Dynamic Range Modification:

Limiters, compressors and expanders can solve many of today's audio processing problems—just so long as you understand how they operate.

By Richard C. Cabot, P.E.

The limitations of the available media have long been a problem in recording and producing high quality audio. The dynamic range of the human voice or of common musical instruments is much larger than that of conventional discs and tape. Typical vinyl discs and analog tapes provide 60dB to 70dB of dynamic range between the noise floor and the maximum output level. Music can easily exceed a 40dB range between soft and loud passages, yielding only a 20dB S/N ratio on the low-level program material.

Devices were developed to *compress* the dynamic range of signals, making the loud sounds softer and the soft sounds louder. This reduced dynamic range fits comfortably on tape, without the low levels being lost in noise and the high-level sound causing distortion.

Unfortunately, some audio processors actually create problems when dealing with the dynamic range of the human voice and musical instruments. In the case of vocals, these problems include amplifier overload, pickup of room noise and excessive sibilants in speech.

Gain changing

Dynamic-range changing devices provide a signal at their output, which *supposedly* differs only in level from the original signal. The device's gain changes with the signal level as neede[d] to alter the dynamic range; the wave[-]form shape theoretically remains th[e] same, but its size (voltage) is made large[r] or smaller as necessary.

The ideal compressor or expander i[s] like a skilled, and very fast, mixin[g] engineer who rides a fader on the signal[,] correcting for undesired changes i[n] level. Nothing is changed about th[e] signal but its level.

Because the characteristic of interes[t] for such units is gain, their steady-stat[e] operation may be described with a grap[h] of input level vs. output level. On a lo[g-] log scale (dB output vs. dB input) we ge[t] a graph similar to Figure 1, which is com[-]monly referred to as a *transfer curve*. Fo[r]

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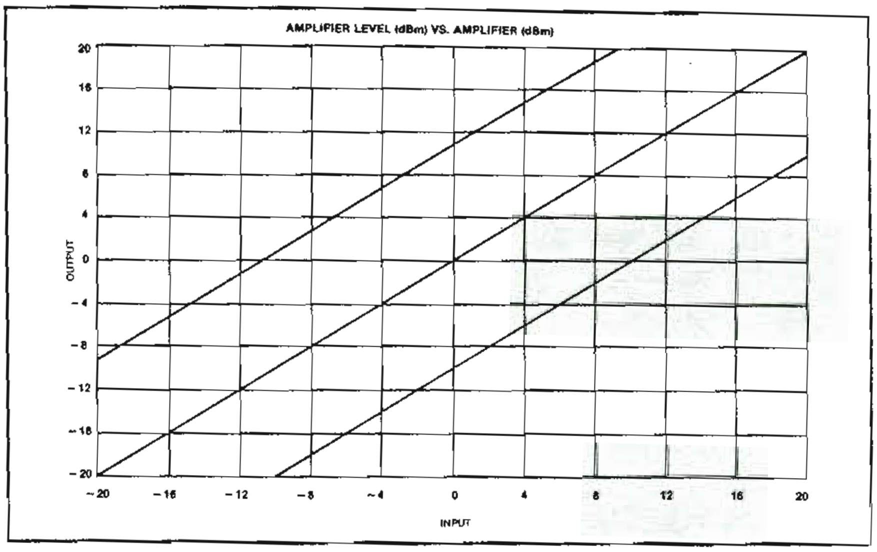

Figure 1. *A basic transfer curve, showing output in dBm vs. input in dBm (log plot). A conventional amplifier produces a line at a 45° angle, with gain determining the actual position on the graph.*

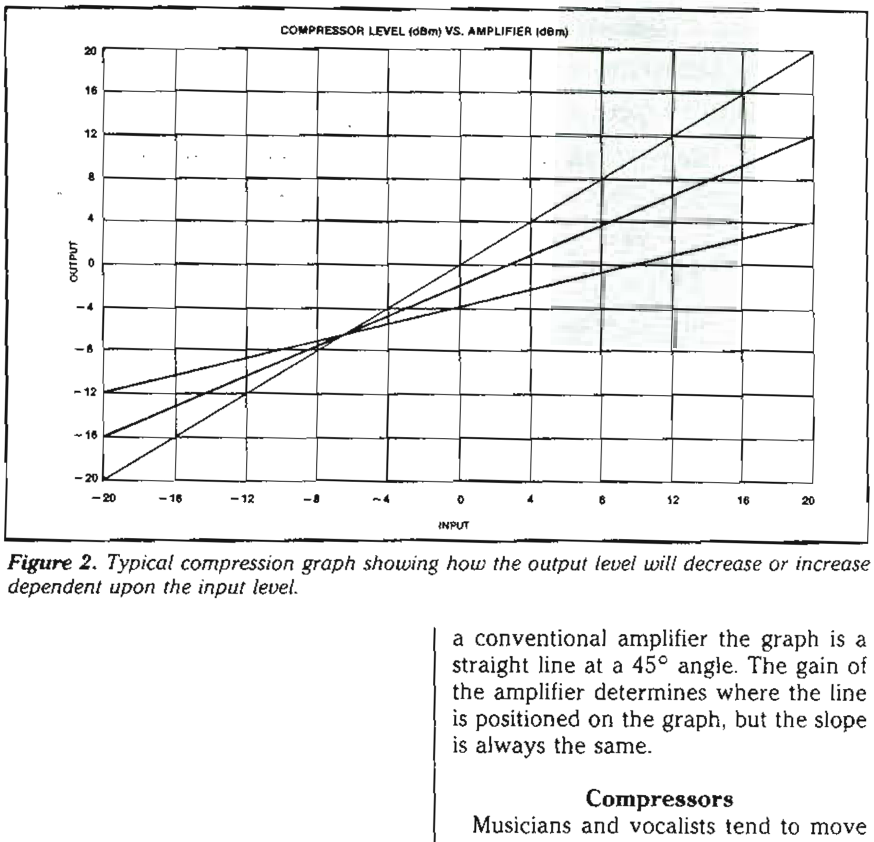

Figure 2. *Typical compression graph showing how the output level will decrease or increase dependent upon the input level.*

a conventional amplifier the graph is a straight line at a 45° angle. The gain of the amplifier determines where the line is positioned on the graph, but the slope is always the same.

Compressors

Musicians and vocalists tend to move around when working into a microphone, causing the signal levels to fluctuate. Sometimes this movement is unavoidable, as with a live performance, and sometimes it is simply due to the vocalist's movement during a take. The solution to such problems usually must be found in the audio processing chain.

When mixing a song for today's competitive airwaves it is often desirable to add more "punch" to the sound in an attempt to capture the listener's attention. The basic philosophy is to make the music fit the listening medium. Often a single's mix, which will get a lot of AM radio play, will be different from the album mix, which will be played on FM or on quality home systems.

This development has led to limiters and compressors that operate in many unusual ways, each optimized for one particular recording application or playback medium.

What we need is a device that will increase the level of the soft sounds and decrease the level of loud sounds, a characteristic that is shown in Figure 2. As the input signal amplitude increases, the output signal amplitude increases by a smaller amount. When the input signal amplitude decreases, the output signal amplitude decreases by a smaller amount. There is always a point where the input level equals the output level, called the unity-gain point.

The slope of the curve is called the *compression ratio.* A compressor whose output level increases by 1dB for every 3dB of input level increase is said to have a 3:1 compression ratio.

The classic block diagrams of a compressor are shown in Figure 3. Compressors come in two flavors: *feedback* and *feedforward.*

The feedback-type compressor is the oldest and most common of the two varieties. In this design, the output signal level is sensed and an appropriate voltage fed back to the gain-control element that precedes it. As the input level is increased, the output level tries to increase. This is sensed by the level-sensor circuit, which in turn drives the gain-control element in an effort to reduce the amplitude. Changing the gain after the level sensor changes the slope of the compression characteristic.

Such circuits are easy to build, and are self-correcting for errors in the gain element or level sensor. However, they guarantee that the output will overshoot its final value when the input level is suddenly increased. This is because the circuits do not see any change in level until the output of the compressor changes, by which time the overshoot has occurred.

Feedforward designs sense the input level and generate the necessary control voltage for the gain element to make the output level change as desired. Because the level sensor sees the signal the instant it is applied; this approach reduces the overshoot problem. However, feedforward designs place more stringent requirements on the accuracy of the level-sensing and control circuitry. Any inaccuracy in these circuits results in a corresponding error in the gain setting. There is no feedback loop to reduce the error.

The graphs shown in Figures 1, 2, 4, 6 and 8 are all straight lines on linear dB scales. This characteristic is obtained with level sensors that output a voltage proportional to the dB signal level. Voltage-controlled amplifiers (VCAs) also exhibit similar characteristics, the gain in dB being proportional to the voltage on the control input. It is possible to design a compressor or limiter that has non-linear curves, although they are not very common today.

Solid-state circuits lend themselves well to the log and antilog functions required for linear dB action. It is possible, however, to build a compressor with level sensors and control elements that are not linear dB action. Such circuits are commonly used only in specialized noise-reduction systems, and in simple feedback-type limiters. In any case, the principles are the same.

Compressors can be quite complex devices. Some provide multiband operation in which the compressor divides the frequency spectrum into several bands and then processes each one separately. Although this approach produces a subjectively louder sound, the result is a frequency response that varies according to output level.

Limiters

Sometimes a recording or production engineer encounters a signal that is normally fairly constant in level, but which occasionally increases suddenly, causing the system to clip or distort. Examples include kick drum, certain vocals, synthesizers and cymbal crashes. To correct such signal fluctuation requires a limiter, a device that operates as a normal linear amplifier for signals below some preset input level but becomes a compressor for signals above that level.

Some units found in the studio are composed of both a compressor and limiter. The compressor is used to reduce the signal's dynamic range, while the limiter prevents tape saturation.

A transfer curve for a typical limiter is shown in Figure 4. The level at which the limiter changes from unity-gain operation to compression is referred to as the

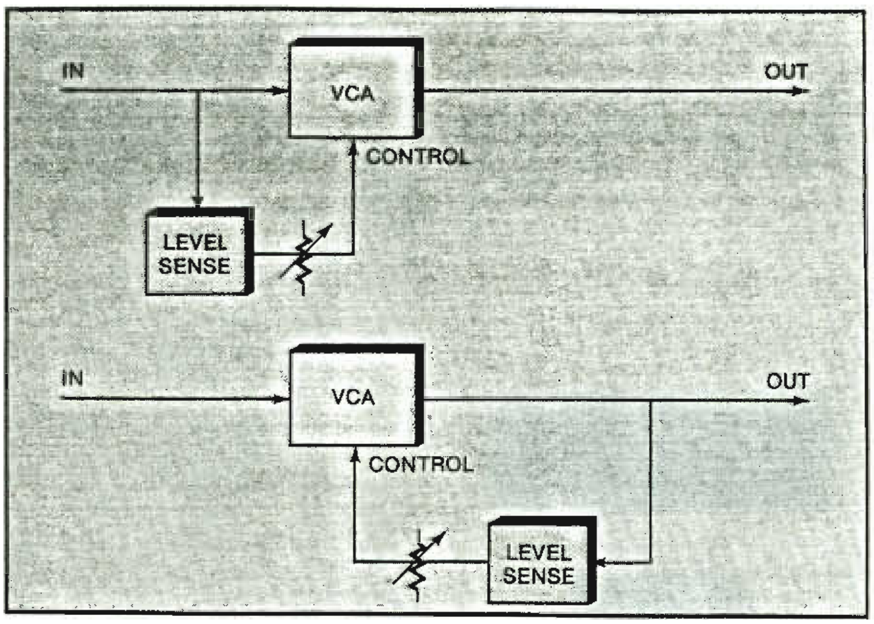

Figure 3. *Two basic compressor block diagrams: a typical feedforward compressor circuit (top); and a feedback compressor circuit (bottom).*

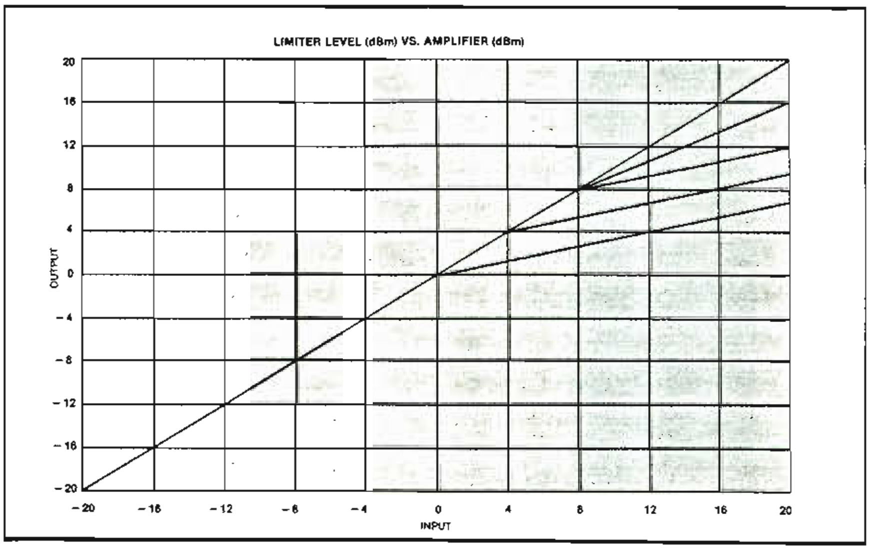

Figure 4. *A transfer curve for a typical limiter, showing the gain reduction in the output signal above the threshold level or turnover point.*

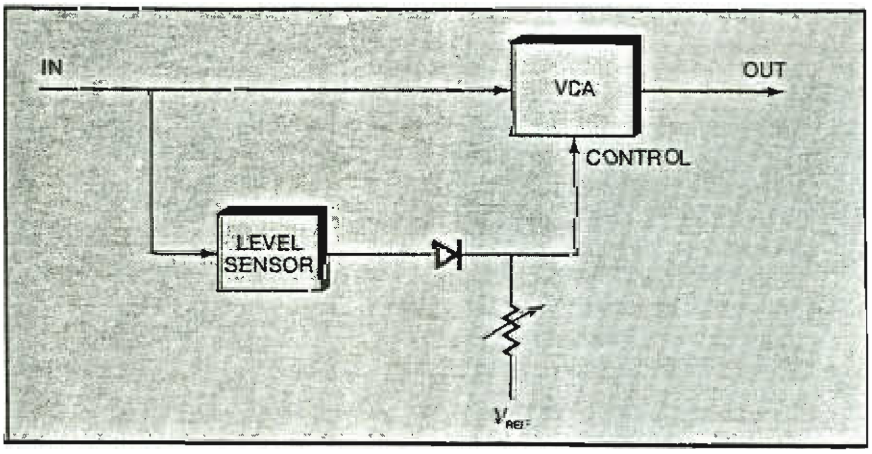

Figure 5. *A basic limiter block diagram. A typical compressor circuit may be essentially converted to a limiter by the addition of the diode as shown, for transistor-based circuits.*

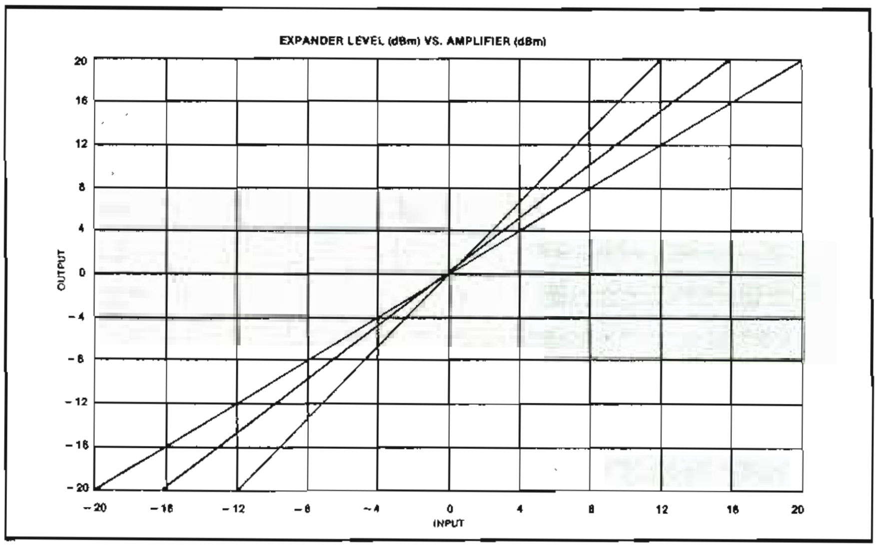

Figure 6. *A typical transfer function graph for an expander circuit, showing the basic function of making loud sounds louder and soft sounds softer.*

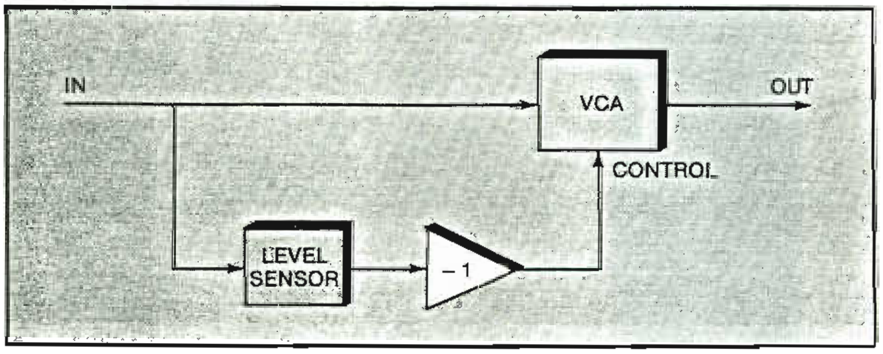

Figure 7. *A basic block diagram of an expander circuit.*

threshold level, or *turnover point*. This point is normally variable, allowing the threshold to be adjusted to match the requirements of the studio operating levels and the program material. As with a conventional compressor, above threshold the compression is characterized by the slope of the transfer curve.

The *knee* in the transfer curve may be sharp, as shown in the graphs, or rounded. Some limiter manufacturers claim that the side effects from a rounded-knee characteristic are less audible; this is a matter for your ears to decide.

A compressor may be converted into a limiter by the addition of a diode before the gain-control element, as shown in Figure 5. The dc voltage from the threshold control is applied to the output side of the diode, forcing the signal level to exceed the threshold before compression can occur.

As with compressors, limiters may be designed in either a feedback or feedforward configuration. A feedforward-type requires predictable characteristics in its level sensor and voltage-controlled element, a straightforward task with transistor-based circuits. If the exact slope of the compression is not of great concern, a feedback-type limiter does not require closely controlled elements.

Limiting thresholds are set by the diode bias voltage, or its equivalent components. The limiting function may be performed with a field-effect transistor (FET), or combination of a light-dependent resistor (LDR) and a light-emitting diode (LED). Such relatively inexpensive designs allow low-cost limiters to be built into power amplifiers or mixing consoles.

Expanders

Expanders are the functional inverse of compressors: They make soft signals softer and loud signals louder, as shown in Figure 6. As can be seen, the slope of the lines is always greater than the 45° slope of a linear amplifier.

If an expander has an increase in output level of 3dB for an increase in input level of 1dB, it is said to have an expansion ratio of 3:1.

In the block diagram of a typical expander, Figure 7, the only change from a compressor is the addition of an inversion stage to make the gain increase with increasing signal level.

Noise gates

Many recording sessions and live performances use open microphones designed to pick up some desired instrument or vocalist. However, when there is no sound from the desired source the microphone continues to pick up ambient noise. If there are very many of these open mics, the background noise can get out of hand. A similar problem arises when trying to add a track from a noisy tape into an otherwise quiet mix.

What is needed is a way to turn down the gain when the signal level drops below some preset value. A device that does exactly this is the noise gate. Noise gates are to expanders what limiters are to compressors. Above the threshold level a noise gate operates as a normal amplifier; below threshold the gain decreases with decreasing signal level, making soft sounds much softer.

This action effectively gates out, or removes, the background noise, but does not affect the desired signal. The noise-gate characteristic shown in Figure 8 looks much like a limiter's transfer curve

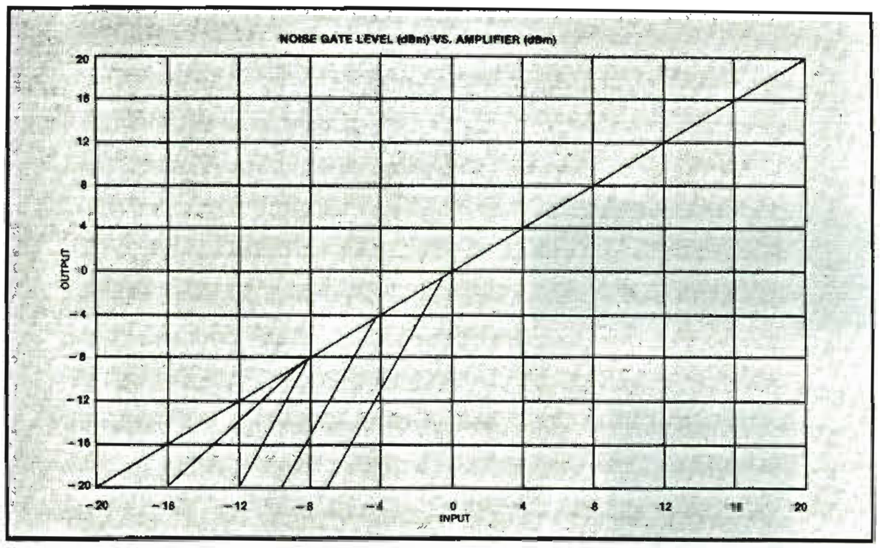

Figure 8. A transfer function for a typical noise gate circuit. The curve resembles the transfer curve of a typical limiter flipped diagonally from what is shown in Figure 4.

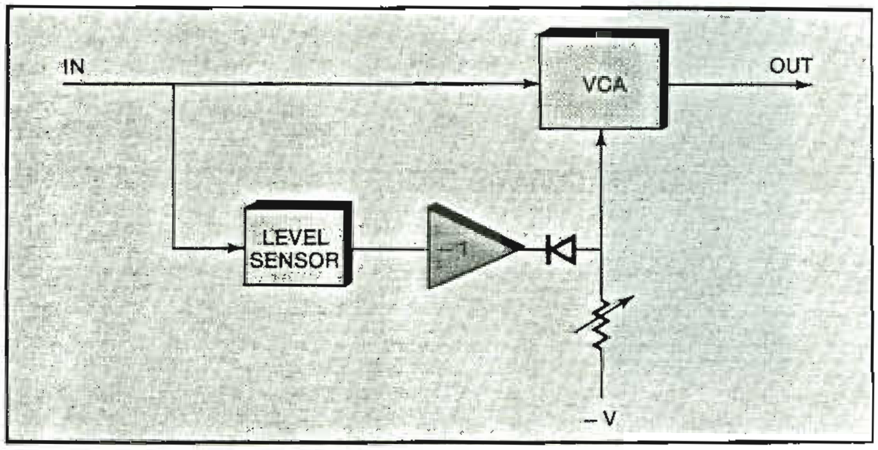

Figure 9. The basic block diagram of a noise gate.

that has been flipped over diagonally.

As with a limiter, a noise gate has two important parameters: threshold and expansion ratio. By adjusting the threshold level, the unit can discriminate between the desired signal and unwanted background noise. If there is insufficient level difference between them there will be erratic changes in gain, as the noise gate switches in and out of expansion mode.

The noise gate block diagram shown in Figure 9 looks much like a cross between an expander and a limiter. The inverter is used, as in an expander, to make the gain increase with increasing signal level. However, the diode now prevents the level sensor's output from exceeding the desired threshold—when this occurs the gain is clamped. Below threshold the noise gate functions as a conventional expander.

Noise gates can also be used for special effects. An excellent example of this is the "Phil Collins" gated reverb drum sound. The signal is fed through a reverb unit that has been set to a fairly long RT_{60}. The recirculation signal on the reverb decays to below an appropriate level, which gives a rich sound that ends abruptly when the noise gate cuts in.

Time effects

So far we have considered only the steady-state behavior of these devices. When the signal amplitude changes with time, however, such signal processing is not so easy. Audio signals are, by their nature, ac waveforms that go positive and negative many times per second. It is essential that the signal amplitude be controlled *without* affecting the waveshape of these ac signals.

For example, if the signal amplitude is adjusted too quickly, the waveshape will be changed, causing audible distortion. If it is adjusted too slowly, on the other hand, some signal peaks will pass through unchanged and the compressor has failed to do its job.

Figure 10 illustrates a typical limiter's response to a time-dependent waveform. A tone burst with an instantaneous 20dB change in amplitude is applied to the input. When the signal amplitude increases, the limiter takes a certain amount of time to respond, resulting in overshoot at the output. As the limiter adjusts to the new gain required, the output amplitude decays to the desired value. When the signal amplitude drops, the output level also drops by the same amount. As the limiter readjusts to the new signal level, the output gradually increases.

Many audible problems are related to the time required by the unit to adjust the gain, including the pumping and breathing sounds heard as the medium-level background material is modulated in amplitude by high-level program material.

If, in an attempt to avoid these problems, a limiter is designed to be slow-responding, it will not be able to prevent peak amplitudes from exceeding the desired level. Any design is a tradeoff between audible side effects and incomplete processing.

This situation has led several manufacturers to provide attack- and release-time controls on their units. Such controls

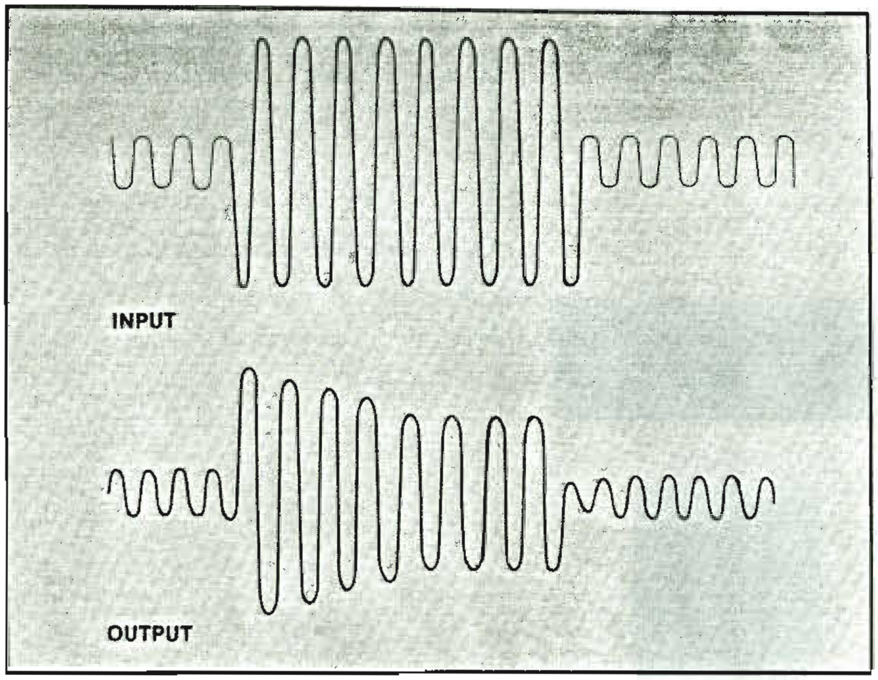

Figure 10. Representative waveforms showing probable tone-burst response for a compression circuit, illustrating the time response of such a unit.

allow an engineer to increase the attack time and reduce the release time until side effects begin to be audible on the particular material being processed.

Adjustable attack and release times also enable the user to operate special effects by adjusting the timing. For example, the attack time may be lengthened to allow the beginning of notes to distort, giving guitars a "gritty" edge to their sound when played hard.

Compressor and limiter features

There are many features available on compressors and limiters that may be important for a particular application. Most professional compressors provide an indication of the gain or gain reduction taking place, and many units also allow monitoring of input or output signal level. This visual aid can be helpful when adjusting drive levels in a system.

Some compressors and limiters allow the control voltages in the level-sense path to be tied to other similar units for use in multichannel systems. If a pair of units is used on the two channels of a stereo mix, tying these points together will prevent the image from shifting because of unequal channel compression.

A few units allow the input to the level-sense circuitry to be patched for special effects. In this way an external equalizer can be inserted into the level-sense path for removing rumble or other low-frequency noise that would otherwise disrupt the level-sensing action. A de-esser is a compressor that has been set to work on high frequencies by the insertion of an HF boost in the level-sense path. Putting a notch filter at the power-line frequency into the level-sense path of an expander or noise gate would allow strong hum to be reduced further than the action provided by a conventional expander. When music is present the filter would mask the hum that would be let through, yet the hum cannot keep the expander gain up.

An unusual, but useful, application for the level-sense input would be to have a microphone picking up background sound to control the level of the main mic channel. With the unit set for a 2:1 expansion, the desired vocal would always be kept a fixed level above the music or noise.

Compressors can sometimes function as remote-controlled attenuators, to provide a remote volume-control function using only dc on the control wires. This might be used, for example, to control the headphone volume in the studio, or could allow a musician to have footpedal control of the instrument level.

Performance specifications

The standard performance specifications of distortion and signal-to-noise ratio are difficult to apply to devices that modify the dynamic range of program material. Because a signal's gain changes with input level, as well as the selected compression or expansion ratio and threshold level, these performance measurements also change.

The noise will generally get worse at high gain values (low signal levels for expanders and high signal levels for compressors). Distortion will sometimes peak at intermediate values or the extremes of gain, depending on the type of gain-control element used. Be wary of units that are specified at only one gain value, or at 1:1 compression ratio. It is best to examine a family of curves of distortion for different gain settings.

Specifications such as frequency response, common mode rejection and maximum input level should be comparable to other types of signal-processing devices. Precautions taken with unbalanced systems apply especially to compressors, which can significantly increase system hum when they have no input signal. Using such a device at different places in the system places differing constraints on residual noise and headroom. Carefully study the system's gain structure before specifying any particular device.

If is difficult to quantify the specifications unique to limiters and compressors in a way that allows meaningful comparison between the audible performance of different units. Attack time and release time are only two aspects of the dynamic behavior. A limiter's distortion performance during the attack portion of its response will significantly alter the perceived distortion for actual program material.

Many compressors and limiters are marginal for headroom, and hard clip on large inputs until the level sensor responds and reduces the gain. Other units are designed with more headroom or an integral softclip circuit, which greatly reduces the level of high-order distortion products during overdrive.

Some compressors and limiters suffer from leakage of the control signal into the output signal path, which results in low-frequency energy at the output during large swings in signal level. This will usually show up in two-tone, difference-frequency intermodulation measurements, if the tones are spaced closely enough in frequency.

Compressors and limiters, more than any other signal-processing device, should pass a listening test before purchase. The necessary performance criteria are not yet well enough understood to simply rely on paper specification. R-E/P