

Improving Loudspeaker Signal Handling Capability

Abstract

Powered speakers are an increasingly important product category in almost every market where speakers are sold — from home theater and multimedia, to custom installations and sound reinforcement.

One advantage of a powered speaker is that the amplifier always drives a known load. Designers can further exploit the opportunity this

presents by providing some “intelligent” control of the signal driving the load to further optimize performance and headroom.

This application note describes a method for increasing perceived headroom by compressing bass frequencies above some threshold signal level. If performed judiciously, the headroom of the speaker system can be extended without noticeable degradation in sonic performance.

Improving Loudspeaker Signal Handling Capability

Introduction

While the signal-handling capabilities of speakers have improved dramatically over the past decade, most systems lack intelligent management of signals at or near maximum levels. This is particularly true of low-cost powered speaker systems used with multimedia computers. These systems are often further compromised by having a single speaker driver handling either the entire audio band or, at least, both the bass and midrange components. These systems may also have limited power handling capability, particularly at low frequencies.

The adaptive low-frequency limiter presented here will allow users to drive these types of powered loudspeakers to higher signal levels without audible clipping or distortion. The result is a low-cost, “intelligent” signal handling circuit that better protects the speaker drivers, while increasing the perceived dynamic range (headroom) of the speaker and amplifier combination.

It should be noted that no practical limiter of any kind will react instantaneously. Therefore, the first step in creating a system with maximum headroom is to ensure that short-term clipping is handled in a well behaved manner. The amplifier may inherently clip cleanly itself, or a simple diode clipper may be employed between the limiter and the amplifier input stage.

Conventional peak limiting circuits are often used to prevent clipping. However, peak detectors are not an optimum choice for many speaker/amplifier systems because they tend to react so quickly to signals which exceed their threshold — often more quickly than needed when the signal is of such short-duration that the clipping would be nearly inaudible. Compounding this problem is the peak detector’s release time, typically set rather slow to minimize distortion induced by the detector ripple associated with more rapid time constants. The usual outcome is a limiter which too quickly reduces system gain even on short-term, transient events, and then takes a while to recover. The audible effect has been described as “punching a hole” in the audio.

Rms-level detection provides a much better compromise because it reacts in response to total energy, rather than waveshape. The true-rms limiter will allow overloads of short duration to be clipped by the amplifier itself or by an external clipper as discussed earlier. Since these transient events are generally not audible, no limiting is required. Should the signal remain in clipping long enough to risk becoming audible, the true-rms limiter will act to reduce the signal level. The result is a limiter which better maintains headroom by not over-reacting to short-term peaks, thereby minimizing the need to reduce the gain of the system.

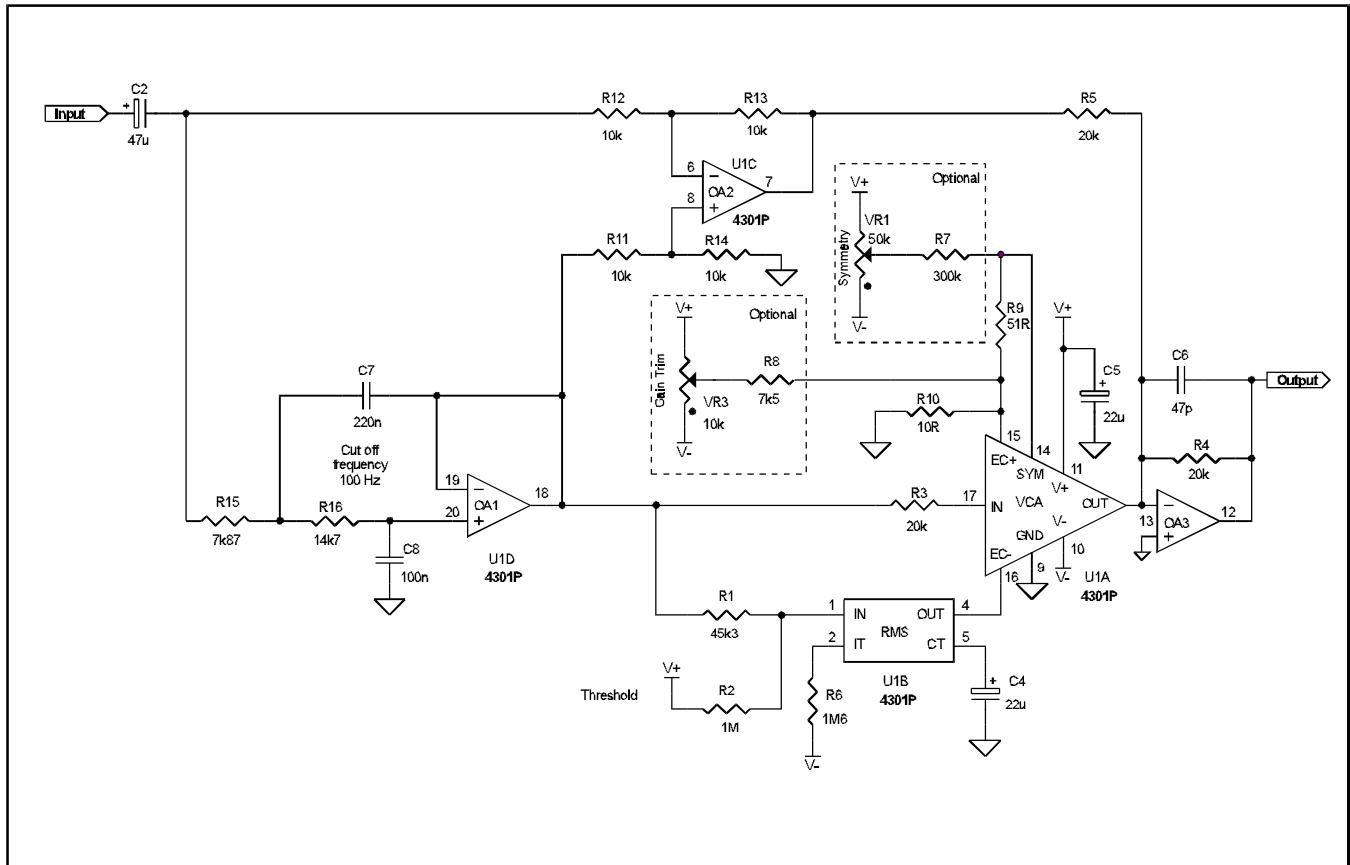


Figure 1. Complete limiter schematic

The circuits presented here, then, provide extended perceived headroom over systems with no signal level management, or systems which use peak limiting. They do this by preferentially limiting low-frequency content (where overload typically occurs), and by ensuring that the system does not over-react to peaks of short duration.

Theory of Operation

The recommended circuit, presented in Figure 1, acts as a low-frequency above-threshold limiter. It comprises a “derived filter”, a true-rms signal level detector, and a voltage-controlled amplifier (VCA). These three circuit elements are all built with a single THAT 4301 “Analog Engine™”, a versatile integrated circuit which includes the rms-level detector, a “decibel-linear” VCA (whose gain changes exponentially with the applied control voltage), and three general-purpose operational amplifiers.

The following sections describe the various functional blocks in more detail.

The Derived Filter

One “derived” filter topology is shown in Figure 2A. The virtue of this type of filter is that when the lowpass output and highpass outputs are summed together at unity gain, the original signal is restored, unaltered in amplitude or phase.

The circuit formed by OA1, R15-16, and C7-8 is a Sallen-Key lowpass filter with a cutoff frequency of 100 Hz, which can easily be adjusted by changing R15 and R16, or C7 and C8 appropriately. See the table labeled “Sallen-Key Lowpass Filter” in Table 1 for the component values covering an appropriate range of frequencies.

The output of this lowpass filter goes to the compressor and to the differential amplifier formed by R11-14 and OA2. This differential amplifier derives the inverted highpass output by subtracting the lowpass output from the input signal. The highpass signal is later summed, uncompressed, with the compressed lowpass signal in the appropriate phase at the output of the VCA. R11-R14 should have tolerances of no worse than 1%, since their toler-

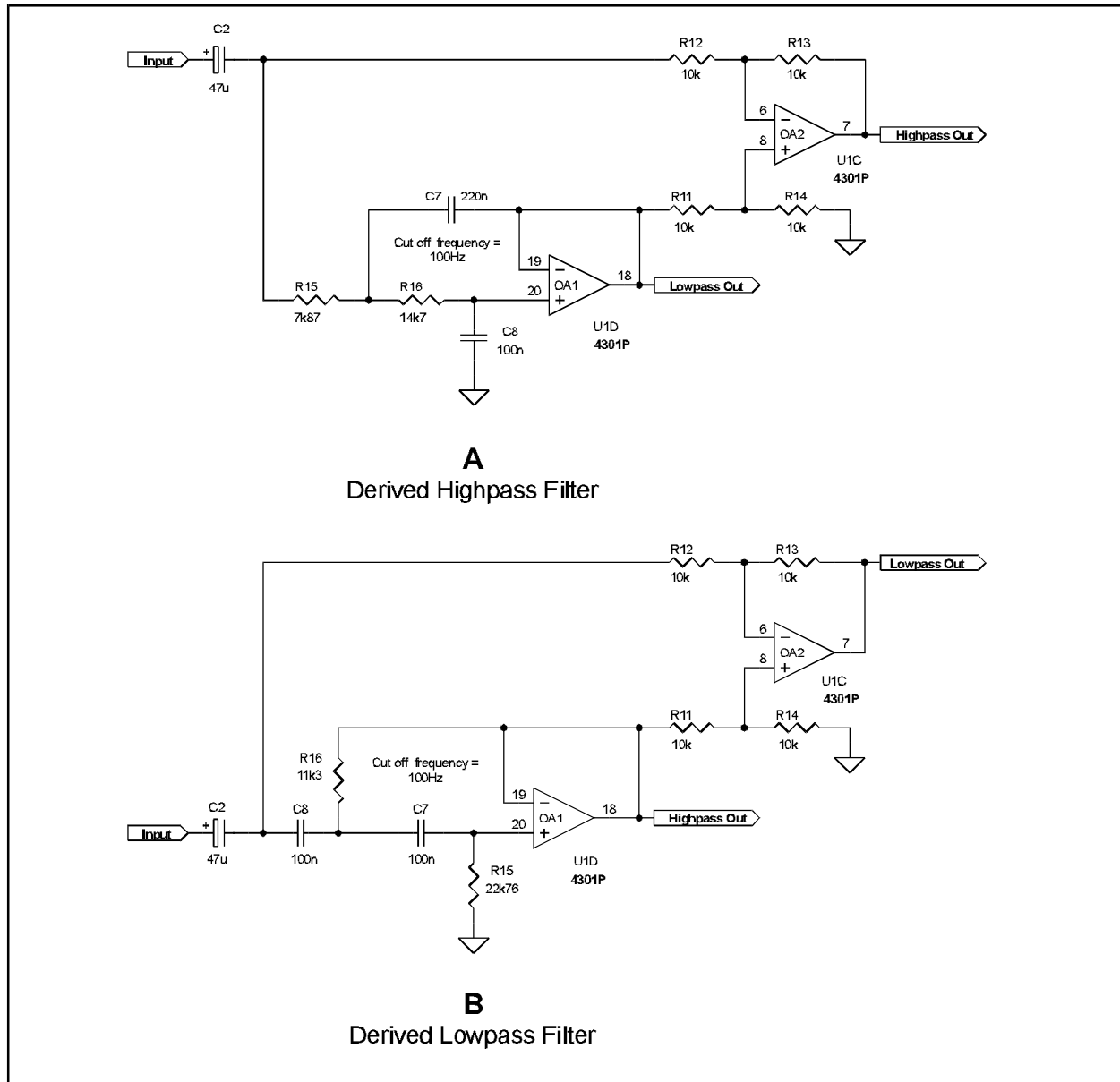


Figure 2: Derived filters

Sallen-Key Lowpass Filter (for use with derived Highpass filter)				
Freq	C7	C8	R15	R16
100	220 nF	100 nF	7.87kΩ	14.7kΩ
126	220 nF	100 nF	6.19kΩ	11.5kΩ
159	220 nF	100 nF	4.99kΩ	9.31kΩ
200	220 nF	100 nF	3.92kΩ	7.32kΩ
252	220 nF	100 nF	3.09kΩ	5.76kΩ
317	220 nF	100 nF	2.49kΩ	4.64kΩ
400	220 nF	100 nF	1.96kΩ	3.65kΩ

Table 1

Sallen-Key Highpass Filter (for use with derived Lowpass filter)				
Freq	C7	C8	R15	R16
100	100 nF	100 nF	22.6kΩ	11.3kΩ
126	100 nF	100 nF	17.8kΩ	8.87kΩ
159	100 nF	100 nF	14.3kΩ	7.95kΩ
200	100 nF	100 nF	11.3kΩ	5.62kΩ
252	100 nF	100 nF	8.87kΩ	4.42kΩ
317	100 nF	100 nF	7.95kΩ	3.57kΩ
400	100 nF	100 nF	5.62kΩ	2.80kΩ

Table 2

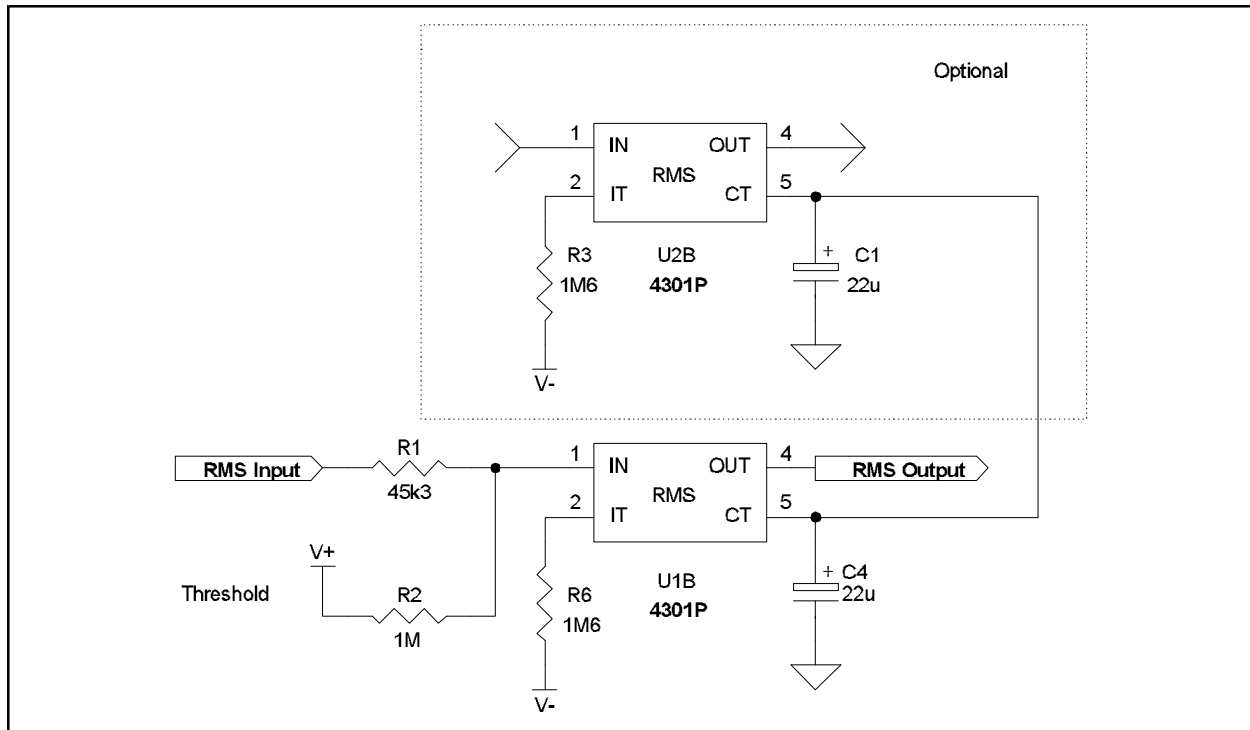


Figure 3. RMS-level detector circuit

ances will essentially determine the CMRR of the differential amplifier, and hence the leakage between lowpass and highpass sections.

Figure 2B shows another permutation of a derived filter. In this circuit, the lowpass portion of the signal is derived, which results in the ability to achieve greater low frequency compression since the derived signal always has a slope of 6dB/octave, regardless of the roll-off of the filter that is used to derive it. This arrangement results in less perceived bass at high levels of compression, and as a result, the designer will usually need to choose a lower cut-off frequency when using the second topology. Table 2 (labeled “Sallen-Key Highpass Filter”) shows component values for tuning this filter over a flexible range of frequencies.

The Level Detector

The level detector in the THAT 4301 is a true-rms detector similar to the popular THAT 2252. This detector takes the absolute value of the input current, logs it, squares it, and then averages it. The square rooting is done implicitly at the control port of the VCA. This action ultimately results in an output sensitivity of 6.5mV/dB.

In the circuit shown in Figure 3, the detector is set for a nominal timing current of 7.5 μ A, and the detector’s time constant is established at 75ms, a reasonable choice since only low frequencies are being compressed. Given that the circuit is designed to operate off ± 12 V, the timing current is programmed by setting

$$R6 = \frac{V+}{I_t} = \frac{12\text{V}}{7.5 \times 10^{-6}} = 1.6\text{M}\Omega,$$

And the time constant is set by making

$$C4 = \frac{I_t}{V_T} (\tau) = \frac{7.5 \times 10^{-6}}{0.026} (0.075) = 22\mu\text{F}.$$

The THAT 4301 VCA will have its lowest distortion at unity gain, so this is where we want the gain to sit when the circuit is not compressing. Unity gain occurs when the VCA’s E_c -input is at zero volts, so this should be the rms-detector’s output when the circuit is not limiting. Referring to the THAT 4301 data sheet, the input current for which the output of the detector is zero volts is calculated as follows:

$$I_{in0} = 1.13 \times I_t = 8.48\mu\text{A}.$$

This also sets the limiter's threshold, the signal level above which the limiter will begin to operate. It should be noted here that this value will usually be derived by experimenting with the speaker system under consideration. For this example, we will assume that we want to begin limiting at an input of -6dBu, or 0.388 V_{RMS} . Using this voltage and I_{in0} , R1 is calculated as follows:

$$R1 = \frac{V_{in0}}{I_{in0}} = \frac{0.388}{8.48 \times 10^{-6}} = 45.3k\Omega.$$

(Note: The portion of the circuit contained in the box marked "optional" shows the connection for using a second THAT Corporation RMS detector in stereo applications. This connection will force both detectors' outputs to be the true-RMS-sum of the input signals, and thus keep the stereo image from wandering as a signal goes into compression. The reference point of the RMS detectors must be matched for this technique to work optimally.)

We now need to ensure that the detector maintains zero volts at its output when the signal drops below the -6dBu threshold. We will do this by separately injecting the detector's input with a current equal to I_{in0} . The threshold current is injected through R2, which is calculated as follows,

$$R2 = \frac{V+}{I_{in0}} = \frac{12V}{8.48 \times 10^{-6}} \approx 1M\Omega.$$

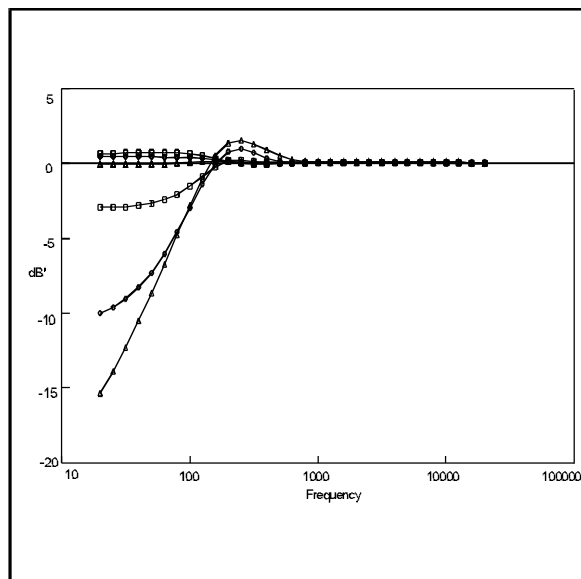


Figure 4. Low frequency compression characteristics at various input levels using derived highpass filter

Now, for input signal levels less than about 388 mV, the current through R2 will dominate and hold the detector's output at zero volts (and hence the VCA's gain at 0dB).

When the input signal exceeds 388 mV, the detector's output will begin to change, resulting in the limiter action described below.

The Compressor / Limiter

The circuit topology shown in Figure 1 has the VCA and rms-level detector configured as a feed-forward compressor/limiter. As the input signal increases above the detector threshold set by R2, the detector's output increases at 6 mV/dB. This will command a corresponding gain decrease from the VCA of 1dB per 6 mV, resulting in the proper "infinite compressor" or limiter behavior. The circuit operates, therefore, to keep the net signal output level constant (for more information, see THAT Corporation's applications notes, Basic Compressor/Limiter Design, and The Mathematics of Log-Based Dynamics Processors, both available on our web site at www.thatcorp.com, or directly from THAT Corporation).

As shown in Figure 1, an external potentiometer (VR1, labeled Symmetry) is used to adjust the THAT 4301's VCA for minimum distortion. For less precise applications where users can tolerate as much as 0.7% THD (worst case) in the bass region, the THAT 4301 can be used without a symmetry adjustment. This may well be acceptable in many designs

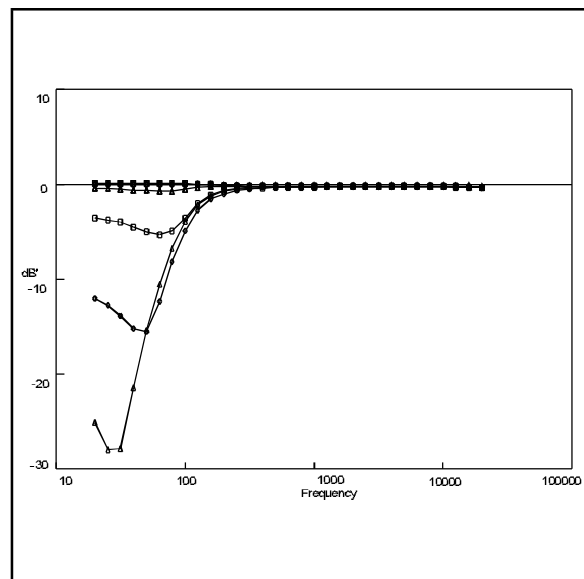


Figure 5. Low frequency compression characteristics at various input levels using derived lowpass filter

where the speaker drivers and/or amplifier may contribute even higher distortion levels at bass frequencies. Further, it is often difficult to discern 2nd harmonic distortion at low frequencies. Indeed, especially with smaller speaker systems with limited bass response, we often use the harmonic content of the lowest frequencies to identify the (unreproduced) fundamental that produced them.

In a similar vein, there is a tolerance on the detector's input current required for 0V output. Below threshold, then, the VCAs gain may vary by as much as +/-3 dB. If this tolerance is acceptable (because, for example, it is made up for in some other stage), the potentiometer labeled VR3 (Gain Trim) may be removed.

Results

The intent of this circuit is to provide protection for powered speaker systems while extending the perceived headroom of these systems in a sonically pleasing manner. The circuit accomplishes this by a) compressing, and ultimately limiting, the low frequency content fed to a speaker driver or system when the signal level exceeds a predetermined threshold, and b) using rms-level detection to ensure that system gain is not reduced unnecessarily (a characteristic of many peak detector circuits).

Figure 4 and 5 show the normalized frequency responses of the compressor outputs at various input levels from -10dB to +15dB relative to threshold. At low input levels the response is essentially flat, but as the signal level increases, the portion of the spectrum below the cutoff frequency is compressed. The slight peaking at the cutoff frequency at higher levels of compression in Figure 4 is a result of the derived filter.

In subjective testing using bass intensive source material, the derived low pass topology was most effective at limiting speaker distortion (at the expense of sounding a bit more "tinny") while the derived-highpass topology provided the least obtrusive compression (though it did allow distortion at a slightly lower level.) Since the derived-lowpass topology rolls off at 12dB/octave versus 6dB/octave for the derived-highpass topology, the cutoff frequencies were adjusted to 100 Hz and 200Hz respectively to achieve the same degree of compression at high levels. Under these conditions in subjective tests, we were able to play the speakers significantly louder than without compression with the same audible levels of distortion.

Final Thoughts

Tone and volume controls should be located before the compressor circuit to ensure that speaker and power amplifier will be fully guarded regardless of the setting of these controls.

Any low frequency equalization should be placed before the compressor as well, though it could come before the tone and volume controls. .

The compressor should come after the variable gain stages but before the stages that it is required to protect. The pole frequency for the compressor's derived filter should be chosen so that the compressor will correctly attenuate excessive bass signal levels.

This approach will allow users to avoid excess low frequencies excursion without unwanted signal pumping, by compressing only the low frequency signals.

For more information or technical assistance, please contact THAT Corporation.

Notes