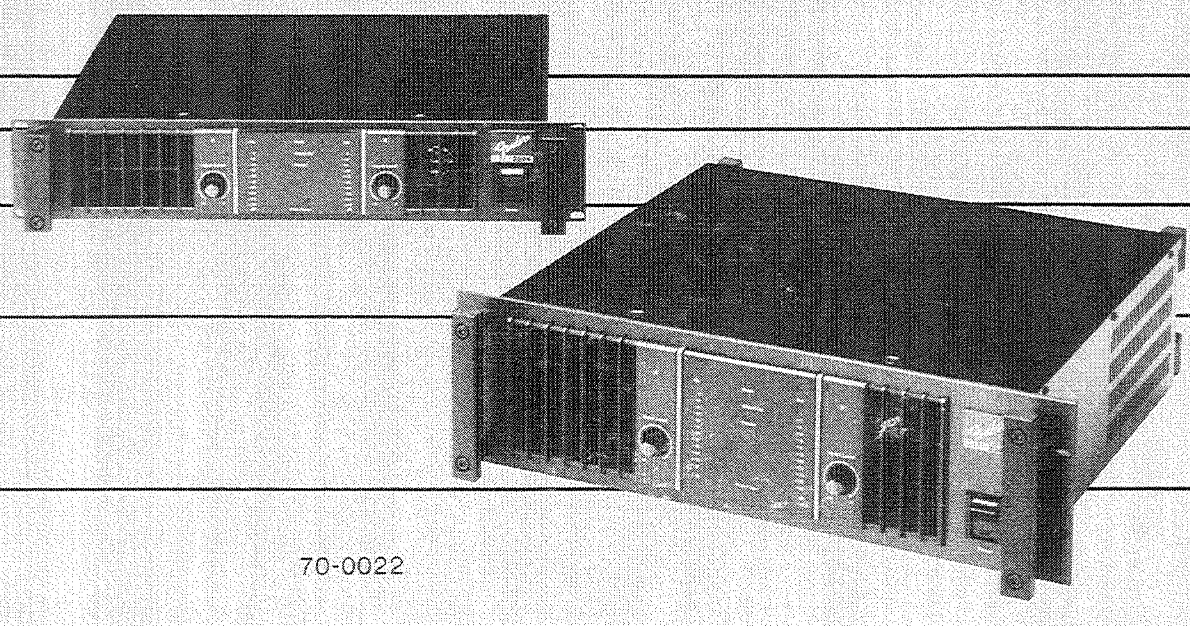
OWNER/APPLICATION MANUAL

$5.00

Fender

Series 2200 POWER AMPLIFIERS

2224
2244



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About this Manual

This manual is designed to be helpful to you whether you're building a new sound system from scratch or putting the 2224 or 2244 into use in an existing system. Since both amplifiers are nearly identical, except for power output, we often refer to the "2224/2244" to save space. We have included the information you're likely to need for normal operation of a power amplifier, plus additional information that may prove useful in both basic and advanced sound system setup and effective use of your sound system budget.

This manual is organized into a number of helpful sections. Use of the Contents page and the Index (located at the end of the manual) to help you quickly locate needed information.

Regardless of your experience level with power amplifiers, we recommend that you at least read the first section of this manual. It will give you all the information that you will need to get your amplifier installed, hooked up and operating.

If you are a relative newcomer to the world of high-performance sound systems, take the time to read the rest of the manual. You'll find operating guidelines, a review of amplifier basics, and the basic electrical and mathematical information necessary to make the best use of your power amplifier. The appendix found at the end of this manual explains the mathematical and electrical units used in the text.

If you are a seasoned veteran, we recommend that you scan the Contents page, then skim the General Information About Amplifiers section, and read the sub-section about How Fender Amplifiers Protect Themselves and the Speaker Load. Additional topics of interest to you may be located via the Index.

SECTION ONE

Description of Features

Front Panel

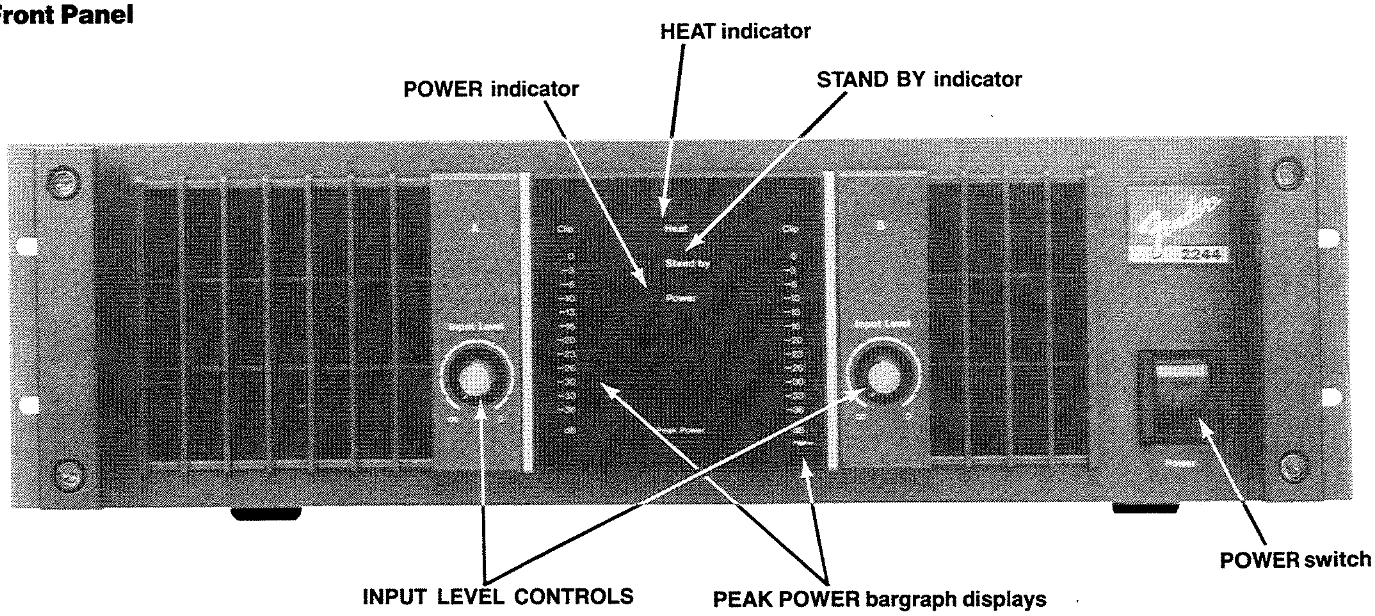


FIGURE 1. 2244 FRONT PANEL (SIMILAR TO 2224)

INPUT LEVEL CONTROLS Two 20-position step-attenuators that reduce input signal fed to the inputs of the 2224/2244's power amplifier section. The control settings can vary between zero attenuation or loss (sensitivity is 0 dBu/0.775 Volt) and infinite loss (at least—85 dB from maximum sensitivity. The controls are detented (click-stepped), making them easier to reset.

PEAK POWER bargraph displays Two multi-segment LED bargraph-type indicators for each channel represent the amplifier's instantaneous output power. When the "0" LED is illuminated, the amplifier is delivering its rated power output into 4 or 8 ohm loads. When the "CLIP" LED is illuminated, the amplifier can no longer increase its output level in proportion to further increases in input level. Reduce the input level (by reducing the input signal or by turning the INPUT LEVEL control down) to avoid distortion. You can find a discussion of clipping in the section titled: "Correlating Amplifier Power With Program Dynamics."

HEAT indicator This LED indicator illuminates to indicate that the amplifier's heat sink tunnel or the amplifier's power transformer has overheated. When this indicator is illuminated, the amplifier's output power has been reduced automatically following a progressive curve, and the fan runs at high speed. If the temperature further increases, the output relay will operate to disconnect the speakers. After a cooling period (the length of the period depends on the ambient temperature near the amplifier), the thermal protection circuits reset automatically, restoring normal operation.

STAND BY indicator This LED indicator illuminates whenever the amplifier's muting circuit is activated. The muting circuit is activated whenever:

- The amplifier is first turned on.
- The amplifier detects DC potential across either of the two channels' outputs
- The amplifier detects strong sub-sonic signals across either of the two channels' outputs
- The second stage of thermal protection is reached.

POWER indicator An LED indicator that illuminates when the amplifier is turned on.

POWER switch This switch is actually a circuit breaker. If the switch "trips" repeatedly (for instance, immediately at turn-on), the amplifier needs service. Seek the assistance of a qualified service technician. Using a circuit breaker switch instead of a more conventional power switch and fuse eliminates the need to carry spare fuses, and provides an instant visual indication when the circuit breaker has been tripped.

CAUTION
Modification or disabling of the circuit breaker/power switch voids all subsequent warranty claims.

Rear Panel

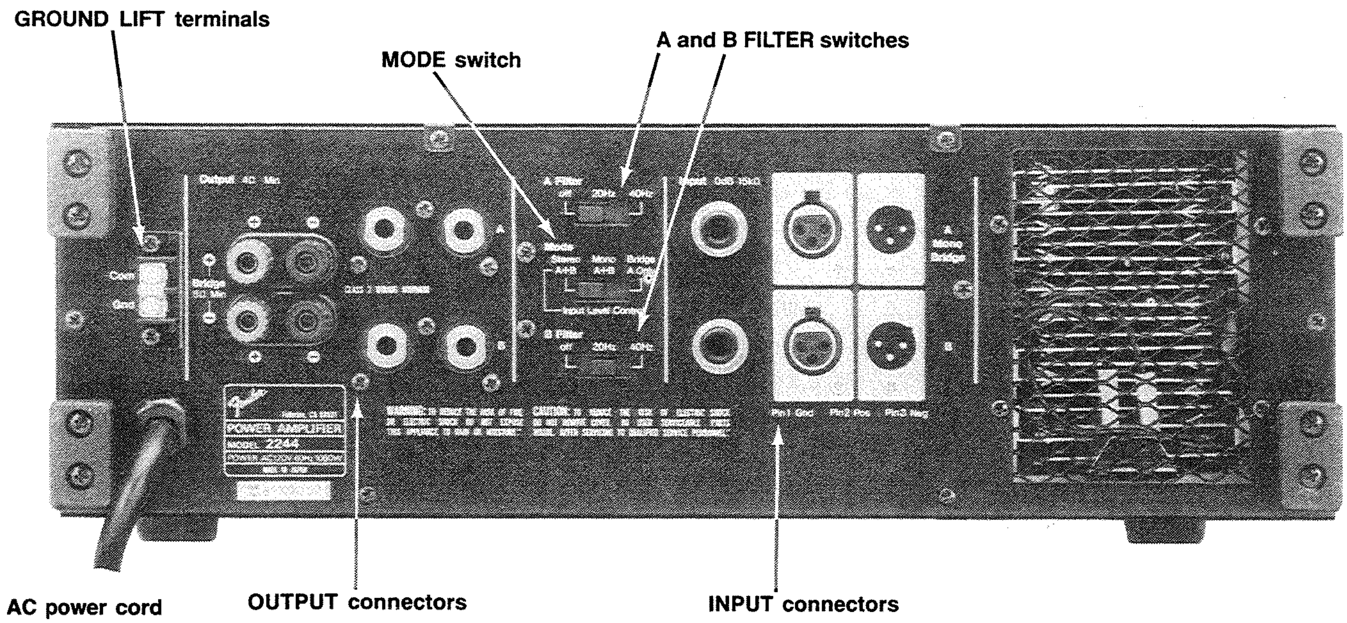


FIGURE 2. 2244 REAR PANEL (SIMILAR TO 2224)

INPUT connectors Each amplifier channel has paralleled XLR-type connectors (one male and one female) and a 1/4 inch tip-ring-sleeve ("stereo") phone jack. All inputs are electronically balanced and accept signals from balanced line sources (such as those with transformer balanced or actively balanced (differential electronic) outputs). The inputs also accept signals from unbalanced circuits.

NOTE

Fender amplifiers are wired so that pin 2 of the XLR connectors is considered "hot." This means that the polarity of the amplifier's output matches the polarity at pin 2 of the XLR connectors. This conforms to the JIS/DIN/IEEE international standards. The input sensitivity remains the same for balanced or unbalanced sources.

As of this date, there is no clear standard in the United States for the polarity of connections on the XLR connector. If opposite amplifier polarity is required, or if the equipment driving the amplifier uses pin 3 as its "hot" lead, the input cable wiring should be reversed from this convention. Use a specially wired cable or (preferably) an adaptor to do this.

Two XLR connectors are provided to facilitate "chaining" or multiple paralleled input connections of several amplifiers without special adaptor cables. Since all of the input jacks (per channel) are parallel connected, you can parallel the amplifier's inputs by interconnecting any jack on channel A with any jack on channel B. if you choose to use the phone jack on one channel and the XLR connector on the other channel, be sure that your cable is wired as follows:

XLR connector	Tip-ring-sleeve plug
pin 1	sleeve
pin 2	tip
pin 3	ring

You can also parallel the amplifier inputs by using the MODE switch.

The input impedance of each channel is 15k ohms, balanced or unbalanced, and the input sensitivity is 0 dBu or 0.775 volts.

OUTPUT connectors Each channel has a pair of five-way binding posts and a pair of paralleled 1/4 inch tip-sleeve phone jacks. The binding post spacing is standard (3/4 inch) to accommodate single or dual banana plugs, spade lugs, or bare wire. The 3/4 inch spacing also applies to channel A and channel B, allowing one dual banana plug to be used for bridged mono operation.

Connect your speakers to these jacks. Ensure that the total impedance presented to any amplifier channel is not less than 4 ohms (speaker loads calculated at 2 ohms will not "blow up" the amp, but 4 ohms is preferable because actual impedances typically drop to values well below rated impedances). Refer to Figures 31 and 32.

CAUTION

When using the amplifier in its bridge mode, ensure that the total load impedance is not lower than 8 ohms.

The output power available varies with the load impedance presented to the amplifier, and also depends on the amplifier's operating mode. Consult the specifications elsewhere in this manual.

A and B FILTER switches Two three-position switches (one per channel) select the low frequency (LF) cutoff point of the amplifier. Each channel may be independently set to:

- OFF. The LF cutoff is below 5 Hz.
- 20 Hz. The LF cutoff is 20 Hz, at 12 dB/octave.
- 40 Hz. The LF cutoff is 40 Hz, at 12 dB/octave.

For most applications, set both switches to the 40 Hz position. Use the 20 Hz position in subwoofer applications or for stereo/hi-fi use. The OFF position should be reserved for test-bench use. A more in-depth discussion of low frequency protection may be found in the sections titled: "Frequency response" and "High pass filters protect speakers and reduce distortion" located elsewhere in this manual.

GROUND LIFT terminals These terminals are normally connected together at the factory. Leave them connected for most applications. The terminals provide a means of separating the connection between the internal ground system of the amplifier (the Com terminal) and the amplifier chassis (Gnd terminal), which is permanently connected to the ground lead of the AC power cord. The ground lift (removing the jumpers) procedure may be useful in large, multiple amplifier, permanent installations.

MODE switch This three-position switch selects the operating mode of the amplifier: Stereo A+B, Mono A+B, Bridge A Only.

Stereo A+B mode selects independent operation of each amplifier channel.

Mono A+B mode connects both amplifier inputs together. In this mode, only the Channel A input jacks are used, but both channels' INPUT LEVEL controls and OUTPUT connections can be used independently to feed the same signal at different levels to different loads.

Bridge A Only mode configures the two channels for mono bridged operation. Only the Channel A control and switches are operative. Connect the speakers to the two RED binding posts (the Channel A post is "hot"). Do not allow either of these wires to contact any other wire, chassis, rack or connector. If you use this mode, it's not good practice to use 1/4 inch phone plugs at either end of the cable.

AC power cord The 2224 and 2244 require nominal 120 Volt, 50-60 Hz AC power. A typical houshold-type output provides a 15 or 20 ampere circuit which is adequate for low power operation (8 ohm load each channel). In heavy duty professional applications, or those applications involving 4 ohm loads (or 8 ohm bridged loads), we recommend checking carefully for adequate AC power. These applications require at least a 20 ampere circuit.

WARNING

DO NOT CHANGE A THE FUSE FOR THE AC POWER DISTRIBUTION SYSTEM TO A HIGHER VALUE UNLESS YOU ARE ABSOLUTELY CERTAIN THE WIRING IS RATED FOR THE HIGHER CURRENT. If a high-current short occurs, the wiring becomes the "fuse," which usually starts a fire before the wiring can "blow." We recommend that you refer to the National Electrical Code before attempting any such change.

If you are driving 4 ohm loads (or an 8 ohm load in bridge mode), as a guideline, you can safely connect two 2244 amplifiers or four 2224 amplifiers to a 20 ampere circuit. If you are driving 8 or 16 ohm loads (16 or 32 ohms in bridge mode), you can connect three 2244 amplifiers (six 2224 amplifiers) to a 15 or 20 ampere circuit. This general guideline applies to amplifiers driving loudspeakers (not resistors) with musical-type input signals (not tones).

Basic Connections and Wiring

Power and audio signal cables are the most common sources of sound system failure. Well-made and carefully maintained cabling is essential to the reliability of the whole system. If long speaker cables are required, make sure the wire is of sufficient size to transfer all the available amplifier power to the speakers rather than absorbing power itself. As a rule of thumb, the larger the wire, the better (larger wire has a smaller "gauge number").

We have listed the smallest wires (the highest numbered gauges) recommended for best results. To make it simple, we'll assume you're operating under worst case conditions, with 4-ohm loads; 8-ohm operation will improve results with the same wire, and 2⁄ohm operation requires still heavier wire because the cable resistance is a higher precentage of the total load on the amplifier.

LENGTH* UP TO	25 FEET	25 TO 50 FEET	50 TO 100 FEET
MINIMUM WIRE SIZE**	#16 AWG	#14 AWG	#12 AWG

**Length of dual conductor cable (i.e., the 100 foot run specified here from amplifier to speaker represents a 200 foot round trip).*
***Small diam. wire=high gauge #, large wire=low gauge #; AWG is an abbreviation for American Wire Gauge*

TABLE 1. RECOMMENDED MINIMUM SPEAKER WIRE SIZES Every three increments in wire gauge either doubles or halves the wire's resistance.

Large diameter (small gauge number) wire is expensive, and long cables made from it are heavy. Rather than running long speaker cables, it is better to locate power amplifiers near speakers and run a line-level signal cable over the long distance to the amplifier. This approach eliminates most of the signal loss due to speaker cable's resistance so the speakers will be fed all the amplifier's power without the need for heavy cables. It can actually save money in many instances.

Always use stranded wire for three reasons:

(1) it is more flexible and less prone to metal-fatigue breakage.

(2) if an end is nicked while insulation is being stripped for connection, only one or two strands will break, not the entire wire, and...

(3) there is some evidence, though disputed, that higher frequency audio signals flow along the outside of each conductor (skin effect); if this is so, the more strands, the lower the effective cable resistance to high frequencies.

In cases where speakers and power amplifiers are located far away from the signal source (be it a mixer or a preamp), "balanced line" signal cables are a wise choice. This is discussed elsewhere in this manual.

CAUTION

NEVER USE COIL CORDS FOR SPEAKER HOOKUP, even in an emergency. Coiled guitar type cords usually have higher internal resistance than the speakers themselves. This is due to the light-gauge wire used to keep the coil cords flexible. These cords will prevent most of the power from reaching the speakers. In high power operation, a coil cord can melt, cause a fire hazard, and possibly damage the amplifier. As a general rule, guitar-type connecting cords, both straight and coiled, make poor speaker cables.

The 2224 and 2244 can produce enough power output to damage electronic equipment connected to their outputs. Beside being capable of destroying speakers, under certain circumstances shock and/or fire hazards are possible. High power amplifiers should always be properly maintained and used with care in clean and dry environments.

Common Sense Operating Precautions

If you've mounted all your sound equipment in a rack or portable case, you can ensure that everything stays calibrated by marking the settings of the necessary controls. Small pointers made from masking tape are visible in dim light, and can be removed with alcohol or rubber cement thinner without damage to the paint on most front panels, including those of the Fender amplifiers. However, be sure to check the finish in an inconspicuous place to determine the suitability of any cleaner.

Assuming you're NOT turning all the equipment on at once with a switched power receptacle "strip," be sure to turn on the power amplifier last. This will prevent turn-on "thumps" from the mixer or other pieces of gear from possibly damaging speakers. The reverse logic should be used – turn OFF the amplifier FIRST – when shutting the system down.

The 2224 and 2244 have a relay which is timed to turn on the speaker outputs after the amplifier's power supply is fully charged up, thus preventing any turn-on noise. Timing of the amplifier's turn-on circuit is usually sufficient to accomodate all the turn-on anomalies from other pieces of gear in a system, making it acceptable to use a single switched power strip in a permanent or semi-permanent system.

CAUTION

The 2224 and 2244 can draw a lot of AC power. Be sure the AC power source for your AC distribution system has adequate current capability to bear the entire load with an extra margin of safety. If you use a power strip with a built-in circuit breaker, make sure the breaker is rated for sufficient current to handle its load as well.

In multiple amplifier installations, we recommend sequential turn-on (either manually or via timed relays) to avoid a sudden, major drain on the AC line.

You should also keep in mind that severe reduction of power line voltage affects the amount of power you can get FROM the amplifier. If you need to run long AC extension cords, make sure their conductors are as large as practical (small gauge number). Just as smaller diameter wire causes speaker line loss, smaller power lines cause loss. However, the effect of small AC lines is one of intermittent clipping under severe conditions.

By "severe" reduction, we mean when the AC power line voltage drops below 105 volts. Normally, a 12.5% reduction in line voltage like this would cause ordinary power amplifiers to lose output capability and a 12.5% reduction in output voltage would drag even a 400-watt amplifier down to 308 watts!

The 2224 and 2244, however, have triac-fired, "PHASE-CONTROLLED BACK SLOPE" power supplies. This exceptional power supply technology applies only enough voltage to the amplifier power supply from the AC power line to keep the power supply at full power, but under "normal" operating conditions this means that the regulating circuitry of the power supply does not need to use the top 15 volts of the AC line's rated voltage. This "normal" operating mode enables the 2224/2244 power supply to deliver rated power at reduced AC power line voltage – all the way down to about 108 volts. YOU WON'T LOSE POWER AS YOU WOULD WITH CONVENTIONAL AMPS, when the AC power line voltage drops.

This procedure applies to stereo, mono or bridged operation into a full-range loudspeaker system which uses a passive high-level crossover (or none at all). If you are using the 2224/2244 in a multi-amplified system with an electronic or low-level passive crossover, the INPUT LEVEL controls on the amplifier are generally set to maximum (zero loss), and all level controlling is done at the crossover (skip step 10). A more precise procedure using an oscillator and voltmeter may be found elsewhere in this manual.

1. Turn all equipment OFF.
2. Plug the amplifier into a source of 120 Volt, 50-60 Hz AC power. Follow the precautions mentioned earlier in this manual about the current capability of the power circuit.
3. Connect the wiring from the signal source(s) to the amplifier's input jack(s).
4. Select the appropriate settings for the FILTER and MODE switches.
5. Connect the speaker(s) to the output terminals, as appropriate for the setting of the MODE switch.
6. Adjust the INPUT LEVEL controls to their minimum (infinity) setting.
7. Turn everything else ON except the amplifier.
8. Adjust the controls on the signal source for "normal" indications on the source's meter or level indicator. If there is no metering, then set the master control at zero (minimum).
9. Turn the amplifier on. After a short delay, the STANDBY indicator should extinguish (you may hear a click sound from within the amplifier as the output relay engages).
10. Adjust the INPUT LEVEL control(s) to maximum. Carefully advance the master control on your signal source until the sound level from the speaker system is just past the "correct" level; i.e. just a little bit too loud. Remove the input signal from the source, leaving the master control (and any input controls on the source) set as they were. if the system is noisy (hissy), reduce the setting of the 2224/2244 INPUT LEVEL control(s) by one "click" and repeat this step. You must "juggle" the settings of the source's controls and the amplifier's controls until you find a combination that gives you the desired amplifier output, freedom from clipping caused by excessive output demands placed on the signal source, and poor signal-to-noise performance caused by excessive amplifier gain.

Specifications

2244

Output Power (Specs meet FTC pre-conditioning criteria.)
Continuous sine wave output power both channels driven, 20 Hz to 20 kHz @ less than 0.05% THD
8 ohms: 220 watts.
4 ohms: 440 watts.
Single Channel driven, 1 kHz, 0.05% THD
8 ohms: 250 watts typical.
4 ohms: 500 watts typical.
Bridged mono operation, continuous sine wave output power from 20 Hz to 20 kHz @ less than 0.05% THD
16 ohms: 440 watts.
8 ohms: 880 watts.

Frequency Response
+0, -1 dB, 10 Hz-50 kHz.

Power Bandwidth
Better than 10 Hz-50 kHz (3 dB down points from rated power.)

THD and IMD (SMPTE, Stereo or Mono)
Less than 0.05% @ full rated power 20 Hz-20 kHz, 4 ohms. Typically 0.006% @ 1 kHz, @ rated power.

Hum and Noise
100 dB below full output.

Damping Factor
Greater than 200 @ 1 kHz, 8 ohms.

Rise Time
Less than 3.5 microseconds.

Slew Rate
Greater than 37 volts/microsecond. (The amplifier is intentionally rise time limited; see page 13 and subsequent text).

Channel Separation
80 dB @ 1 kHz.

Input Impedance
15 kohms, balanced.

Sensitivity
0 dB (0.775 V) for full rated power.

Power Requirements
108-132 volts, AC, 60 Hz only.

Weight
57 lbs. (24 kg) net.

Dimensions
Height: 5.25 inches (133 mm)
Width: 19.0 inches (483 mm)
Overall Depth: 17.5 inches (445 mm)
Depth Behind Front Panel: 15.5 inches (394 mm)

Accessories
Rear support, adjustable rear rack mount kit (Part # 70-2201)

2224

Output Power (Specs meet FTC pre-conditioning criteria.)
Continuous sine wave output power both channels driven, 20 Hz to 20 kHz @ less than 0.05% THD
8 ohms: 120 watts.
4 ohms: 240 watts.
Single Channel driven, 1 kHz, 0.05% THD
8 ohms: 160 watts typical.
4 ohms: 320 watts typical.
Bridged mono operation, continuous sine wave output power from 20 Hz to 20 kHz @ less than 0.05% THD
16 ohms: 240 watts.
8 ohms: 480 watts.

Frequency Response
+0, -1 dB, 10 Hz-50 kHz.

Power Bandwidth
Better than 10 Hz-50 kHz (3 dB down points from rated power.)

THD and IMD
(SMPTE, Stereo or Mono)
Less than 0.05% @ full rated power 20 Hz-20 kHz, 4 ohms.
Typically 0.006% @ 1 kHz, @ rated power.

Hum and Noise
100 dB below full output.

Damping Factor
Greater than 200 @ 1 kHz, 8 ohms.

Rise Time
Less than 3.5 microseconds.

Slew Rate
Greater than 42 volts/microsecond. (The amplifier is intentionally rise time limited; see page 13 and subsequent text).

Channel Separation
80 dB @ 1 kHz.

Input Impedance
15 kohms, balanced.

Sensitivity
0 dB (0.775 V) for full rated power.

Power Requirements
108-132 volts, AC, 60 Hz only.

Weight
41 lbs. (19 kg) net.

Dimensions
Height: 3.5 inches (89 mm)
Width: 19.0 inches (483 mm)
Overall Depth: 17.5 inches (445 mm)
Depth Behind Front Panel: 15.5 inches (394 mm)

Accessories
Rear support, adjustable rear rack mount kit (Part # 70-2201)

Performance Graphs for 2244 (typical of 2224, also)

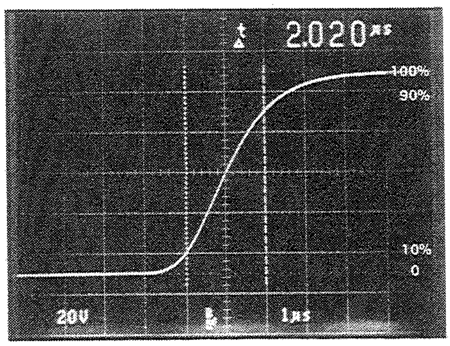


FIGURE 3. RISE TIME.
The amplifier is delivering 100 volts peak to peak (625 watts) into a 4 ohm load at 10 kHz. Rise time is defined as the time to rise from 10% to 90% of the maximum peak amplitude (or peak-to-peak if AC) of a linear system. The measured time is 2.020 µsec. Vertical scale is 20 V/division, Horizontal scale is 1 µsec/division.

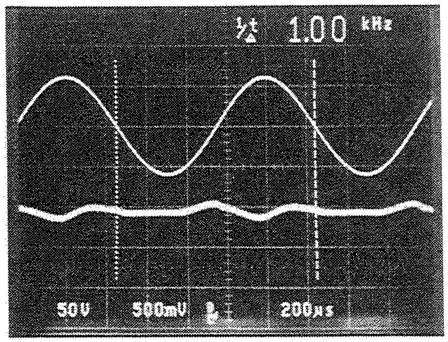


FIGURE 4. HARMONIC DISTORTION AT FULL OUTPUT, 1 kHz.
The top trace shows the output signal, the full rated 440 watts. The bottom trace, highly magnified by the distortion analyzer, shows the residual harmonic distortion (0.0046% THD).

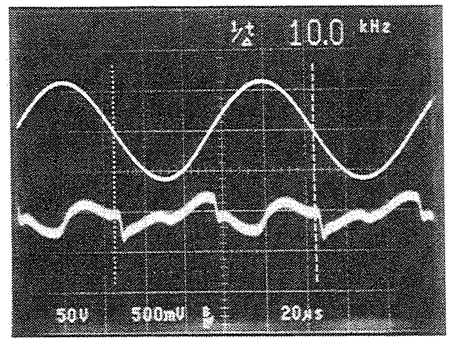


FIGURE 5. HARMONIC DISTORTION AT FULL OUTPUT, 10 kHz.
The top trace shows the output signal, the full rated 440 watts. The bottom trace, highly magnified by the distortion analyzer, shows the residual harmonic distortion (0.0131% THD).

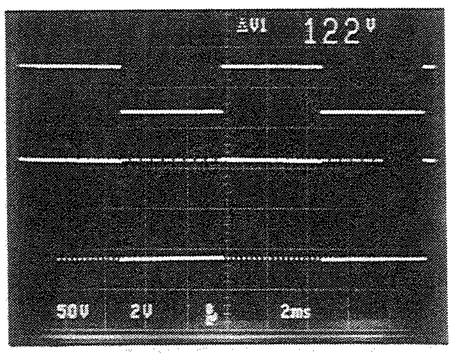


FIGURE 6. LARGE SIGNAL SQUARE WAVE RESPONSE, 100 Hz.
The top trace shows the amplifier input (2 V/div). The bottom trace shows the amplifier output (50 V/div). The dotted cursor on the bottom trace indicates a peak-to-peak output of 122 volts. The amplifier is delivering 930 watts into a 4 ohm resistive load (single channel driven).

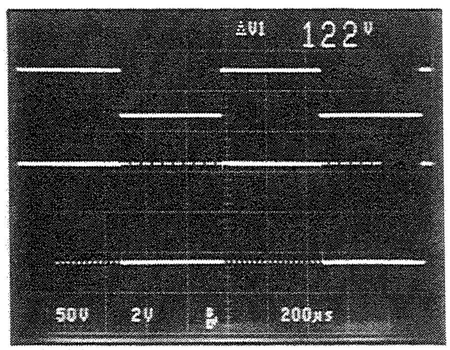


FIGURE 7. LARGE SIGNAL SQUARE WAVE RESPONSE, 1 kHz.
This graph is similar to Figure 6, but is measured at 1 kHz. Again, the output remains at 122 volts P-P.

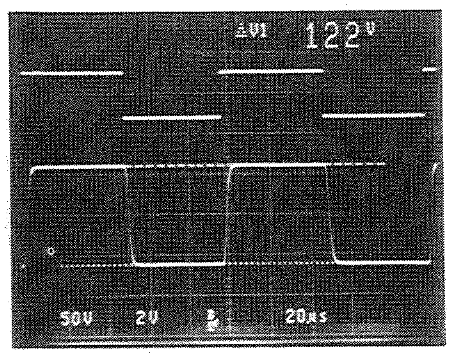


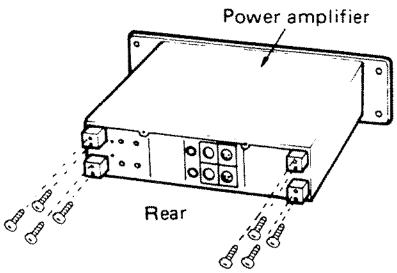
FIGURE 8. LARGE SIGNAL SQUARE WAVE RESPONSE, 10 kHz.
This graph is similar to Figures 6 and 7, but is measured at 10 kHz. Again the output is 122 volts P-P, demonstrating excellent power bandwidth. Note that the maximum output with a 10 kHz square wave is rise time limited, indicating a slew rate greater than 42 volts per microsecond. Note also that there is no overshoot, ringing or other abberations on the leading and trailing edges, indicating excellent high frequency performance. Most amplifier square wave photos are done at the 1 to 10 watt level, rather than the high power used here, because the typical amplifier's power response is worse than its small signal response. Fender amplifiers perform well at all power levels.

Equipment Rack Installation

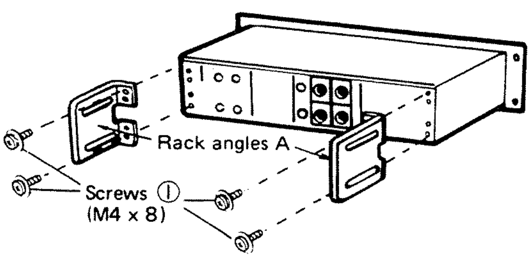
PARTS LIST

Rack angle A 2
Screw ① (M4x8, with spring washer and plain washer). 4
Rack angle B 2
Screw ② (M5x12, with spring washer and plain washer). 4

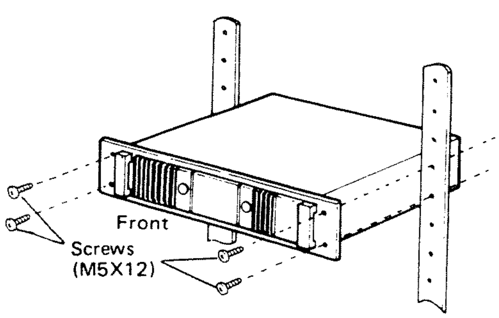
1. **Remove the rubber feet (for cord winding) on the rear of the power amplifier.**



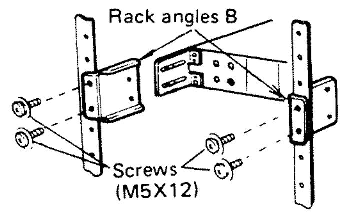
2. **Mount the rack angles A to the holes from which the rubber feet were removed, using screws ① (M4 x 8) supplied.**



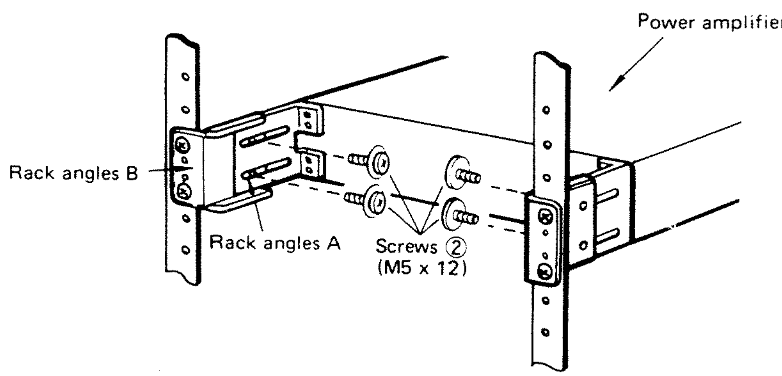
3. **Mount the power amplifier in a rack with four screws in the normal manner.**



4. **Fit the rack angles B on the rear side on the rack with screws (M5 x 12) and plain washers. Align as necessary. Fit rack angles B to rear rack rails, with similar screws as used for front mounting.**



5. **Secure the rack angles A mounted on the power amplifier and the rack angles B mounted on the rack to one another using ② screws (M5 x 12) supplied.**



6. **When alignment adjustments have been completed, firmly tighten all screws.**

Note: When mounting the rack angle A, use Holes C to mount the model 2224 power amplifier, and use Holes D to mount the 2244 amplifier.

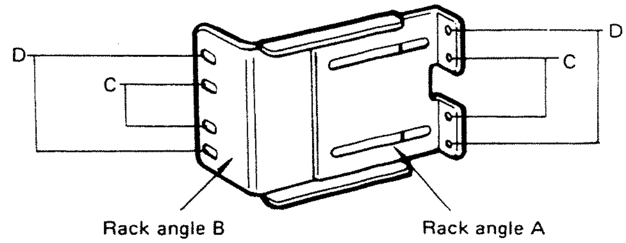


FIGURE 9. INSTALLATION OF REAR SUPPORT FOR RACK MOUNTING.

Frequency response

Frequency response is a measure of an amplifier's ability to amplify all frequencies equally from low to high frequencies; it describes any change in the amplifier's output level due to changes in frequency. "Flat" frequency response refers to the flat or straight portion of the line on a frequency response graph. The portion of the response that is flat should extend slightly beyond the limits of useable audio frequencies to ensure that all audio signals being amplified are unchanged with respect to their frequency balance. Although a figure of merit of 20 to 20,000 Hz is often used in conjunction with audio tests and specifications, few people actually hear as high as 20,000 Hz and few practical speaker systems respond down as low as 20 Hz with the exception of some large subwoofers (more on subwoofers later). For your reference, we've included a chart that shows the frequency of the fundamentals on the piano keyboard and how they relate to the audible frequency spectrum. Harmonics account for almost all the energy heard in musical signals above 4,000 Hz, and only organ pedals and a few piano and synthesizer tones fall below 41 Hz, or "low E" on the open E-string of the Fender bass. Piano and other acoustic instruments make little fundamental output from their lowest notes, and you hear mostly higher frequency harmonics when these instruments play in their lower registers. Even electric guitars have a very limited frequency range.

Even though musical signals are somewhat limited in frequency, it's important that the amplifier be capable of covering more than the desired frequency range, and it's also important that the amplifier be able to supply flat frequency response at any power level – something sorely lacking in some so-called high power amplifiers (the 2244 can supply 440 watts from one channel all the way up to 50 kHz, more than two times the upper frequency limit of the best ears).

As an example, let's consider percussion instruments. Percussion sounds often exhibit waveforms characterized by sharp pulses and steeply rising wave shapes. Even a single pulse can be defined in terms of frequency, by inverting frequency and time to obtain "period". A 1,000 Hz sine wave (pure tone) is said

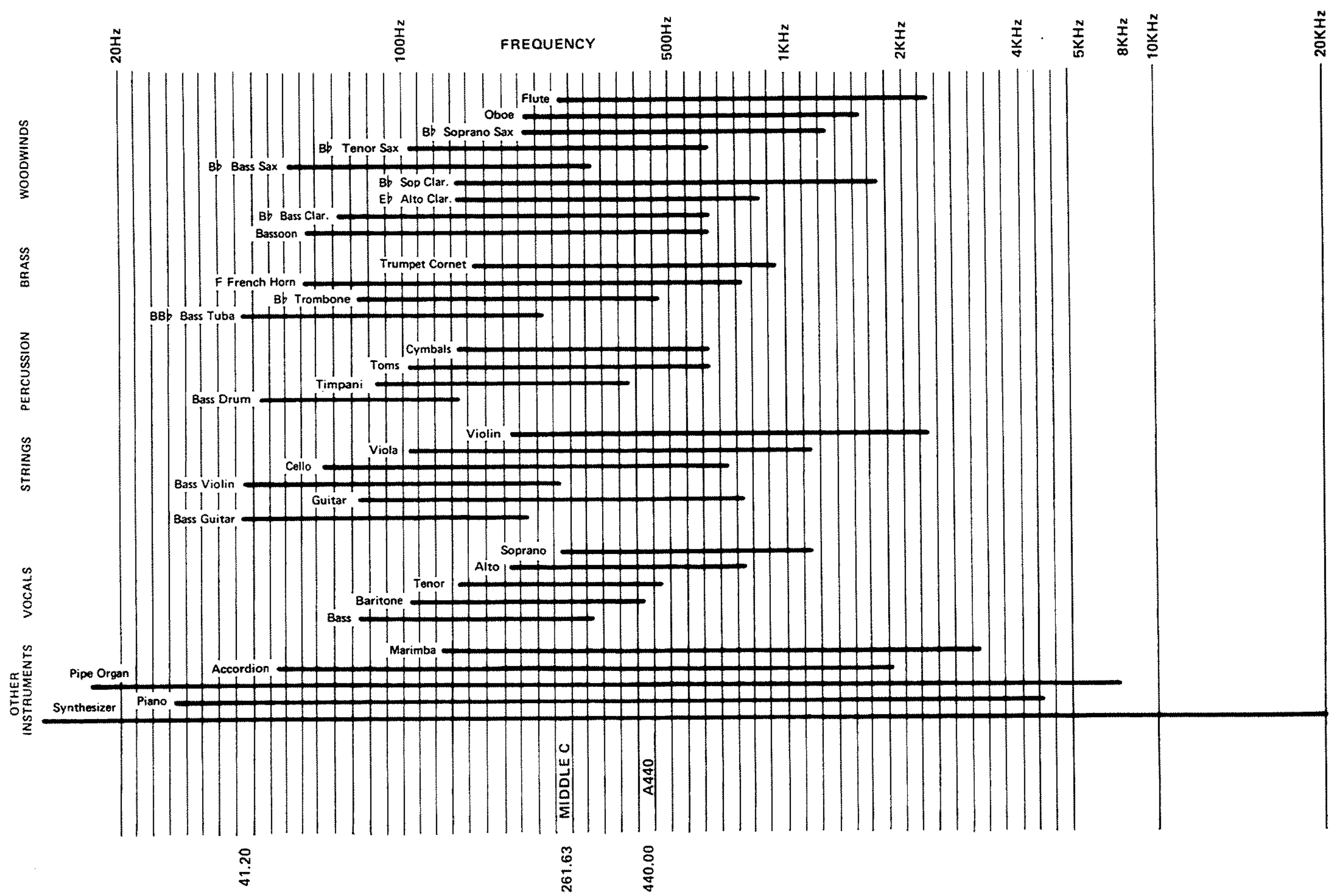


FIGURE 10. FREQUENCY RANGE OF TYPICAL INSTRUMENTS.
While the fundamentals are relatively low in frequency, harmonics extend up several octaves.

to produce 1,000 cycles per second. Therefore one of the cycles takes 1/1,000 of a second or 1 millisecond. A pulse from a percussion instrument such as a triangle may be as short as 10 to 50 microseconds (millionths of a second). Translated into frequency, 50 microseconds turns out to be one cycle of a 20,000 Hz frequency, 10 microseconds is equivalent to one cycle of a 100,000 Hz frequency, and a pulse is actually more like a half-cycle, which causes the amplifier to amplify only to one side of zero volts--a tricky waveform to reproduce accurately. Such pulses cannot be accurately reproduced by an amplifier which rolls off rapidly above 20,000 Hz. It is necessary that the amplifier's frequency response extend beyond merely audible limits to ensure the proper reproduction of steep pulses and upper musical harmonics. While you may not listen to 20,000 Hz sinewaves, if a "live" instrument produces high frequency energy in these upper frequency ranges, faithful reproduction of the original sound can only be achieved if the amplifier is capable of supplying solid output at these high frequencies. The amount of additional bandwidth required (if at all) is a matter of some debate. The small-signal (1 watt) frequency response of the 2224/2244 amplifiers extends to 100 kHz.

The 2224 and 2244 have a switchable filter designed to eliminate very low frequencies at the amplifier's input. The filter's cutoff frequency (−3dB point) is selected by the rear-panel FILTER switches. In the OFF setting, the frequency response is nearly flat to DC (zero hertz). The other two settings select a 20 Hz or 40 Hz rolloff. These two settings provide speaker protection against DC shifts and transients, such as "turn-on thumps", from equipment plugged into the amplifier's input. A 40 Hz filter may be selected for a greater measure of protection, and for conservation of valuable amplifier power. With it, musical sounds, most of which occur above 40 Hz, will suffer less degradation from unwanted sub-sonic signals such as wind noise, turntable rumble, stage vibration and microphone handling noise; such signals merely serve to heat voice coils and create distortion. Since compression drivers are particularly sensitive to low frequency signals, the 40 Hz filter is always recommended when the amplifier is being used to drive high-frequency loudspeakers or horns. Even a small "thump" can permanently deform a compression driver's diaphragm, causing early failure. It is advisable to select one of the two built-in high-pass filters (low frequency rolloff), either 40 Hz for standard woofers, or 20 Hz for subwoofers; one of these two settings is recommended at all times. If you don't hear a marked effect with the higher frequency high pass filter engaged, leave it selected.

CAUTION

Using the amplifier without one of its high pass filters will increase the risk of damaging your speakers from extreme cone excursions due to inaudible very low frequency signals such as record warp or low frequency "envelopes" from effects devices.

Power response (power bandwidth)

Bandwidth refers to the frequency range covered by an amplifier or signal processing device, from its lower frequency limit to its upper limit (the limits being defined by the −3 dB points). When measured at a nominal or low signal level (typically 1 watt in a power amplifier), the bandwidth is called "Frequency Response". When measured at maximum power, it is called "Power Response" or "Power Bandwidth."

Power response defines the FREQUENCY LIMITS of the amplifier's ability to deliver at least HALF the power (−3 dB) that it can normally deliver near the middle of the audio frequency spectrum, which is typically taken to be 1,000 Hz. The bandwidth for the power response is determined by a test where the amp gain and input signal level are set for rated power with a midband signal, then the input frequency is gradually raised until the amplifier's output is no longer "flat", but instead falls by 3 dB. The 2224/2244 have the same power response and frequency response.

Most contemporary music has substantial energy in the high frequency range compared to other types of music. For accurate reproduction at high levels, it's important that the amplifier be able to deliver large amounts of power at high frequencies as well as mid and bass frequencies, otherwise the music will suffer a progressive loss of treble as the volume is increased. The 2224 and 2244 have a power bandwidth of from 10 Hz, to 50,000 Hz, assuring that reproduction will be natural sounding and unrestricted at any volume all the way up to full rated power output.

How wide a bandwidth is adequate?

Human beings hear over a frequency range of from about 16 Hz at the lower limit, up to about 16,000 Hz at the upper limit, except for young children and a few rare adults who sometimes hear as high as 20,000 Hz. Some pipe organs have 32-foot pipes used to produce "low C" at 16 Hz, but most of the very largest pipe organs have only 16-foot pipes for the lowest notes – around 32 Hz (the frequency changes by a factor of two depending on whether the pipe is open or capped.) The average kick drum used for rock music is tuned between 50 and 125 Hz, and the lowest note on the bass guitar is E at 41.2 Hz. At the upper limit of audibility, 15,000 Hz is sufficient to provide all the "air" of cymbals, wind chimes, triangle, trumpets, violins, and voices so they all sound lifelike.

Many people involved deeply in consumer audio equipment insist that frequency response must extend far beyond the limits of audibility, and their arguments are right on the mark for carefully selected, expensive consumer audio equipment placed in exactly the right spot in a carefully prepared listening room. According to an IHF survey, only about one person in 10,000 in the U.S. has ever heard audio of sufficient fidelity to perceive the subtle differences audiophiles argue about. In concert use, a total bandwidth from about 40 Hz to 15 kHz is all that is necessary so that most of the audience won't notice any frequencies lacking. This does not conflict with the advisability of limiting the low frequency response of a sound reinforcement system to save power and protect speakers.

Phase response

Phase response refers to the TIMING RELATIONSHIP between high and low frequency signals being amplified. A poor circuit design is capable of shifting the phase of waveforms passing through it. When this happens, the shape of complex waveforms is altered, which some people claim changes the sound. Figure 11 shows one cycle of a sine wave along with the names and numerical relationships of the parts of that cycle. Note the markings at 90, 180, 270, and 360 degrees. For one sine wave to be 90 degrees "out of phase" with another means that the two waves start and finish at different times, even for the same frequency.

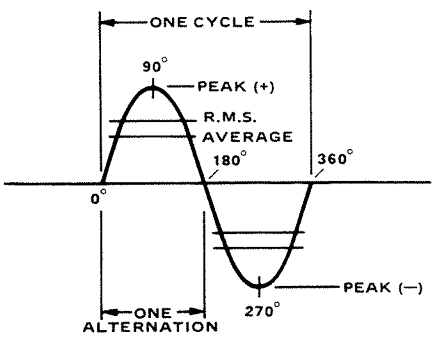


FIGURE 11. BREAKDOWN OF A SINE WAVE.

In Figure 12, which shows two sine waves that are out of phase by differing values, the third example (180 degrees) is unlikely to occur in an amplifier. Rather, it is typical of a polarity reversal, as if one of a pair of speakers were wired backward (black for red), so that one speaker pushes while the other one pulls. While polarity reversal means that one speaker would always be "out of phase" with the other (an extreme case of phase shift), phase shift is the change (in degrees) of the amplifier output to its input, versus frequency. Typically as the input frequency is raised, the high frequencies take more time to pass through the amplifier than do the low frequencies. High frequency phase shift is a fact of life in any amplifier other than a theoretically ideal unit that does not exist in the "real world." Excessive phase shift is an indication of poor amplifier design, and is generally accompanied by poor frequency response.

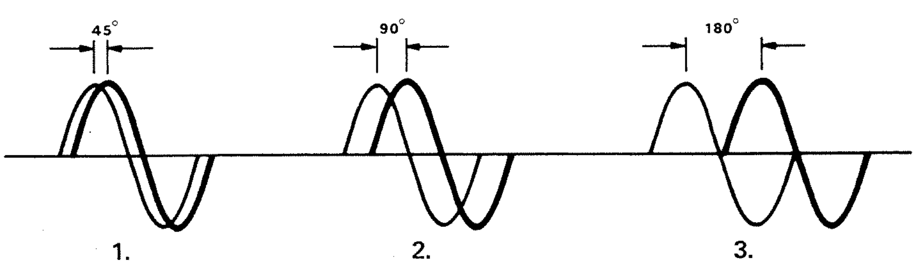


FIGURE 12. OUT OF PHASE SINE WAVES.

Phase shift is probably not a major concern with a quality amplifier such as the Fender 2224/2244, especially since most speaker systems used in professional applications will exhibit much more phase shift than the amplifiers driving them. Many other factors, such as listening room sound absorption, standing wave addition and cancellation, etc., affect phase at low frequencies by much greater amounts than amplifier-contributed phase shift.

Rise time and slew rate

Although rise time and slew rate seem to be two names for the same thing, they are not. Slew rate is the MAXIMUM RATE OF CHANGE OF VOLTAGE that an amplifier can produce (measured in volts per microsecond). Rise time describes the time it takes for an amplifier's output voltage to rise from one given point on the leading edge of a square wave to another point on the same edge (when the amplifier is operating within its linear region). The two points used for rise time testing are from 10% to 90% of the square wave's height. The area of interest is darkened in this diagram:

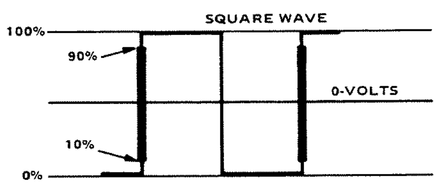


FIGURE 13. HOW RISE TIME IS MEASURED (OVERVIEW).

The test signal is fed through the amplifier and into an oscilloscope. Since the oscilloscope is much faster than the amplifier, the sweep speed of the oscilloscope is adjusted to be fast enough to show the leading edge of the displayed square wave tilt. Then the horizontal displacement of the 10% and 90% points is read and converted to time, i.e., to RISE TIME. (See Figure 3 for actual rise time measurement).

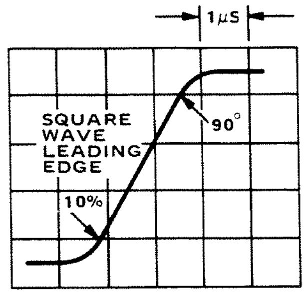


FIGURE 14. HOW RISE TIME IS MEASURED (DETAIL).

Square wave test signals must be close to perfect, with leading and trailing edges that are as nearly vertical as possible. This implies very fast rise and fall times for square wave test signals. These times must be much faster than the amplifier being tested so as not to influence the measurement. The oscilloscope, too, must be much faster than the measurement being made. In the example pictured, the oscilloscope's electron beam is traversing the screen at a rate of one screen division every microsecond, and the test signal's leading edge is taking one and one-half divisions to rise from 10% to 90%, so the rise time is said to be 1.5 microseconds. The rise time measurement is made with the amplifier's output adjusted to 1 watt because it's important to keep the output and the test reading free of the effects of "slew rate limiting", which will appear to slow down the rise time of many amplifiers.

The 2224/2244 are intentionally "rise time limited" to minimize distortion by not allowing the amplifier to try to increase its output voltage faster than the circuit topology permits. If the input were not rise time limited, then a "splattering" known as "T.I.M" (Transient Intermodulation Distortion) could occur due to successive stages of amplification that are "slower" than the signals being fed through them.

The term SLEW RATE relates directly to the amplifier's ability to follow waveforms with steep leading or trailing edges. Slew rate is the maximum rate of change of voltage (or current) with respect to time; it usually is measured with the amplifier clipping while using a very high frequency test signal. When the amplifier reaches its slew rate limit (it runs out of speed), it can no longer make its output a verbatim copy of the input signal. Ultimately, the output signal resembles a triangle (see Figure 15) when the frequency is sufficiently high. If the input frequency is raised further, the triangle waveshape remains, but with decreasing amplitude. When these triangles come to a point, it is clear that the amplifier is spending all its time slewing its output voltage up and down in response to the input, but since it can't match the input signal for speed, no time can be spent at the top or bottom of the wave. This phenomenon is known as "triangulation", and interestingly, when an amplifier is "triangulating", the input waveform shape can be changed without affecting the output waveform.

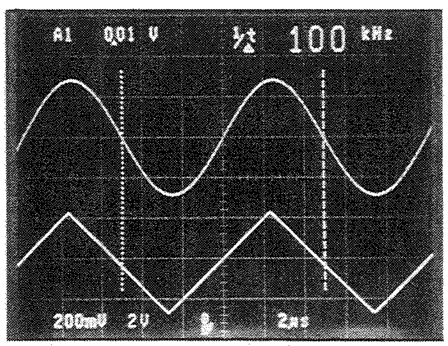


FIGURE 15. SLEW RATE LIMITING: SMALL VS LARGE SIGNAL.
This actual distortion analyzer photo depicts the performance of an amplifier during linear, small-signal operation at 100 kHz, and the non- linear slew rate limiting during large signal operation. The top trace is the undistorted output of the system with a 600 mV P-P (212 mV rms) signal. At large signals of 5 volts P-P (bottom trace), the amplifier is slew rate limited to a value of 1 volt per microsecond rate of change. This causes the triangulation since the output cannot track the sine wave input. Mathematically, the slew rate is defined as dV/dt, or change in voltage divided by change in time.

Gain

"Gain" is the essence of what an amplifier does—and the specification tells you how much it amplifies. Gain is an amplitude scaling factor, referring either to power or voltage. Thus we use the term "voltage gain" or "voltage amplification" for the 2224/2244 because they are voltage source amplifiers. Stated in dB, voltage amplification (gain) is 20 times the base 10 logarithm of the ratio of the output voltage and input voltage of the amplifier. (Refer to the section of this manual titled "The Decibel; what is it, and why is it used?") Amplifier circuits, including those of the 2224 and 2244, are usually designed to give a fixed amount of voltage amplification. Most amplifiers then add an attenuator (volume control) to effectively reduce the fixed gain from maximum by throwing away some of the input signal. Amplifiers do not amplify more when their volume control is turned up; they attenuate less. The attenuators on most amplifiers are capable of shutting off the flow of input signals altogether. With no attenuation, controls

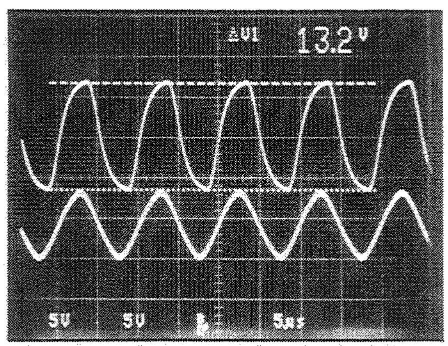


FIGURE 16. SLEW RATE LIMITING: LABORATORY VS "REAL WORLD" CONDITIONS.
The top trace shows the undistorted 100 kHz squarewave output of an electronic crossover delivering 13.2 volts P-P (+15.3 dBv in this case) into a 15 kohm load resistor, a load typical of many amplifier inputs. The 100 kHz square wave is rise time limited, indicating a slew rate in excess of the 8 microvolt per second specification of the 5532 IC operational amplifiers used in the crossover. The bottom trace clearly shows the effects of inserting 200 feet of Belden 8451 shielded cable between the crossover and the amplifier. Note that the peak amplitude is reduced as a result of the finite output impedance of the crossover. Under these conditions, the crossover is now slew rate limited to just 3 volts per microsecond. This phenomenon occurs because the cable capacitance and the 5532 IC together determine the actual slew rate. Remember SR = dV/dt = i/c. Clearly the crossover output stage must deliver 2.6 times more output current to maintain the original slew rate of 8 volts per microsecond. Fender mixers are designed to avoid such problems.

set at "0", wide open, the 2224 and 2244 require 0.775 volts to drive them to maximum rated power output with a 1,000 Hz sine wave input signal.

NOTE
Reducing the setting of the input attenuator does not reduce the amplifier's maximum output power. Rather, it increases the voltage (signal level) necessary to produce full power output.

The 2224 provides a maximum of 32 dB of voltage amplification (40:1 output to input voltage ratio). The 2244 provides 34.7 dB of voltage amplification (54:1 output to input voltage ratio).

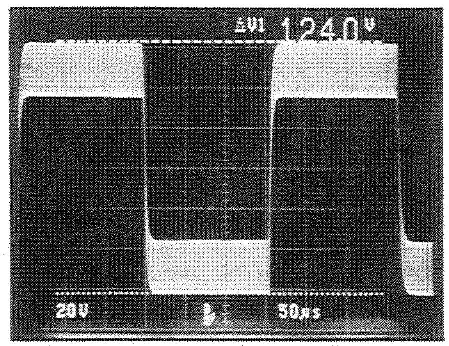


FIGURE 17. TRANSIENT INTERMODULATION DISTORTION TEST.
This is a TIM 100 test signal as measured at the output of the 2244. This represents the maximum undistorted output level of 317.13 watts into 8 ohms or 634.26 watts into 4 ohms. The test signals are a 3.18 kHz square and a 15 kHz sine wave mixed in a 4:1 ratio. The -3 dB point is 100 kHz. Transient Intermodulation (TIM) distortion for both 8 and 4 ohm loads is below 0.01%, the limit of the spectrum analyzer used.

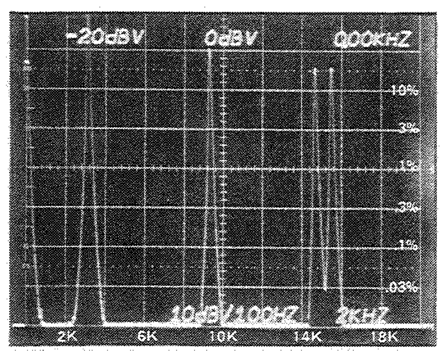


FIGURE 18. SPECTRAL ANALYSIS OF TIM TEST.
This graph depicts the frequency spectrum of the signal shown in Figure 17. The TIM is actually below the limit of the analyzer, which is -80 dB (0.01%). The vertical scale is 10 dB/division, the horizontal scale is 2 kHz/division. The frequency range covered is from DC to 20 kHz. TIM remains below 0.01% for both 4 ohm and 8 ohm loads. The maximum SMPTE Intermodulation distortion is 0.05% for the 2224 and 2244.

An overview of system wiring

When electrical signals pass through wire, magnetic fields are created. When fields external to the wire are strong enough, they can in turn, induce signals in any nearby wires. Keeping signal-carrying wires in groups that share the same signal levels is better than bundling all wiring together. Massive bundles of dissimilar cables permit signal "cross-talk" (wire-to-wire signal leakage) which is worst between wires carrying signals of greatly different levels, such as speaker wiring and microphone wiring. In fancy permanent installations, mic, line, and power cables are often placed in separate steel conduit pipe. You can apply the same wiring logic in a portable sound system rack by running your wiring in neat, separated bundles, each tied or wrapped together with cable ties or electrical tape.

Once your wiring exits the system rack enclosure, no extraordinary care is required except to make sure the wire suits the signals it's carrying, and of course, common sense things like keeping the stage wiring neat and taping down and/or covering wiring wherever lots of foot traffic is expected over the wires.

In most "rock-n-roll" sound systems, it's fairly common practice to run the line level returns to the power amplifiers down the same multi-pair snake used to carry mic signals to the mixer. If you anticipate having to operate the mixer at or near its maximum gain capability, or if you experience oscillation problems, use a separate snake or group of cables to keep the high level and low level signals separated physically.

Whatever you do, never run the AC power for your mixer or other equipment up a microphone snake (someone actually tried this once). At the very least, you would be tempting fate. At worst, equipment could be destroyed and a performer or technician could suffer lethal shock.

Proper sound system wiring is expensive. When designing the system be sure you figure in the cost of wiring and connectors, cable accessories, ties, hardware and such. Remember, THE MOST COMMON CAUSE OF SOUND SYSTEM FAILURE IS POOR WIRING. Be conscientious about wire and connectors – it pays.

Microphone cable generally gets the most abuse. Choose cable that has a sturdy jacket and copper/bronze inner conductors if possible. Size is not critical except for two points: the amount of tugging and abuse the wire will get, and the length. If the length is going to be very long – over 100 feet – it's a good idea to use a slightly stronger cable; some types have steel or copper-clad steel reinforcing strands. Color-coded cable is handy, it helps the engineer visually locate a particular microphone on stage. Belden 8412 is a rubber covered, robust cable. Microphone cables made from it (or another manufacturer's equivalent) are likely to survive the rigors of constant setup, use and teardown longer than less costly cables. Other manufacturers offer cabling of different types, colors, and wire gauges to suit almost any specific application. Local or regional distributors of cable products can be very helpful when you need more detailed information.

If you're running cable in permanent installations, mic cable or "balanced line" wire with hard plastic outer insulation and foil shielding is better because it has a smaller outer diameter so many more wires will fit inside the same conduit and can be pulled through conduit more easily. This type of wire is usually not very flexible, but in permanent installations that's not a problem. If you're using unbalanced wiring in high impedance circuits (like a guitar pickup or high impedance microphone), look specifically for low capacitance cable, to avoid high frequency signal losses. (This is a good idea for balanced wiring, too.)

If you make your own cables, the extra time and effort that you spend doing a first-class job will pay you back by giving you a reliable cable that lasts and lasts. Good soldering technique is a must. Take your time, especially when preparing (stripping and trimming) the cable. Remember to pass the connector's shell down along the cable (in the proper orientation) before you solder the connector in place. Be neat.

Connectors are the most common source of reliability problems. You can greatly improve the reliability of the terminations by using one or more layers of heat-shrink tubing under the connector strain relief. This tightens the grip of the strain relief and prevents severe flexing close to the connector and helps to avoid premature cable failure. In the long run, this kind of common-sense preventive maintenance will prove to be a big time and money saver, and may very well save a performance.

Remember, for speaker cabling, use the largest wire size (smallest gauge number) practical. Speaker cables, especially for portable use, should be rugged. Wire normally used for heavy-duty AC power cables and utility extensions such as those for 240-volt industrial power tools is a good choice. It has heavy rubber or vinyl outer insulation, stranded conductors, and is usually fairly flexible. Wire gauge is, however, the most important consideration, especially for long speaker cables. Remember to always calculate the resistance based on twice the distance to the speaker since the signal must travel up one conductor and back the other to complete the circuit.

WIRE GAUGE	4 OHMS	8 OHMS	16 OHMS
10 Gauge	−0.44 dB	−0.22 dB	−0.11 dB
12 Gauge	−0.69 dB	−0.35 dB	−0.18 dB
14 Gauge	−1.07 dB	−0.55 dB	−0.28 dB
16 Gauge	−1.65 dB	−0.86 dB	−0.44 dB
18 Gauge	−2.49 dB	−1.33 dB	−0.69 dB

TABLE 2. SIGNAL LOSS IN 30 M (100 FT) SPEAKER CABLE.
This chart shows nominal losses (in dB) for a 30 meter (100 ft) cable run of different gauges driving a 4, 8, and 16 ohm load. (Note: since there are 2 conductors, and the signal flows through both, the actual round trip cable run is 60 meters or 200 feet.)

For a fixed "ideal" load, these relationships are logarithmic. However, for typical speaker impedances and cables, a linear calculation will yield reasonably close approximation of the signal loss. Thus 15 m of wire (50 feet) would give about half the loss shown on the chart, and so on. For example, ten feet of 10-gauge wire driving an 8-ohm studio monitor will produce a loss of 0.022 dB. If the wire's resistance becomes significant compared to the overall load impedance, heating will occur in the wire during high power operation.

Signal levels

Different types of audio circuits have different operating levels. Engineers usually divide these operating levels into three categories:

1. MIC LEVEL OR LOW LEVEL from no input up to about −20 dB (77.5 mV). This would include microphones, guitar pickups, phono cartridges, and tape heads, prior to any form of amplification (i.e., before any mic, phono, or tape preamps).

2. LINE LEVEL OR MEDIUM LEVEL from –20 dB to +30 dB (24.5 V). This includes preamp and mixer outputs, and most of the inputs and outputs of typical signal processing equipment such as limiters, compressors, time delays, reverbs, tape decks, and equalizers. Basically, this includes the outputs of any equipment not designed to drive speakers directly.

3. SPEAKER LEVEL AND HIGH LEVEL +30 dB (24.5 V) and higher. These levels would include speaker lines, power lines, and DC control cables carrying more than 24 volts. (Note: As used here, dB refers only to voltage; no power level or specific load is implied.)

Balanced and unbalanced lines

Certainly the wire you use to connect the speakers to the amplifier can make a big difference in system performance. However, it is just as important to pay attention to the wiring which carries signal up to the amplifier? Improper cables between the signal source and the amplifier can result in exaggerated or deficient high frequency response. The line-level signal carrying wiring in an audio system is as much an audio "component" as any other part of the system, and taken for granted, it can cause trouble and degrade system performance.

If we were to simplify the possible wiring techniques, we could say there are two types of signal transmission systems – the balanced line, and the unbalanced line. Either type can be used with high or low impedance circuits. The UNBALANCED LINE is simply a "two-wire" system where the ground acts as one wire, and the signal "hot" wire is enclosed within that ground – shielded by the ground "wire" itself. The ground wire is typically a braided shell of fine wires surrounding the insulated inner "hot" conductor.

The BALANCED LINE is a three-wire system where two signal "hot" wires share an equal amount of potential or voltage WITH RESPECT TO the ground wire, but with opposite electrical polarity from each other. Usually the ground wire is used as a shield surrounding the two signal "hot" wires.

Balanced wiring is more expensive and more complicated, but offers important advantages. It is the application that dictates whether one system or the other is needed to yield satisfactory results. In electronics laboratories, where critical measurements are made and high-precision test equipment is used, UNBALANCED wiring is often used. Unbalanced wiring works well when high-quality wire is used for relatively short distances, as with test gear, and where it's possible to use one power main to feed all the gear. Radio transmitter feeds, computer high-speed data transmission, and ultra-wideband television signals are also fed over unbalanced lines exclusively. In short, there is nothing inherently "unprofessional" about unbalanced wiring. However, outside the laboratory, TV or recording studio, or other such controlled environment, things get complicated by radio transmitters, motors, fluorescent lighting, lamp dimmers, automobiles, and other sources of magnetic and electrostatic "noise."

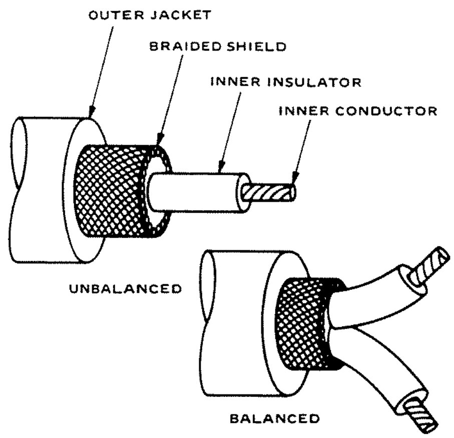


FIGURE 19. CABLE FOR UNBALANCED AND BALANCED LINES.

In electrically "noisy" environments balanced wiring serves to eliminate noise in an ingenious way; the two wires of the "balanced" cable carry the same signal, but each wire is opposite in signal polarity to the other. Balanced inputs are designed to recognize only the DIFFERENCE in voltage between the two wires, and by nature, interference or noise will appear equally – in the same polarity – on both wires, and is thus ignored or "rejected" by the input circuit. Balanced wiring is used in telephone systems where wires run many miles at a stretch without any shielding (that much shielding is too expensive). Phone lines sometimes run hundreds of miles and there are many thousands of lines. Simple two-wire lines are by far the least expensive way to run signals through wire over long distances. Out in the open wires are subjected to radio interference and hum fields set up by power lines. Balancing the two signal hot wires with respect to ground gives long lines immunity to external interference. Using just two simple wires twisted together makes the two signal hot wires subject to exactly the same amount of bombardment from hum and radio sources, so a balanced input (either transformer or active, differential amplifier) can be used to cancel out unwanted signals on the line while passing the desired audio signal. Figure 20 illustrates the principle of balanced-line interference rejection.

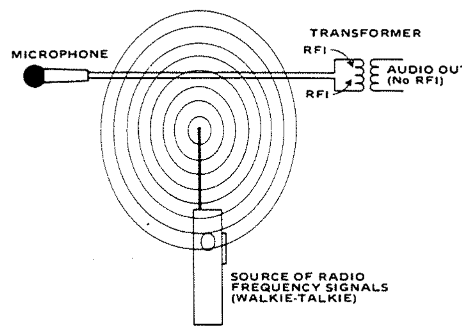


FIGURE 20. NOISE REJECTION IN A BALANCED LINE.

The RFI (radio frequency interference) cuts across both conductors, inducing equal voltages in the same direction. These voltages "meet" in the transformer, and cancel out, while the signals generated by the microphone flows in opposite directions in each conductor, and hence do not cancel out. Thus, in a theoretically perfect balanced system, only the desired signal gets through the transformer.

Interfacing balanced to unbalanced equipment

Special "bridging" transformers are available to interconnect balanced and unbalanced audio equipment. Such transformers are useful for isolating the ground connections of audio equipment.

For example, if no transformer is used to connect a balanced output to an unbalanced input, then the balanced line com-

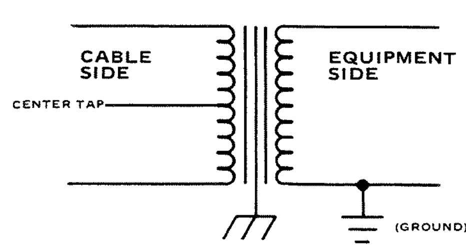


FIGURE 21. SCHEMATIC OF BRIDGING TRANSFORMER.

ing from the output will be made unbalanced. At the unbalanced input, one of the two active or "hot" wires must be connected to ground, and the other to the signal "hot" terminal. In the case of balanced-line outputs, which wire becomes "hot" and which becomes "ground" determines the signal polarity fed to the input of the unbalanced unit. Therefore, it's a good idea to check the operating manual of the balanced unit for advice as to which wire to use. If you are connecting balanced equipment to unbalanced equipment directly, use adaptors designed for that job. It's too easy to make up special cables for specific situations, and have these same cables find their way into other system setup situations where they won't work. For hum rejection, the balanced wiring is always brought out to the unbalanced unit, and the appropriate inner conductor of the balanced cable is connected together with the cable shield, inside the adaptor used to plug into the input connector of the unbalanced unit.

It's worth mentioning that systems with no ground at all are common, and they can deliver perfectly interference-free performance. Our portable cassette machine example is one such "system" (recorder, preamp, amplifier and speaker in a box). Professional audio systems too, can be set up without a ground, but, for AC safety reasons, the practice is not recommended.

Why cable capacitance can affect the signal

Whether balanced or unbalanced wiring is used to connect an audio system together, it's important to take into account the properties of the cabling itself. Cable has electrical properties; resistance, capacitance and inductance. The electrical properties of cable (R, C and L) behave in much the same way resistors, capacitors, and inductors would behave if they were substituted into the signal transmission path in place of the wire. The cable and associated inputs and outputs form a FILTER that removes some of the high frequencies from the signal. The cable thus can significantly alter the frequency response of the system as a whole. Let's look at the filter characteristic of cable: a brief equation describes the loss of high frequency as a function of the capacitance in the cable combined with the resistance of the associated input/output circuit.

$f_O = 1 / (2\pi RC)$
$\pi = 3.1416$
R = Resistance (ohms)
C = Capacitance (farads)

The biggest part of the problem with signal transmission circuitry is cable capacitance. Raising the value of "C" lowers the -3 dB frequency (f_O) as the equation shows. The f_O frequency is the frequency where the loss is 3 dB, THE LOSS GETS WORSE WITH HIGHER FREQUENCIES, at a rate of 6 dB every octave. In extreme cases, enough high frequency can be lost to make the system sound defective – as though someone had turned down a treble control all the way. The internal output resistance of the preamp or mixer, combined with the cable capacitance and input resistance of the next amplifier in the signal chain create a circuit that functions as a low-pass filter. The capacitance of the cable shunts higher frequencies to ground by an amount dictated by the capacitance itself AND by the output resistance (impedance) of the driving source.

In the circuit shown in Figure 22, we'll assume that the output impedance Z_O is a lot lower than the input impedance Z_i, so the latter can be disregarded for practical quick calculations. The capacitance of the cable (C cable) is lumped together for easy visualization. That simply means that even though the cable has a certain amount of capacitance distributed over its entire length, the total capacitance can be treated as one term. The equation, $f_O = 1/(2\pi RC)$ can now be used to determine roughly how much high frequency loss we'll suffer given a certain output impedance and cable capacitance. For the sake of our example, let's use some typical values. We'll say our preamp has an output impedance of 200 ohms, and our cable is 100 feet of 8451-type wire whose total distributed capacitance lumped together is about 6.7 nF. (Belden 8451 cable in 100 foot lengths or so has 67 pF/foot capacitance between one conductor and the other when the other is connected to the shield, and 34 pF/foot capacitance between conductors when the shield is not connected.) To calculate the loss:

$f_O = 1/(6.28 \times 200 \times 6.7 \times 10^{-9}) = 118{,}772$ Hz
(R = ohms, and C = farads)

At a little over 100 kHz, the frequency response at the input of the amplifier would have dropped off by 3 dB and would be losing another 6 dB every octave higher. A frequency response curve showing this sort of response looks like this:

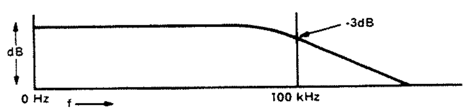


FIGURE 23. HIGH-PASS FILTER EFFECT OF CABLE.

The nature of capacitors, passing higher frequencies and blocking lower ones – depending on the capacitor's value in farads – dumps high frequency energy to ground when the capacitor is connected in parallel as it is with cabling. The danger of high frequency loss from long cables is exaggerated by preamps or mixers that have higher output impedances: if we make the output impedance just 2000 ohms instead of 200, the -3 dB point on the frequency response curve would be just under 12,000 Hz – an audible loss of high frequency "sizzle" that would make the system sound dull. The problem is compounded by slew rate degradation of the output. This is one reason the outputs of Fender mixers all have low actual output impedance and high slew rate capability, but for the best results use the wiring rule

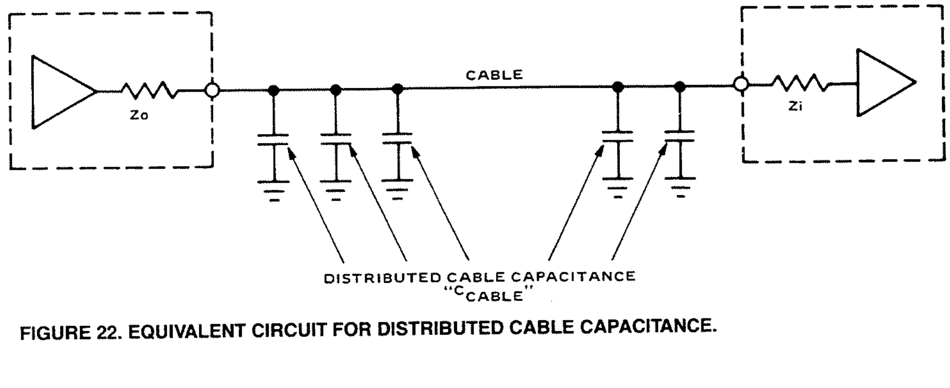


FIGURE 22. EQUIVALENT CIRCUIT FOR DISTRIBUTED CABLE CAPACITANCE.

of thumb: "shorter is better" whenever possible.

Professional audio equipment, including Fender mixers, is often distinguished by lower impedance outputs which are capable of driving long line without high frequency loss. In most applications you won't be bothered with losses between pieces of professional audio equipment, however, if you plan on wiring permanent installations with hundreds of feet of cable, you would do well to consult the operator's information supplied with the equipment to find out what kind of impedances you're working with and what if any, special terminations are required to properly interface each piece to the other equipment in the system. An example of this would be an audio console used to drive long signal lines through a multichannel cable ("snake") to power amplifiers located on stage near speaker systems. Some older consoles (especially tube-types) require a standard 600-ohm terminating resistance at the end of driven signal lines when the load at the driven end of the lines is high impedance, such as the 15,000 ohm load presented by the 2224 and 2244. Check with the manufacturer of the console to be sure; omission of the required loading resistor can result in exaggerated high frequency response, poor transient response, higher distortion, or even electrical oscillation which could possibly overheat amplifiers or burn out high frequency loudspeaker drivers.

Cable impedance calculation in balanced and unbalanced lines

One can determine the cable capacitance indirectly by measuring the frequency at which the output level drops 3 dB. The capacitance appears to change with different source and load impedances, too. For example, with the equivalent balanced circuit diagram shown in Figure 24, the total distributed capacitance calculates to be 46 pf/ft with a 150 ohm source (unterminated), 42.02 pF/ft with a 600 ohm source (unterminated), and 41.65 pF/ft with a

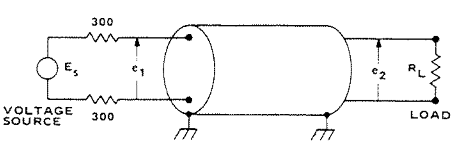


FIGURE 24. EQUIVALENT CIRCUIT FOR CABLE CAPACITANCE, BALANCED OPERATION.

600 ohm source terminated by a 600 ohm load. For balanced operation, the cable capacitance is about 25% greater than the figures specified by the cable manufacturer.

If, for example, a mixer has an output impedance of 600 ohm, and the power amplifier has a 15 kohm input impedance, the frequency response will be down −3 dB at 32,841 Hz, assuming a 100 foot balanced cable. In this case, the total resistance (source in parallel with termination) is calculated as follows:

$$\frac{600 \times 15{,}000}{600 + 15{,}000} = 576.92 \text{ ohm}$$

The −3 dB frequency is calculated as follows:

$$f_{-3\,dB} = \frac{1}{2\pi R_{total} C_{total}}$$

Since R(total) is 576.92 ohm and C(total) is 8.4 nF, the −3 dB frequency is 32,841 Hz. If the cable is terminated by a 600 ohm load instead of 15 kohm, the −3 dB point moves out to 63,156 Hz.

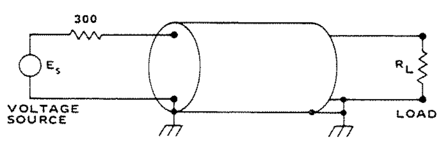


FIGURE 25. EQUIVALENT CIRCUIT FOR CABLE CAPACITANCE, UNBALANCED OPERATION.

For unbalanced operation, the total cable capacitance for 200 feet was found to be 77.4 pF/ft using a 600 ohm source, a 16% increase over specification book values. Capacitance ratio for unbalanced to balanced operation is as follows:

$$\frac{15.47 \times 10^{-9}}{8.4 \times 10^{-9}} = 1.84 : 1$$

This means that unbalanced operation of a 2 wire cable, such as Belden 8451, increases the cable capacity 1.84 times and lowers the −3 dB frequency about one octave. Belden specification book values for the unbalanced case shows a 67/34 = 1.97 or nearly 2 times increase in capacitance! It may be a good idea to check the specs for single conductor cable if you're not using a balanced line.

Clearly, when estimating the effect of cable capacity on frequency response, the total resistance must be known. a good approximation would be to increase the specification book value of cable capacitance by approximately 20%, and calculate the total equivalent resistance of both the source and termination as one lumped value. To be completely accurate, the cable should be treated as a transmission line, which requires considerably more complex calculations. However, such effort is not necessary for the small increase in accuracy.

Impedance

It's a good idea to know what the total load being driven by your mixer or preamp is, so that you can avoid overloading its outputs. If the load is too great, i.e. the impedance of the load too low, it's possible to cause significant loss of level in the system as a whole. In the worst case, you might raise the distortion of the mixer or preamp.

In most systems, other than "terminated" low impedance transmission line systems, the most desirable way of doing things is to have an output impedance as low as possible, and an input impedance as high as is practical. In fact, the German (DIN) standard, a widely accepted standard in Europe, calls for driving from a nominal 100-ohm output (source) impedance into a nominal 10,000-ohm input (load) impedance.

We have prepared a chart that shows the approximate losses you might expect from different input impedances being driven from a more or less typical output impedance of 200 ohms – similar to many solid state outputs.

OUTPUT (200 OHMS)	INPUT Z	LOSS IN dB
200	200	6.02 dB
200	600	2.50 dB
200	1k	1.58 dB
200	2k	0.83 dB
200	6k	0.28 dB
200	10k	0.17 dB
200	20k	0.09 dB

TABLE 3. EXPECTED SIGNAL LOSS DUE TO LOADING.

Normally, a mixer would only be used to drive a few power amplifiers, or an electronic crossover, and driving numerous electronic crossovers or power amplifiers is of concern only when very large sound systems are built up, such as concert systems or distributed institutional systems. When systems of this type are built, it sometimes becomes necessary to drive the signal lines with additional "line amplifiers." Such a line amplifier is usually a low voltage amplification but high current amplification device, with high power gain, that increases the output current and lowers the output impedance down to a value around that of a power amplifier.

Ground loops and grounding techniques

WARNING

In any audio system installation, governmental and insurance underwriters' electrical codes must be observed. These codes are based on safety, and may vary in different localities; in all cases, local codes take precedence over any suggestions contained in this manual. Fender Musical Instruments shall not be liable for incidental or consequential damages, including injury to any persons or property, resulting from improper, unsafe or illegal installation of a Fender power amplifier or of any related equipment; neither shall Fender Musical Instruments be liable for any such damages arising from defects or damage resulting from accident, neglect, misuse, modification, mistreatment, tampering or any act of nature. (IN PLAIN WORDS... IF YOU LIFT A GROUND, THE RESULTING POTENTIAL FOR ELECTRICAL SHOCK IS YOUR OWN RESPONSIBILITY!)

As a further caution, we advise you to never trust any potentially hazardous system, such as an AC power system of any type, just because someone else tells you that it's okay. The situation could be compared to an "unloaded" gun. Before you "trust your life," check things out yourself! People do get killed by "unloaded" guns and a faulty piece of audio equipment can be just as dangerous.

You can find a more lengthy discussion of AC power, grounding, and safety elsewhere in this manual under "AC Power Requirements and Safety."

Ground loops are perhaps the most insidious, and common problems in audio practice. They are difficult to trace and eliminate and are often a source of frustration to even experienced audio engineers. A "ground loop" is a multiple electrical path between audio components formed by the ground wiring, the chassis of the components themselves, or by combinations of these two main elements. Electrical current that generally induces hum and noise, is allowed to "loop" around from one piece of interconnected equipment to another by means of paths that are not intended to carry signals. These currents modulate the potential of the signal-carrying wiring, producing hum and noise voltages that can't be distinguished from signals by the affected equipment, and get amplified right along with the program material.

The first thing to suspect when you hear hum from a sound system is a ground loop. Sometimes, in poorly designed or cheap audio gear, ground loops occur INSIDE the gear itself, and little can be done to get rid of the hum short of having a good audio engineer redesign the ground wiring inside, which is usually a waste of time. We'll assume you want to avoid this kind of equipment, and talk about the causes of ground loops in sound systems using well-designed equipment.

The AC powerline ground, the green wire and the third pin on the AC plug, serves to connect electronic equipment chassis to a wire in the wall power service that leads through building wiring to an "earth" ground. The earth ground is required by electrical codes everywhere, but sometimes is omitted or faulty.

One myth about grounding is that you MUST ground your equipment to prevent noise from entering the system. Anyone who owns a portable cassette machine knows that simply isn't true. The PRIMARY reason we ground our audio equipment is for SAFETY. An important secondary reason is that, with AC powered equipment, under some conditions, proper grounding CAN help reduce external noise pickup. The third reason that we must pay attention to grounding is that, while proper grounding won't always reduce external noise pickup, POOR grounding can unquestionably INCREASE external noise pickup!

Here's a drawing of a typical ground loop situation; a mixer feeding a power amplifier. Note that this situation looks like "normal operating procedure" – everything is connected according to owner's manual instructions.

The ground loop occurs between the mixer and the power amplifier even though there is only one audio cable connecting the two devices. A second ground connection, through the AC cables, and the chassis of the two units, makes the "return" connection and forms a continuous conducting loop for current to flow. One commonly used method to break this ground loop is to "lift" the AC ground on the power amplifier with a two-wire to three-wire AC adaptor (leaving the loose green wire on the adaptor unconnected). Because this practice is in conflict with the AC safety ground and CAN BE HAZARDOUS, here are some rules to minimize the safety conflict:

1. Don't lift the safety ground on any piece of equipment unless it demonstrably reduces noise levels.

2. NEVER defeat the AC safety ground on your mixer or any other piece of gear connected directly to your microphones. The microphones come first in grounding safety.

3. Try to plug all affected equipment into a common AC service. In fact, all sound equipment and related accessories such as guitar amps, keyboards, etc., should be connected to a common AC system to avoid safety hazards.

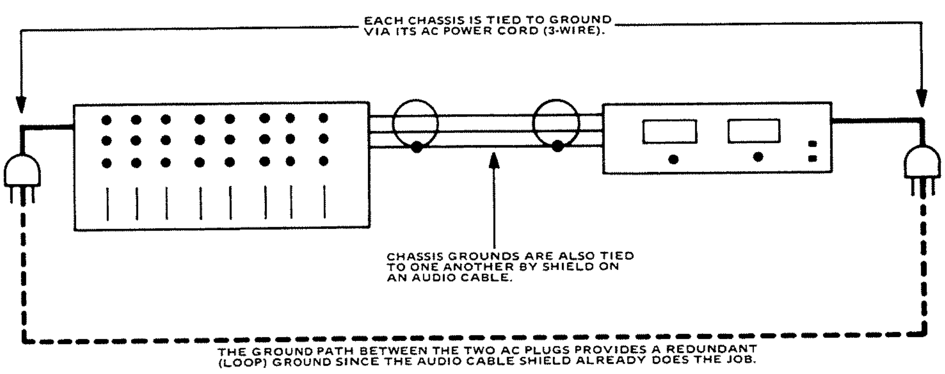


FIGURE 26. A TYPICAL GROUND LOOP IN A SOUND SYSTEM.

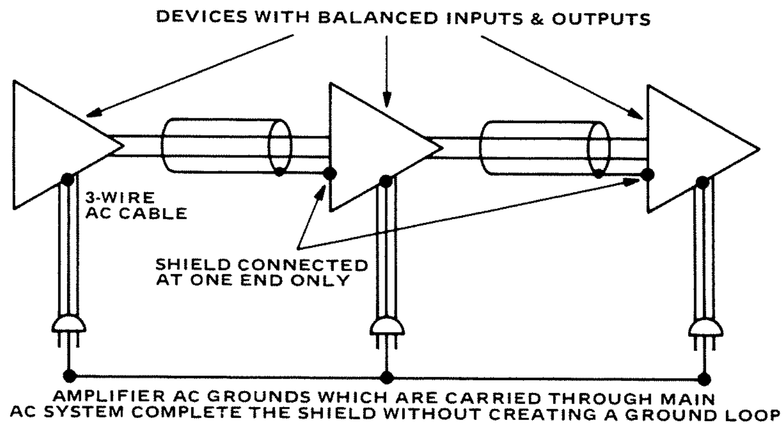


FIGURE 27. TELESCOPING SHIELD APPROACH TO GROUNDING.

It's worth noting that, by using balanced connections between two pieces of audio equipment, you can lift (disconnect) the shield at the output (sending) end of audio cables to cut the loops that carry ground loop currents like the one shown in the drawing. In a balanced line, the shield does not carry audio signals, but only serves to protect against static and RFI, so you can disconnect the shield at one end without affecting the audio signal on the two inner conductors of the cable, and with little or no effect on the shielding. Unfortunately, this is not a very practical solution to the ground loop problem for portable sound systems because it requires special cables with shields disconnected on one end. This is illustrated in Figure 27.

CAUTION

Microphone cases typically are grounded to the shield of the cable, and connect to the mixer chassis via pin 1 of the XLR connector. If there is any electrical potential on any external equipment, such as a guitar amp chassis, a performer holding the mic and touching the other equipment can be exposed to a lethal electrical shock!

That's why it is best to avoid "ground lift" adaptors on AC power connections if there is any other conceivable way to eliminate a ground loop.

Sometimes, you'll find pieces of audio equipment or audio accessories that are designed to anticipate ground loops. These pieces have "ground lift" switches next to their XLR or three-wire phone jack outputs. The ground lift switch makes and breaks the connection between the connector shield pin and the chassis of the particular device. This sort of feature is most evident on devices like "direct boxes," which are used when an electric musical instrument is to be plugged directly into a mixer (whose inputs are not designed to accommodate direct connection of such instruments).

Probably the best way to keep noise out of a microphone input is to start with a high-performance, low-impedance microphone (including any of the Fender microphones) and to connect it to a mixer with a low-impedance, balanced (or "floating") input using high-quality microphone cable that utilizes XLR connectors. Keep microphone cables as short as possible within the constraints of a performer's needs, and keep them physically separated from line-level (mixer output) cables, loudspeaker cables and AC cables.

BEWARE OF RUNNING MIXER INPUT AND OUTPUT SIGNALS IN THE SAME "SNAKE." It is a common, but risky practice to feed the output of a mixer back down a microphone "snake" cable (a "snake" is a multi-pair shielded cable) to a power amplifier on stage. While it is possible to "get away" with this practice in some popular music systems (because output levels from the stage are very high, so mixer gain can be reduced), it is always much better to provide a separate cable from the mixer output to the stage amps. While many of the signals generated on stage may be at or near line level (i.e., an electronic piano output or a mic inside a kick drum), a soloist in a quiet passage may require a higher fader level at the mixer, and that can cause the mixer to oscillate due to crosstalk between the conductors in the snake. Often, the first evidence of the problem occurs when suddenly all the VU meters on the mixer "peg" at the top of their scales, and the system goes into heavy distortion or "motorboating." If this occurs, reduce the mixer gain immediately to avoid possible equipment damage. The only "fix" is to run a separate cable (or cables) for the mixer outputs, and to not run them down the same snake as the input lines.

Series and parallel connection of loudspeakers

Individual loudspeaker components of a given power handling capability, sound dispersion angle, efficiency or other characteristic are often connected in groups to make a "system" that collectively produces "engineered" results beyond the capability of the individual units in that system.

There are two basic ways to hook up loudspeakers or speaker systems so you know the value of the resulting load, and so the power is evenly distributed. This is important when you must use multiple loudspeakers in one enclosure to make up a speaker system, or when you connect more than one speaker system to a single amplifier source (i.e., connecting several loudspeakers in a PA column, or stacking cabinets to form a guitar lead stack). Bass players often combine loudspeakers or systems to produce more output, and PA companies usually run

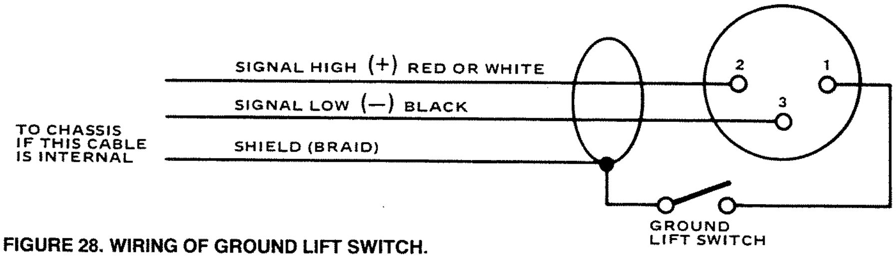


FIGURE 28. WIRING OF GROUND LIFT SWITCH.

several loudspeakers from each power amplifier output to make the best and most economical use of system components. The easy way to figure how much power a group of loudspeakers, or "drivers" can handle is to multiply the power handling specification of one of the drivers by the number of drivers. That part of the process is always the same as long as all the connected drivers are identical. It is not recommended, nor is it common practice to place different types of bass drivers in a single enclosure, or to connect different types of bass drivers in series, or parallel. The reasons for avoiding dissimilar bass drivers in individual enclosures applies equally to the use of series or parallel connection of ENCLOSURES with different types of bass drivers inside, or enclosures of different rated impedances. One of several problems encountered by mixing dissimilar speakers is that the available power will not be distributed evenly among the various speakers connected together.

A group of loudspeakers that are connected in series presents a higher impedance to the amplifier than any one of them alone. Specifically, the total impedance is the sum of the individual impedances. A higher total impedance represents a lesser load to the amplifier, and consequently draws (or receives) less power.

If all the loudspeakers connected in series are the same impedance, then the total impedance is the impedance of one speaker multiplied by the number of loudspeakers. The power delivered to any one speaker is the total power delivered to the group, divided by the number of speakers. Figure 29 shows four eight-ohm speakers connected in series for a total impedance of 32 ohms. If the impedances are dissimilar, the highest impedance driver receives the most power.

Loudspeakers connected in parallel present a load to the amplifier that is lower impedance than the impedance of any one of the drivers. This represents a greater load to the amplifier, and also draws more power. Parallel wiring can be thought of as a pair of parallel wires where individual drivers are all connected as shown in Figure 30.

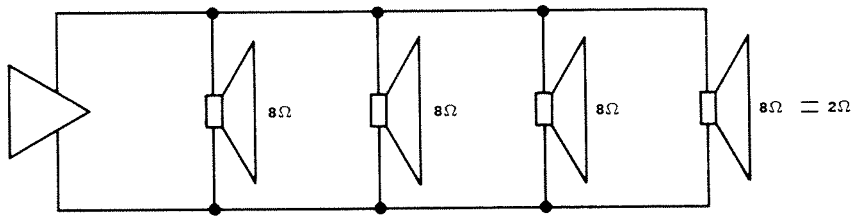


FIGURE 30. PARALLEL CONNECTION OF SPEAKERS.

If all the drivers are identical, then the total impedance of the array is simply the impedance of any one of the drivers divided by the number of drivers. For example, three 16 ohm drivers connected in parallel represents a total load impedance of 16/3, or 5.33 ohms.

It is generally accepted that only drivers of the same impedance (ohms) are wired in parallel. For 2 unequal impedances in parallel, use the following equation:

$$\text{TOTAL OHMS} = \frac{(a \times b)}{(a + b)}$$

The 2224 and 2244 are designed to drive 4-ohm loads connected to each channel (although they will safely drive a pair of 2-ohm loads under certain circumstances, as described elsewhere in this manual). Driving 8 or 16 ohm loads does no harm, but produces less power since the 2224/2244 behave like voltage source amplifiers (see POWER OUTPUT AND AMPLIFIER LOAD). If you are driving one pair of 8-ohm speakers, you'll get the same number of watts using only one amplifier channel with the speakers connected in parallel, or using both channels to drive one speaker each.

If you have enclosures containing several low frequency drivers, or several identical enclosures to connect to one channel of the 2224/2244, wire the individual speakers so that the total load is about 4 ohms or is higher than 4 ohms.

Theoretically speaking, any combination of parallel and series wiring is permissible in a loudspeaker array as long as it results in the distribution of the same amount of power to each driver in a given system. Practically speaking, though, loudspeakers that are connected in series do not perform the same as those connected in parallel. The complete reason involves a lot of math, but in simple terms, the reason is that the speakers connected in series operate at a much different damping factor than those connected in parallel. This results in somewhat different bass performance. Furthermore, if any one speaker in a series circuit fails (open-circuit), the rest of the speakers connected in series with it will cease operating. If you can arrange it, it is generally much better to use parallel connection exclusively for your speakers.

Once you have interconnected a group of loudspeakers in a single enclosure (an array), you can consider that enclosure to be a single large speaker having the combined impedance and power handling characteristics of all the drivers (as calculated for their series/parallel arrangement). This is especially handy when a large number of cabinets are combined into a huge sound system. All you have to do is consider the cabinet's impedance, not the impedances of each driver in each cabinet.

Figure 31 consists of diagrams of 4, 8, and 16-ohm drivers wired in logical order. We stop at the point where the power handling capacity of the speaker system exceeds the power available from the 2244. In the examples given, we are arbitrarily using drivers rated at 100 watts each for quick calculation.

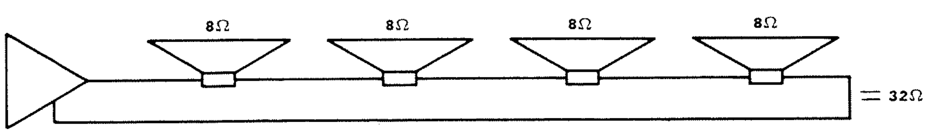
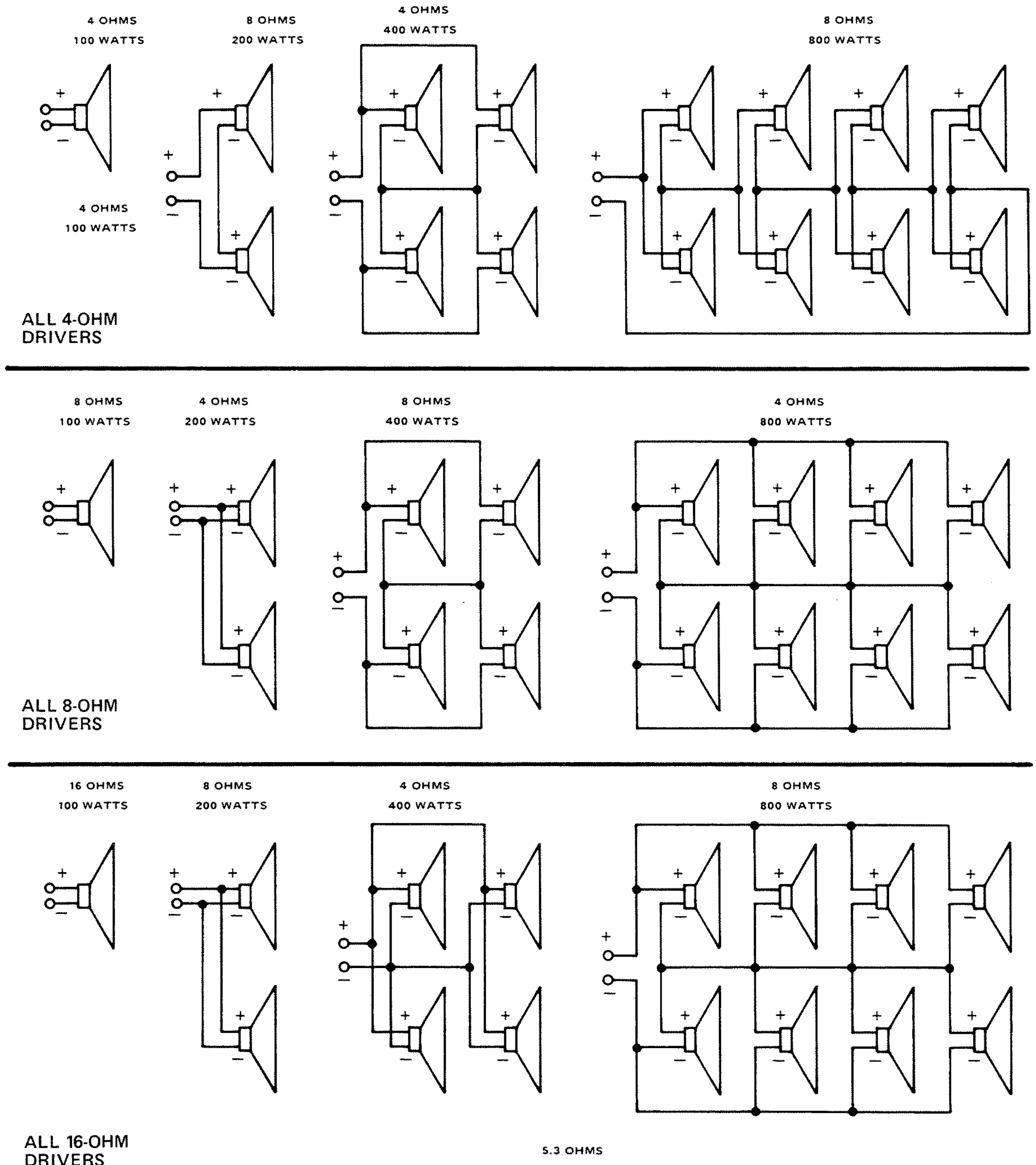


FIGURE 29. SERIES CONNECTION OF SPEAKERS.



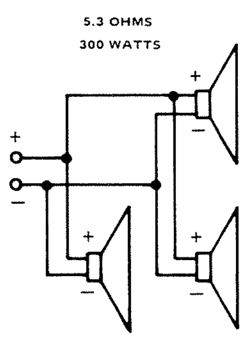


FIGURE 31. MAXIMUM POWER DISSIPATION BY GROUPS OF 4, 8 AND 16 OHM 100-WATT DRIVERS WIRED IN VARIOUS SERIES AND PARALLEL CONFIGURATIONS.

Bridging amplifier outputs

"Bridging" describes the wiring of two amplifiers (or two channels of a stereo amplifier) so that the outputs add together to produce twice the voltage of one amplifier. Since power is proportional to the square of the voltage, bridging can develop up to four times the output power given the same load impedance... assuming that the amplifier can handle it. A more useable rule of thumb is twice the power into twice the impedance. Typically, however, the total output power in bridged mode would be somewhat less than the theoretical 4X increase (at full power operation) because of power supply losses. For example, the 2244 in stereo mode delivers 440 watts per channel into two 4-ohm loads; in the bridge mode, the total power output is 880 watts into a single 8-ohm load. Notice the power is doubled, not quadrupled, because we have specified a load of twice the impedance.

Most "monitor" speakers and also consumer "Hi-Fi" speakers have an 8 ohm impedance rating. To get more power and dynamic range from 8-ohm speakers, two 2244's used in bridged configuration can be used to provide outstanding headroom and dynamic range needed for accurate reproduction of peaks and transient sounds.

Special wiring and gain setting circuits are incorporated into the 2224/2244 to allow the amplifier to be converted to bridging operation. This is done when the rear panel mode switch labeled "BRIDGE." One input (channel A) is all that's required to drive the 2224/2244 when it is operated in its bridged mode. The rear panel switch "rewires" the input circuits to feed the two channels of the amplifier simultaneously. In the bridged operating mode, the channel B output of the 2224/2244 is electronically placed in reversed polarity with the channel A output. The channel B "hot" output terminal is then used as the "low" side of the output, and the channel A "hot" output terminal is used as the "high" side of the output. THE BLACK TERMINALS ARE NOT USED IN BRIDGED MODE OPERATION.

Just like speakers wired in series, the 2224/ 2244 outputs are essentially placed in series operation when the amplifier is switched to its bridged mode. Two identical speakers wired in series have double the impedance of each speaker. Two amplifier channels wired in series (bridged mode) have twice the output voltage of one channel. This would tend to deliver four times the power, though such is not possible in practical terms. Instead, the minimum speaker impedance must be increased; if the minimum speaker impedance were not increased, too much power would be "called for" which could overload the amplifier; by increasing the minimum speaker impedance, one avoids overload and related problems from power supply limitations and output transistor "Safe Operating Area" limitations.

In the case of the 2224/2244, the minimum load impedance must be doubled. This might seem to give a net "break-even" at first sight, but the Ohm's Law-based equation for power (from "Impedance" section) shows how power is doubled along with voltage in bridged operating mode:

$$W = E^2/R \text{ or,}$$

watts = volts squared, divided by ohms

Where each channel might give an output of 42 volts into 4 ohms during normal operation, in bridged operation, the output would be 84 volts into 8 ohms. Here is the numerical result of these two situations:

$$W = 42 \times 42 / 4 = 441$$

(per channel in normal operation; given two channels, this equals 882 watts total)

$$W = 84 \times 84 / 8 = 882$$

(total power output is the same)

Bridged operation of the 2224/2244 will enable driving 8-ohm speaker systems to the full output power capability of the amplifier, and will allow the maximum output VOLTAGE available from the amplifier for driving loads that require more voltage such as 70 volt and distributed speaker systems (see "Distributed and 70 volt speaker systems" on page 24). Note that in bridged mode, the amplifier is effectively 6 dB more sensitive, so you may wish to turn down the input level.

CAUTION

NEVER CONNECT THE BLACK OUTPUT TERMINALS TO ANYTHING WHEN THE AMPLIFIER IS IN ITS BRIDGED MODE. In bridge mode, both of the RED output connectors are "hot." Do not allow them to short together, or to any other connections in your sound system (including those from other amplifiers). Quarter-inch phone plugs are a poor choice for speaker connectors, especially in a bridged system, because these plugs cause a momentary short circuit as they are plugged in or pulled out.

When operating in the bridged mode, the extra voltage warrants extra care to avoid touching speaker wiring since the amplifier can easily deliver a lethal combination of voltage and current.

WARNING

TREAT BRIDGED AMPLIFIER OUTPUT WIRING WITH THE SAME RESPECT YOU TREAT AC POWER LINE WIRING!

Make sure that no return path between speaker wiring and equipment chassis or rack cabinets, which are probably grounded, exists any time, especially when you're using the 2224/2244 in the bridged mode of operation. In bridged mode, the output terminals of the 2224/2244 are both "hot," and are not referenced to ground but only to each other. Thus a return path connected to ground would short one or both sides of the amplifier and could cause the amplifier to shut down or might even damage it.

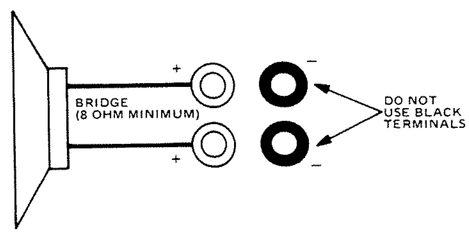


FIGURE 32. REAR PANEL CONNECTIONS FOR BRIDGED OPERATION.

For "mono" bridged operation, patch the input signal into the channel A input only. The channel B input should remain unused. Connect the speaker wires to the two red output terminals as shown, with the channel A red terminal used as the "+" and the channel B red terminal used as the "-." Be careful not to connect anything to the black terminals of either channel to avoid unwanted imbalancing of the two channels' outputs. When the 2224/2244 amplifiers are set to bridge mode, only the Channel A input connectors, controls and switches are operative. The Channel B input connectors, controls and switches are disabled. For reasons of clarity (avoiding clutter), you may wish to remove all unused wiring from Channel

B, and turn its INPUT LEVEL control down as well. If you choose not to do this, there is no danger or possibility of harm to the equipment, but the operator may become confused. The gain of the amplifier effectively increases in bridge mode. You should recalibrate your sound system gains when switching the amplifier to bridge mode.

Distributed and 70-volt speaker systems

In large office buildings, hospitals, and other industrial installations, many small loudspeakers are used in a "distributed" wiring scheme. The purpose of distributed speaker systems is mainly to cover large areas for paging and background music, and to make the most efficient use of the available hardware (amplifiers and loudspeaker drivers) so that the cost of the overall system is reasonable. In large meeting rooms or halls with low ceilings, distributed speaker systems may be the only way to cover an audience with an even sound level.

Generally, large distributed sound systems are not played at concert levels, and don't require very much power from each loudspeaker, especially when there are enough individual loudspeakers so that most everywhere in a building is covered at least partially, by one or more of the loudspeakers.

Like most common drivers, the drivers used in individual ceiling speakers are usually 8 ohms. Since sound contractors base the design of distributed speaker systems around coverage of desired areas, it isn't often that the number of individual drivers allows either a 4 or 8 ohm load to be easily wired to the driving amplifier. Since one failure in a series string of speakers knocks out all the speakers in the string, neither series or series/parallel wiring is used. Only parallel wiring can be used where each speaker must function regardless of the condition of the other speakers in the string. The impedance of ten 8 ohm speakers in parallel is 0.8 ohms so how can you parallel 10, 50, or 100 individual drivers without loading down the amplifier? By using transformers.

In a "70 volt" system, one amplifier is used to drive one single pair of wires that are run to each speaker location along a single line. This line is called a "70 volt" line because 70 volts represents a the rated operating level (maximum voltage output) on the line. A 5 watt transformer like the type found on the garden variety ceiling speaker, will draw 5 watts when the audio line is at 70 volts RMS. A 10 watt transformer like the type found on larger ceiling speakers will draw 10 watts when the audio line is at 70 volts RMS. Thus the name "70 volt line" used with this type of distributed system. With the 70 volt line, any number of speakers can be used, up to the point where the total power that is drawn by all the speakers and transformers on that line equals the power output capability of the amplifier driving the line.

Most of the commercially available drivers for distributed use come with an integral line transformer that has several "taps," or internal wiring choices, that permit the installer to choose how many watts of power will be drawn by the individual speaker as it is connected onto the existing line. If one 120 watt amplifier is used to feed a 70 volt line, then 24 individual drivers with their 5 watt transformer tap can be connected to the line. Thirty drivers with 4 watt taps, 60 drivers with 2 watt taps, or 120 individual drivers with 1 watt taps can all be attached to the single 70 volt line. The standard small 8 inch ceiling speakers usually have transformers that offer 1, 2, 3, 4, and 5 watt taps (or some variations ranging from 0.25 watts to as high as 100 watts for specialized high-power 70 volt lines) to give a wide range of wiring possibilities which include such variations as single 70 volt lines that have quiet and loud "zones" where some individual drivers are hooked up using their 1 watt taps while others on the same line have their 5 watt or other taps hooked up – any combination is possible as long as the total number of watts from adding up all the connected transformer taps totals no more than the available amplifier power.

The impedance of very small transformers falls off at lower frequencies, so sound system designers often specify a high pass filter to reduce the low frequency content of music and signals fed to distributed speaker systems which use small inexpensive speakers. In distributed systems with high fidelity, the speakers and transformers are bigger and don't require as much protection. Larger, more expensive speakers sometimes are used in distributed systems, and transformers with excellent frequency response are available at a higher additional cost. These higher quality 70 volt line transformers are also used where very long speaker wiring must be used or where electrical safety considerations prevent connecting amplifiers directly to loudspeakers.

The ideal transformer would have no impedance of its own, but rather, would act like an electrical "lens," merely magnifying or reducing voltages fed to it, and, in the process, transformer impedances too. Larger, more expensive 70 volt line transformers are more efficient than their smaller counterparts, and behave much more like the ideal. In some cases they can save money on higher quality installations where many good quality speakers are used in place of a larger number of inexpensive smaller speakers.

Applications of the 70 volt line vary from supermarkets to amusement theme parks, but wherever multiple loudspeakers are used, especially where large distances are covered by speaker wiring, 70 volt type systems are popular, economical, and reliable.

In supermarkets, one line is used for all of the speakers since any paging being done is essentially in one large room. Several lines (and several amplifiers) may be used in hospitals, office buildings or industrial installations where different paging areas are needed, for example, on different floors. A page is then sent to desired areas from a central communication station or through automatic switching equipment installed by telephone companies and controlled by a special telephone dial code.

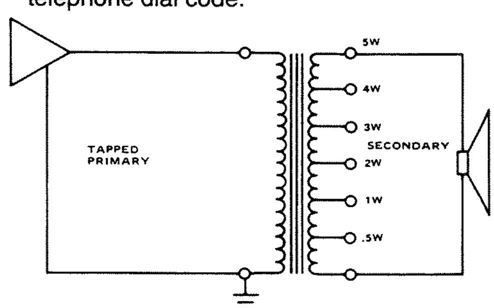


FIGURE 33. SCHEMATIC DIAGRAM OF SPEAKER DISTRIBUTION TRANSFORMER FOR 70-VOLT LINE.

Audio connectors

In most situations, the audio equipment you have dictates what type of connectors are needed to mate one piece of equipment with another. The ideal way to connect audio components together is by permanent techniques such as soldering, wire wrap, punch block, or crimped spade lugs and screw terminals. Since that would prevent easy portability and component patching, these techniques are used only where the connections are not likely to be changed, (i.e., within individual pieces of gear). The next best thing to permanent techniques is a reliable connector. There are many different types of connectors, and each has its particular virtues and liabilities, depending on the particular application.

Let's take a close look at the various popular connectors used in audio gear. Although connectors seem fairly straightforward, there are many things to consider when choosing the best connectors for each job.

PHONE CONNECTORS. These are an audio industry standard, which take two main forms; quarter-inch (6.33 mm) diameter TS (tip-sleeve), and TRS (tip-ring-sleeve). These are called "phone plugs" because they were developed many years ago by the telephone company. In addition to these two main phone plug types, there are also smaller diameter "mini phone plugs" with TS and TRS configurations. The sleeve part of the phone plug usually connects the end of the cable shield to ground. The ring on the TRS types connects the (−) minus or "low" side of a balanced cable, and the tip connects the (+) plus or "high" side for either unbalanced TS or balanced TRS type phone plugs. There are good and bad phone plugs, ranging in price from expensive to cheap. AVOID CHEAP PHONE PLUGS; they invite system breakdowns. Trying to save money on connectors is a good thing as long as the connectors you save money on are good quality connectors. Heavy duty, military style, brass phone plugs are designed for telephone switchboards. For mic and line level signals, brass plugs are fine, as long as you keep them clean and free of skin oils and fingerprints. Brass plugs can make it difficult to apply strain relief, and speaker cables made with brass plugs are generally less reliable than, for example, the standard "Switchcraft 280" plated plug with its machined one-piece brass collar and machined brass shell and crimp strain relief, or the similar "ADC AP280."

If you must use phone plugs on speaker cables, check the tip/sleeve insulator periodically to make sure it's not cracked or partially missing. This insulator can deteriorate from normal use – packing up, stepping on plugs, etc. A cracked tip insulator can cause the tip to short to the sleeve or ring, thus presenting a short circuit to whatever audio gear is connected at both ends of the cable. Even though phone plugs are often used for speaker connections for convenience, they are far from ideal. A phone plug's tip contacts only a very tiny area of metal on the jack spring, which causes high contact resistance, and with an extremely high current capability amplifier like the 2244, this could produce audible distortion, heating of the jack or other strange effects.

XL or XLR CONNECTORS. These are alternately called "Cannon" connectors (named for the company that invented them) or just mic connectors. XLR connectors are available with 2, 3, 4, 5, and 6 pins, although the three-pin version is by far the most common connector in professional audio – especially for portable sound reinforcement, microphones, and field recording. Pin number 1 on the typical XLR female makes contact with pin 1 on the male before pins 2 and 3 make contact. This serves to discharge any stored electric charges in the cable shield, and thus reduces "pops" that might otherwise occur when mating or disconnecting cables while the system is

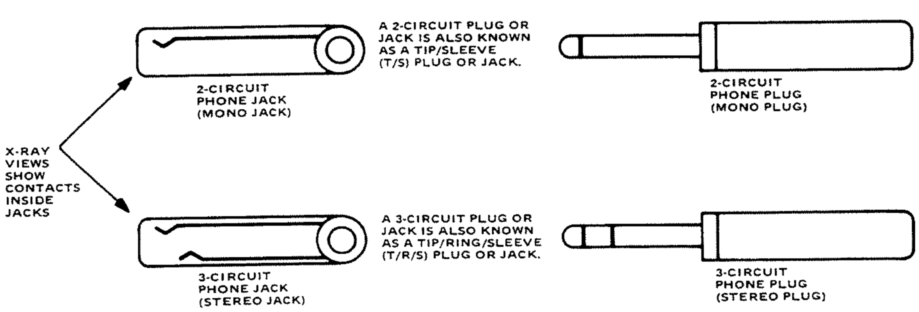


FIGURE 34. MALE AND FEMALE PHONE CONNECTORS.

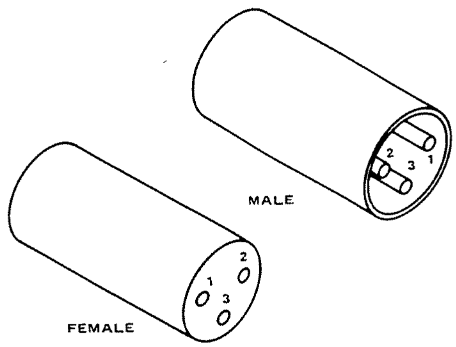


FIGURE 35. MALE AND FEMALE XLR-3 CONNECTORS.

in use. There are a few other significant advantages to the XLR type connector; the built-in locking latch that prevents accidental disconnection, recessed contacts that stay relatively free of oils from handling and provide good electrical connections with low contact resistance, and, on the Switchcraft "A3" and ADC "P3" connectors (XLR equivalents), the option of connecting pin 1 to the shell to provide a ground path from shield to equipment chassis as well as circuit common. WE RECOMMEND THAT THE SHELL OF THE XLR NOT BE CONNECTED TO THE SHIELD; while this exposes a few inches of cable to an unshielded condition, it keeps the XLR connector from creating a "ground loop" should it touch another piece of equipment or even a damp floor. Some equipment designers use the XLR type connector for unbalanced inputs or outputs because of the connector's inherent ruggedness and because it makes the ground connection first. The chief disadvantage of the XLR is its cost, from 2 to 3 times that of a comparable quality phone plug or jack.

BANANA CONNECTORS. These connectors are typically paired "dual bananas," sometimes refered to as "GR" plugs, after General Radio Co., a manufacturer of electronic test gear, which pioneered their use, or "Pomonas" after another manufacturer of the plugs. Banana plugs offer fairly low contact resistance as long as the connections are physically tight. The connectors themselves offer distinct advantages and disadvantages. Among the advantages are higher current capability than phone or XLR connectors, ease of wiring, quick-connect ability (which can be used to change polarity (+ for −) in a hurry), and

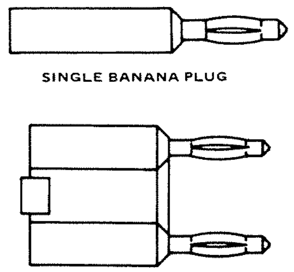


FIGURE 36. BANANA CONNECTORS.

the ability to "stack" banana plugs for parallel connections. The chief disadvantages of the banana plug are the possibility of connecting in the wrong polarity, relatively poor strain relief for the wire, and exposed contacts on plugs that can prove to be hazardous both to the equipment and sometimes to the operator, and the possibility of accidental disconnection. The 2224/2244 and other amplifiers have five-way binding posts that are designed to accomodate bananas as well as direct wire connections. The bananas should be used only when frequent connections and disconnections must be made.

PHONO (RCA) CONNECTORS. Originally designed to connect internal chassis subsections in radios and televisions, the RCA pin plug, also called a "phono plug," is widely used in consumer electronic equipment because of its low cost and extremely good performance with very low level signals like those from phono cartridges. The RCA has several big advantages and a few disadvantages too. The biggest advantage of this little connector is its cost – only a few cents in its most basic solder-on configuration. High quality RCA plugs are available, like the Switchcraft 3502, which has a strain relief and machined brass shell. The RCA connector bears a physical similarity to the BNC connector, which is widely used on laboratory test equipment that must measure very low level and/or high frequency signals. The small size of the RCA enables equipment designers to cram lots of connectors onto a small panel, and indeed some modern mixers have hundreds of these ubiquitous connectors on their rear panels where only a few XLRs might fit. The small size is, at the same time, a major disadvantage of the RCA connector since this makes it hard for the connector to support a heavy wire. It's too delicate for rough road use, more difficult to plug in than a phone plug, as well as somewhat more difficult to solder properly. The RCA plug also mates the "hot" signal wire first, which allows potentially embarassing "pops" or worse in a live show situation (the phone plug isn't any better in this respect). Finally, like the banana connector, the RCA type can only be used for unbalanced wiring.

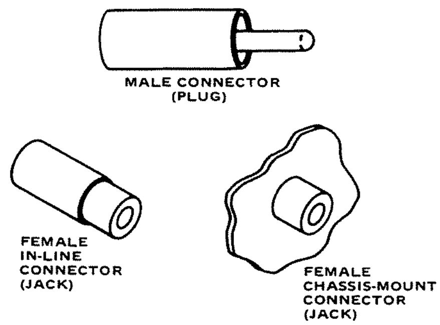


FIGURE 37. MALE AND FEMALE PHONO CONNECTORS.

MISCELLANEOUS CONNECTOR TYPES. A variety of special connectors are also used in audio practice; you'll find crimp and solder lugs attached to screw terminal strips on many professional "telephone-style" pieces of audio gear. There are large audio installations, like recording studios, where much of the wiring is "hard wired" using actual telephone punch blocks – blocks containing rows of common connections with spring-loaded wire "grabbers" that cut into the wire as it is fed by a special punch. (CAUTION: such connectors assume the use of solid conductor cables, and are intended for fixed, not portable, installation. If stranded cable is used in a punch block, an unreliable connection may result.) Some PA companies use heavy-duty twist locking power plugs and sockets for speaker wiring (and they hope nobody plugs them into the power line). A connector gaining popularity for speaker connections is the old Cannon "P-type" connector, which is available with from 2 to 8 pins. On high quality snake cables, you're likely to find multi-pin connectors with up to 100 pins that have screw-on or latching retainers to protect the delicate pins from twisting; you may also find this kind of multi-pin or similar connectors on audio consoles, multi-track tape decks, and so on.

Connector choices should be based on reliability, ease of repair or replacement, and standardization where it applies.

Where to get cables and connectors

Wire manufacturers like Alpha, Belden, Canare, Columbia Mogami, and National just to name a few, can supply audio cable suitable for any amplifier installation. Many cables are available pre-wired with connectors. If bulk cable is purchased, connectors are available separately from such manufacturers as ADC, ITT/Cannon, Neutrik, Switchcraft, and others. Catalogs offered by these companies and others are usually free for the asking, if your local electronic parts store doesn't carry the required cable and/or connectors.

Level adjustment and headroom

Volume control and fader settings throughout an audio system affect the noise and headroom performance of the system. The dynamic range of an audio device or system is the difference between the "noise floor" of the system and the maximum output level. To provide the best overall system performance, level settings should be optimized as much as possible for each system component. As an example, it is usually considered wise to start with the signal source end of a signal chain (the microphone preamp) and set the input level at the maximum that will not produce overload. The next step is to set the level of the microphone module or mixer channel so that it properly drives the bus section of the mixer (the circuits that come just before a "master" volume control). On mixers with VU meters (like Fender mixers), the meters will show you the levels at the console output. If line amplifiers, electronic crossovers, equalizers or other signal processing equipment are used next in the signal chain, the signal levels driving these units should be set so their dynamic range is optimized as well, by setting the input level of each item in the chain so that input signals are as high as possible without producing overload (actually "overdrive" is the appropriate term, rather than the commonly used "overload").

In order to follow these guidlines, you may have to check the owner's manual of outboard pieces of equipment you're using to find out what those pieces need for input levels. An external voltmeter will probably come in handy for this part of your system setup. The last item in the signal chain is the power amplifier. The input levels of the 2224/2244 power amplifiers should be set using the input attenuators so that maximum program levels from the driving source equipment won't drive the amplifiers to clipping. It's a good idea to make all the equipment in the signal chain work as hard as possible while avoiding overload, so that by the time you get to the power amplifiers, you will need to turn down the volume by using the amplifier's attenuators. This keeps overall system noise as low as possible.

A diagram of the relative voltage levels in typical system components hooked up in signal chain order, will help you to visualize what we mean about the settings within the "gain ranges" of various individual pieces of audio equipment.

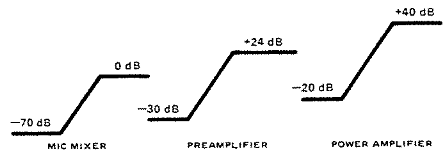


FIGURE 38. GAIN ADJUSTMENT RANGES OF VARIOUS EQUIPMENT IN A SOUND SYSTEM.

Remember that music program material has a certain dynamic range. Program material spans a wide volume range from its quiet passages to its volume peaks. The average volume of signals being amplified should fall somewhere in the middle of the capability of each piece of audio equipment in your signal chain, but you will need to allow for the peaks in the program material. In order to run the program material as high as possible without clipping and distortion, you will have to choose a target amount of system "headroom" for the peaks to operate in and set your volume controls and faders accordingly.

Rock music has much less dynamic range than classical music (the quiet passages are not as quiet), and you can generally run system levels (VU meters) higher than you would with classical music. In rock music, the peaks are not as far above the average levels as they are with classical music, so you don't need as much headroom.

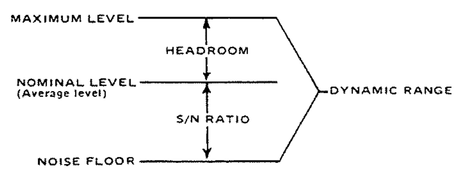
Most rock music audiences don't miss extreme dynamics since the recordings of the music designed for air play have compressed dynamics in the first place. In fact, when program dynamics are intentionally limited or "squashed" a bit by audio signal compressors or limiters placed in the signal chain of the sound system used for rock music, very few audience members will notice, and these devices (limiters) can greatly reduce peaks and thereby allow the average level to increase while affording some speaker protection (they can keep amplifiers from clipping).

To illustrate the importance of headroom, let's use an imaginary sound system that has the following specifications (we're allowing 20 dB of headroom and that makes even classical music sound fairly natural):

DESIRED HEADROOM: 20 dB
SPEAKER SENSITIVITY:
100 dB/1W/@1m
AUDIENCE AVERAGE DISTANCE:
10 m (33 feet)
MAXIMUM AMPLIFIER POWER:
240 watts at clipping/4 ohms (one channel of a 2224)

Begin by assuming that the average sound level in the audience will be around 90 dB SPL. Here's what we get: The average listener distance reduces the sound level from 100 dB (1 W/1m) down to around 80 dB at 10 meters (assuming an out-of-doors non-reverberant environment), so we have to supply the speakers an average power of 10 watts (10 dB over 1 watt) to make our average sound level 90 dB at the listener's position. To that, we add our 20 dB of desired headroom, and right away we see that we'll need 1000 watts of amplifier power to achieve the full 20 dB of headroom! Only 240 watts are available.

At this point there are several things we can do; we can start building up the sound system, adding amps or speakers or both. We can simply reduce our dynamic range and limit the amount of headroom we allow by using a limiter. Or we can turn down the levels by 10 dB, which usually is an unacceptable alternative, especially for a rock audience. While increasing the system's dynamic range may be the "ideal" way to do things, logistics and economics sometimes dictate the use of a limiter to restrict the program dynamics.



When large numbers of amplifiers and speakers are used in a sound system, it is usually sufficient to allow about 10 dB of headroom. However, when a sound system is used for background music and paging in noisy environments such as factories, or when high sound levels must be avoided such as in hospitals, it may not be necessary to allow much headroom at all. Here, limiters are often set to allow only 5 or 6 dB of headroom, "squashing" anything in the program material that tries to peak beyond that. A squashed sound may not be "Hi-Fi," but usually the paging message is all-important, not the sound quality in such systems.

The cost of sound system components becomes an important factor when big sound systems are asked to produce that little bit of extra headroom for clean sound. Remember, every time we want just 3 dB more sound in the air, we must double the power in watts fed to the speakers, or, if we have only a fixed amount of watts, double the number of speakers we use.

Where budgets permit, it is excellent "insurance" to overbuild just a bit – especially in the amplifier department.

Specific techniques for headroom optimization

In order to properly set up a sound system, it is essential to have some basic test equipment. An oscillator and a voltmeter are the two most needed items. The oscillator can be inexpensive, so long as it's capable of producing a tone that can be adjusted up and down in level. If you've got a single frequency oscillator, its frequency should be somewhere around 1 kHz. The voltmeter needs to be "flat," responding equally to voltages over the range of frequencies you expect to use for testing. Many digital multimeters (DMM's) are not suitable for audio voltage measurement because the frequency response is very limited. Although some models may include advanced features like "true RMS reading" voltage over a wide frequency range, you must be sure to check the frequency response specification in such units. Your voltmeter also needs to be sensitive enough so that you're not limited to reading voltages from low level devices such as mixers and preamps, with the meter indication near the bottom of a meter scale.

Each piece of audio gear in your system should have data available that tells you what the maximum input and output levels are, and what the "nominal" (average) levels are. Here's a hypothetical setup example, designed to allow 20 dB of headroom:

1. Start by feeding a "nominal" level test signal to the first piece of gear in your audio signal chain. For this example we'll use a simple stage PA setup. We'll feed a mic-level signal, about -50 dBu (0.003 V), into the mic inputs of the mixer. (0 dBu =0.775 Vrms). The exact voltage isn't critical as long as it's somewhere near a typical microphone output level. However, keep in mind that a mic with a rated −50 dB output can produce anywhere from −70 dB when placed several feet in front of a quiet acoustic guitar, all the way to 0 dB or more when placed inside a kick drum or near a loud singer's mouth. This is a good reason to use mixers with adequate input headroom, like the Fender 42 Series.
2. On the mixer, set the mic channel's GAIN control, ATTENUATOR, and/or PAD so no overload lights flash in response to your input levels, and THEN set the mic channel's INPUT LEVEL CONTROL (typically its slide fader) to its nominal setting.
3. Now, adjust the mixer's MASTER level control so that the output from the mixer is 20 dB BELOW THE MIXER'S RATED MAXIMUM OUTPUT. This ensures that the remaining 20 dB of mixer output is available as "headroom."
4. Let's use an equalizer as an example of an item that might be patched between the mixer and the power amp. We'll keep the example simple but remember that all the pieces in any audio system should be treated the same way when you're setting levels. Let's assume that the mixer's maximum output level is +24 dBu so we set its output level to +4 dBu in order to establish our 20 dB of headroom.

   The maximum input level of our example equalizer is +14 dBu, which is 10 dB less than the mixer's maximum output level. Setting the input level on the equalizer at nominal +4 dB would result in our running out of headroom 10 dB too soon. At this point, we need to pad down (attenuate) the output of the mixer by 10 dB so the equalizer receives a −6 dB input level; this will preserve the 20 dB of headroom through the equalizer. Very often the equalizer will have a built in gain control to accomplish the necessary input attenuation. If not, the same job can be accomplished with a pad made of a few resistors. It may seem almost intuitive to make up the level difference between the mixer and equalizer by simply decreasing the setting of the mixer's master gain control. This is not satisfactory, however, since the mixer now must operate at a 10 dB noise disadvantage. Consider that 10 dB lower output level means 10 dB more headroom, but the signal level is 10 dB closer to the residual noise of the mixer.
5. The meters in audio equipment sometime require calibration to make their readings agree with nominal levels. Therefore, at each step of these gain settings you may wish to check the output voltages with your voltmeter to be sure signals are at the correct level at the output jacks; don't trust any front panel meters until you have verified their accuracy.
6. Once you have set the equalizer's input level to agree with the nominal +4 dBu output level of the mixer, you can consider the output of the equalizer, which must drive the power amplifier.
7. The 2224 and 2244 have a specification that might appear confusing in light of the "maximum" and "nominal" values given for the mixer and the equalizer. They state: "Sensitivity equals 0.775 V input for rated power output." This means that the output from any device whose output voltage is 0.775 V OR MORE, will be capable of driving the 2224/2244 to its full rated power output. If the source driving the 2224/2244 offers more than 0.775 V, then the input attenuators on the front panel of the 2224/2244 should be adjusted down to reduce (attenuate) the input voltage; otherwise clipping may occur.

   The 2224 and 2244 have an active electronic input which should not be fed more than +25 dBu (13.78 V). This input circuitry is ahead of the input attenuators. Thus it is possible to overload them if you exceed the +25 dBu maximum input level specification. This represents 25 dB of headroom in the input circuit, which is more headroom than is needed in any reasonable audio system.
8. The equalizer in our example has a maximum input rating of +14 dBu, or 3.88 volts. For the sake of simplicity, let's assume that the equalizer provides unity gain: that is, its output always equals its input (assuming no boost or cut is affecting the signal). Let's further assume that the equalizer's maximum output level is also +14 dBu. The 2224/2244 require 0 dBu (0.775 V) to operate at maximum. A simple calculation, subtracting 20 dB of headroom from the equalizer's +14 dB maximum level provides a nominal level of −6 dBu. This is still 14 dB more than we need to drive the

power amplifier at a −20 dB average input (which provides 20 dB of headroom for the amplifier). *To match the levels, we can turn the amplifier's input attenuator DOWN 14 dB and get 14 dB less noise and hiss from the speakers.* The whole system will still operate all the way up to full power on peaks that are 20 dB above nominal! What's more, once it's set up, we can operate the entire system we've described here with one single volume control – the "master" fader on the mixer!

NOTE Remember, you can always turn down the level of an amplifier which is very sensitive (as are these Fender amps). If an amplifier is insensitive, you may "run out" of signal level in some cases, so you may not be able to drive the amp to its maximum output level without purchasing an extra "booster" amp. Think of excess sensitivity as an opportunity for noise reduction at your discretion; turn down the attenuators when you don't absolutely need the sensitivity.

Mains requirements

The 2224/2244 requires an AC power line voltage of 120 volts, plus or minus 10%. That is, it will operate normally with AC power input between 108 volts and 132 volts. We recommend that you provide AC power as close to 120 volts as possible, because if the voltage on the powerline falls too far below 120 volts, or surges above 132 volts, the 2224/2244 will not operate properly and may be damaged.

The 2244 can draw up to 30 amperes peak from the AC power line during periods of high power demands such as those created by heavy bass notes in the program. If you plan to run several 2244's at or near full power from a single AC power line, check to make sure your AC power line can supply enough current (allow a minimum of 25 amps for each 2244).

In large commercial portable sound systems, like those used in rock concerts, house electricians should be able to provide as many different AC power line feeds as needed. In arenas that hold 10,000 people or more, between 5000 and 50,000 watts of amplifier power may be supplied to speaker systems "flown" (hung) over the stage area. The AC power lines used for these huge arrays should be rated at twice the "worst case" load conditions anticipated (though they rarely are). Even so, there are still occasional "disasters" where limited or out of date power line feeds are shared by lights and sound, and a surprise surge or instantaneous power demand can shut down the whole show.

AC outlet wiring

AC power outlets can be wired in different ways. There is only one way, however, that 120-volt AC power outlets can be properly wired. The power line usually enters a building from a pole-mounted transformer. The downstream or "secondary" winding of this "service entrance transformer" has a "tap" at the middle of the winding called the "center tap." In household wiring, the voltage appearing at either end of the winding of the 240-volt secondary is thus split into two separate 120-volt "legs," with a "neutral" or "common" (the center tap) and two "hot" wires. In commercial wiring, a 3-phase, high voltage line is usually converted down to 240 or 220 volts, then down again to 120 or 110 volts.

The center tap common is connected

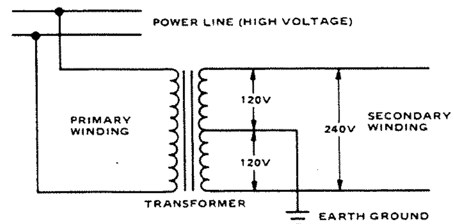


FIGURE 39. 240V CENTER-TAPPED SECONDARY FEEDING TWO OUTLETS.

to ground at three places (National Electrical Code as of this writing). First, the transformer's center tap runs down the power pole and connects to a large copper bar driven into the earth at the base of the pole. Second, the center tap runs to the building entrance box, connects to the "neutral bus bar" inside the box, and splits to another copper bar driven into the earth at the building, and also to the cold water main pipe at the power service entrance. Metal "flex" conduit, solid metal conduit pipes or a separate third wire such as in "romex" type cable, is then connected to metal wall boxes in the building (metal screws connect the third or "ground" prong of the outlet to the metal box). This third prong is called the "safety ground." The white wires (neutral) are always connected directly to ground, and never bypassed or interrupted by a circuit breaker or fuse. The black wires are first connected to circuit breakers or fuses in the power service box and then routed to the wall outlets in the building.

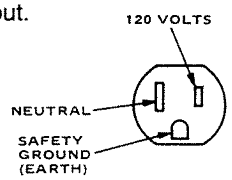
WARNING
IN AC POWER WIRING, BLACK IS HOT, WHITE IS NEUTRAL – the opposite of most audio signal wiring and speaker wiring. It is safer to treat all AC wiring as potentially lethal; how much do you trust a stranger who may have miswired the system in the past? Be wary, and test the voltages yourself. You may prevent a tragedy.

Although the white wires and the ground in a building are technically at the same potential (voltage), and should measure the same potential using a voltmeter, the ground prong connections at the outlets are connected to the grounding bar that was driven into the earth as an additional safety precaution in case something should happen to the wires running from the service entrance transformer to the building or within the equipment itself. If a short should occur within the equipment, hopefully the electricity will find its way to ground via the safety ground, instead of via a person's body. When checking AC power lines at the outlet, be sure you have proper testing tools and some familiarity with the danger of shock hazards from AC power. Follow the diagram shown here, being careful not to touch metal with your hands or short the test leads together:

WARNING

Cold water pipes no longer afford a reliable means to ground an electrical system. This is because water companies and even local building codes now favor the installation of a length of PVC (plastic) pipe prior to the water meter for the express purpose of electrically isolating the water system from any electrical grounds. In the past, with dry, poor-conductivity soil, some water company crews had experienced dangerous electrical shocks at distances up to 1/4 mile from the facility where a water pipe had been used to sink electrical currents. In fact, for this reason some areas now have ordinances AGAINST using water pipes for grounding. If necessary, use your own copper or chemical ground rod, and wet the earth around it.

If you detect any voltage between the larger slot (white wire) in an outlet and the round ground terminal (round third pin) when there is no load on that line, you should contact a licensed electrician to check it out.



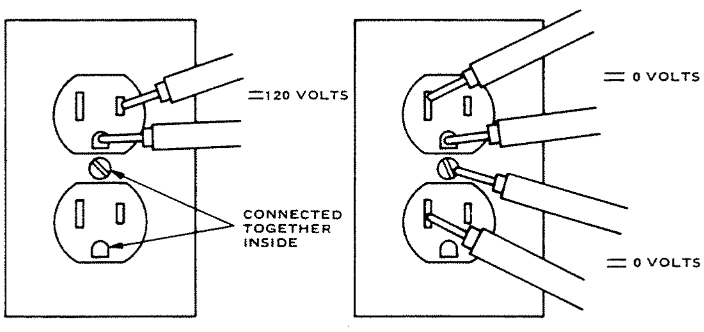


FIGURE 40. TESTING AC OUTLETS FOR PROPER WIRING.

Improperly wired AC outlets

No matter whether you consider yourself a technician or not, there are two items you ought to have with you every time you set up your system in a new location. One of these is a commercial outlet tester, the second is a neon lamp AC voltage tester. You should be able to buy these inexpensive items at most hardware stores, electrical stores and some lighting stores.

The three-prong outlet tester will tell you if the outlet is properly wired. An improperly wired outlet may have its two AC wires reversed ("polarity reversal") or it may have a disconnected ground. ANY fault in the wiring of the AC outlet is potentially hazardous and thus, THE BEST, AND PERHAPS ONLY SAFE WAY TO DEAL WITH AN IMPROPERLY WIRED AC OUTLET IS TO SIMPLY REFUSE TO USE IT UNTIL IT HAS BEEN REPAIRED.

Dealing with two-wire outlets

The problem with two-wire AC outlets is they don't have that very important third ground prong. Thus, to use one of these two-wire outlets you have to "adapt" it to the three-wire AC plug on your sound system (your mixer) with a two-wire to three-wire AC adapter. PROPERLY USED, these adapters maintain a safe ground for the sound system just as well as a three-wire outlet. To make this two-wire adapter work properly, you MUST connect the loose green wire on the two-wire end of the adapter to a GROUNDED screw on the two-wire outlet. How do you know whether or not the screw is grounded? Easy! First connect the adapter's green wire to the screw on the two-wire outlet; then plug in the two-wire adapter into the outlet. Now, plug in your three-wire AC outlet tester into the adaptor. If the screw is grounded, your AC outlet tester will tell you. (Most three-wire AC outlet testers either have a "good" light, or they don't light at all on a good receptacle.) If the screw is not grounded, the outlet tester will indicate this too. In this case, you must connect the adapter's green wire to some other grounded screw in order to maintain a safe ground for your system.

If the outlet tester shows a good ground but reversed polarity on your two-wire to three-wire adapter, sometimes you can reverse the adapter in the outlet by pulling it out, twisting it a half turn and reconnecting it.

A troublesome form of "lifted ground" is the accidental disconnection of the ground or green wire from the outlet's safety ground. In older wiring, the heavy green wire was sometimes omitted from internal wall wiring in favor of letting the metal flex conduit or pipe suffice as the ground path from the electrical service entrance. Inspectors usually consider this approach all right. Normally, there is no problem as long as the metal conduit in the wall remains intact and all the screws holding its joints together stay tight – and there might be dozens of those joints between the water main and the individual outlet boxes in the walls. A barely touching or loose screw in some conduit joint inside a wall can cause the ground of the next outlet box in the line (and all the subsequent boxes on that same line) to become disconnected.

Polarity reversal in the wall, with the wires connected black-for-white, can cause hum or noise in some equipment as well as producing a shock hazard. With some improperly designed equipment, polarity reversed power wiring could cause blown fuses, circuit breakers, or at worst, a fire or LETHAL shock hazard. For example, consider a typical piece of sound equipment with a 2-wire AC cord with a bypass capacitor installed between chassis and the neutral (ostensibly for hum reduction). Leakage across the capacitor, or a shorted capacitor, connects the chassis to the AC neutral leg. If the neutral and hot lead are reversed in the outlet in this situation, the chassis is actually hot. This can also occur if the "ground" switch is set so that the capacitor is connected between the "hot" side of the AC line and the amplifier chassis. Effectively, this is the same as plugging the amplifier into a miswired outlet (and just as dangerous, too). If a performer touches a grounded microphone case, or any other grounded item, and touches the hot chassis, a lethal shock can occur; it has happened before.

If the neutral becomes lifted at a power outlet, it is possible that items plugged into the outlet can absorb the full 240 volts available from the power service. Figure 41 shows how.

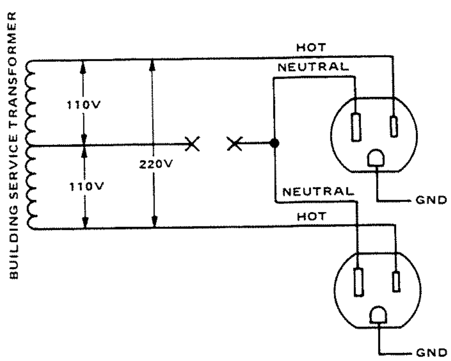


FIGURE 41. 110 V AC OUTLETS WITH LIFTED NEUTRAL.

Such outlets may operate, but the voltage can swing from 0 V to 220 V AC, creating a shock hazard and possibly damaging your equipment.

If a 2244 were plugged into one socket of one of the two outlets in Figure 41, and a rack of signal processing equipment plugged into the other, fuses would probably blow upon turning on the system, and some of the signal processing equipment could be destroyed.

TIPS

1. Get an AC outlet tester for your tool box.
2. Be conscientious, use your tester on power lines, and use a neon voltage tester to check for voltage present between microphones and guitar amps.
3. Use common sense; while using a clip-on ground lead may do the job in your equipment rack, it may not be the safest way to get the job done. AC power must be regarded with respect.
4. Don't use questionable power outlets; anticipate the worst and carry a long extension made of heavy (small gauge number) wire with you at all times. (An extension should be #12-3 (12 gauge, 3 wires), and no longer than 15 meters (50 feet).)
5. If you can't find suitable power at a job site, refuse to plug your equipment in. Besides being a hazard, it could destroy your equipment. Don't risk shock hazards or destruction of your equipment.

You can build a small 0-250 volt AC voltmeter into a box to be plugged between the AC power mains and your equipment racks. It's not a bad idea to be able to monitor the line voltage from the wall and check each of the three wiring legs supplying your power for proper potential with respect to ground. Some racks have a panel meter mounted so that line voltage can be read at a glance. A "soft" power line that "sponges" up and down in voltage as the load changes will affect the sound of the system, often sounding like a bad limiter. More important, should the voltage drop too low, it is possible that equipment can be damaged so monitor your line voltage *at the equipment end* of any long power extensions you do use.

SECTION TWO

General Information about Amplifiers and Sound Systems

What is an amplifier?

An amplifier is a device that makes more of something; with an electronic audio amplifier, the "something" is an audio signal. An amplifier can raise the power of an audio signal by raising its voltage, current, or both. By this definition, any electronic device that supplies a net power increase to an audio signal fed into it is an amplifier (whether it's called a mixer, a preamp or a power amp).

The ability of amplifiers to increase the power of their input signals is called "gain." Often, but incorrectly, "gain" is called "amplification" or "volume." When we "turn up the gain," the result is an increase in the volume or sound level from a speaker, or an increase in signal level or "amplitude" in an electronic circuit.

Amplifiers increase signal strength using circuitry to control some source of power. In the case of these Fender amplifiers, power is obtained from the 60 Hz AC power line, and converted to a usable form (DC) by power supply circuitry inside the amplifier. The DC power is then controlled by the relatively weak audio input signal and converted to a more powerful audio output signal. The circuitry can be configured in any of thousands of different ways to do the job intended by the designer.

Typical power levels in sound systems

Many typical audio sources may not provide meters to display their output power levels, but there are industry standards (or widely accepted practices) that allow us to estimate the levels. The lowest power level in a typical audio system is generated by microphones and phono cartridges. Normal speech at about a meter from the "average" dynamic microphone produces a power output from the microphone of only about a TRILLIONTH of a watt. Phono cartridges produce as much as a thousand times this much power, but the signal is still very weak, measured in BILLIONTHS of a watt. Microphone and phono preamps supply the gain needed to raise the feeble output of mics and cartridges up to a power range called "line level." Line levels are between 10 MILLIONTHS of a watt and 250 THOUSANDTHS of a watt (1/4 W). While some mixers and preamps boost signal levels a bit further, such as for headphones, line levels measured in milliwatts cannot drive speakers to useable levels. A POWER AMPLIFIER must be added to deliver tens or hundreds of watts to the speakers.

Dynamic range requirements for speech and music reproduction

Signal Dynamics vary greatly with different types of program material. The human voice usually produces a fairly steady volume level (little dynamic range) when speaking compared to the wide dynamic range of music signals, though there are lively exceptions. Sound systems which are required to amplify only speech often can be designed using components with less "headroom" than those necessary to reproduce music for the same size audience. ("Headroom" is the system's remaining capability to produce levels over and above the nominal or average level.)

Speech is usually considered loud enough if the entire audience can understand what a person is saying, whereas music systems are designed to deliver some degree of excitement by virtue of sheer sound levels – and to do this with enough headroom to properly reproduce musical program energy peaks without a lot of distortion. For example, an acoustically "good" meeting room or restaurant banquet room can be well covered by a dozen inexpensive ceiling-mounted speakers and a 25 watt amplifier so that 300 people can hear a speech clearly. To hear the same degree of detail in recorded music, rather than speech, the same 300 people might need a pair of the largest studio monitor speakers and a 240 watt per channel amplifier (about 13 dB more peak power capability).

Why does music sometimes require so much more power? Consider the brief "attack" of a plucked or hammered string; it may not seem much louder than the sound which follows, but it typically does require much more power to be accurately reproduced. In some cases the difference or ratio between the peak and average values of sound pressure level (SPL) from percussion instruments can be as high as 40 dB, which expresses a power ratio of 10,000-to-1. 10,000:1 is the ratio of the highest to the average power, but what of the difference between the highest peak levels and quietest levels in a live musical performance? (Be aware that a 10 dB power ratio is 10 times the power, whereas a 20 dB voltage or SPL ratio is 10 times to voltage or sound pressure; this factor sometimes causes confusion for dB comparisons.) Theoretically, the live "quiet-to-concert fortissimo" sound pressure level ratios may be upwards of 120 dB (a ONE TRILLION-to-1 ratio). That is, if the *average* sound were to be reproduced with just 1 watt, the *loudest peak* sound would require 10,000 watts, and the *quietest* sound just 0.00000001 watts! (This is one reason we use the dB to describe sound levels... the actual numbers with watts are often too large or too small for convenient reference.) The theoretical dynamic range and practical dynamic range will differ, however. Typically, the ambient noise level (the "noise floor") is between 30 and 40 dB SPL in a quiet hall, and since the threshold of pain is about 120 dB, that leaves a useable dynamic range of about 80 dB at best. In fact, 60 dB or 70 dB more often works out to be the actual dynamic range of a performance. That's still, of course, a 1,000,000:1 or 10,000,000:1 power ratio. To be completely capable of reproducing percussion sounds then, it is necessary to provide plenty of headroom in the audio system.

In addition to dynamic considerations, there are other differences between sound systems for speech and music. Speech covers a frequency range or "bandwidth" that is much less than that needed to reproduce music. Specifically, amplified speech which is perceived as being of high quality can be produced by a sound system with two octaves less frequency range than a music system perceived as having equal quality. Since more power is needed by the increased bandwidth, music reproduction requires more power than speech reproduction; since the program peaks usually determine amplifier size for a given sound system, and music has wider dynamic range, still more power is required to reproduce music with comparable quality to a speech-only system.

The Decibel (dB): what is it, and why is it used?

The decibel (dB) is engineering shorthand for a comparison of two numbers; it is a ratio of two numbers... an exponent... a logarithm to the base 10. Alexander Graham Bell found that sounds perceived by humans as "twice as loud" had about ten times as much power as the sounds to which they were being compared. The term "Bel" was named after him for his accomplishments. The "Bel" represents a large value difference, so the "decibel" or dB, a tenth of a Bel, has become the standard term. It turns out that a 1 dB change in sound is about the limit of our ability to recognize a change in volume of normal speech. The human hearing mechanism can distinguish sounds over volume levels that vary more than a trillion-to-one (120 dB), from the quietest whisper at 50 feet, to the physical sensation in our ears at 25 feet from the stage of a loud rock concert.

It's one thing to compare small power ratios like two-to-one or ten-to-one, and another matter altogether to compare ratios like 1,000,000,000,000:1 (a trillion-to-one). Using dB, the comparison is put in perspective this way:

2:1 = 3 dB

10:1 = 10 dB

1,000,000,000,000:1 = 120 dB

The math for this may seem simple, if you know algebra, or somewhat more complex if you don't. In any event, the equations are presented here.

Decibels for power is equal to 10 times the common Log (base 10) of one power divided by the other, or:

$$(\text{watts})\ \text{dB} = 10\text{Log}_{10}\ (\text{P1/P2})$$

To find the dB difference in volts or the difference in SPL, use 20Log instead of 10Log:

$$(\text{volts})\ \text{dB} = 20\text{Log}_{10}\ (\text{V1/V2})$$

$$(\text{SPL})\ \text{dB} = 20\text{Log}_{10}\ (\text{SPL1/SPL2})$$

EXAMPLES

Power The difference in dB between 50 watts and 100 watts is:

$\text{dB} = 10\text{Log}_{10}\ (100/50)$ which is 3.01 dB

Voltage The difference in dB between 50 volts and 100 volts is:

$\text{dB} = 20\text{Log}_{10}\ (100/50)$ which is 6.02 dB

Sound pressure level is usually expressed as dB SPL. 0 dB SPL is the threshold of hearing for the best human ear. Saying "100 dB SPL" means that the sound pressure is 100 dB or 100,000 times the sound pressure of the smallest audible sound. That 0 dB threshold is a ridiculously small number, 20 micropascals or 0.0002 microbar, which is 0.00000000000000011 watts (at the eardrum)! Computing the power of 100 dB SPL gives you 1.1 microwatts on the eardrum, and 120 dB SPL, the threshold of pain, calculates to be 110 microwatts, still only 0.00011 watts of acoustic power pushing on the eardrum.

You can see from this example how much easier it is to use dB to describe the huge differences, both in terms of tiny numbers and big ratios. The accompanying chart shows how dB can describe both power and voltage (sound pressure levels work just like voltage).

TABLE 4. THE DECIBEL RELATED TO POWER AND VOLTAGE.

Given the value in dB, this chart helps you find power or voltage (or SPL) ratios. For positive (+) values of the decibel, both voltage and power ratios are greater than unity; use the two right-hand columns.

For negative (−) values of the decibel, both voltage and power ratios are less than unity; use the two left-hand columns.

		Power Ratio	Pressure Ratio
EXAMPLE-Given: ±9.1 dB	**Find: +9.1dB**	**8.128**	**2.851**
	−9.1dB	**0.1230**	**0.3508**

← −dB			+dB →	
Voltage Ratio	**Power Ratio**	**dB**	**Voltage Ratio**	**Power Ratio**
1.000	1.000	0	1.000	1.000
.9886	.9772	.1	1.012	1.023
.9772	.9550	.2	1.023	1.047
.9661	.9333	.3	1.035	1.072
.9550	.9120	.4	1.047	1.096
.9441	.8913	.5	1.059	1.122
.9333	.8710	.6	1.072	1.148
.9226	.8511	.7	1.084	1.175
.9120	.8318	.8	1.096	1.202
.9016	.8128	.9	1.109	1.230
.8913	.7943	1.0	1.122	1.259
.8810	.7762	1.1	1.135	1.288
.8710	.7586	1.2	1.148	1.318
.8610	.7413	1.3	1.161	1.349
.8511	.7244	1.4	1.175	1.380
.8414	.7079	1.5	1.189	1.413
.8318	.6918	1.6	1.202	1.445
.8222	.6761	1.7	1.216	1.479
.8128	.6607	1.8	1.230	1.514
.8035	.6457	1.9	1.245	1.549

← −dB			+dB →	
Voltage Ratio	**Power Ratio**	**dB**	**Voltage Ratio**	**Power Ratio**
.7943	.6310	2.0	1.259	1.585
.7852	.6166	2.1	1.274	1.622
.7762	.6026	2.2	1.288	1.660
.7674	.5888	2.3	1.303	1.698
.7586	.5754	2.4	1.318	1.738
.7499	.5623	2.5	1.334	1.778
.7413	.5495	2.6	1.349	1.820
.7328	.5370	2.7	1.365	1.862
.7244	.5248	2.8	1.380	1.905
.7161	.5129	2.9	1.396	1.950
.7079	.5012	3.0	1.413	1.995
.6998	.4898	3.1	1.429	2.042
.6918	.4786	3.2	1.445	2.089
.6839	.4677	3.3	1.462	2.138
.6761	.4571	3.4	1.479	2.188
.6683	.4467	3.5	1.496	2.239
.6607	.4365	3.6	1.514	2.291
.6531	.4266	3.7	1.531	2.344
.6457	.4169	3.8	1.549	2.399
.6383	.4074	3.9	1.567	2.455

← −dB			+dB →	
Voltage Ratio	Power Ratio	dB	Voltage Ratio	Power Ratio
.6310	.3981	4.0	1.585	2.512
.6237	.3890	4.1	1.603	2.570
.6166	.3802	4.2	1.622	2.630
.6095	.3715	4.3	1.641	2.692
.6026	.3631	4.4	1.660	2.754
.5957	.3548	4.5	1.679	2.818
.5888	.3467	4.6	1.698	2.884
.5821	.3388	4.7	1.718	2.951
.5754	.3311	4.8	1.738	3.020
.5689	.3236	4.9	1.758	3.090
.5623	.3162	5.0	1.778	3.162
.5559	.3090	5.1	1.799	3.236
.5495	.3020	5.2	1.820	3.311
.5433	.2951	5.3	1.841	3.388
.5370	.2884	5.4	1.862	3.467
.5309	.2818	5.5	1.884	3.548
.5248	.2754	5.6	1.905	3.631
.5188	.2692	5.7	1.928	3.715
.5129	.2630	5.8	1.950	3.802
.5070	.2570	5.9	1.972	3.890
.5012	.2512	6.0	1.995	3.981
.4955	.2455	6.1	2.018	4.074
.4898	.2399	6.2	2.042	4.169
.4842	.2344	6.3	2.065	4.266
.4786	.2291	6.4	2.089	4.365
.4732	.2239	6.5	2.113	4.467
.4677	.2188	6.6	2.138	4.571
.4624	.2138	6.7	2.163	4.677
.4571	.2089	6.8	2.188	4.786
.4519	.2042	6.9	2.213	4.898
.4467	.1995	7.0	2.239	5.012
.4416	.1950	7.1	2.265	5.129
.4365	.1905	7.2	2.291	5.248
.4315	.1862	7.3	2.317	5.370
.4266	.1820	7.4	2.344	5.495
.4217	.1778	7.5	2.371	5.623
.4169	.1738	7.6	2.399	5.754
.4121	.1698	7.7	2.427	5.888
.4074	.1660	7.8	2.455	6.026
.4027	.1622	7.9	2.483	6.166
.3981	.1585	8.0	2.512	6.310
.3936	.1549	8.1	2.541	6.457
.3890	.1514	8.2	2.570	6.607
.3846	.1479	8.3	2.600	6.761
.3802	.1445	8.4	2.630	6.918
.3758	.1413	8.5	2.661	7.079
.3715	.1380	8.6	2.692	7.244
.3673	.1349	8.7	2.723	7.413
.3631	.1318	8.8	2.754	7.586
.3589	.1288	8.9	2.786	7.762
.3548	.1259	9.0	2.818	7.943
.3508	.1230	9.1	2.851	8.128
.3467	.1202	9.2	2.884	8.318
.3428	.1175	9.3	2.917	8.511
.3388	.1148	9.4	2.951	8.710
.3350	.1122	9.5	2.985	8.913
.3311	.1096	9.6	3.020	9.120
.3273	.1072	9.7	3.055	9.333
.3236	.1047	9.8	3.090	9.550
.3199	.1023	9.9	3.126	9.772

← −dB			+dB →	
Voltage Ratio	Power Ratio	dB	Voltage Ratio	Power Ratio
.3162	.1000	10.0	3.162	10.00
.3126	.0977	10.1	3.199	10.23
.3090	.0955	10.2	3.236	10.47
.3055	.0933	10.3	3.273	10.72
.3020	.0912	10.4	3.311	10.96
.2985	.08913	10.5	3.350	11.22
.2951	.08710	10.6	3.388	11.48
.2917	.08511	10.7	3.428	11.75
.2884	.08318	10.8	3.467	12.02
.2851	.08128	10.9	3.508	12.30
.2818	.07943	11.0	3.548	12.59
.2786	.07762	11.1	3.589	12.88
.2754	.07586	11.2	3.631	13.18
.2723	.07413	11.3	3.673	13.49
.2692	.07244	11.4	3.715	13.80
.2661	.07079	11.5	3.758	14.13
.2630	.06918	11.6	3.802	14.45
.2600	.06761	11.7	3.846	14.79
.2570	.06607	11.8	3.890	15.14
.2541	.06457	11.9	3.936	15.49
.2512	.06310	12.0	3.981	15.85
.2483	.06166	12.1	4.027	16.22
.2455	.06026	12.2	4.074	16.60
.2427	.05888	12.3	4.121	16.98
.2399	.05754	12.4	4.169	17.38
.2371	.05623	12.5	4.217	17.78
.2344	.05495	12.6	4.266	18.20
.2317	.05370	12.7	4.315	18.62
.2291	.05248	12.8	4.365	19.05
.2265	.05129	12.9	4.416	19.50
.2239	.05012	13.0	4.467	19.95
.2213	.04898	13.1	4.519	20.42
.2188	.04786	13.2	4.571	20.89
.2163	.04677	13.3	4.624	21.38
.2138	.04571	13.4	4.677	21.88
.2113	.04467	13.5	4.732	22.39
.2089	.04365	13.6	4.786	22.91
.2065	.04266	13.7	4.842	23.44
.2042	.04169	13.8	4.898	23.99
.2018	.04074	13.9	4.955	24.55
.1995	.03981	14.0	5.012	25.12
.1972	.03890	14.1	5.070	25.70
.1950	.03802	14.2	5.129	26.30
.1928	.03715	14.3	5.188	26.92
.1905	.03631	14.4	5.248	27.54
.1884	.03548	14.5	5.309	28.18
.1862	.03467	14.6	5.370	28.84
.1841	.03388	14.7	5.433	29.51
.1820	.03311	14.8	5.495	30.20
.1799	.03236	14.9	5.559	30.90
.1778	.03162	15.0	5.623	31.62
.1758	.03090	15.1	5.689	32.36
.1738	.03020	15.2	5.754	33.11
.1718	.02941	15.3	5.821	33.88
.1698	.02884	15.4	5.888	34.67
.1679	.02818	15.5	5.957	35.48
.1660	.02754	15.6	6.026	36.31
.1641	.02692	15.7	6.095	37.15
.1622	.02630	15.8	6.166	38.02
.1603	.02570	15.9	6.237	38.90

← −dB			+dB →	
Voltage Ratio	Power Ratio	dB	Voltage Ratio	Power Ratio
.1585	.02512	16.0	6.310	39.81
.1567	.02455	16.1	6.383	39.81
.1549	.02399	16.2	6.457	41.69
.1531	.02344	16.3	6.531	42.66
.1514	.02291	16.4	6.607	43.65
.1496	.02239	16.5	6.683	44.67
.1479	.02188	16.6	6.761	45.71
.1462	.02138	16.7	6.839	46.77
.1445	.02089	16.8	6.918	47.86
.1429	.02042	16.9	6.998	48.98
.1413	.01995	17.0	7.079	50.12
.1396	.01950	17.1	7.161	51.20
.1380	.01905	17.2	7.244	52.48
.1365	.01862	17.3	7.328	53.70
.1349	.01820	17.4	7.413	54.95
.1334	.01778	17.5	7.499	56.23
.1318	.01738	17.6	7.586	57.54
.1303	.01698	17.7	7.674	58.88
.1288	.01660	17.8	7.762	60.26
.1274	.01622	17.9	7.852	61.66
.1259	.01585	18.0	7.943	63.10
.1245	.01549	18.1	8.035	64.57
.1230	.01514	18.2	8.128	66.07
.1216	.01479	18.3	8.222	67.61
.1202	.01445	18.4	8.318	69.18
.1189	.01413	18.5	8.414	70.79
.1175	.01380	18.6	8.511	72.44
.1161	.01349	18.7	8.610	74.13
.1148	.01318	18.8	8.710	75.86
.1135	.01288	18.9	8.811	77.62
.1122	.01259	19.0	8.913	79.43
.1109	.01230	19.1	9.016	81.28
.1096	.01202	19.2	9.120	83.18
.1084	.01175	19.3	9.226	85.11
.1072	.01148	19.4	9.333	87.10
.1059	.01122	19.5	9.441	89.13
.1047	.01096	19.6	9.550	91.20
.1035	.01072	19.7	9.661	93.33
.1023	.01047	19.8	9.772	95.50
.1012	.01023	19.9	9.886	97.72
.1000	.01000	20.0	10.000	100.00
3.162×10^{-1}	10^{-1}	10	3.162	10
10^{-1}	10^{-2}	20	10	10^{2}
3.162×10^{-2}	10^{-3}	30	3.162×10	10^{3}
10^{-2}	10^{-4}	40	10^{2}	10^{4}
3.162×10^{-3}	10^{-5}	50	3.162×10^{2}	10^{5}
10^{-3}	10^{-6}	60	10^{3}	10^{6}
3.162×10^{-4}	10^{-7}	70	3.162×10^{3}	10^{7}
10^{-4}	10^{-8}	80	10^{4}	10^{8}
3.162×10^{-5}	10^{-9}	90	3.162×10^{4}	10^{9}
10^{-5}	10^{-10}	100	10^{5}	10^{10}

Common forms of the dB

The dB is always used to express a ratio. However, it has been adopted to express many different quantities, or to express similar quantities in terms that are better suited to particular applications. You may have noticed different suffix letters tacked onto the dB. Here is a listing of the most common forms, with their 0 dB reference values:

0 dBm = One milliwatt. The dBm is an expression of POWER, and has come to be regarded as a "standard." Any load can be used, but remember that with a more-or-less fixed voltage, the power, and hence the number of dBm, will change as the load changes.

(0) dBV = One volt. No load or other conditions are implied here. This figure is the International (ISO) standard. Many manufacturers use this figure when discussing noise, since it creates a 2.2 dB "advantage" for the specification, i.e., "EIN = −132.2 dBV" is the same as "EIN = −130.0 dBv." dBV is commonly used to specify microphone open-circuit output voltage.

(0) dBv = 0.775 volt. No load or other conditions implied.

(0) dBu = 0.775 volt. No load or other conditions implied. This form supercedes dBv to avoid confusion with dBV.

(0) dBW = 1 watt. The dBW denotes the power level referred to 1 watt into 8 ohms.

(0) dB SPL = 20 micropascals. The SPL value denotes decibels of sound pressure level above the threshold of audibility, defined as 20 uPa (micropascals) R.M.S. at 1000 Hz. This does not directly relate to power.

dBA = Specially weighted dB SPL. dBA is a measurement, in decibels, made using a filter (an "A" weighting filter) that simulates the non-linear sensitivity (poor low-level frequency response) of human hearing. "dBA" usually refers to readings made with sound level meters for the purpose of determining how the measured sound is perceived, rather than its absolute value in "unweighted" dB SPL. Unfortunately, "dBA" measurements include a low frequency rolloff, and therefore do not reflect the presence of hum. However, "dBA" is also used by OSHA and other government agencies that monitor sound levels in industrial or public environments.

There are many other forms of the dB;

you can make them up to suit your need to compare any two quantities – "double your pleasure" could read; "+3 dB Ps" if you wanted to go that far. The terms listed here are the ones you're likely to find most often in audio literature.

Fender has purposely chosen to use "dB" to represent the voltage levels assigned with our products. Through the widespread use of 0 dB = 0.775 volts in the U.S. the term "0 dB" has attained the status of a generic reference voltage, thus, we feel it will avoid some confusion to simplify "dBu" to just "dB," and ignore load impedances except in specific discussion of load and impedance requirements.

There are some differences that are important to you, especially when you are comparing specs between pieces of equipment that use different forms of the dB. The two major differences are in the VOLTAGES of dBV, dBv and dBm. The use of dBm is most widespread although it is improperly used when a voltage is being described. "dBm" can only specify power, and no particular load impedance is implied, though 600 ohms is commonly used.

ANY FIGURE LISTED IN dBv OR dBu IS 2.2 dB LOWER THAN dBV IN VOLTAGE; that is, "0 dBv" (or dBu) equals "−2.2 dBV." The dBm term used to describe voltage is technically incorrect unless the line impedance is 600 ohms; where only voltage is concerned, dBu should be used. The dBm term is a POWER RATING. If you're specifying a voltage, use dBu or dBV. dBm can be misleading unless the impedance is also specified and calculations are made to confirm the voltage.

Most modern electronic audio components have high impedance inputs. The 2224 and 2244 have an input impedance of 15k ohms, this means that a single 600 ohm (minimum rated load Z), +4 dBu (0.775 V) output should be capable of driving twenty five 2224/2244 power amplifiers.

How speaker sensitivity affects required amplifier power

There are some 10,000 watt audio systems, but they are found primarily in very large concert sound systems or in large, permanently installed arena sound systems. Therefore, exactly what are we talking about in terms of the average power and the "extra" power required for adequate headroom in a typical high-quality sound system? A great deal depends upon the sensitivity of the speakers being used.

Actually, the amplifier requirements depend on the speaker system's *efficiency* (the total acoustic output power for a given electrical input power), but that value is seldom specified; instead, *sensitivity* specifications are readily available (sensitivity is the on-axis sound pressure level for a given electrical input power at a given distance, usually 1 watt at 1 meter). Sensitivity is not the same as efficiency, but we'll make a gross generalization and assume that if a speaker is more sensitive, it is more efficient (this is only true if the speakers being compared have exactly the same dispersion pattern at all frequencies).

Let's explore a typical "sound column" type speaker that might be used in a small club or meeting room. (A similar discussion is presented on page 27, but this further example is provided here to help make a point.) Such a speaker can produce an output of 100 dB SPL at a distance from the speaker of 1 meter when 1 watt is fed to its input (this sensitivity may vary plus 5 to minus 10 dB with different column speakers). The rated sensitivity specification must be put in perspective. Our audience won't be sitting one meter away from the speaker, and every time the distance away from the sound source doubles, the sound pressure level drops as much as 6 dB. (The 6 dB drop assumes a non-reverberant outdoor type environment; in reverberant environments, the sound level does not drop off as rapidly). Lets assume the closest seats in the audience are 8 meters from the speaker on stage – about 25 feet. This means that with 1 watt input power, the audience hears about 82 dB SPL, not 100 dB. For rock music, we might want to be able to produce 120 dB peaks in the front row, so let's start adding power until we're up to 120 dB. Given that there is an increase of 10 dB SPL for a tenfold increase in watts, lets see how the numbers work:

Given 100 dB for 1 watt at 1 meter, and given a speaker-to-audience distance of 8 meters (25 feet): 1 watt = 82 dB
10 watts = 92 dB
100 watts = 102 dB
1000 watts = 112 dB
6309 watts = 120 dB

We perceive a 10 dB increase in sound pressure level to be about twice as loud. If amplifier power alone is used to obtain that 10 dB, it takes ten times the power. Simple math tells us that to make the sound 4 times louder, 100 times the power is required. Given the above example, where 100 watts produces an average level of 102 dB SPL at 8 meters, it would take 10,000 watts to make the sound 4 times (20 dB) louder at 122 dB SPL. If we were to use a speaker that is just 3 dB less sensitive, the power requirements would be doubled to 20,000 watts. If the speaker were 10 dB less efficient, it would take ten times the power, or 100 kilowatts, to produce the same 122 dB. On the other hand, if the speaker were 10 dB more efficient, we could obtain the same 122 dB with a tenth the power, or about 1,000 watts. It becomes clear why, in order to allow a sufficient margin of power for headroom, the largest possible amplifier and the most sensitive speakers are desirable.

Ways to obtain adequate levels without burning up speakers

In order to produce 120 dB SPL at 25 feet with our reasonably sensitive speaker (100 dB/1W/1m), we're going to need a 6,300 watt amplifier. Unfortunately, the speaker will probably burn up when the amplifier power exceeds more than a hundred watts or so. More speakers must be utilized in order to handle the requisite 6 kilowatts. When speaker systems are properly stacked together in "clusters," their efficiency at certain frequencies increases so the total amplifier power requirements sometimes are reduced.

There are many ways of adding amplifiers and speakers to increase SPL. For simplicity we will apply the same amplifier VOLTAGE to each speaker in the following example; this implies that two identical speakers connected to the same amplifier output will draw twice the amount of power from the amplifier that one speaker would draw (in reality, this is not always so at maximum power output if protection circuits limit the output). In our example, one stereo amplifier and two speakers will be our reference, the starting point we define as "0 dB," and we assume a constant output voltage power amplifier:

QUAN. AMP. CHANNELS	QUAN. SPKRS ON EACH CHAN	TOTAL QUAN. SPEAKERS	RELATIVE LEVEL AT REFERENCE POSITION
1	1	1	0 dB
1	2 (parallel)	2	+6 dB
2	1 (per channel)	2	+6 dB
2	2 (parallel/CH)	4	+12 dB

TABLE 5. THE EFFECTS OF ADDING MORE SPEAKERS AND/OR MORE AMPLIFIER POWER TO INCREASE SOUND LEVEL.

High Pass filters protect speakers and reduce distortion

Where possible, the 2224/2244 built-in high pass filters should be switched into the circuit. They block "useless" low frequencies from being amplified and thereby avoid "wasting" power, so more power is available to reproduce the desired musical signal. Equally important, the low frequency driver (woofer) suspensions are not over-stressed trying to reproduce noise and spurious signals. Still another benefit is that Doppler distortion is reduced. This distortion, named for the Doppler effect, is a form of intermodulation induced when a speaker cone is simultaneously reproducing treble and bass notes. Given bass and treble note of the same volume level, physics dictates that the cone must move as much as 4 times further, which requires twice the power, for every octave lower in pitch. The travel of the cone as it reproduces the bass notes causes the treble notes that "ride on the cone" to move up and down in pitch, much like a train whistle changes pitch as the train passes by the listener. This Doppler distortion (intermodulation sidebands) is reduced by blocking unnecessary low frequencies at the amplifier input.

Speaker non-linearities may require the use of more speakers

At high power levels, speakers can require disproportionately more power to generate a given increase in sound pressure level. When voice coils are heated by high power inputs, their electrical resistance increases. The higher resistance prevents the amplifier's full power from being converted into a magnetic field, so speaker sensitivity drops. Better quality commercial speakers have larger voice coils that dissipate heat more effectively than poor quality speakers. Even better speakers, when operated with a continuous 100 watt power input, typically experience a drop in sensitivity of about 0.5 dB. In poor designs, the drop can be as much as 3 dB; in this case the heating has the same effect on overall system performance as throwing away half the amplifier power. In order to reduce the losses due to overheated voice coils, better speakers can be used, and more of them. (Even if more amplifier power is available, it may be better to use more speakers since heat-induced non-linearity is a form of distortion that power alone cannot overcome.)

It takes more than amplifier power and speaker sensitivity to generate enough level in a live sound system

It's relatively simple to raise the available SPL by adding speakers and amplifiers, but no amount of amplifiers and speakers will produce significantly higher *useable* sound levels in a sound system with "live" microphones if the system is already at the point of feedback. Raising the available gain (without causing feedback) requires careful attention to system layout, smooth system frequency response, and correct phasing (polarity) of mics and speakers.

Are the 120 dB sound pressure levels and 6 to 10 kilowatt power levels previously cited realistic?

Yes, basically they are, though practical values may be somewhat lower. In terms of decibels, the microphone's output might be 90 dB below 1 milliwatt given an input of normal speech. The mic preamp may add 60 to 90 dB of gain to produce "line level," and then the power amplifier adds another 30 to 40 dB to obtain a total 130 dB of gain – a power ratio of 10 TRILLION to one. These levels are illustrated in Figure 42.

The amount of power gain shown here is 130 dB. If for example, it were necessary to amplify normal speech at 2 feet from a microphone up to a level sufficient to satisfy a large stadium full of listeners, the gain required would approach 130 dB. However, this is an extreme amount of gain for an audio system to produce, and would probably be impractical due to noise and the likelihood of feedback. Instead, microphones are held closer to the talker, and/or the talker uses a higher voice level so that less overall gain is required. Certainly 50 to 90 dB of gain in an audio system is common.

Correlating amplifier power with program dynamics

The difference, in decibels, between a program's quietest and loudest passages is its *dynamic range.* Ideally, a sound system should be designed using amplifiers that are capable of matching, or slightly exceeding, the program material's dynamic range. The amplifier's dynamic range is defined as the difference in dB between the amplifier's maximum output (clipping point) and its "absolute noise floor" ("noise floor" is the residual amplifier hiss and noise under normal operating conditions with no signal present). While it can be influenced by many factors, the noise floor is proportional to the gain of the system.

If program dynamic range is *greater than* amplifier dynamic range, the loudest

WHAT IS DOPPLER EFFECT?

You've probably observed how the sound of a truck horn or a train whistle appears to increase in pitch as the vehicle approaches, then decrease in pitch as soon as the vehicle passes by. Actually, the horn or whistle itself does not change pitch, but the sound reaching you does! This is because the sound waves are "squeezed together," actually compressed in time, as the moving sound source comes toward the listener. Then the sound waves are "spread out," or expanded in time, as the sound source moves away. This always occurs when the listener is in motion relative to the sound source, or vice-versa, but the effect is only observable when the speed is a significant percentage of the speed of sound; a closing speed of 5 miles per hour is not sufficient to create an audible effect.

Sound travels through air at some 1125 feet per second (at sea level, 68 degrees Farenheit). Let's say a train continues to sound its whistle, and that sound travels outward in all directions at 1125 foot per second. As the train approaches at 15 miles an hour, the sound waves move toward the listener at an additional 44 feet per second, or nearly 4% of the speed of sound. The pitch of the music from the speaker will shift about 4%, or about a half step! If the train is moving at 60 miles per hour, the pitch changes nearly two steps. Similarly, when the train passes, its whistle now is moving away from the listener, so the sound waves arrive that much more slowly, lowering the pitch by a corresponding amount. This phenomenon is called "Doppler shift."

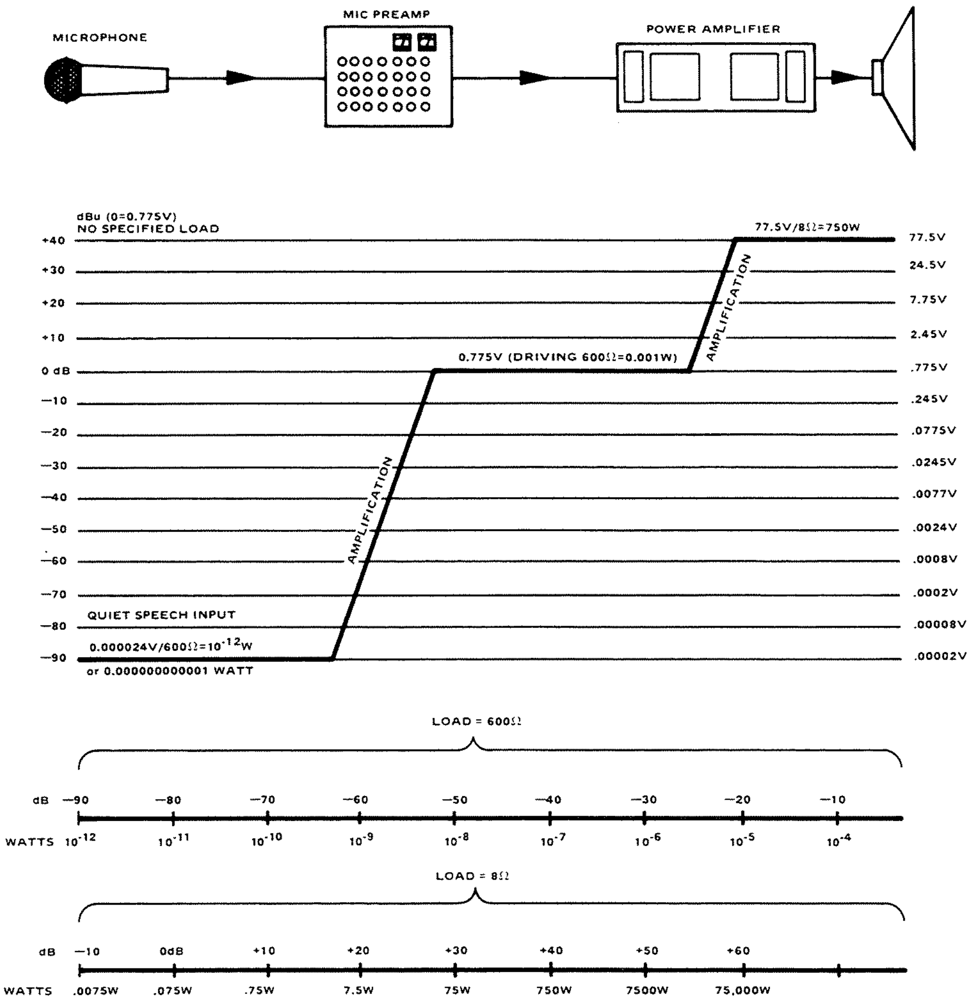


FIGURE 42. POWER LEVELS WITHIN THE AUDIO SYSTEM.

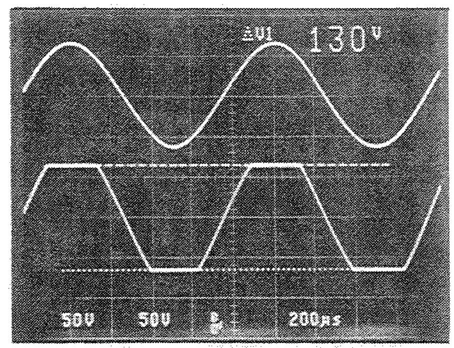


FIGURE 43. UNCLIPPED AND CLIPPED OUTPUT WAVEFORMS.

The upper trace shows channel A of the 2244 operating just within the limits of the amplifier's power supply, delivering 45.5 volts rms (258.78 watts) at 1 kHz into an 8 ohm resistive load. Lower trace is the channel B output with the input level increased 3 dB (41%). This overdrive (3 dB into clipping) causes the truncated sine wave. As you can see, both channels produce the same maximum peak-to-peak voltage, but the waveform in channel B cannot be reproduced accurately because the amplifier is attempting to produce more output voltage than its power supply can provide.

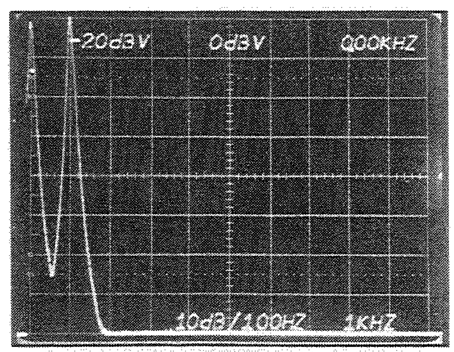


FIGURE 44. FREQUENCY SPECTRUM OF UNDISTORTED OUTPUT.

This spectrum analyzer photo shows the output of the unclipped (channel A) output from Figure 43. The vertical scale is 10 dB/division, and the frequency span is from DC to 10 kHz at 1 kHz/division. Note that the harmonic distortion is more than 80 dB below the fundamental 1 kHz frequency (below 0.01%). Actual THD measured 0.0043% with distortion analysis.

passages in the program material will exceed the amplifier's output capability (causing distortion), or the quietest passages may be "buried" in amplifier's residual hiss, depending on the relative levels and gain settings throughout the entire sound system. If the program's dynamic range is *about the same* as that of the amplifier, everything will work well. However, if the dynamic range of the program and amplifier are matched but the level is turned up to overcome ambient room noise or provide more amplification to very quiet sounds, then the louder passages in the program will require more output power than the amplifier can supply, leading to distortion. This is why we suggest that the amplifier have more dynamic range capability than the program material.

What is the distortion that occurs when the program level asks the amplifier to exceed its rated power? When you attempt to drive any amplifier beyond its full power (that is, the maximum clean power that the amplifier is capable of delivering to the load it is driving) it will exhibit an undesirable operating characteristic called "clipping." An oscilloscope photo of clipped and unclipped waveforms is illustrated in Figure 43.

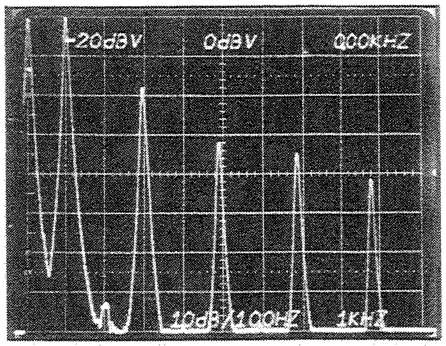


FIGURE 45. FREQUENCY SPECTRUM OF CLIPPED OUTPUT.

This spectrum analyzer photo shows the output of the clipped (channel B) output from Figure 43. Note that the harmonic spectrum is considerably different than that shown in Figure 44. Here the third harmonic is 18 dB below the fundamental, the fifth is 32 dB down, the seventh is 34 dB down, and the ninth 42 dB down. The overall rms load voltage increased from 45.5 V to 51.6 V rms, and the power from 258 watts to 332 watts (even though peak-to-peak voltage is the same as the unclipped waveform). Calculated %THD from the first 9 harmonics is 13.01% (the square root of the sum of the squares of each harmonic). Overall THD to 300 kHz measured 13.1%. Of significance is the fact that the second harmonic is down 74 dB (0.05%), clearly showing the excellent balance and symmetry of the amplifier because a theoretically clipped sine wave produces no even order harmonics! In the case of a clipped *single frequency,* the increase in power occurs due to the addition of the higher order harmonics.

More powerful amplifiers can be safer for speakers!

A given speaker, when operated at the same peak sound level, may last longer when driven by a higher power amplifier. Contrary to apparent logic, this tendency actually does make sense because higher power amplifiers will supply more "clean" power before the onset of clipping, and clipping can be disastrous to speakers. Clipping, which implies exceeding the full power operation of a given amplifier, causes rapid heating of speaker voice coils. The heating, coupled with the high displacement of the speaker's moving assembly away from the normal centered position and toward its mechanical limits due to the high power levels, is a catastrophic event for most drivers, including woofers. High frequency compression drivers and tweeters, however, are perhaps more susceptible to clipping. Clipping produces harmonics, sounds not present in the original signal, and at higher frequencies. This abundant, clipping-induced high frequency energy tends to exceed the normal design limits of the high frequency drivers, which are not expected to be fed such high levels of continuous power. The result is decreased service life or instant burn out.

NOTE There are limits to the benefits of larger amplifiers. A larger amplifier, when driven into clipping, will certainly destroy speakers faster than a smaller amplifier driven into clipping. Also, a single loud transient (dropped microphone, for example) can do more damage to driver suspensions if more amplifier power is available, which is why high pass filters should be used whenever practical. In any sound system, there is no substitute for careful operation.

Although clipping is sometimes desirable in musical instrument amplifiers when an "overdriven" or "fuzz" sound is desired such as with rock electric guitar styles, it is NEVER desirable in amplifiers used to drive sound reinforcement speakers.

A related, but not indentical problem occurs when V-I (voltage-current) limiting is triggered by protection circuitry within the amplifier. The protection mode of the amplifier can cause high frequency (around 5 kHz) "hash" noise to be superimposed on a high power, lower frequency waveform. In a system where an amplifier handles full-spectrum sound and is connected to a passive, high level crossover network, the instantaneous load on the amplifier can be far greater than the rated speaker impedance; certain assymetrical waveforms can cause all the drivers to simultaneously load the amplifier, essentially "in parallel" such that an 8-ohm system temporarily looks like a 2-ohm system. This condition can trigger many an amplifier's V-I limiting, even though the amplifier is supposedly large enough for the demands placed on it by a theoretically "well behaved" load. (The Fender 2224 and 2244 are built so they can handle such loads without problems.)

Sometimes occasional clipping or protection circuit V-I limiting can be tolerated in order to avoid the cost of doubling or quadrupling the power amplifier capacity of a system. Remember, however, that due to the spectral imbalance which occurs, an amplifier which is regularly driven into clipping or whose protection circuits are often triggered can burn out speakers whose "sine wave" or "program" power rating is two or three times the rated output of the amplifier.

If more volume is needed and the amplifiers are already operating at their maximum level before clipping, or if clipping is becoming excessive at a given sound level, then more or larger amplifiers will be required. Operating a larger amplifier so it produces the same sound level as the smaller one may not use more actual voltage, but the extra power available for peaks will prevent clipping when all the other operating conditions stay the same. This is the meaning of "headroom."

Alternatives to using more amplifier power

One alternative to raising amplifier power to avoid clipping is to reduce amplifier output, which reduces the sound level. To bring the sound level back up where you want it, the speakers must then be replaced with higher efficiency (higher sensitivity) models – often a costly proposition. If you are already using the most efficient speakers available, you will either have to accept lower sound levels when you turn down amplifier gain to prevent clipping, or else increase the number of speakers and amplifiers to regain some of the sound level.

Another alternative to raising total amplifier power is to set up a bi-amplified or tri-amplified speaker system. Such systems can utilize the same total amplifier power (or less) to deliver more SPL before objectionable clipping. Even when the low frequency amplifiers in such multi-amped systems do clip, they cannot damage sensitive mid and high frequency compression drivers.

Amplifier power rating methods

There are a number of different ratings or specifications that describe amplifier power. It has been said that "a test can be devised to produce any desired specifications," and under lab conditions, this is nearly true. A number of years ago, the United States Federal Trade Commission (FTC) established standard methods of rating amplifiers sold in the consumer (hi-fi) market to avoid the abuse of non-technical consumers by unscrupulous amplifier manufacturers who cited misleading or meaningless tests to persuade consumers that their amplifiers were more powerful than others. However, the FTC ratings themselves are simplistic and do not necessarily describe the "real world" performance of an amplifier. Since professional sound system designers can understand and make better use of several types of power ratings, these other ratings continue to be used. In fact, amplifiers intended for professional use are not required to carry FTC ratings (though some do, anyway). As a result, you may find any of the following power ratings listed for a given amplifier:

PP or PEAK POWER. This is a measurement of the maximum UNDISTORTED power an amplifier can deliver to its load. Usually the load specified in such measurements is a non-reactive, laboratory load resistor (an ideal load). The test is usually done with with one channel operating, and with a 1000 Hz sine wave input signal (i.e., a pure tone in the middle of the audio frequency spectrum). Peak power is useful to determine the ability of the amplifier to reproduce peaks and transient signals that occur in musical program material. Since peak power is calculated using the peak-to-peak (or zero-to-peak) output voltage instead of the RMS output voltage of the amplifier, the peak rating can be several times higher than the Continuous Average (RMS) power even if there is no difference between the maximum and continuous output power levels.

CONTINUOUS AVERAGE (RMS) POWER. "Continuous Average Power" used to be known as "RMS Power." "RMS" is an abbreviation for "root mean square," a mathematical term used to denote the voltage or current value of an AC electrical signal that can create the same HEATING as an equivalent DC voltage or current in a resistor. (The RMS value of a sine wave signal is the peak value multiplied by half the square root of two; this relationship may be considerably different for more complex waveforms.) Continuous Average Power ratings are used in amplifier testing as a convenience; test equipment measures RMS values of AC signals. Mathematically, there is no such thing as "RMS power." If RMS current is multiplied by RMS voltage, the result is NOT RMS power. Besides, power is generally calculated using the RMS voltage and a given load impedance; such calculations yield "average" power, not "RMS power." "Average sine wave power" is the proper way to express "RMS power."

IHF POWER. The IHF (Institute of High Fidelity), a trade organization, defined a power rating method that is essentially the same as "Continuous Average Power" or "RMS Power." They also defined a secondary rating known as "Dynamic Headroom." The Dynamic Headroom of an amplifier is based on a 20 millisecond, 1 kHz burst of higher power within a half second after operating for 480 milliseconds at the nominal level (approximately the same as the old "music power" method). Instead of specifying a value in watts, however, the IHF calculates the ratio of that midband peak burst power output to the continuous average power output and states the result in dB of Dynamic Headroom. Given equal continuous average power ratings, the amplifier with a larger dynamic headroom specification *may* sound louder – if the difference is sufficient. About 3 dB of difference is needed in order to hear a clearly audible advantage. For example, an amplifier rated at 100 watts average power, and having 3 dB of dynamic headroom, would probably have carried an old-style music power rating of 200 watts.

Although professional power amplifiers are not subject to FTC rules concerning specifications and testing methods, the 2224 and 2244 are tested in a very rigorous way that approximates the real-world use the amplifier will get. Their specifications easily meet or surpass FTC requirements for consumer amplifiers. The 2244, for example, will deliver 28 amperes per channel into a 2 ohm load for a short period. (A sophisticated protection circuit senses a very low impedance load, assumes a short circuit, and reduces the power output accordingly.) The amplifier is thus capable of at least 784 watts per channel for short periods.

How loads can affect amplifier output

At any power up to the maximum rated output the 2224 or 2244 amplifier behaves like a "voltage source," that is, a source of steady voltage that is unaffected by the load impedance – what engineers like to call a "stiff" output. The voltage from an ideal "voltage source" remains constant whether it is driving into a load or not, and regardless of the load's impedance (as long as it is 4 ohms or higher). Therefore, as the load impedance decreases, the power output increases because the voltage stays the same; If a 4-ohm load is substituted for an 8-ohm load, the power from the amplifier will double as long as nothing else is changed. An Ohm's Law-based equation shows why this happens:

WATTS = VOLTS SQUARED DIVIDED BY OHMS, or $P = E^2/Z$

The 2224/2244 protection circuits automatically limit current from the amplifier's power supply if the load impedance drops below a safe limit (under 2 ohms). Remember that speakers do not behave like laboratory test resistors. Elsewhere in this manual we describe how speaker voice coils change impedance when high power signals are applied to them. More significantly, voice coil impedance changes with the FREQUENCY of the signal applied (i.e., they are REACTIVE loads). This means that speakers, unlike resistors, will draw different amounts of power at different frequencies.

How can a speaker "store" power? At some frequencies, the speaker stores electrical energy by converting it to mechanical energy, then reconverting the mechanical energy to electrical energy. This process takes a finite amount of time, and the delayed (stored) electrical energy returned to the circuit must flow somewhere. The "somewhere" is back into the amplifier's output stage. This energy flow can overheat the amplifier, trigger its protection circuits, cause raspy sound, and so forth... unless the amplifier is designed to handle such "reactive" loads. This means the amplifier must have many output transistors and good heat sinks, as do these Fender amps.

At the lowest point on a typical "impedance curve," given a steady-state audio signal, the impedance value is just slightly higher than the static DC RESISTANCE of the voice coil. The maximum impedance, however, may be as much as 40 times the speaker's rated. "Raw frame" cone type drivers will exhibit different impedance curves than the same drivers in an enclosure, but even when installed in well-designed speaker enclosures, the speaker's peak impedance can be many times higher than its nominal rated impedance.

NOTE We specified a "steady-state" audio signal, such as a swept sine wave, because the actual load impedance a passive, high-level crossover network presents to an amplifier can be considerably lower than its rated impedance. This occurs with certain pulsed or assymetrical waveforms (which are not *that uncommon), wherein the charging current for the reactive speaker/-crossover loads add simultaneously as though several drivers were connected in parallel across the amplifier. This little-known behavior, which further taxes audio power amplifiers, was documented in a paper presented to the Audio Engineering Society's 1983 European convention. The paper is titled "Input Current Requirements of High-Quality Loudspeaker Systems," authored by Ilpo Martikainen and Ari Varla of Genelec Oy (Finland) and Matti Otala of the Technical Research Centre of Finland. The paper is Preprint number 1987 (D7), available from the Audio Engineering Society, 60 East 42nd Street, New York, NY 10165.*

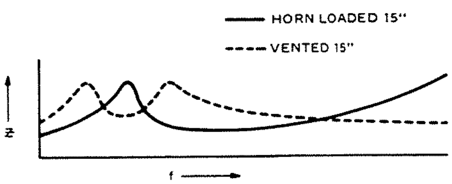


FIGURE 46. TYPICAL LOUDSPEAKER IMPEDANCE CURVE.

This graph shows an impedance curve for a typical high-efficiency speaker: impedance on the vertical scale versus frequency on the horizontal scale.

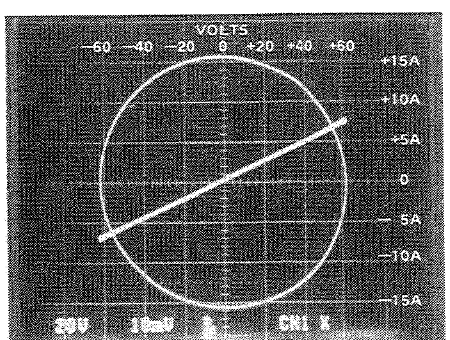


FIGURE 47. EFFECTS OF RESISTIVE AND REACTIVE LOADS ON AMPLIFIER OUTPUT VOLTAGE AND CURRENT.

This photo is a double exposure of an oscilloscope with an 8 ohm resistive load (26.5 degree line) and a 4 ohm reactive load (circle). This shows the load voltage and current relationships. The horizontal scale is in volts (20 V/div) and the vertical scale is calibrated to a current probe (5 A/div). The 8 ohm resistive load trace (straight line) shows that when the amplifier output is +20 volts, the current is 5 amps. Similarly, a −20 volt output results in a −5 amp output. The reactive load (circle) includes 26.5 uF capacitance and has an impedance of 4 ohms at 1500 Hz. Of special significance is the fact that the voltage and current are NOT in phase with each other, as in the resistive case. When the amplifier output voltage is at zero, the current has two maximum values of plus and minus 15.5 amps! With a reactive load, all of the output power is dissipated in the amplifier, not in the load. In this example, the 2244 is dissipating over 660 watts.

Many of the better quality, commercially available speaker systems rated at 8 ohms have a minimum impedance of about 6 ohms, those rated at 4 ohms drop to about 3 ohms minimum, and 16-ohm speakers will exhibit about a 12 ohm minimum. Speaker manufacturers derive impedance ratings by using the average value of the impedance curve within the frequency range in which the speaker is intended to operate. Manufacturers usually specify this frequency range for individual drivers. When a packaged, multi-way speaker system's impedance is given, it is often the nominal impedance at the crossover network input. AES standards now suggest listing the minimum impedance as well.

Ohm's law and calculating power

The first law of electrical circuits, "OHM'S LAW," describes the relationship of electricity doing work in a circuit. Voltage by itself is only potential, it does nothing unless given a load to work into. There are THREE BASIC UNITS necessary to form a working electrical circuit, besides a continuous pathway for electricity to flow. These three units are VOLTS, OHMS and AMPERES. In order to produce a working circuit, we need only to feed some voltage into a continuous resistance (ohms), which produces a current flow (amps). This can be done as simply as connecting a piece of wire from one end of a battery to the other end, or some other items may be inserted into that simple circuit such as a lamp or a motor.

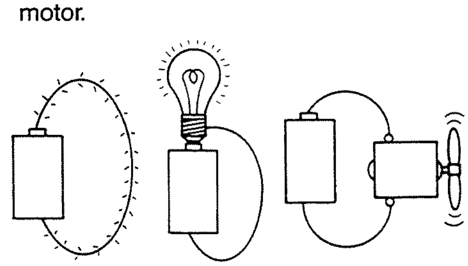


FIGURE 48. CURRENT FLOW FROM AMPLIFIER TO SPEAKER IS MUCH LIKE CURRENT FLOW FROM A BATTERY TO A HEATING ELEMENT (WIRE), A LIGHT BULB OR AN ELECTRIC MOTOR.

A loudspeaker is basically a motor whose motion is directed in a straight line forward and backward. By EIA standards, connecting the "+" terminal of a battery to the "+," red or marked terminal of a loudspeaker and the "−" terminal of the battery to the "−," black or unmarked terminal of the speaker, usually pushes the cone forward (away from the magnet of the cone type loudspeaker, or toward the magnet of a compression driver). An exception is JBL, whose speakers all move in the opposite direction. Reversing the polarity of the leads reverses the direction of cone travel. This can be easily seen with woofers whose cone suspensions aren't too stiff.

Batteries are a source of DC (direct current). Ohm's Law for DC circuits is simple and straightforward, using only simple arithmetic:

VOLTS = AMPS x OHMS
OHMS = VOLTS/AMPS
AMPS = VOLTS/OHMS

Some derivatives of Ohm's Law can be used to find any missing quantity in a circuit as long as two other factors are known. Here are two equivalent charts, each showing the various possible solutions derived from Ohm's Law quantities.

E=IR	**R=E/I**	**I=E/R**	**W=EI**
E=W/I	**R=E²/W**	**I=W/E**	**W=E²/R**
E=(WR)$^{1/2}$	**R=W/I²**	**I=(W/R)$^{1/2}$**	**W=I²R**

E=volts, R=ohms, I=amperes and W=watts

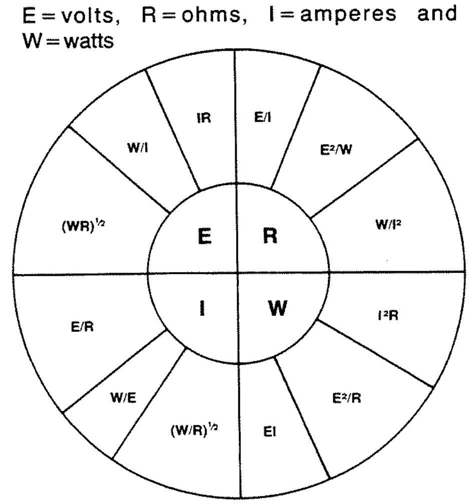


FIGURE 49. OHM'S LAW AND ITS DERIVATIVES.

Power (watts) is not, strictly speaking, part of Ohm's Law, though it is related. Ohm's Law is used to determine power

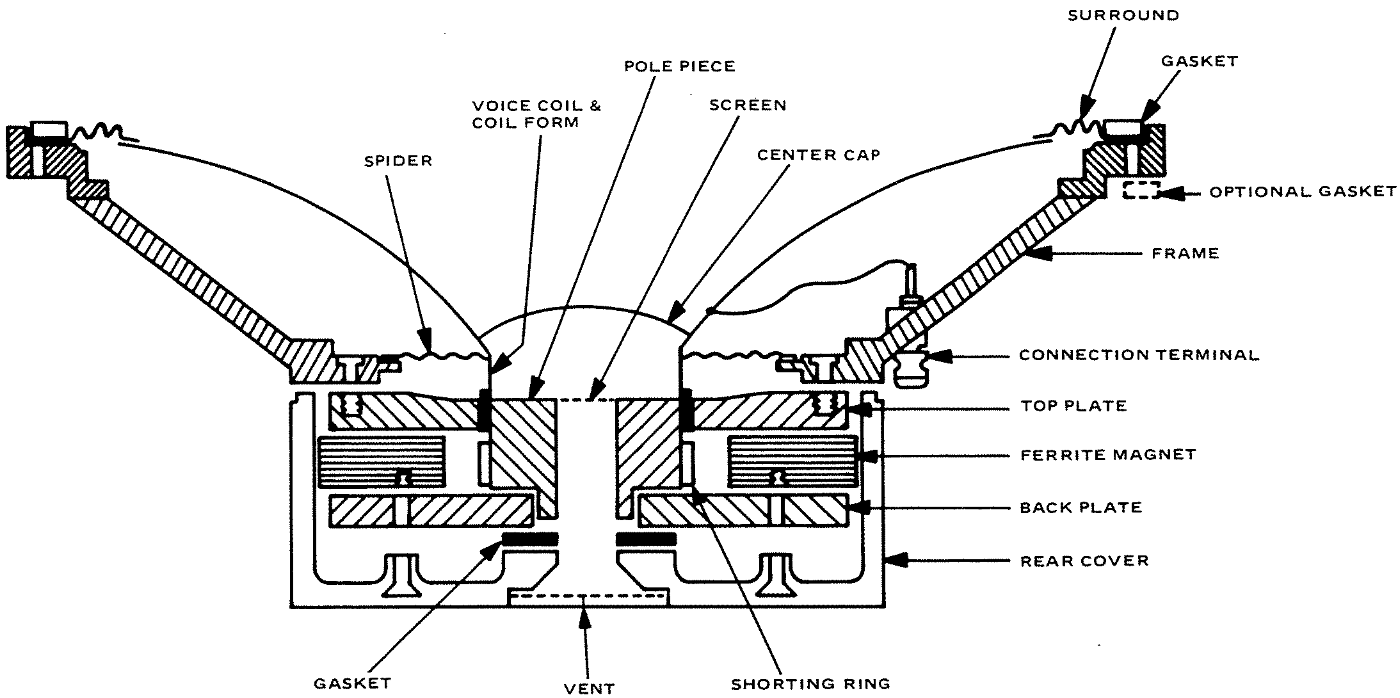


FIGURE 50. CONE TYPE SPEAKER CROSS-SECTION.

when the only information about what's going on in a circuit does not include power, but does include at least two of the other necessary quantities.

Loudspeaker cone motion is dependent on the current flow through the coil, the length of the conductor in the magnetic field, and the strength of the magnetic field supplied by the magnet structure which forms an air gap in which the coil moves.

The more current and magnetic flux interaction, the more cone motion will be produced at a given frequency. According to Ohm's Law, the more voltage fed in, the more current, and so the POWER also increases.

AC impedance versus DC resistance

Impedance is usually rated in ohms, but there are other electrical factors included in impedance that contribute to its complex nature. In our battery example, we see Ohm's Law at work in a DC circuit. Sound however, is made by causing a loudspeaker's cone to move very rapidly – mimicing the vibration of the original sound source being amplified and reproduced. The cone may, however, vibrate the air with greater energy to create more powerful sound waves. An audio amplifier is, in fact, a power source which can be modulated with enough speed and precision to feed the loudspeaker an electrical signal that produces those vibrations.

AC, or alternating current, changes its polarity (direction of current flow) at the same rate as the sound vibrations that formed the original electrical signal at the microphone, and in so doing, calls upon the amplifier to reverse its electrical output very quickly, depending on the frequency of the signal. This also means the voltage from the amplifier's output terminals will reverse direction from positive to negative or vice-versa, AND PASS THROUGH ZERO VOLTS on its way. In fact, in just one simple sine wave cycle, voltage (and current) hit one positive peak, one negative peak, and three zero-volt points. The mechanical analog produced by the loudspeaker fed this signal is: starting at rest, the cone pushes forward, returns to the rest point as it passes zero, pushes backward and returns to the starting position where it comes to rest again.

Some of the other electrical factors in impedance include "inductance" (which is a sort of electrical "springiness"), capacitance (an electrical storage characteristic), and a combination of various phase angles or voltage and current "vectors." While we won't discuss these factors in detail, they do affect an amplifier's ability to drive a complex load such as a speaker, and they must be considered when designing protection circuitry that works properly in "the real world."

$E=IZ$	$Z=E/I$	$I=E/Z$	$P=ER$
$E=P/I\cos\phi$	$Z=E^2\cos\phi/P$	$I=P/E\cos\phi$	$P=EI\cos\phi$
$E=(PZ/\cos\phi)^{1/2}$	$Z=P/(I\cos\phi)^2$	$I=(P/Z\cos\phi)^{1/2}$	$P=E^2\cos\phi/Z$

Where ϕ is the current/voltage phase lag of the AC signal

Loudspeakers are built using actual springs. The springs take the form of the compliance surrounding the outside circumference of the cone, and the spider at the inside circumference. These springs serve to resist the force of the loudspeaker's motor (the coil and magnet) and return the cone to center rest position after it is driven forward or back by a signal from the amplifier. This is called "restoring force." A loudspeaker is also capable of storing an electrical charge, as mentioned previously; now we'll go into more detail as to how it does so. When the cone/voice coil assembly is driven away from its center rest position, its inertia represents stored energy imparted to it by the signal from the immediate past. Although the cone would naturally tend to return to rest instantly, propelled by its "springs," it is limited by the ability of the springs to accelerate the mass of the assembly. Then, after the cone mass has been accelerated, it requires outside force to "put on the brakes." This kind of "tail wagging the dog" behavior of the loudspeaker assures that the voltage and current fed into the device will be out of phase by some degree most of the time. In fact, there are only two conditions where voltage and current will be in phase: (1) when the loudspeaker is driven by a signal whose frequency falls at the loudspeaker's minimum impedance or (2) at loudspeaker's maximum impedance.

Remember that any loudspeaker's "RATED IMPEDANCE" is really theoretical; in actual operation, it exhibits that impedance only rarely. Most amplifiers like to work into resistive loads (like simple resistors) which are free of inductance and capacitance. The voltage and current are always in phase in a simple resistor. This means all the power the amplifier is capable of delivering out of its output terminals can be dissipated in the resistor and not in the amplifier's output transistors, as can happen when the amplifier is driving reactive loads. Fender amplifiers are designed to work well with reactive loads; they are able to "put on the brakes" and damp cone motion without overheating or premature triggering of protection circuits.

Amplifier damping and speaker performance

Moving any mass requires force to overcome inertia. In the case of low-frequency loudspeakers, the cone/voice coil assemblies have lots of mass – up to around 200 grams (close to a half pound in the Earth's gravity). Tight control of massive loudspeaker cone assemblies not only requires power from an amplifier to get things moving, but also some sort of built in braking ability to overcome unwanted leftover motion. This ability is called "damping," and here's how it works: the loudspeaker is set in motion by a signal from the amplifier. If the loudspeaker were called on to stop dead (by disconnecting the amplifier) while the cone was in motion, the "springs" in the loudspeaker's cone suspension would cause the cone to bounce. An unwanted overshoot of the resting position and a "ringing" would occur. What is needed to avoid the overshoot is a short circuit across the loudspeaker's terminals at the instant the signal from the amplifier is removed. In essence, this is what happens in the case of an amplifier with a high "damping factor." Shorting the coil of the loudspeaker causes the coil to generate a counter-force – a force in the opposite direction of the cone motion – due to the motion of the coil through the magnetic field of the magnet gap. This produces a braking effect from a phenomenon called "back EMF."

Damping factor is the ratio derived by dividing the amplifier's rated load impedance by the actual amplifier output impedance, in ohms. This implies that an amplifier with a damping factor of 100 and a rated load impedance of 8 ohms will have an actual output impedance of 0.08 ohms. The smaller the amplifier's output impedance, the more that amplifier will "look like" a dead short from the loudspeaker's point of view when no signal is present (when the amplifier's output voltage is zero).

Damping factor is said to be one of the specifications that produce the characteristic "tight" or "punchy" sound from bass and kick drum, although this may be somewhat exaggerated in importance. For example, damping factor can be radically altered by adding any resistance to the circuit made by the amplifier and loudspeaker. If as in the above example, the amplifier has an actual output impedance of 0.08 ohms, and a speaker cable of the same value, 0.08 ohms, is used to connect an "ideal" load (like an 8-ohm resistor), then the damping factor is effectively cut in half. Of course, amplifier output impedance changes with changing frequency, which is why damping factor is often specified below a certain frequency, or at several frequencies. (In fact, damping is based more on resistance than impedance, and the voice coil's internal resistance can become a limiting factor. Even with a short circuit across the speaker terminals, the voice coil resistance limits current flow, and hence limits damping.) If damping factor is to be kept high, large wire (small gauge NUMBER) speaker cables are needed. Were it not for the weight, cost and stiffness, it might be a good idea to use automotive battery cable instead of speaker wire.

The damping factor of an amplifier is usually reduced drastically if the amplifier is driven into clipping...exactly when it is needed the most! If a Fender amplifier is driven to clipping, the behavior is well-controlled, with only a slight reduction in output impedance so that damping remains adequate.

NOTE The specified damping factor for Fender amplifiers is based on the actual source impedance of the amplifier at its output terminals. This includes a 0.03 ohm resistive component from the output protection relay. As a result, the damping factor is specified to be 200. Internally, ahead of the relay, the damping factor is in excess of 1000. This is not particularly significant, since 200 is a respectable figure, but you should know that some manufacturers quote their amplifiers' internal source impedance and ignore the degradation that occurs in relays, solder joints, internal cabling and the output connectors.

In most loudspeakers (woofers) which are subject to damping from high quality amplifiers, the voice coil overhangs the magnetic air gap of the magnet structure. The portion of the voice coil wire outside the influence of the magnet's magnetic field becomes simply added resistance, even though it's still part of the overall speaker impedance, and this can serve to swamp out the effects of good damping when there is enough coil overhang. As an example, let's say a particular loudspeaker has a magnet top plate 7 mm thick, and a voice coil 14 mm long. There would be 50% of the coil outside the gap at any given time, some of which will be located in a weak magnetic field "fringe" area, so in an 8 ohm loudspeaker, nearly 4 ohms might be added as a series resistance, and this limits the effect of damping from the amplifier to the point where wire size increases reach a point of no return for the additional investment. Maximum power transfer between amplifier and speaker is the important thing at this point, and boosting wire size beyond a certain point provides no additional benefit. Overhanging coils are not necessarily bad, however, since this design contributes to speaker linearity (a nearly constant amount of voice coil remains in the primary flux field of the magnet gap).

In the typical magnet structure of larger loudspeakers there is a phenomenon called "fringing" that exhibits a non-linear influence on the voice coil. The magnetic field follows "lines" of force, or magnetic flux, straight across the magnetic air gap of the magnet structure. Above and below the gap, the flux lines bend to form weaker fringes whose strength decreases as the inverse square of the distance from the gap.

The familiar pattern of magnetic fringes is much the same as that for the

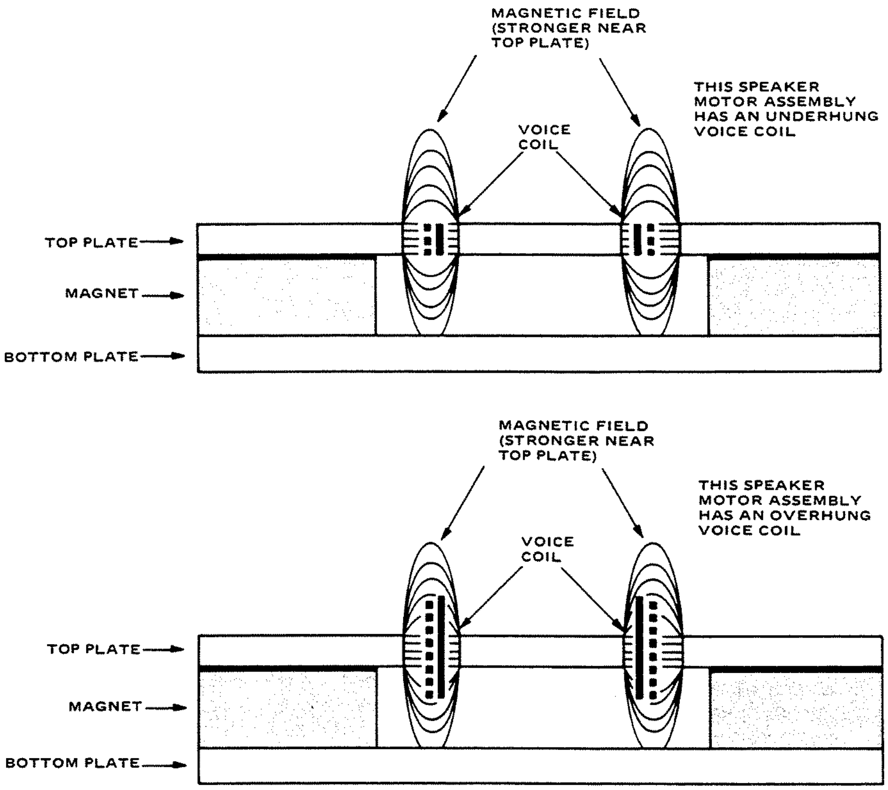


FIGURE 51. UNDERHUNG VERSUS OVERHUNG VOICE COILS.

classic school room demonstration of filings and bar magnet, sprinkling filings on a paper placed over the magnet reveals the shape of the magnetic field surrounding the magnet. The loudspeaker's ultimate behavior will depend on the interaction of the current in the voice coil and the magnetic field around the coil. Getting the current into the coil is the job of the amplifier. The amplifier's ability to deliver its maximum current to the coil depends on the connecting cable, the internal impedance of the amplifier itself, and on numerous other amplifier characteristics which may ultimately be the most significant factors. You might tend to think of an amplifier/loudspeaker circuit in its simplest terms.

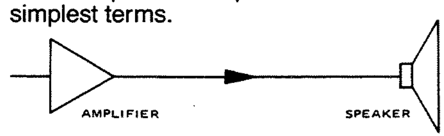


FIGURE 53. BASIC CIRCUIT MODEL FOR AMPLIFIER/SPEAKER SYSTEM.

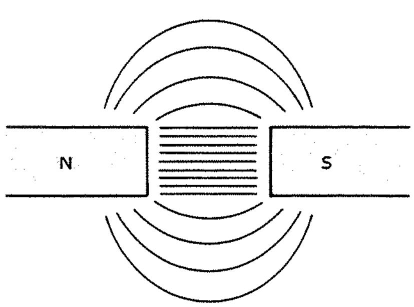


FIGURE 52. DECREASING MAGNETIC FIELD BEYOND THE VOICE COIL GAP OF A SPEAKER MAGNETIC ASSEMBLY.

However the actual performance is more complicated and can be represented by the circuit in Figure 54.

The various terms in this circuit include:

e_g = Open-circuit output voltage of the amplifier.

R_g = Output resistance of the amplifier

R_E = DC resistance of the voice coil.

C_{MES} = Electrical capacitance due to cone assembly mass.

L_{CES} = Electrical inductance due to cone assembly compliance(springs).

R_{AS} = Electrical resistance due to cone assembly suspension losses.

There is a major additional source of electrical resistance due to the loading of the air on the cone of the loudspeaker. If you think the loudspeaker load looks complicated, take a look at just a few of the components of a high-level passive crossover such as you're likely to find inside the cabinet of a speaker system, illustrated schematically in Figure 55.

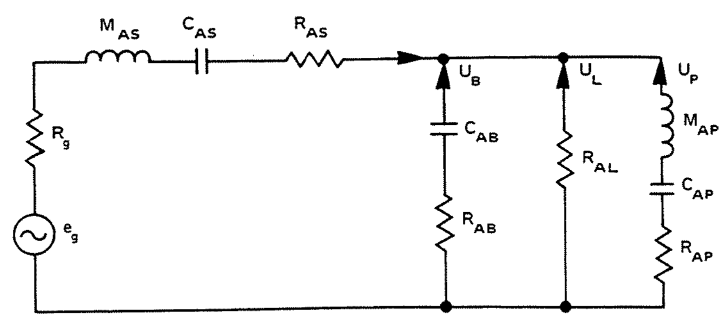


FIGURE 54. COMPLEX CIRCUIT MODEL OF AMPLIFIER/SPEAKER SYSTEM.

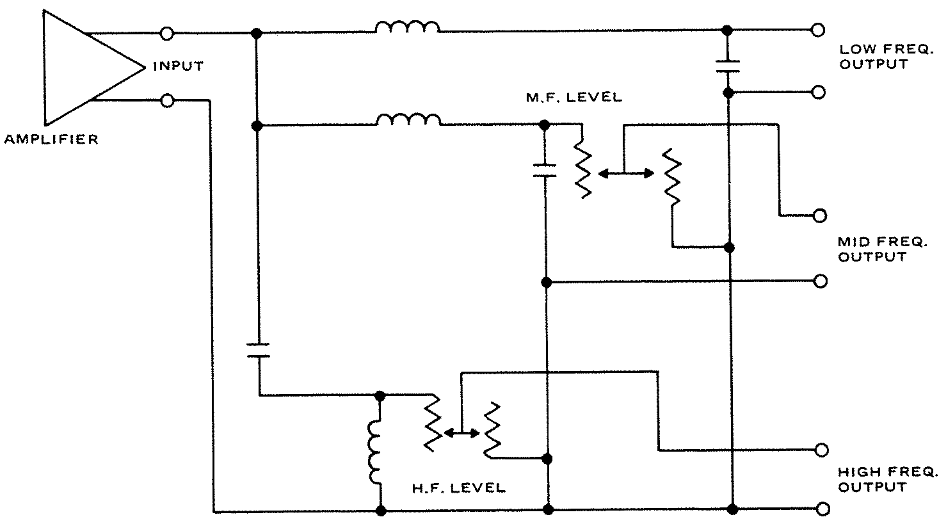


FIGURE 55. CIRCUIT DIAGRAM OF TYPICAL 3-WAY, HIGH LEVEL, PASSIVE CROSSOVER IN A SPEAKER SYSTEM.

Compare the components of the speaker system crossover to to the circuit of a typical laboratory test setup, as shown in Figure 56.

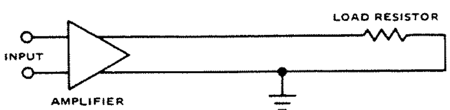


FIGURE 56. CIRCUIT DIAGRAM OF SIMPLE LABORATORY LOUDSPEAKER TEST/EVALUATION SYSTEM.

An amplifier should be designed to handle complex loads and ignore the peculiarities of compliance "springs" and crossover components. It's relatively easy to build an amplifier that performs well under laboratory conditions – yielding excellent specifications – and have it fail miserably with loudspeaker loads. Fender amplifiers work properly in the laboratory and in the field.

Back EMF

"Back" EMF is an abbreviation for back "electromotive force," a counteracting force. Back EMF is the electricity generated by the restoring force of the loudspeaker's compliance, the air load, and other factors while they are forcing the cone/coil assembly to move back to its rest position after displacement by the amplifier. The voice coil is not only the motion-producing "armature" of a linear-motion motor, it is also the armature of an electrical generator. In fact, if you connect a loudspeaker to the INPUT of an amplifier, it will function as a microphone. This ability of the loudspeaker to act as a generator and can also cause back EMF. Back EMF normally occurs whenever the loudspeaker cone is moving and the inertia/suspension cause it to keep moving beyond that motion which is "dictated" by the input signal. The extra motion generates the back EMF. Amplifiers with high damping and low output impedance provide a "path" for back EMF to short to ground, almost as if a short circuit is momentarily introduced to the loudspeaker's terminals. The result of this, as we've already mentioned, is a "braking" effect on the cone's motion and consequently a reduction of cone overshoot. It is also important that the amplifier's outputs not be affected by the introduction of back EMF into them. When solid-state amplifier designs first began to appear in the 1960's, back EMF was occasionally to blame for destroying output transistors – which were much less robust than today's transistors.

In addition to the use of modern high-power transistors like those used in the 2224/2244, protection from the effects of back EMF and accidental shorts on the outputs of the amplifier can be improved by paralleling output transistors to increase the amount of current that can safely be handled by the output circuits. The 2244 uses twelve output transistors in the output circuit of each of the two channels of the amplifier. Each side uses six NPN and six PNP transistors in a true complimentary symmetry output circuit. The transistors are very conservatively rated at 200 watts, or a maximum of either 10 amps or 200 volts each. Multiplying 200 watts by 12 transistors gives the amplifier a total (theoretical) output device capability of 4,800 watts at 25 degrees Centigrade (77 degrees Farenheit) case temperature.

Types of distortion

HARMONIC DISTORTION is the appearance of whole-number multiples of the input signal's frequency at the amplifier output. Even-order harmonics (2, 4, 6 and so on times the input frequency) tend to be less harsh-sounding than odd-order harmonics (3, 5, 7 and so on times the input frequency). Sometimes individual harmonics are specified, i.e., "3% 2nd harmonic distortion." More often, however, all the harmonic distortion is measured at once, without breaking out the individual harmonics' contribution to the value. This number is called TOTAL HARMONIC DISTORTION, or THD.

Specifications for distortion are given in percent (%). These percent figures refer to the percentage of the output signal which was not present in the original input signal – or how much distortion is added by the amplifier under test. Distortion EQUAL in level to a signal is said to be 100% distortion, distortion 20 dB lower in level than the signal is said to be 10% distortion (-20 dB is equal to

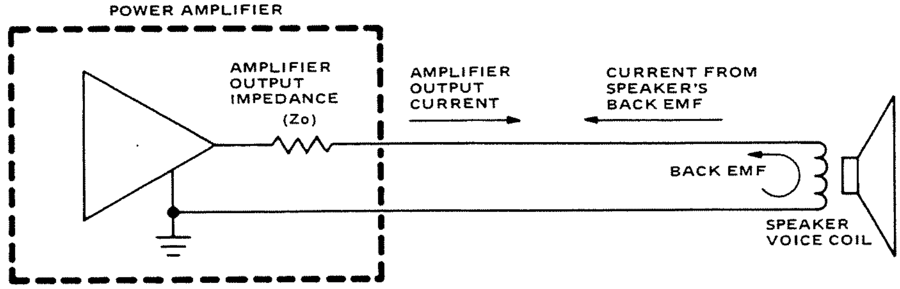


FIGURE 57. BACK EMF IN AN AMPLIFIER/ SPEAKER SYSTEM.

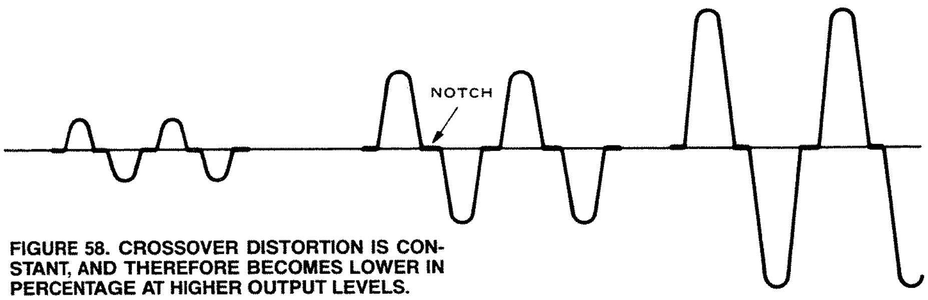


FIGURE 58. CROSSOVER DISTORTION IS CONSTANT, AND THEREFORE BECOMES LOWER IN PERCENTAGE AT HIGHER OUTPUT LEVELS.

one-tenth in voltage). Here's a reference between percentage and decibels for the voltage output of an amplifier:

100% = 0 dB (unity)
10% = −20 dB
1% = −40 dB
.1% = −60 dB
.01% = −80 dB

By specifying a THD rating for the 2224 and 2244 of 0.05% or less, we are indicating that the harmonic distortion products added by the amplifier are always at least 66 dB below the output signal. Minus 66 dB is 1/2,000th of the total output of the amplifier, and this holds true at any output level up to the rated maximum output. However, the typical THD is 0.006%, which is 0.026 watts. This is 1/17,000 of the 2244's rated output.

CROSSOVER DISTORTION has nothing to do with crossover networks. Instead, it refers to a common problem in improperly biased amplifiers. The problem occurs as the amplifier's output voltage passes or "crosses over" zero volts. If the bias isn't right in the amplifier's circuits, the output waveform is "squeezed" together while the transistors are waiting to "turn on" (bias keeps the transistors barely on at zero output). One interesting thing about crossover distortion is that it stays constant whether the amplifier output is large or small. The audible effect DECREASES as the volume goes up. Most other forms of distortion increase right along with increased amplifier output.

Crossover distortion has a very high peak value, while the average value is quite low. For example, 0.01% crossover distortion, as measured on a THD meter, can be audible. In a listening evaluation with a signal having typical harmonic distortion of 0.01% THD, the 0.01% distortion is inaudible. The reason two similar-measuring distortions sound different is that we tend to hear the peak value, not the average (RMS) values derived by the measuring equipment.

Crossover distortion is a sign of a poorly designed or defective amplifier. However, since it is a form of harmonic distortion, whatever crossover distortion is present is included into the amplifier's specification for total harmonic distortion. The irony of this is that even though the crossover distortion may be very obnoxious at low output levels, specs for THD are usually run at or near full power, and so an excessive amount of crossover distortion might not be a significant part of the THD figure in the specs of a fairly powerful amplifier. If you were to plot the THD versus power output for the 2224/2244, you would see that the distortion is ALWAYS LOW.

INTERMODULATION or "I.M." distortion is not a type of harmonic distortion. It is generated by the interaction of signals being amplified. Music is made up of complex signals – many frequencies of sounds being produced simultaneously. An artificial I.M. test signal composed of two widely spaced frequencies designed to simulate worst-case music intermodulation is used to drive an amplifier under test to produce intermodulation components which are "beats" or sum/difference signals not present in the original input signal. The amount of unwanted intermodulation components added by the amplifier is measured and expressed as a percent of the output signal in the same way harmonic distortion is specified. (Unavoidably, some harmonic distortion is also created by the test, so the I.M. figures measured are never "pure" intermodulation, but really a mixture of harmonic and intermodulation distortion.) Since I.M. distortion is NOT harmonically related to the input signals (except for testing anomalies, as noted above) but may be tones, sounds or pitches in a different key and register altogether, I.M. distortion is often more audible than harmonic distortion. I.M often is manifest as a gritty sound that can cause listener fatigue. The 2224 and 2244 keep I.M. distortion products at an extremely low 0.05%...again, −66 dB compared to the amplifier's output.

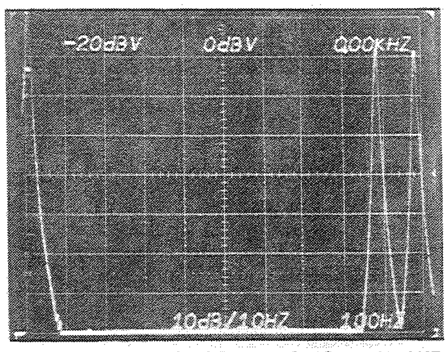


FIGURE 59. I.M. TEST OF 2244 USING 900 Hz AND 1 kHz JUST BELOW CLIPPING LEVEL OF THE AMPLIFIER.

Each frequency is being reproduced at 1/2 the maximum single-channel output into 4 ohms (252 watts per frequency). Intermodulation components are off-screen since they are at least 72 dB below the fundamental 900 Hz and 1 kHz tones. Vertical scale is 10 dB/div, horizontal scale is 100 Hz/div from DC to 1 kHz.

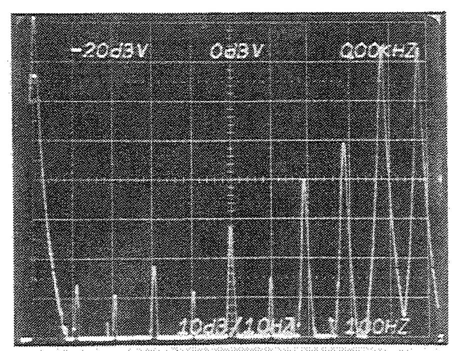


FIGURE 60. I.M. TEST AS IN FIGURE 59, BUT WITH EACH SIGNAL RAISED 3 dB (INTO THE CLIPPING AREA).

Here the same 900 Hz and 1 kHz tones are used, but the power of each tone would be raised by 3 dB if measured individually. Note that unlike the clipping of a single sine wave, the intermodulation now causes frequencies to appear below as well as above the two fundamental frequencies. The distortion components occur at 100 Hz intervals. This spectrum analyzer photo clearly demonstrates that clipping of music does not produce just higher-frequency harmonics... it also produces lower frequencies. The vertical scale is 10 dB/div, and the horizontal scale is 100 Hz/div from DC to 1 kHz.

One-way versus multiple driver speaker systems

A speaker "system" may consist of a single driver (speaker) in some sort of box (enclosure), but usually, a speaker "system" is thought of as an enclosure containing a "woofer" or bass driver, a midrange driver, and a tweeter – or some variation on this basic theme. The midrange driver and tweeter may be of the "direct radiator" variety (i.e., cone-type), or they may be compression drivers with horns. Why are multiple drivers used? Primarily to obtain higher efficiency, as much as 10 to 20 dB more than "comparable" full-range drivers. Physics dictates that efficient drivers must have narrower usable frequency bands, which means several types must be used in order to cover the full audio spectrum. A secondary advantage is that more uniform dispersion can be obtained across the audio spectrum than with a single full-range driver, which will exhibit narrower dispersion at higher frequencies.

In multi-driver speaker systems, a CROSSOVER electrically divides the full-range audio signal so the appropriate frequency range (or "band") can be fed to each driver: woofer, midrange and tweeter in 3-way systems, or woofer and midrange/tweeter in 2-way systems. Crossovers are available that will separate audio bands sharply or gradually (see Figure 61) to suit the characteristics of the particular drivers.

How crossover networks operate and why they are used

Crossovers are really pairs of filters which roll off higher frequencies so they don't reach low frequency drivers, and roll off low frequencies so they don't reach high frequency drivers. While we sometimes speak of a "crossover point," in fact crossovers are not magic, and are not capable of producing abrupt "brick wall" stops and starts in frequency response. The crossover point is merely the middle of a region of overlaps between two drivers where neither is producing full output.

To visualize how a crossover functions, let's consider a simple two-way speaker system with a woofer and tweeter. If we examine the low frequency output of its crossover, which is connected to the woofer, the signal "rolls off" gradually as the frequency increases. This is essentially a low pass filter which blocks unwanted high frequencies from reaching the woofer (in order to prevent too-narrow dispersion). If we examine the high frequency output of the crossover, which is connected to the tweeter, the signal level increases gradually as frequency increases.

In the middle, the "crossover region," the level fed to the woofer is down about 3 dB (half the power) relative to the level at a frequency well below the crossover region. Similarly, the level fed to the tweeter is about 3 dB below the level fed to it at a frequency well above the crossover region. A frequency that is right in the middle of the crossover region is being produced equally by the woofer and tweeter, so the combined "half power" signals add up to full power for smooth frequency response – at least in theory they do. This frequency where both drivers are fed equal level is the "crossover frequency" or "crossover point."

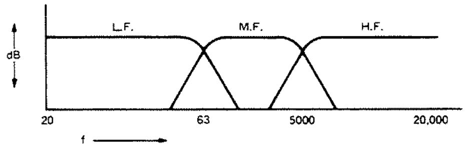


FIGURE 61. FREQUENCY RESPONSE OF A TYPICAL 3-WAY CROSSOVER NETWORK.

Observe that while there are 3 outputs (low/mid/high), there are two crossover points. Each crossover point requires two filters, one high pass and one low pass.

Given its function, you can understand why a crossover network is also known as a "dividing network" or a "frequency dividing network." Two-way and four-way crossovers operate on the same principle, but with fewer or more outputs and a crossover point between each pair of outputs.

High level versus low level crossover networks

Crossover networks can be placed in either of two "locations" in the signal chain of a sound system. The most common location is inside the speaker enclosure. In this case, the crossover network divides the output of the power amplifier just before it is fed to the individual drivers. Such crossover networks are called "high level" since they operate with speaker-level signals. They are also called "passive" because they require no power source to operate; the circuit of a passive, high-level crossover network typically consists of inductors (coils) and capacitors, but no transistors or integrated circuits. When high power operation is involved, the coils in the passive crossover can heat up, increase their resistance, and cause significant loss of power to the drivers; larger, more expensive coils help, but don't eliminate the losses. Good quality, passive high-level crossovers can cost several hundred dollars, weigh 10 to 25 pounds each, and one such unit will be required for each speaker enclosure.

The other location for a crossover network is just before the power amplifiers. In this case, the crossover network divides the output of the mixer (or other line- level signal source) just before it is fed to power amplifiers, which, in turn, feed the drivers in a speaker system. Such crossover networks are called "low level" since they operate with line-level signals, not the output of power amplifiers. While low-level crossovers can be passive, most are "active," which means they require a source of power, and use transistors or ICs. Another term for an "active, low-level crossover network" is "electronic crossover network." When fed by a low-level crossover network, each power amplifier is used to drive just one band of frequencies – one type of driver. For example, one amplifier can be fed

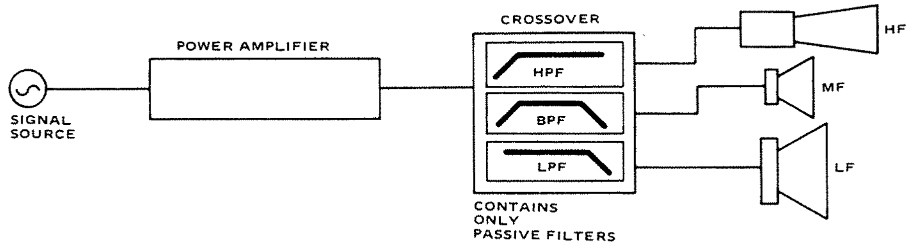


FIGURE 62. BLOCK DIAGRAM OF PASSIVE, HIGH-LEVEL CROSSOVER.

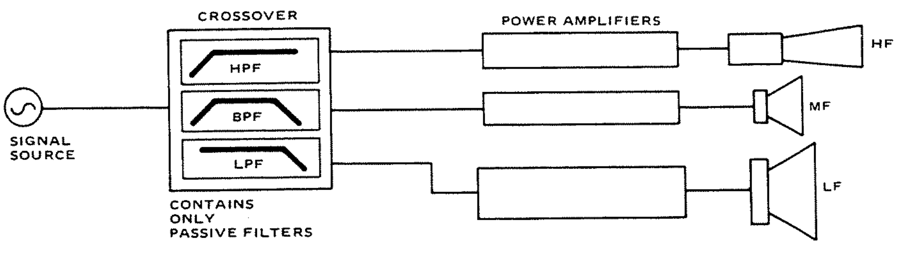


FIGURE 63. BLOCK DIAGRAM OF PASSIVE, LOW-LEVEL CROSSOVER.

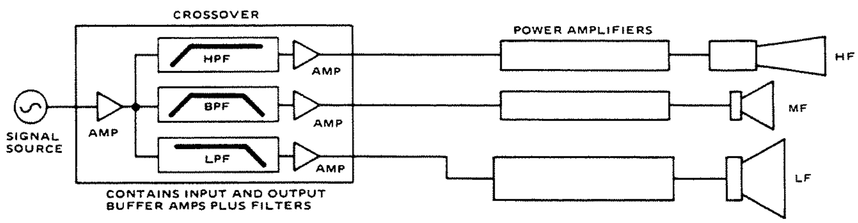


FIGURE 64. BLOCK DIAGRAM OF ELECTRONIC (ACTIVE, LOW-LEVEL) CROSSOVER.

bass frequencies, and its output goes to the woofers, while another amplifier is fed high frequencies, and its output goes to high frequency compression drivers.

Overall, this method of using a low-level crossover to feed two or more amplifiers with different bands of frequencies is called "multi-amping." Specifically, a two-way multi-amped system is said to be "biamplified," a three-way system is "triamplified," and so on. There are numerous advantages to multi-amped systems with low-level crossover networks, as opposed to systems with high-level, passive crossover networks. Particularly in large sound systems, the sound quality can be improved, overall amplifier power requirements can be less for a given amount of headroom, and total system cost and weight can be reduced; these factors are explained further in subsequent text.

Active (electronic) and passive low-level (before-the-amplifier) versus passive high-level crossover networks

Passive types (non-powered units with no IC's or transistors) use small inductors and capacitors in a metal box with connectors on it. The major drawback of passive low-level crossovers is their fixed characteristics; neither the crossover frequency nor the "slope" (the rate at which the internal filters roll in and out (in dB per octave) can be changed easily, unless many more components are added, which eliminates the cost advantage of the passive circuit. Passive low-level crossovers are used primarily where they are designed to complement a specific speaker system, so the exact characteristics are known and no adjustability is required. More often, however, low-level crossovers have active electronic circuits that accomplish the filtering. Because an IC, a capacitor and a resistor (plus a few peripheral components) can serve the same electronic function as a inductor, the electronic crossover networks do not need to use inductors. Without inductors, it becomes feasible to provide a control that alters the "inductance" of the electronic "inductor equivalent" circuit. This makes it relatively simple to adjust the crossover frequency, as well as the slope or steepness of the filters.

Other practical advantages are obtained when the large inductors of a passive, high-level crossover network are eliminated. The series resistance in the signal path to the drivers decreases, which increases the damping factor. Moreover, since large, low-loss inductors of copper wire are very expensive, smaller iron-core copper inductors or air-core copper inductors of marginal size are often used in commercial high-level passive crossovers; such inductors produce greater distortion at high power levels. Worse yet, if the signal level exceeds that for which the crossover was designed, the inductors can saturate. This means that they stop being inductors, resulting in drastically altered crossover characteristics, and possibly "blown" drivers. Using a low-level crossover instead of a high-level crossover network avoids this potential source of distortion or failure.

Advantages of multi-amplified systems

By using a low-level crossover and multiple power amplifiers, instead of individual amplifiers with a high-level crossover in each speaker enclosure, overall system headroom can be increased without necessarily increasing total amplifier power capacity. When a single amplifier has to reproduce the full audio spectrum, the low frequencies can "use up" the headroom of the amplifier (remember, it typically takes more power to reproduce low frequencies at a given sound level than it does to reproduce high frequencies). When higher frequencies share the available amplifier power, they "ride" along on the low-frequency waveforms (see Figure 65). As a result, they are subject to being "chopped off," or "clipped," even though the low or high frequency signal alone would not be sufficient to cause clipping. Although the low frequency portion of the signal may not yet have reached the clipping level, the peaks of the superimposed high frequency signal try to push the amplifier beyond the voltage limits of its power supply. In order to eliminate the problem with a full-range amplifier/high-level crossover system, it is necessary to turn down the volume or to use a larger power amplifier.

Separating the highs from the lows with a low-level crossover and feeding each frequency band to a different amplifier means that the treble will not ride atop the bass. A moderately high power amplifier will then be sufficient to reproduce the bass without clipping, while a smaller power amplifier can be used to reproduce the treble without clipping. The numbers work out, particularly in larger sound systems, such that the combined power handling capacity of all amplifiers in the multi-amped system can be less than the power capacity that would be required with a system that relies on full-range amplification.

With a full-range system, clipping is very audible since higher frequency harmonics caused by the clipping are fed to midrange and high frequency drivers which do an excellent job of reproducing the distortion. With a multi-amplified system, when the low frequency amplifier clips and generates higher frequency (annoying) harmonic distortion, those frequencies can only reach the low frequency driver. Since the low frequency driver is less sensitive to high frequencies, and does not reproduce them well, less distortion gets to the listener's ear due to the system's inherent rolloff. The mid and high frequencies, which are reproduced by another amplifier, remain clean.

CAUTION

When a power amplifier is connected directly to a high frequency transducer, as it is in a multi-amplified system, there is the possibility that clipping of that amp can damage the transducer. That's because the clipped amplifier will produce full bandwidth distortion products, including low frequencies which can rapidly damage diaphragms due to over-excursion. For this reason, some sound contractors and rental companies (who are not always able to control proper use of their systems) may place protection capacitors between the amplifier and the high frequency drivers. See the subsequent subhead titled "High frequency driver protection."

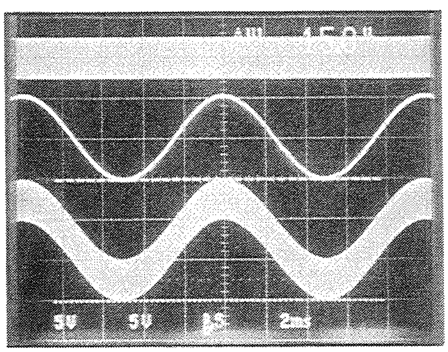


FIGURE 65. HOW TWO DIFFERENT FREQUENCIES CAN "ADD UP" IN AN AMPLIFIER.

Top Trace: A high frequency (10 kHz) sinewave at 5 volts peak-to-peak amplitude.

Middle trace: A mid frequency (1 kHz) waveform at 10 volts peak-to-peak amplitude.

Bottom trace: The combined signal produced when the 10 kHz and 1 kHz signals are both applied to the amplifier. Note that the peak amplitudes here are 15 volts peak-to-peak. With musical program material, the presence of a strong low frequency signal component can cause clipping of lower-level high frequency signal components unless the amplifier has adequate headroom. Biamplification provides one means to avoid such problems, or using a much larger amplifier to handle the full-range signal is another alternative.

With multi-amplified systems, it is also possible to place compressor/limiters at the output of each band of the low-level crossover. In this way, too high a level low-frequency signal or too high a a level high frequency signal can be "pulled down" without "ducking" the entire program. Each compressor/limiter can be adjusted for a different threshold, and, with more sophisticated compressor/limiters, the time constants can be tailored to each frequency band, which minimizes compression-induced harmonic distortion. This approach makes it possible to operate the sound system at higher average levels, without suffering from "over-compressed" sound and without encountering clipping, thus further increasing the useful dynamic range of the system.

When to use a high-level passive crossover

For small to medium sized rooms and clubs, high quality speaker systems that come equipped with an internal high-level passive crossover, like the Fender Models 2841 or 2851, can often do an excellent job. Even for large speaker systems that are very efficient, but will not be used at extremely high power levels, a high-level passive crossover may be the best choice. These crossovers also make sense in existing speaker systems that are doing the job satisfactorily. They make sense where cost is an important design factor in some self-contained speaker systems. Last but not least, if only one amplifier channel is available to drive a speaker system, then a passive, high-level crossover is the only reasonable choice.

Crossover characteristics and features

To explore crossover characteristics, it is helpful to have a clear understanding of the frequency distribution of musical program material. There are 3 decades of frequency in the audio spectrum:

20 Hz to 200 Hz
200 Hz to 2,000 Hz
2,000 Hz to 20,000 Hz

This same 3-decade range covers about 10 octaves:

20 Hz to 40 Hz
40 Hz to 80 Hz
80 Hz to 160 Hz
160 Hz to 320 Hz
320 Hz to 640 Hz
640 Hz to 1,280 Hz
1,280 Hz to 2,560 Hz
2,560 Hz to 5,120 Hz
5,120 Hz to 10,240 Hz
10,240 Hz to 20,480 Hz

Forty Hz is about the lowest USEFUL frequency for high-power sound reinforcement of rock music where subwoofers are not utilized; very low pipe organ note or synthesizers can go lower in pitch. Low "E" on a bass or organ is 41.2 Hz, and C4 (high C) on the piano is at 4,186 Hz. These are the fundamental frequencies of the notes; the harmonics extend upward in frequency to beyond audibility, but at a greatly reduced energy level compared to the fundamentals. The highest frequency that most untrained, adult ears can perceive is around 15,000 or 16,000 Hz. Sensitive, trained ears can perceive sounds as high as 20,000 Hz, but most sound reinforcement systems do not try to deliver much above 16,000 Hz to the audience (it would be difficult to distribute the last few thousand Hertz uniformly to the audience due to the high rate of attenuation in air at such high frequencies). The job of the crossover network, then, is to select one or more frequencies between about 41 Hz and 16 kHz, and to divide the sound to be fed to the appropriate drivers.

Providing high frequency capability in a sound system is usually a matter of using lots of tweeters to cover the listening area. Because tweeters tend to have relatively narrow sound dispersion angles, many tweeters are needed to provide adequate levels and dispersion at high frequencies. Fortunately, there is not very much energy in the musical program at the uppermost end of the frequency spectrum, so total power handling capacity can be less for tweeters than for bass and midrange drivers.

In a three-way system, the crossover frequency for the tweeters might be set at 5,000 to 7,000 Hz, and the amplifier power required to drive an array of tweeters operating in that range would typically be just 25% of the power required to drive the midrange of the system. The low frequency end of the audio spectrum is another matter. Most of the energy in music is contained in the bottom 5 octaves of the audio frequency range; in rock music, the spectral energy is shifted down even further. There are likely to be more low frequency drivers than high frequency drivers in a typical large sound system, and often the efficiency of different sections of the speaker system will be different. Since there are different sensitivities and amplifier power capabilities for each band of the speaker system, most electronic crossovers have controls for setting the output level of each frequency band. Such level setting features are useful but redundant when using the 2224 and 2244, which have stepped input attenuators that offer a quick and easy means of balancing the gain between sections of a multi-amped system.

The crossover slope (or rolloff rate) determines the degree of separation of frequencies on either side of the crossover point. A 6 dB per octave slope gives a smooth transition between frequency bands with minimal phase shift. However, this gradual rate does not adequately protect high frequency drivers against damaging diaphragm excursions caused by low frequencies; too much low frequency can get through "on the skirts of the filter slope," and undesirable "comb filtering" may occur, too. Typically, electronic crossovers offer a choice of 12, 18 and 24 dB/octave slopes. These high slope rates afford better driver protection and are more commonly used.

Different drivers sometimes require different slopes to blend properly with the rest of the drivers used in any given system. Sound system designers with access to frequency response measurement equipment and well trained ears can refine the choice of crossover slopes and frequencies.

The selection of a crossover frequency is not an arbitrary choice; it must be based on the driver specifications and available manufacturer's recommendations. If a certain high frequency driver's specification is stated as "Power capacity: 20 watts pink noise from 2,000 to 20,000 Hz," then a crossover of 2,000 Hz or higher should be used. If a higher crossover frequency is used, the power handling capacity of the high-frequency driver may increase somewhat. For example, suppose a midrange driver that operates up to 5,000 Hz is used with the aforementioned high frequency unit which has a minimum 2,000 Hz operating frequency; if the mid/high transition is set to 4,000 Hz instead of 2,000 Hz, the power handling capacity of the high frequency unit may be increased due to the reduced displacement requirements at higher frequencies. However, the thermal capacity of the driver is not changed, so the practical effect may be only a slight increase of power capacity.

In other words, if the drivers are blowing due to cracked diaphragms, as suggested by a phase plug imprint on the diaphragm, then raising the crossover frequency will help by reducing the overexcursion. If the drivers are blowing due to burned out voice coils, as indicated by charring, the thermal limit is the problem. Excess temperatures are caused by continuous power at too high a level. In some cases, this is due to DC components (which are prevented by protection circuits in Fender amps) or due to RF or oscillation in the amplifier output circuit, which would be beyond the audio range. More often, however, thermal problems are caused by clipping. Fender amplifier input circuits have filters to limit rise time, which protect against amplifying incoming ultra-high frequencies that would serve only to increase distortion and burn out tweeters.

A common method of simplifying a three-way system is to biamplify it using an electronic crossover for the low-to-mid frequency transition, and to then provide a high-level passive crossover for the mid-to-high frequency transition. This approach is a logical and effective cost saving means of putting together a three-way system: the parts required to make a higher frequency passive crossover are small and inexpensive, the amount of energy in the music signals is small, and the real "problem" with passive high level crossovers is with the bass anyway. The same logic may be used if a supertweeter is added.

The octave below the bottom of the usual musical frequency range (20 Hz to 40 Hz) contains information that can enhance the listening experience in certain applications. Better movie theaters now use subwoofer systems to augment low frequencies recorded on the soundtracks of more sophisticated films; this is done either by means of an electronic crossover to isolate and separately amplify low frequency information mixed into the sound track, or by triggering an auxiliary low frequency synthesizer from a code tone printed on the film soundtrack. Discotheques utilize subwoofers and octave dividers (units that synthesize bass an octave below that which exists in the recorded program) to create a richer, more visceral experience on the dance floor. Where live mics are involved, however, the very low bass response can add to rumble and feedback problems. Thus subwoofers are somewhat less common in live sound reinforcement systems.

Subwoofer systems, properly applied to sound reinforcement, can create a feeling of greater system power while adding excitement and audience involvement through physical sensation. The skin actually does some of the "hearing" at extremely low audio frequencies, and it is this frequency range that is reproduced by subwoofers.

Modular subwoofers

Except in permanent installations, subwoofers are easiest to utilize if they are designed with a modular approach. Where possible, it is desirable to include an electronic crossover capable of both simultaneous low-pass and high-pass functions to separate the system's elements. At the very least, a low-pass network should be provided to complement the natural high-pass function of the rest of the speaker system. Any speaker system will roll off below some low-frequency limit, influenced by the enclosure/woofer combination and how they are tuned, and by the efficiency of the woofer(s) in the enclosure. However, a high pass filter (low cut) is desirable to prevent uncontrolled woofer motion below the resonant frequency. A two-way system (in this case, subwoofer and full-range system speaker) theoretically should utilize a subwoofer crossover frequency and slope that forms a mirror image of the natural roll-off at the bottom of the full-range system.

NOTE Artistic judgement is necessary when considering subwoofer use. In practice, the subwoofer output level may be increased to obtain the proper "feel," even though the resulting response curve is not "flat." Conversely, in difficult environments, or when used with "tubby" or bad-sounding program source material, a subwoofer can be a detriment to the quality of the reproduced sound; in such instances, it may be better to reduce the subwoofer level or not use the subwoofer at all. One of the most critical factors with subwoofers is the crossover characteristic. A 3-pole (18 dB/octave) filter is the minimum that should be used, and 4-pole (24 dB/octave) is even better. When adding a subwoofer to a full range system, the upper crossover frequency should be between 50 Hz and 80 Hz, and the lower end of its range should be high pass filtered.

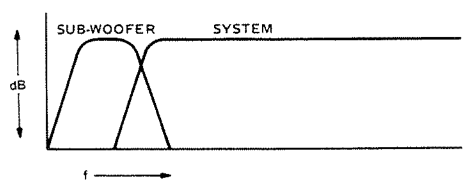


FIGURE 66. FREQUENCY RESPONSE OF SUBWOOFER BAND PASS FILTER AND HIGH PASS FILTER FOR THE REST OF THE SPEAKER SYSTEM (A 2-WAY CROSSOVER).

While the inherent high-pass characterstic of the full-range speaker system may be sufficient to properly separate it from the subwoofer, the addition of a high-pass section in the subwoofer's crossover can be beneficial. It makes possible a steeper transition between drivers, thereby saving power, avoiding "comb filter" effects, and limiting cone excursion in the full-range system's bass speakers.

Delay

Sometimes, delay can be useful for REDUCING the echo of a sound reinforcement system in large spaces or outdoors! Since sound travels at a relatively slow 1125 feet per second, speakers placed more than about 10 meters (33 feet) from the main speakers on stage, produce an artificial, but clearly audible, "pre-echo" for listeners nearby. When a delay line is inserted ahead of the amplifiers that feed the remote speakers, and set for a delay a few milliseconds longer than the time it takes for sound to get from stage to remote speakers, the audience near the remote speakers will hear sounds arriving from the stage first, and then the remote speakers. This sound will then be perceived as happening simultaneously, and as emanating from the stage (not from the remote speaker). Such delays are routinely used in theatres and arenas where balconies make it difficult for the main stage speaker array to cover audiences underneath. The sound delay setting on most digital delay units is labeled in "ms" or milliseconds. This is handy as a reference for estimating proper delay settings for the distance, since 1 ms is a thousandth of a second, or about the amount of time it takes sound to travel just over a foot (13.5 in).

Effects

The term "effects" can be used to describe anything that produces an audible effect on the sound of a system other than just making it louder. Echo, delay, reverb, phasers, flangers, gates, compressors, limiters, parametric and graphic equalizers, exciters, filters, and envelope generators can all be classified as effects devices.

Effects often are thought of as small, battery-powered boxes that are used by guitarists. Many of these "electric guitar" style effects units can be used in a sound system signal chain, just like a rack-mounted "high-tech" effects device, as long as their their noise and distortion are acceptable... and provided their input and output characteristics (levels, impedances, connectors) allow for proper interface with the sound system.

Effects used in sound systems tend to cause a certain amount of unpredictability. One important aspect of some effects is their ability to cause radical changes in gain (volume) or equalization, which can lead to serious overloading of other system components like mixer or preamp circuits, electronic crossovers, limiters, power amplifiers and speakers. If you plan to regularly use "radical" effects, become a meter-reading fanatic. Keeping an eye on your levels (VU meters and amplifier power indicators) can, in the long run, cut down maintenance expenditures, especially for expensive items like speaker reconing and compression driver diaphragms.

Compressor/limiters

A Compressor is a line-level amplifier that decreases its gain as the level of the input signal increases. It does so to reduce the dynamic range of the program material. A compressor may operate over the entire range of input levels, or it may operate only on signals above or below a certain threshold level. When a signal is being compressed, the ratio between the change in output level (in dB) versus the change in input level (in dB) is known as the compression ratio.

A limiter is merely a class of compressor with approximately an 8:1 or greater compression ratio, although the ratio itself does not define whether a unit is a compressor or a limiter. The threshold and application defines the unit. Limiters are used primarily to prevent the signal level from exceeding a set threshold. Often a single unit can be adjusted to function as a compressor or a limiter. Originally, the term "compressor" described any device which increased the level of low-level signals, and decreased high-level signals, "squeezing" them toward the threshold. A limiter, on the other hand, was defined as a device which acted only to decrease levels above a threshold. Today, these distinctions are not always made so the compression ratio and threshold settings become the major differences between the devices. Compressor/limiters can be useful devices for protection against excess signal levels. Compressor/limiters are not a panacea for every audio ill; they cannot stop feedback or make cheap microphones or speakers sound good, nor can they make up for the lack of sufficient amplifiers and speakers in a system.

When a compressor/limiter is working, it "decides" to start compressing based on the front panel settings you choose so you have to choose carefully. For example, the circuitry often does not know the difference between bass and treble. Since the highest signal levels in music program material usually are produced by low bass notes, compressing full-range music could cause all the music to be modulated whenever there is a heavy bass note. The resulting "pumping" is one reason to use compressor/limiters AFTER the crossover; then, a loud bass note will not be able to "duck" the midrange or highs. However, post-crossover compression can cause other problems, so be sure to carefully evaluate such a hookup during a "sound check." (Sometimes the compressor is designed to operate differentially with frequency; a high-frequency sensitive compressor is known as a "de-esser" and is used to reduce speech sibilance.)

In studio monitor and sound reinforcement systems, one way to avoid speaker damage in the event of a malfunctioning or misused effects device is to place a limiter just ahead of the power amplifier input.

In factories and stores, compressor limiters are used to level out the differences between the voices of people who use the paging system, making it easier to use the microphones and also making the pages more audible under noisy conditions. Where conditions are unpredictable, a compressor/limiter can

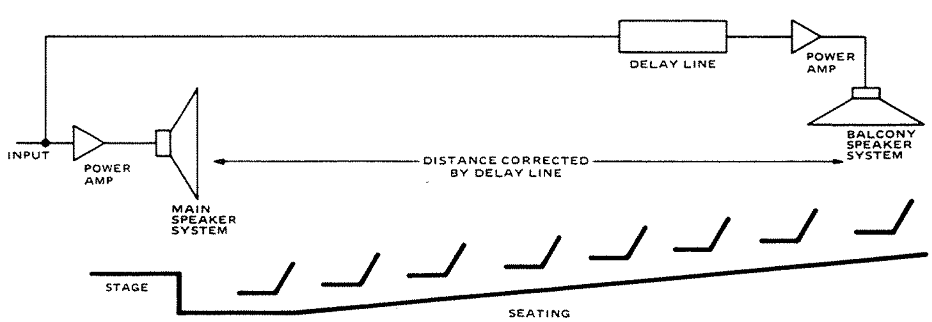


FIGURE 67. BLOCK DIAGRAM OF TIME-DELAYED REMOTE SPEAKER SYSTEM.

help prevent system overloads that might otherwise cause damage, blow fuses, and generally render the paging system inoperative.

Overuse of even the best compressor/limiters can be as obnoxious as use of a low-quality unit. Aside from the unpleasant sonic quality, excess compression creates substantially higher average power levels, which increases thermal stress on the speakers, which can burn out the speakers. Compression should be avoided in high-fidelity music reproduction and sound reinforcement systems, though in paging systems it's often very useful to "level out" the sound of background music; in such systems, it may be better to accept some audible compression side-effects than to allow amplifier clipping or speaker overload.

Equalization

In the early days of the telephone, the word "equalizer" was coined to describe a device used to "equalize" the deficient frequency areas, and make them "equal" to the unaltered areas of frequency response.

Originally the term "equalize" meant "boost." Today, there are equalizers that boost only (in the traditional sense), that cut only (attenuate), or that boost and cut. An equalizer set for boost increases the voltage of signals when those signals have frequency components that fall into the range being boosted. An equalizer set for cut reduces voltage in the affected frequency range.

There are sweep type equalizers with adjustable center frequencies, and parameteric equalizers, where the center frequency and "Q" (filter width) are adjustable, making it possible to "tune in" on the smallest frequency anomaly. Graphic equalizers have many different fixed frequency bands which can all be adjusted more-or-less independently. Selectable multi-frequency equalizers are found on many recording consoles, and of course "tone control" type equalizers with simple bass and treble "shelving" filters are widely used. Each type of equalizer is useful for achieving a certain effect, or for correcting perceived sonic problems.

Each type of room has its own distinct sound – or at least, it affects the character of a sound system. Thus, the room itself provides a form of equalization. The best rooms, such as recording studios, are fairly flat in their sound absorption and/or reflection characteristics. Some rooms are downright terrible, like a concrete and glass nightclub or a typical gymnasium. Equalization can help to an extent, but should never be expected to make up for all the deficiencies in room acoustics.

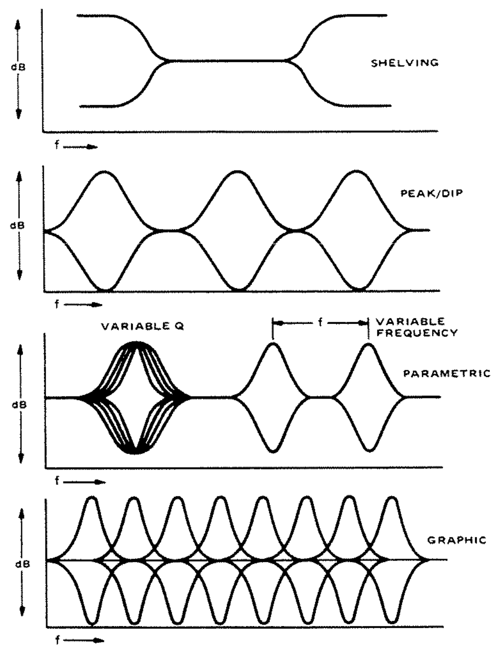


FIGURE 68. RESPONSE CURVES OF VARIOUS TYPES OF EQUALIZERS.

Speakers have their own frequency and polar (angular dispersion) characteristics. No speaker has the exact same frequency response on-axis (in front) as it does off-axis; the sound sometimes varies greatly as one measures it at the front of the speaker, to the side, and at points between. This non-uniform frequency response at different angles can vastly complicate room equalization problems, sometimes making them insoluble. Equalization can, however, help to correct overall speaker system response deficiencies. Peaks can be lowered with a narrow-band graphic equalizer or a parametric equalizer, thus making it possible to increase the gain before feedback occurs. Dips in the middle of the speaker's range, or gradual roll-offs at the ends of its range, can be boosted to obtain "flatter" response... or perhaps the ends of the spectrum may be intentionally cut, depending on the desired effect and the room acoustics.

Intelligibility is a very important factor in sound system design and operation. Intelligibility refers to the system's ability to make speech understandable or to make singing, or individual musical instruments sound clear and distinct. Equalization, using the tone controls on a mixer or an external equalizer plugged into an individual input channel, can be judiciously applied to individual sound sources within a mix so that each sound being amplified is distinct from the other sounds; this can go a long way toward improving intelligibility regardless of the sound system which is used. Additional equalization of the sound system itself may still be required.

There are several aspects to "protection" in a power amplifier. Normally, protection circuitry should do nothing. However, there are many ways for something to be abnormal, and when that happens, protection circuitry can prevent catastrophic damage to speakers, speaker wiring and the amplifier itself.

Designed and built for professional use

What distinguishes "professional" sound equipment from "Hi-Fi" or, as it is now known "consumer audio" equipment? Most of the time, professional equipment is nothing more than equipment that is widely used by professionals, either because it is functionally unique, or has gained a reputation for being appropriate for the job.

In the world of professional audio, especially sound reinforcement, power amplifiers are exposed to abusive conditions like frequent setup and transportation on a day to day basis. Touring sound crew members are usually rushed, many people on a crew will handle the audio equipment, and rarely are conditions on the road like those in a home environment. Few consumer amplifiers can stand up to these conditions. Larger professional high-efficiency loudspeaker components, crossovers and speaker systems present loads that are much more "reactive" than the loads encountered in home situations, with rare exceptions, including "audiophile" consumer systems. An extremely important aspect of professional audio equipment design is reliability. Consumer audio enthusiasts don't lose income if their super-high-power amplifiers quit in the middle of playing a direct-to-disc record, but professional sound systems are often set up on a contracted basis, by professional sound contractors whose income depends on their equipment, and loss of sound during a live show can mean loss of revenue and possibly legal action against the sound contractor.

The design criteria of some "audiophile" amplifiers places sonic results ahead of all else. While there is nothing inherently wrong with the use of esoteric circuitry, sometimes protection circuitry is omitted, which can compromise long-term reliability. The 2224 and 2244 are designed not only to deliver superb sonic quality, but also to operate satisfactorily in a wide range of environmental conditions and with severe loads that are common in professional sound systems.

Amplifier output stage protection

The output stage of an amplifier consists of the main power transistors that drive the speakers, plus related electronic components. Transistors are most susceptible to failure when their internal junctions are overheated...typically 150 to 200 degrees Centigrade is the maximum allowable junction temperature. To an extent, the more overall power being delivered to the speakers, the greater the power dissipated by the transistors, and the greater the heat. However, maximum heat dissipation occurs at 40% of maximum power. Unless heat at the transistor's internal junction is reduced, the transistor will fail.

One way to avoid transistor failure is to use more transistors so that the overall power dissipation is shared. The 2224 has six output transistors per channel, three each NPN and PNP. The 2244 has twice the number of output transistors. Another approach is to use power transistors which have high power ratings, and therefore do not heat up as much when a given amount of power flows through them. Each of the six or twelve output transistors in Fender amplifiers is rated at 200 watts. The combined power handling capacity of these transistors (1200 or 2400 watts) far exceeds the requirements for the maximum rated output of the amplifiers.

There are good reasons for us to build in this seemingly extreme reserve power handling capability. Transistor power ratings are predicated on junction temperature. For every degree Centigrade the junction rises, the number of watts that can be safely accommodated goes down. When these transistors heat up, they can still handle plenty of power. However, in order to preserve the highest possible power handling capacity and avoid amplifier failure, heat must be dissipated. For this reason, we created exceptionally effective heat sinks. Their large, staggered "fingers" (fins) carry heat away from individual transistors and efficiently radiate it into a tunnel-shaped duct. The fins are copper (for good heat conduction) and are chrome plated to avoid oxidation. A fan pulls cool air through the duct via a port in the front of the amplifier. The heat sinks create turbu-

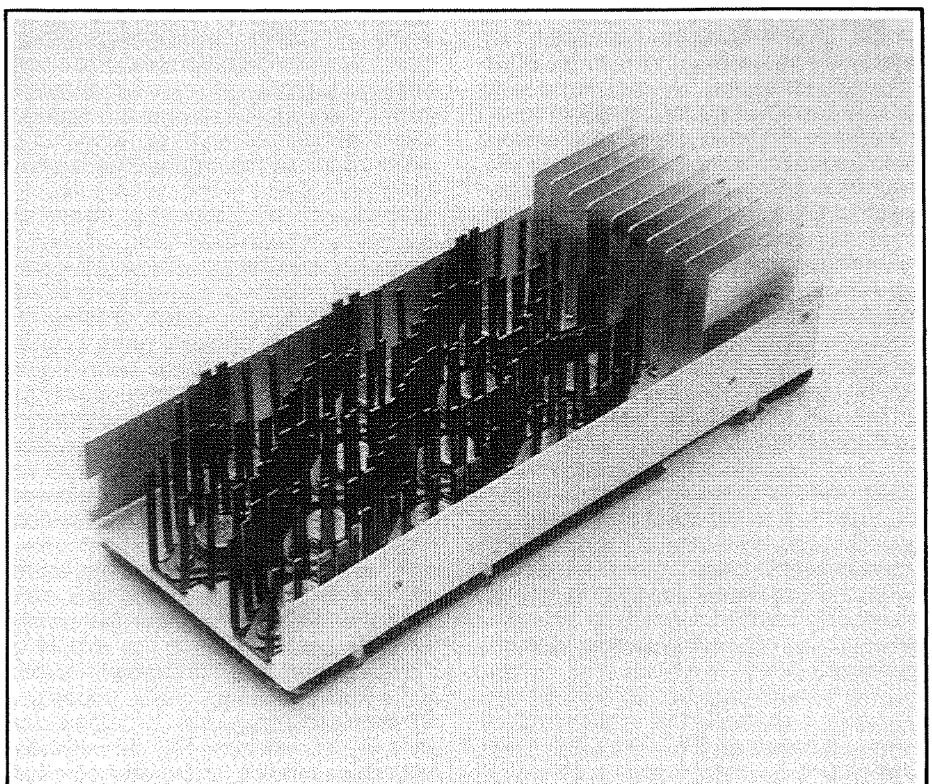


FIGURE 69. HEAT EXCHANGER IN FENDER AMPLIFIER.

lent air flow, which is the most efficient means to transfer heat from the metal fingers to the air. The warm air is then exhausted out the back of the amplifier – a more sensible scheme than pulling warm air from the rack and exhausting it to the front, as is done by some amplifier manufacturers. This design is so thermally efficient that the output transistors almost always remain cool, retaining their maximum power handling capability.

In the event of very high ambient temperatures or very high power demands on the amplifier, the heat sink temperature may rise. A thermal sensor "tells" protection circuitry when the temperature reaches a near-critical point, and the circuitry automatically ramps down the power supply output voltage so that less power will be delivered to the load, allowing the transistors and heat sink to cool without actually interrupting the audio output. As the amplifier cools down, the output power will automatically be returned to normal. In the very rare event that a severe overtemperature condition develops despite these protection measures, the speaker output will be automatically disconnected until adequate cooling occurs.

Efficient heat sinks work well with long-term high power levels, but they offer little protection for shorter term power surges in the order of a hundred milliseconds to several seconds; in these instances, the junctions can reach very high temperatures before the heat transfers to the transistor case, then the heat sink, then the forced air. Electronic energy sensing protection circuitry is provided for this reason. The specific circuit has been carefully designed to avoid "clamping" brief peaks that are necessary for accurate reproduction of musical transients. A plucked string or hammered drum head, for example, may create a transient that is 20 or 30 dB above the rest of the sound, though it only last for a few milliseconds; the transistors and power supply will have no difficulty in delivering as much as 1500 watts per channel for this brief time. However, to prevent damage to the transistors, the output current is automatically reduced if the excess power level lasts more than 100 milliseconds. When the power demands are reduced, the "clamp" is released. The time constants are such that "chatter" or current limiting noise will not occur unless the effective load impedance is well below 2 ohms.

Dealing with complex loads

At a speaker's two impedance extremes (the peak and the minimum impedance), voltage and current will be in phase with each other in the circuit from the amplifier output, to the speaker. In this instance, the term "phase" refers to the relative position of the current and voltage with respect to each other. A single cycle of a sine wave (pure tone) can be divided into "phase angles" by degrees, as in Figure 11 on page 13. At every other frequency beside the two frequencies where maximum and minimum impedance occur, the voltage and current phase angles in the amplifier/speaker circuit either lead or lag each other.

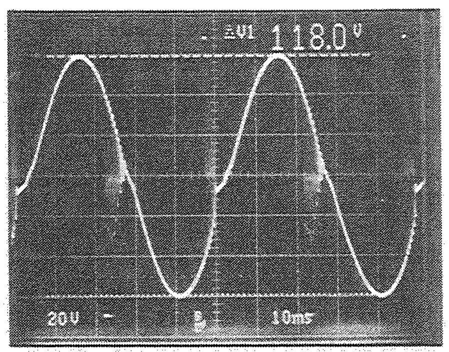


FIGURE 70. PREMATURE TRIGGERING OF PROTECTION CIRCUITS DUE TO AN INDUCTIVE LOAD IN A TYPICAL (NON-FENDER) AMPLIFIER.

Many amplifiers are designed using resistive test loads, and are not designed to work well into the typical reactive speaker load. Here the test frequency is 20 Hz and the load is purely inductive with a phase angle of 90 degrees. The inductive load causes triggering of the amplifier's VI protection circuits. Note that the protection spikes are actually greater in amplitude than the actual output level (which is indicated at 118 volts peak-to-peak). These protection spikes are far more offensive-sounding than clipping because they generate a wide frequency spectrum, as shown in Figure 72. The Fender 2224 and 2244 protection circuits are designed to eliminate this problem for 4 ohm loudspeaker loads (or even lower impedance loads); 2 ohm 60-degree load operation is possible.

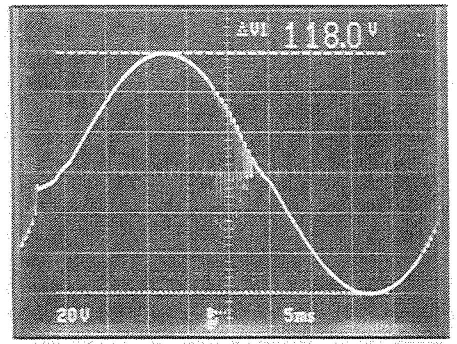


FIGURE 71. EXPANDED VIEW OF FIGURE 70, WITH HIGHER OSCILLOSCOPE TRACE SPEED (5 ms/div) FOR SINGLE CYCLE.

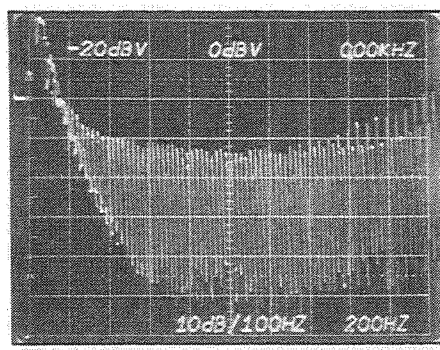


FIGURE 72. FREQUENCY SPECTRUM OF PREMATURE PROTECTION CIRCUIT TRIGGERING (FROM SITUATION IN FIGURE 70).

This spectrum analyzer photo illustrates the considerable output between 20 Hz and 2 kHz. Although the pulses are very narrow, the frequency spectrum generated is great. As far as the ear is concerned, the perceived sound (buzz) is louder than the fundamental tone of 20 Hz. Simple clipping is much less audible than VI limiting because the VI limiting is heard as a "crack" or a buzz. Vertical scale is 10 dB/div, Horizontal 200 Hz/div. The fundamental at 20 Hz is at 0 dBv on the display.

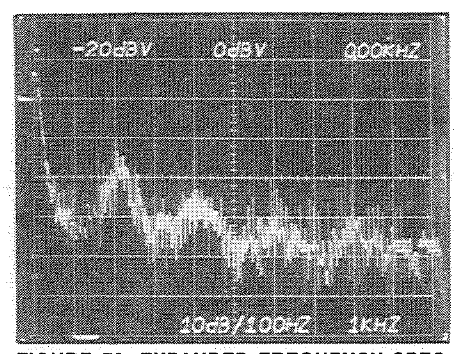


FIGURE 73. EXPANDED FREQUENCY SPECTRUM FROM FIGURE 72, CONTINUED OUT TO 10 kHz.

Note that the 20 Hz bass note activating the amplifier's protection circuitry generates considerable energy beyond 10 kHz! The vertical scale is 10 dB/div, and the horizontal scale is 1 kHz/div.

The "complex impedance" characteristic of speakers can cause the protection circuits in many amplifiers to trigger falsely. Such unwanted "protection" constitutes a problem because it occurs before the amplifier is actually producing maximum power into its load. If the protection is intermittent, it can cause "chatter" – unwanted noise. The 2224 and 2244 will supply maximum power into extremely difficult loads, much more than other similarly-rated amplifiers. The 2224 and 2244 enter protection mode rarely, and then gradually and only when the output conditions approach a pre-set value (still well below the "SOA" or "safe operating area" of the amplifier's output transistors).

Power amplifier protection

Given all this power handling capacity, it is certainly possible for the 2224/2244 to deliver a staggering amount of power to a near-dead short circuit or a low impedance load. For example, even with a pure resistive 2-ohm load, 28 amperes (784 watts) would flow. Given reasonable ambient temperatures, the output transistors and heat sinks could probably handle this power indefinitely, but the power transformer might not be able to operate indefinitely without overheating, or the power mains themselves may be inadequate. Obviously, some form of protection is required.

A fuse can be of some value in protecting the power supply against AC powerline surges and long-term overloads at the outputs. For this reason a circuit breaker type On/Off switch is utilized; if it "trips," the operator can see the problem and reset it from the front panel. *Every effort has been made in the design of the 2224/2244 power supplies, to prevent unwanted shutoff during a live performance.*

Since the breaker-type switch is not designed to be an overly sensitive "watch dog," it might still be possible, under the wrong circumstances, for a continuous high-power output condition to severely overheat the power transformer without tripping the breaker. For this reason, a thermal sensor is mounted on the power transformer itself; when it gets too hot, the power supply output is ramped down to facilitate cooling. (This is separate from the output stage thermal sensor circuit.) If the temperature continues to increase, the load is automatically disconnected.

Speaker protection

There are a number of conditions that can damage speakers. A turn-on transient (a low frequency "thump") may occur as the power supplies in signal processing equipment activate internal circuits, but before the circuits reach a stable operating status. These "thumps" are then amplified by the power amp. *The safest approach here is to disconnect the speakers from the amplifier until all equipment has stabilized, which is what has been done in the Fender 2224 and 2244.* The relay is an extra heavy duty industrial type whose large contacts avoid arcing or resistive losses. The relay is closed after power has been on a few seconds and everything is stable.

If DC were to appear at the amplifier outputs, it would cause speaker voice coils to move to one position and stay there, which would probably cause severe overheating and premature failure of the speaker. Even relatively low levels of DC can cause this problem. DC can appear at a direct coupled amplifier's output when the input signal has just a few millivolts of DC offset; in AC coupled amplifiers, DC can still appear with assymmetrical, very low frequency input signals. High DC levels, as might be caused by a short between the power supply rails and the output, can destroy a speaker suspension from over-excursion even before the voice coil burns out. *For this reason, circuitry in the 2224/2244 senses the presence of any DC at the output (including a non-symmetrical low frequency waveform or infrasonic signals under 1 Hz), and opens the speaker relay immediately.* The relay will again close when the DC is removed from the output.

Certainly the presence of continued, excess power (even if it is clean, unclipped and free of DC) can destroy a speaker. The 2224/2244 can deliver high power to almost any speaker load so proper system design and operation are important. However, if a short circuit in the speaker or cable, or a very low impedance speaker load, or improper operation causes excessive power to be drawn from the amplifier for too long a period, *thermal sensors in the output stage and power transformer will eventually reduce the output level or even disconnect the speaker load, if necessary.*

As you can see, the Fender 2224 and 2244 have many built-in safeguards to protect your speakers. Nonetheless, careful system design and operation should always be observed.

Use adequately rated speakers

The best speaker protection is obtained using amplifiers of sufficient power to avoid clipping, loudspeaker components designed to handle large amounts of power, and using these loudspeakers within their rated frequency band. In the case of studio monitors and especially home Hi-Fi speakers, it isn't always possible to have drivers of the highest efficiency (sensitivity), so more power will be needed to obtain a given sound level. For professional portable sound systems and most installed sound systems, it is essential to choose high-efficiency drivers to help maximize the sound-for-watts aspect of the system. High efficiency drivers reduce the number or size of amplifiers, and the number of system components, while allowing the greatest amount of headroom, and reducing distortion because the system can be operated at levels below its maximum capability. A common sense approach to system design and operation means not asking system components to operate at full capability on a continuous basis.

Speaker fuses

It is risky to use speaker fuses in an attempt to run speakers and amplifiers at or near full power all the time. Fuses are good protection against known quantities of amperes that flow for a certain amount of time, but these two things are exactly what is unknown in a music signal. One reason for the uncertainty is the changing frequencies of music signals which, in turn, change the current (amps) in loudspeaker voice coils. You may have intended that a certain amount of power should blow a protection fuse, calculating the power based on an 8-ohm load. However, speaker loads are frequency dependent, and the fuses may keep blowing every time a certain note of the scale is played on a bass or synthesizer. You may get what seems to be just the right fuse until a time-related power surge, such as feedback or a highly limited signal, is put through the system. Slo-blo fuses offer a little bit better system stability against time-related surges at the cost of poor voice coil protection; they are too slow to protect compression drivers and tweeters against large transient power surges which occur frequently in music signals. Fast blowing fuses can be used to prevent amplifier DC faults from destroying speakers in the event an amplifier "goes DC," *an almost impossible event in the case of the 2224 and 2244.* These fast blowing DC protection fuses should be selected for a current rating needed to pass the speaker's rated driver power handling capacity. If the driver is rated at 100 watts and 8 ohms, that corresponds to 3.5 amps. If the speaker is a multiway system containing a high level passive crossover, then system power capacity for the entire system should be used as a guideline. Remember that while the fuse adds a certain amount of protection for the woofer; it offers questionable protection for a compression driver or tweeter in the same system because those components are generally capable of handling less power than the woofer, and the amount of padding for the high frequency drivers provided by the crossover may or may not balance the decreased power handling capacity of those drivers.

Filter networks for high frequency drivers

A compression driver or tweeter can be protected against transient DC surges, DC shifts in amplifier output due to non-symmetrical waveforms, and turn-On/turn-Off "thumps" from system electronics, by inserting a capacitor in series with the driver-amplifier hookup, and possibly a resistor shunt across the driver (thus consitituting a half section filter). Even with a multi-amplified system, such protection may be necessary to prevent low frequency energy from damaging the diaphragm or suspension due to over-excursion; low frequency energy can be generated when clipping occurs (see note below). With the Fender 2224 and 2244, the danger to high frequency drivers is minimized, but it can never be eliminated with any amplifier capable of generating high power levels.

NOTE Clipping a single sine wave generates only harmonics that are higher in frequency than the clipped signal. However, when two sine waves are simultaneously clipped, they not only produce higher frequency harmonics, they also produce intermodulation products which include lower frequency components. When complex musical signals are clipped, it is possible for full-bandwidth energy to be generated. The extent of this effect depends on the amplifier itself and the degree to which it is overdriven. In actual "real world" applications, some amplifiers exhibit better behavior than others in this regard.

When constructing the protection network, select a capacitor which is rated at a minimum of 200 volts. NON–POLARIZED mylar, polystyrene, polypropylene, motor-start or oil-filled types are preferred, but may not be available in high values, so they may have to be paralleled with other types of capacitors. The value should be chosen to yield a high pass frequency which is an octave below the setting of the electronic crossover's high-pass frequency for the drivers.

NOTE If you're using a self-contained high level crossover network, the capacitor is not necessary since one is already present in that network where it will protect the driver. Low level networks that are ahead of the power amp require this type of protection for the driver(s). The resistors are used in addition to the capacitor to damp the resulting resonant circuit (caused by the inductance of the driver and the protection capacitor); such a resonant R-L circuit could actually deliver more voltage to the driver at certain frequencies than is present at the amplifier output!

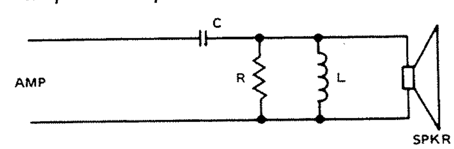


FIGURE 74. SCHEMATIC OF HIGH PASS FILTER & SPEAKER.

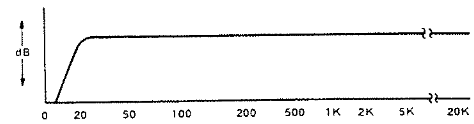


FIGURE 75. FREQUENCY RESPONSE OF HIGH PASS FILTER.

This circuit can provide a 12 dB/octave roll-off. The inductor constitutes a short circuit for DC components, so if the amplifier output "goes nuts" and creates low frequency charging currents in the capacitor, the inductor blocks the low frequency components from reaching the driver. The equation for a constant K network is $X_L = 2\pi FL$ or $X_C = \frac{1}{2\pi FC}$ where X is the driver impedance.

In order to determine the value of the "blocking" capacitor in microfarads, use this little equation based on the filter corner frequency equation $f_3 = 1/(2\pi RC)$. C = 500,000/(3.14 x frequency x impedance) $= \frac{0.159}{fZ}$

This equation will give you a 6 dB per octave high pass filter whose -3 dB point is at the specified frequency in the equation. If you want the filter to commence an octave below the crossover frequency, be sure to use that lower frequency in the equation. These capacitors will end up being quite large and fairly expensive, but in the long run, can prove worthwhile to protect drivers, especially expensive compression drivers.

The resistor R should be about 1.5 times the driver's rated impedance. For example, use a 12 ohm resistor for an 8 ohm driver. The power rating of both resistors should, preferably, be equal to the driver's power rating. One should bear in mind that the resistor, while it damps the resonant circuit, also reduces the effective driver efficiency and places a greater load on the amplifier. For example, an 8 ohm driver in parallel with a 12 ohm shunt resistor constitutes a 4.8 ohm load. Two of these driver/filter circuits in parallel constitute a 2.4 ohm load on the amplifier, not the 4 ohm load that one might casually assume if only the driver impedances are considered.

Limiters

Limiters aren't usually thought of as speaker protection devices, but when limiters are used to prevent clipping by reducing the level of the waveforms a bit before they can get up to the clip point of the amplifiers, the speakers absorb far less heat and are subjected to much less mechanical stress from working against their mechanical limits.

Equalizers

Equalizers, similarly, can be used to limit the amount of useless energy being fed to speakers the same way that high pass and low pass filters limit inaudible frequencies, thus limiting the overall amount of power to the speakers. The 2224/2244 have built-in high pass filters that are switchable for operation at 20 Hz or 40 Hz (or "off" to enable DC amplification, as might be used by a repair technician for bench testing to confirm proper amplifier operation).

Transformers

Transformers, like those found in distributed systems, can sometimes limit both low and high frequencies, not so much by design, but by physical limitations. Some transformer's inability to pass very low frequency information is due to core saturation, which is directly related to core size. High frequency response may roll off due to inductive coupling losses from poor winding design, low quality core material, or high coil capacitance. With DC coupled amplifiers, DC offset can be amplified and quickly overheat the transformers, although this is not a consideration with Fender amplifiers, which have a maximum DC offset of about 6 millivolts (typically 1.5 to 3.5 mV). If transformers are to be used, only the best quality ones should be considered. In general, however, we suggest using transformers for allocating amplifier output and reducing current flow in distributed systems, when necessary, not to restrict low frequency energy.

Mathematical and Electrical Units and Symbols

Throughout the text, various mathematical and electrical terms are used. For the purpose of this manual, they are decribed here.

Mathematical units

Quantities used in electronic measurements can vary over an extremely wide range. Generally speaking, there are three systems of denoting the value (size or amount) of a quantity. These are:

A. The absolute value, as in one million ohms.

B. A the Greek alphabet, as in one megohm (1,000,000 ohms, where mega is the abbreviation for a million).

C. Expressing the value as a power of 10. This is known as "scientific notation." For example, one million ohms also can be written as:

1×10^6 ohms (1 times ten to the sixth power).

For relatively small numbers, say from 1 to 1000, the absolute value (A) is generally used. When numbers become much larger than that, or smaller than 1, the latter two methods (B & C) are more commonly employed.

Table 6 shows the more commonly used Greek prefixes, and their values. Suppose, for example, you wish to convert 100,000 Hz to value using a Greek prefix. First find the prefix that is the largest value which is still smaller than the number you wish to convert. Divide your number by the value of the prefix. Now you can express the result of the division followed by the prefix, and finally followed by the name of the measurement unit. Specifically, to convert 100,000 Hz:

1. Look 100,000. It is 1000, and so the prefix is "kilo" (for thousand).
2. Divide the answer is 100.
3. The value is expressed as 100 kilo Hertz, or, more commonly, 100 kHz.

Scientific notation is a system that uses the powers of 10 to represent very large or very small numbers. Scientific notation is convenient to use because the power of 10 instantly tells us the relative size of the number with which we are dealing.

To convert a number to scientific notation:

1. If the number is larger than one, then count the number of digits to the left of the decimal point, then subtract 1. If the number is smaller than one, then count the number of zeros to the right of the decimal point, then add 1. The value you derive after adding/subtracting 1 is the exponent to be written after the 10. If the original number was less than 1, then the exponent is negative.
2. Shift the decimal point until the original number has one significant digit (that is, non-zero) to the left of the decimal point. Discard any leading zeros.
3. Write the power of 10 after the number obtained in Step 2. For example:

Convert the following numbers to scientific notation:

12345 .00000123 1000 543210987654

$12345 = 1.2345 \times 10^4$
$0.00000123 = 1.23 \times 10^{-6}$
$1000 = 1 \times 10^3$
$543210987654 = 5.43210987654 \times 10^{11}$

numeric value	prefix	notation	english	symbol
1,000,000,000,000	tera	1×10^{12}	trillion	T
1,000,000,000	giga	1×10^9	billion	G
1,000,000	mega	1×10^6	million	M
1,000	kilo	1×10^3	thousand	k
0.001	mili	1×10^{-3}	thousandth	m
0.000001	micro	1×10^{-6}	millionth	µ
0.000000001	nano	1×10^{-9}	billionth	n
0.000000000001	pico	1×10^{-12}	trillionth	p
0.000000000000001	femto	1×10^{-15}	quadrillionth	f

TABLE 6. SYMBOLS, NAMES, AND PREFIXES FOR POWERS OF 10.

Electrical units

Each of the parameters used to describe an electric (or electronic) circuit is listed here. For each one, its abbreviation and meaning are given.

Parameter: voltage
Unit of measurement: Volt
Abbreviation: V
Description: The Volt is the unit of measurement of electro-motive force (EMF), which is the same as electrical potential.

Parameter: current
Unit of measurement: Ampere (Amp)
Abbreviation: I
Description: The Ampere is the unit of measurement of the flow of electrical current.

Parameter: resistance
Component: resistor
Unit of measurement: Ohm
Abbreviation: R or Ω
Description: The Ohm is the unit of measurement for resistance, which is the opposition of flow to an electric current in a DC circuit.

Parameter: impedance or reactance
Unit of measurement: Ohm
Abbreviation: Z
Description: Impedance is the total opposition of flow of an electric current in an AC circuit. Reactance describes the effects of capacitance or inductance, and is part of impedance (which can include opposition to flow due to DC resistance plus AC reactance. Impedance

is dependent upon frequency. Impedance is a more general term than reactance; it is often used when describing circuits that have only resistance or reactance, although this is usually stated.

Parameter: frequency
Unit of measurement: Hertz
Abbreviation: Hz
Description: The Hertz is the unit of measurement of frequency, which is the number of times per second an event occurs. The "event" is typically a single cycle of a regular (periodic) waveform. The old unit of measurement of frequency was CPS (Cycles per second).

Parameter: capacitance
Component: capacitor (or condenser)
Unit of measurement: Farad
Abbreviation: C or F (C for capacitance; typically μF, nF or pF for micro, nano or pico Farad)
Description: The Farad is the unit of measurement of capacitance. Capacitance is the property of storing an electrical charge, and capacitors do just that. They also serve to allow AC currents to pass, while blocking DC currents. The reactance (opposition to flow of AC current) decreases as the frequency increases. A Farad is a very large amount of capacitance, which is why component values almost always are expressed in microFarads or smaller units.

Parameter: inductance
Component: coil or inductor or choke (same)
Unit of measurement: Henry
Abbreviation: H
Description: The Henry is the unit of measurement of inductance. Inductors (coils) allow DC currents to flow with low resistance, but offer increasing reactance (opposition to flow of AC current) at increasingly high frequencies. In this sense, the inductor is the "opposite" of the capacitor, which exhibits decreasing reactance at higher frequencies, and blocks DC current.

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