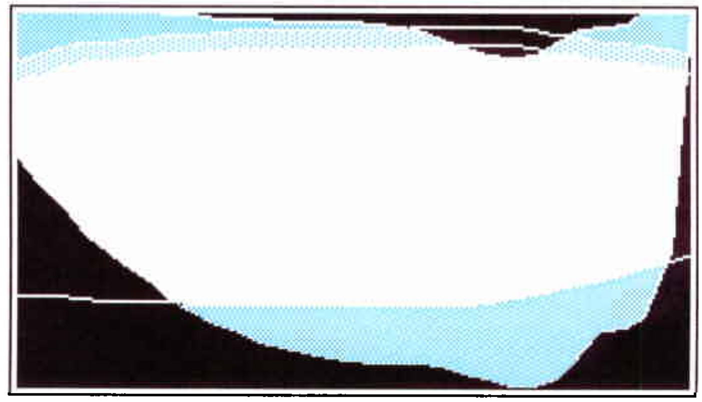
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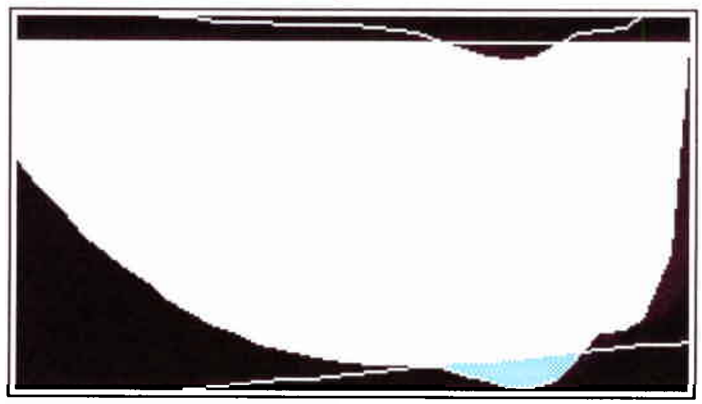
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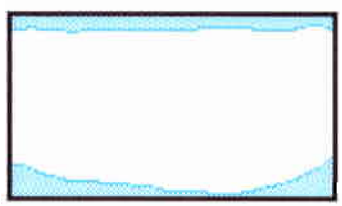
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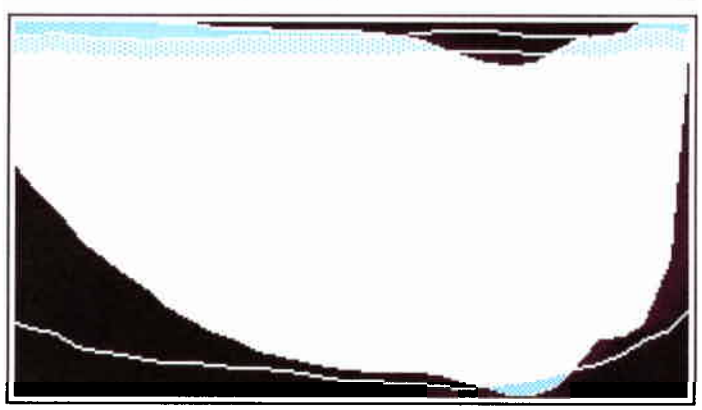
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Dolby SR works with every modern recorder

Dolby SR processing can be used with any modern professional analog tape recorder or high quality audio communication system. This means that nearly every recording studio and communications facility in the world is already equipped to install and use Dolby SR. Often, changing over to Dolby SR will only require the removal of the Dolby A-type noise reduction modules already installed and their replacement by Dolby SR modules of the appropriate type. Its practical design makes recording with Dolby SR easier than recording without it; and editing, mixdown, copying, maintenance and other studio procedures are generally simpler because of the features of the new system.

Full information on Dolby spectral recording equipment is available on request from Dolby Laboratories, Inc.

Technical note

The following list contains sources used to prepare the illustrations in this booklet and suggestions for further reading. Copies of publications marked (*) are available on request from Dolby Laboratories, Inc.

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