Audio BASICS PART 2

by Ray Marston

Ray Marston continues to look at audio-system basic principles in the concluding part of his special feature article.

Signal Compression

The 'signal compression' method of dynamic range manipulation operates in the basic way illustrated in Figure 1. Here, the initial input signal is applied to the input of a dynamic range compressor unit, which, in this example, has a 2:1 dynamic compression ratio and can convert an input signal with a 90dB dynamic range into an output with a 45dB dynamic range. This 'compressed' output is applied to the input of the 'poor-dynamic-range' transmission/recording medium, and when required, is converted back into its original (90dB range) form by a matching dynamic range expander unit, which has characteristics that are the exact inverse of those of the compressor. This type of compressor-expander system is generally known as a 'compander' (or 'compandor') system, and was originally devised to improve the quality of various voice communication systems.

Note in Figure 1, that the '2:1' compressor/expander ratio applies to the systems dynamic range in terms of dB, and not to its actual range in terms of input and output voltages. This point is made clear in the table of Table 1, which shows that the compressor and expander work by giving highly non-linear variations in voltage gain to different input signals. The gain varies over a 175:1 range, from ×0·57 to ×100 in the compressor unit, and from ×1·75 to ×0·01 in the matching expander unit.

Practical compressor and expander circuits are both built around the basic dynamic range expander circuit that is shown in (a) descriptive and (b) symbolic forms in Figure 2. Here, the audio input signal is applied to the inputs of a current-controlled variable-gain cell (a high grade operational transconductance amplifier, or OTA) and an electronic rectifier that converts the mean input signal voltage into a proportional DC output current, which controls the gain of the variable-gain cell. The action is such that if the signal input rises by 10dB (from, say, -40dB to -30dB), the gain also rises by 10dB, to give an overall increase in output voltage of 20dB, i.e., a 2:1 ratio of dynamic expansion. The gain-control attack and decay times are controlled by capacitor C_T.

The gain cell's output signal appears in the form of a current, rather than a voltage, but can by converted into a proportional voltage via a suitably wired op-amp. Figure 3 shows ways of using the basic dynamic range expander to make practical voltage-in to voltage-out (a) expander or (b) compressor units. In the expander circuit, the gain cell's output current is simply fed directly into the inverting input terminal of the op-amp, which gives direct current-to-voltage conversion. In the compressor, the audio input signal is fed into the op-amp's inverting input via R1, and the basic expander is wired in series with the op-amp's output-to-input negative feedback path, causing the overall circuit to act as a voltage-in to voltage-out dynamic range compressor with dynamic characteristics that are the exact inverse of those of the expander circuit.

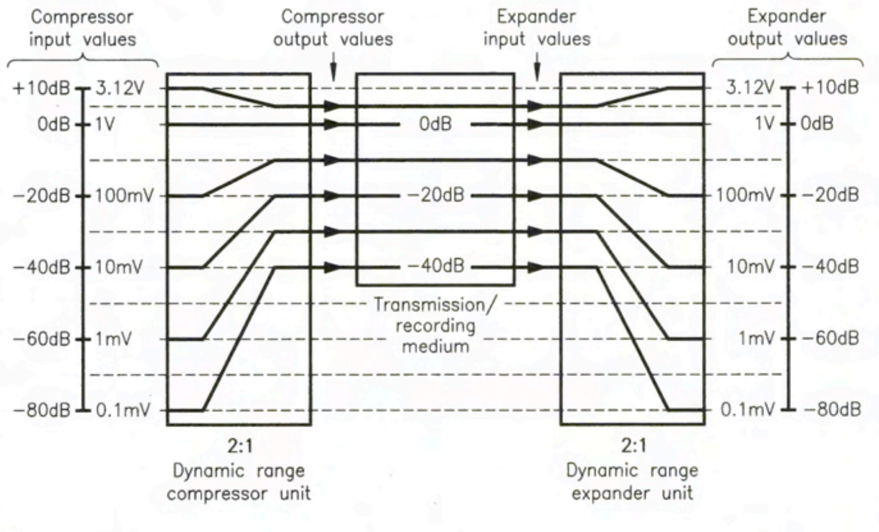


Figure 1. Diagram illustrating the basic principle of dynamic range companding, using a 2:1 companding ratio, over a 90dB operating range.

Table 1. Table showing the voltage gain variations of the basic 2:1 dynamic range compressor/ expander.

INPUT VALUE dB	INPUT VALUE Volts rms	OUTPUT VALUE dB	OUTPUT VALUE Volts rms	VOLTAGE GAIN	
+10dB	3.12V	+5dB	1.78V	x 0.57	2:1 Dynamic range compressor unit
0dB	1.0V	0dB	1.0V	x 1	
−20dB	100mV	−10dB	316mV	x 3.16	
−40dB	10mV	−20dB	100mV	x 10	
−60dB	1mV	−30dB	31.6mV	x 31.6	
−80dB	0.1mV	−40dB	10mV	x 100	
+5dB	1.78V	+10dB	3.12V	x 1.75	2:1 Dynamic range expander unit
0dB	1.0V	0dB	1.0V	x 1	
−10dB	316mV	−20dB	100mV	x 0.316	
−20dB	100mV	−40dB	10mV	x 0.1	
−30dB	31.6mV	−60dB	1mV	x 0.0316	
−40dB	10mV	−80dB	0.1mV	x 0.01	

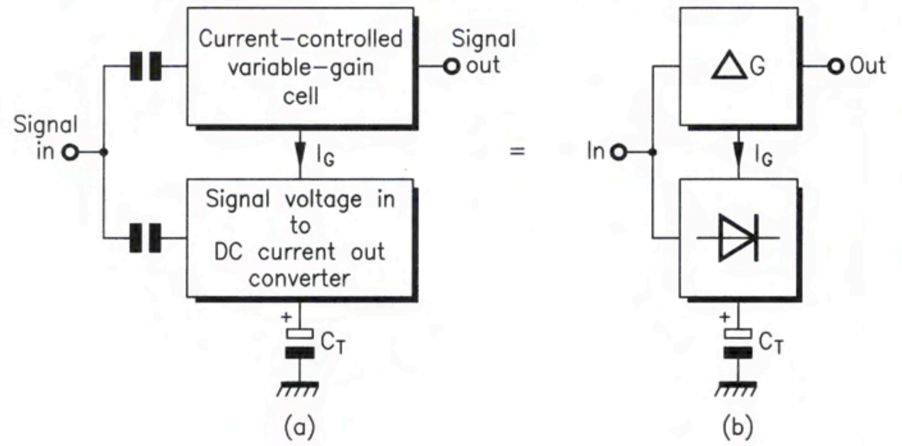


Figure 2. Diagram showing (a) descriptive and (b) symbolic versions of a basic dynamic range expander.

Hybrid Dynamic Range Control

At first sight, pre-emphasis and compression methods of dynamic range control seem to offer huge practical advantages, but in reality, both systems work by introducing various types of gain distortion and thus have weaknesses that limit their practical value in many Medium-Fi to Hi-Fi applications. The high-frequency gains of pre-emphasis filters should, for example, always be limited to about 20dB (in the manner shown in Figure 11 of Part 1, Issue 113) to avoid overdriving unusually large high-frequency signals that are generated as fundamental – rather than harmonic – waveforms.

Compander systems should be treated with special caution. They originated as strictly 'Low-Fi' voice processing units, as typified by the popular NE570 IC, and usually generate rather high levels of noise, THD, tracking distortion and output DC-tracking shift; these defects become magnified if the system's compression ratio is artificially raised above its basic 2:1 value. The system's greatest defects are caused by a problem known as 'breathing' or 'pumping', which occurs when a large transient input waveform is followed by a near-zero input signal, causing the expander gain to drop sharply on the arrival of the transient, but (because of the system's AGC action) to rise sharply to its maximum value a few moments later, to produce the distinct hissing sound of system noise.

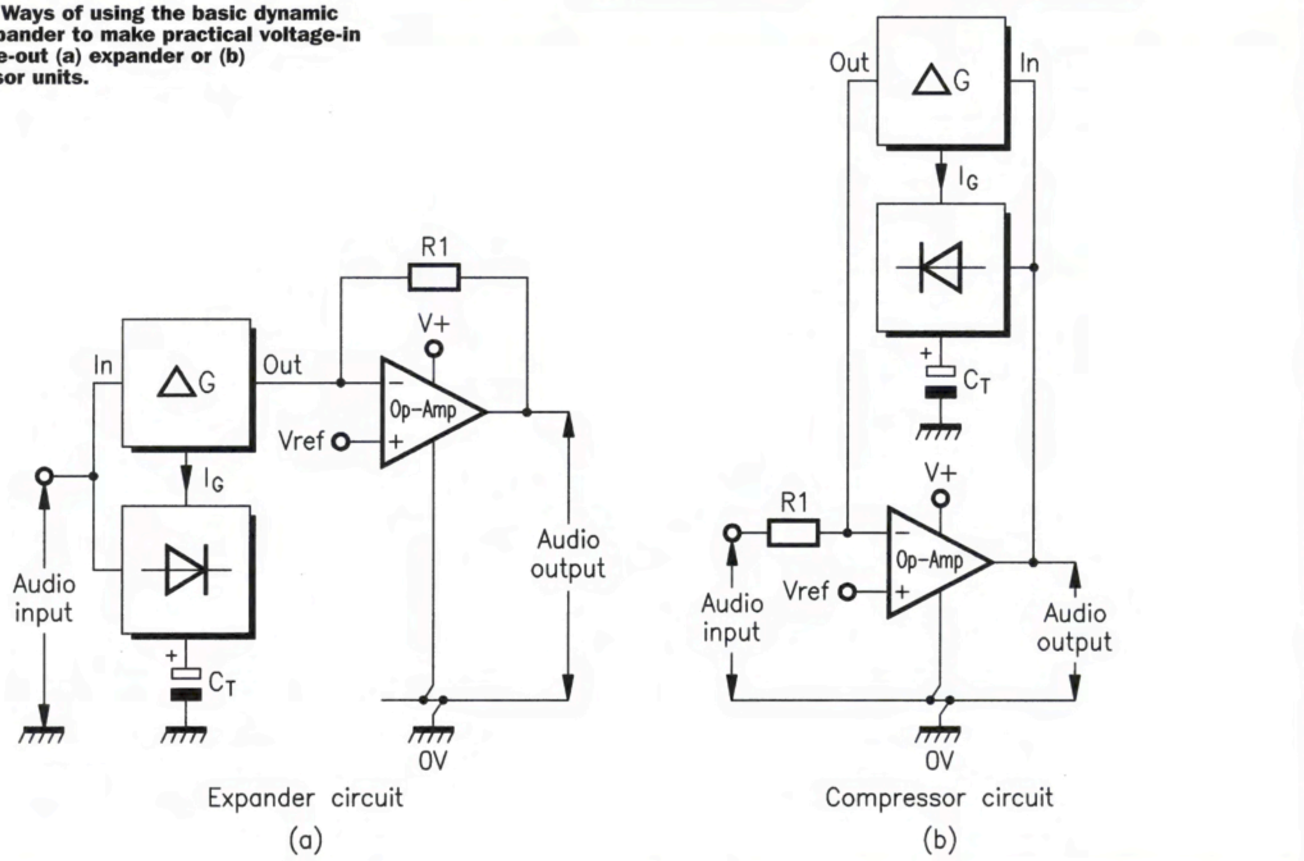
The best way to obtain really good dynamic range manipulation in Hi-Fi applications is to use a hybrid system that uses a subtle combination of pre-emphasis and/or filtered companding techniques, as in the cases of the dBx and Dolby magnetic recording/playback systems, which each give a large increase in useful dynamic range but without suffering 'pumping' problems.

Digital Audio

Audio systems are designed to convert acoustic input signals into remotely located acoustic output signals, and are basically analogue systems. In modern audio electronics there is, however, one area of the system in which digital electronic techniques offer definite advantages over analogue ones, and that is in the area of high-quality signal storage (as in CDs). This advantage occurs because digital signals have only two amplitude levels – either 'high' or 'low' – and thus (unlike analogue signals) cannot be corrupted by normal levels of system noise or non-linearity.

Figure 4 shows the basic elements of a digital audio-signal 'storage' and 'replay' system. On the 'storage' side of the system, the audio input signal is subjected to normal signal processing (amplification and/or filtering, etc.). It is then applied to the input of an ADC (Analogue-to-Digital Converter) unit, which converts the analogue input signal into a digital equivalent, which is then superimposed on the system's storage medium (a CD or tape, etc.). On the 'replay' side of the system, the

Figure 3. Ways of using the basic dynamic range expander to make practical voltage-in to voltage-out (a) expander or (b) compressor units.



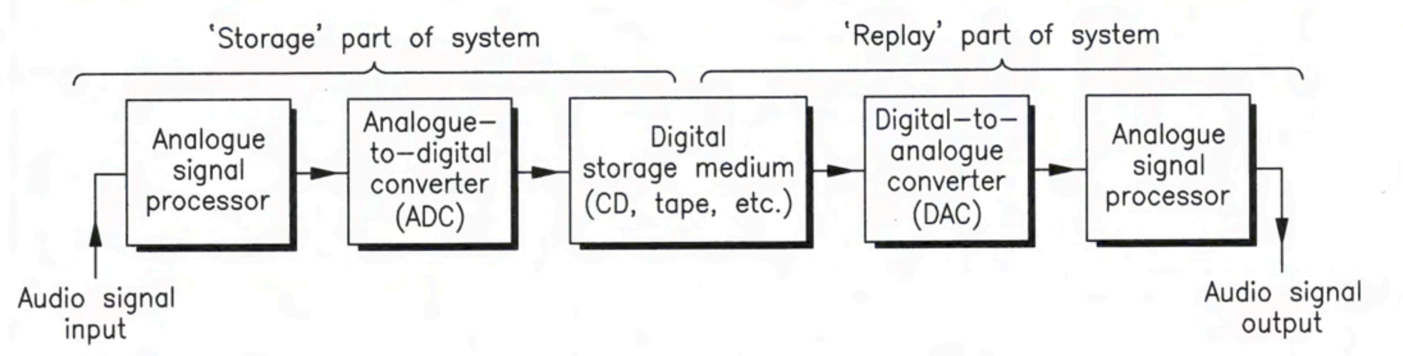


Figure 4. Diagram showing the basic elements of a digital audio-signal 'storage' and 'replay' system.

storage medium's digital signals are inspected by a DAC (Digital-to-Analogue Converter) unit, which converts them back into analogue form and passes them onto the outside world via another signal processing analogue circuit.

Figure 5 illustrates some of the basic operating features of the ADC part of the system, as applicable to a normal 16-bit CD. Here, the system repeatedly takes high-speed samples of the audio input signal's instantaneous analogue amplitude, converts each new sample's amplitude measurement into a multi-bit digital output word and passes it onto the recording media before carrying out a similar operation on the next sample. To be effective, the system's sampling frequency must be at least double that of the highest signal frequency of interest. Modern CD systems are designed to handle signal frequencies of up to 20kHz, and to attain this, they use a standard sampling frequency of 44·1kHz and thus execute 44·1 sampling operations during a 1kHz signal cycle, 14·7 samplings in a 3kHz cycle (see Figure 5(a)), and 3·67 samplings in a 12kHz cycle (see Figure 5(b)).

The effective signal-to-noise ratio and dynamic range of an ADC unit's output is directly proportional to the unit's 'bit' size, and can be simply calculated from the equations:

(1). Signal-to-noise ratio $= 6 \times n$ *dB*

(2). Useful dynamic range $= 6 \times (n - 1)$ *dB*

Where n is the ADC's bit size. Thus, 16-bit ADCs have S/N-ratios of 96dB and have useful dynamic ranges of 90dB. The current generation of CDs are recorded via 16-bit ADC, which generate 16-bit outputs in the basic format shown in Figure 5(c) and can thus generate up to 65,536 different codes or level-measurement values. The next generation of CDs (for which players are already available) are scheduled to use 20-bit data recording, which offers S/N-ratios of 120dB and useful dynamic ranges of 114dB and can generate up to 1,048,576 different codes.

The above explanation of CD encoding is, of course, much simplified, and merely illustrates the basic principles. Figure 6 shows a more realistic picture of the actual coding system used in 16-bit CDs. Here, each 16-bit data word is made up of two 8-bit symbols, as shown in (a). All CDs give stereo outputs, so two words are generated at the same time, one for the L/H channel and one for the R/H channel, and are applied to the CD encoding system in series, as a 32-bit sample, as shown in (b). Each of these samples is modified and enlarged by the CD encoding system and is then incorporated in a CD frame. Each of these frames commences with 8-bits of display data, followed by three of the modified samples, which are followed by a group of parity bits, plus three more samples and another group of parity bits, and ends with a synchronisation code. The CD frames each contain 588 data bits, and are generated at a 7350 per second rate. The CD data is thus processed (at both the record and replay ends of the system) at a rate of 4·3218M-bps.

At the 'player' end of the CD system, the decoder circuitry copies and stores each frame of data as it becomes available, then converts its samples and words back into their original analogue forms (via a 16-bit or larger DAC) and passes them to the appropriate channels of the audio Hi-Fi unit – in the correct time sequence – before moving on to the next frame, and so on.

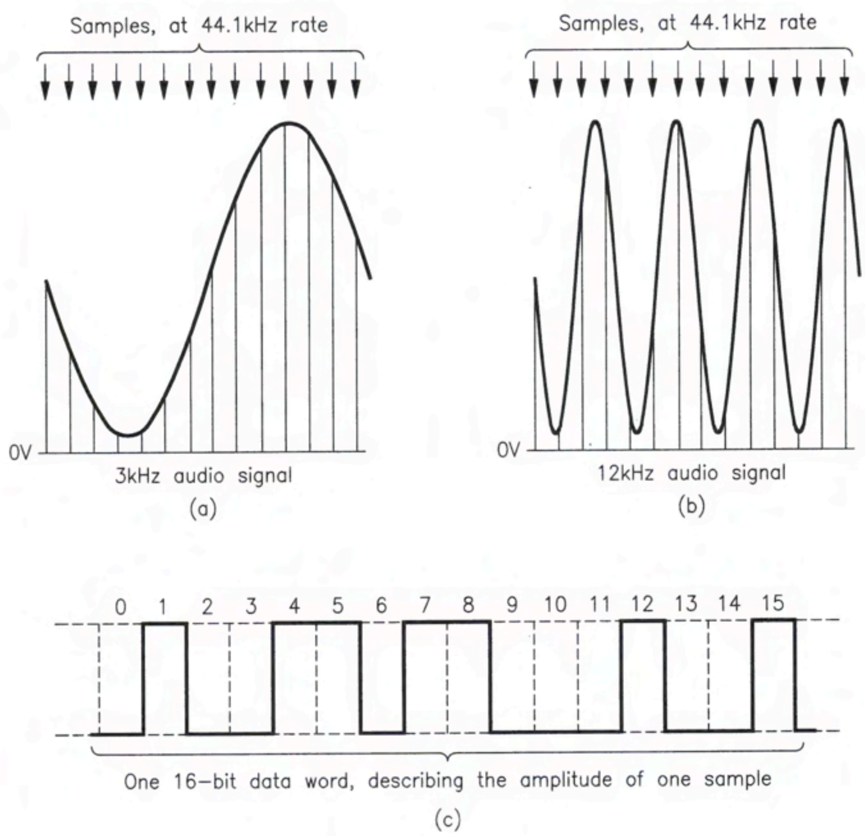


Figure 5. Diagram illustrating some basic features of analogue-to-digital conversion in 16-bit CD systems (see text for explanation).

The Hi-Fi Unit

In its simplest form, a Hi-Fi unit may consist of little more than an input selector, a pre-amplifier, a power amplifier, and a pair of loudspeakers. Usually, however, the unit is fairly elaborate, and typically may take the form shown in Figure 7, which (in the main amplifier unit) shows just one channel of a stereo system. Thus, the main amplifier contains an input selector switch (S1), plus tone and volume control circuitry that feeds the input of the speaker-driving power amplifier. It also houses a power supply unit that can power the main amplifier and (usually) the other Hi-Fi units (tuner, CD player, etc.), plus one or two LED display units that give visual output indications of parameters such as the volume control setting and the instantaneous output amplitude values of the power amplifier.

In most cases, switch S1 can select inputs from an AM/FM tuner, a CD player, a cassette player unit, a phono pick-up and pre-amplifier/equalizer unit, and from an 'Auxiliary' input terminal. This auxiliary input may be driven from a source such as an audio mixer unit, a TV sound-channel tuner, or a remote sound monitor such as a baby alarm, etc. Often, the main amplifier is designed for operation via a remote control unit, in which case, the tone and volume control circuitry usually take the form of voltage-controlled units that are driven via the remote decoder, and S1 takes the form of a multi-way electronic switch that is driven via the decoder. If the Hi-Fi is a really elaborate one, it may also in corporate one or more audio delay lines, in the form of an ambience synthesiser or an echo-reverb unit.

The two most important basic items in the Hi-Fi system are its loudspeakers and its power amplifier. Inferior loudspeakers can make even the very best power amplifier sound bad, and an inadequate power amplifier can make even the very best of loudspeakers sound awful. Assuming, however, that a Hi-Fi unit's loudspeakers and power amplifier are both of excellent quality and are well matched, it will still be found that different listeners will each gain their own individual subjective impressions of the system's quality, and will probably tweak the tone controls until the system 'sounds right'. And 'sounding right' is, of course, the most important function of the entire Hi-Fi system. ELECTRONICS

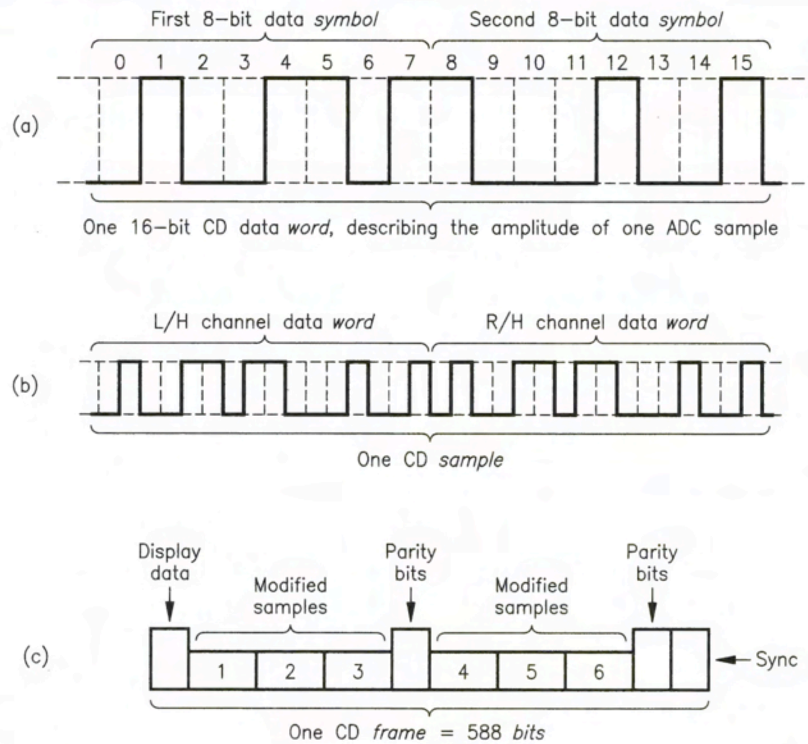


Figure 6. Diagram illustrating the formatting of a standard frame in a 16-bit CD; these frames are generated at a 7,350 per second rate.

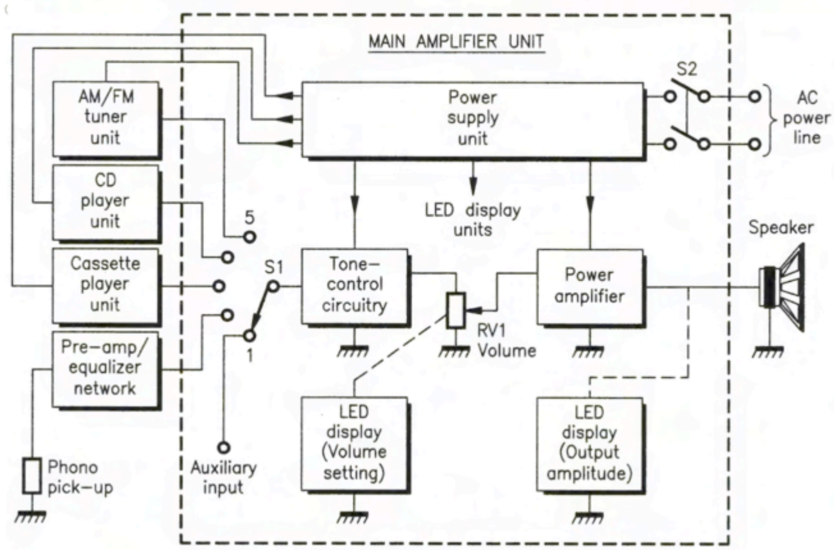


Figure 7. Typical form of a modern Hi-Fi unit, showing only one of the stereo channels of the main amplifier unit.