Yorkville

PROFESSIONAL AUDIO BASICS AND USER TIPS GUIDE

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PROFESSIONAL AUDIO BASICS AND USER TIPS GUIDE

Although we are aware of the fact that the vast majority of our users would rather not have to learn anything about Professional Audio, we are also aware that it is a necessary evil. We offer this document for reference by those seeking more information. It is by no means a finished work. Nor does it cover all topics necessary. However, it is a starting point. We hope that you can make use of it and encourage your feedback.

If it gets too technical in spots, we encourage you to keep moving and read the document several times. We also encourage you to print this series of documents out and keep them for future reference. For those of you who are looking for something real basic, check out our "P.A. Basics".

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SYSTEM SETUP

GETTING STARTED

The following is a simplified set of instructions for general applications. The aim is to get a basic system up and running with as few problems as possible.

[1] A BASIC STAGE SETUP GOES AS FOLLOWS:

- (a) Place the front monitors near the edge of the stage, aiming back at the performers.
- (b) Place the mics and stands in front of their respective monitors at a distance of 1 to 3 feet, depending on the cabinet's up-facing angle - the speakers should be aiming directly at the backs of the mics. Other monitors should be located as closely as possible to the performers, also aiming at the backs of their mics. This is to reduce feedback potential.
- (c) Place the main FOH (front-of-house) speakers at stage front at the far corners aiming straight out at the audience. Do not aim them in at the audience in front of centre stage unless the stage is deep enough for the mics to be set farther back to reduce feedback.
  *{ **TIP** - If stage-front-centre audience coverage is a problem, perhaps because they are too close to the stage to hear the FOH speakers, try turning a spare monitor around to face them}.*

[2] CONNECTIONS GO AS FOLLOWS:

- (a) Connect all mics, line-level signal sources - i.e., tape decks, CD players, instrument amp line outputs, etc.- also processors and external effects units, to their respective channel inputs or send and return jacks on the mixer (for more information see INPUTS and OUTPUTS under THE MIXER).

  *{ **TIP** - It is also a good idea to identify the various channel sources, perhaps with small stick-on labels at the bottoms of the channels, e.g., "lead vocal", "guitar", "drum vocal", etc.}*

- (b) Do not connect speakers yet. Transients from things being plugged into the mixer and switched on can cause speaker damage eventually if not immediately. Connect speakers last.
- (c) Similarly, when you are connecting one or more external power amplifiers and speakers, be sure to connect the speakers to the power amp after it has been connected to the mixer and the mixer has been powered up. The reason for this is because some mixers do not have turn-on transient suppression built in. Any power amp(s) which are connected to the mixer and running with their speakers connected when the mixer is switched on can amplify this large burst of signal voltage with a resounding "pop" or "boom" and the speaker system may be damaged. Even if the speakers survive this type of accident the first time, repeated accidents will eventually take their toll.

  *{ **TIP** - If there is a power failure during the job, try to switch off the power amps immediately, before the power comes on again. Mixers with built-in power amps usually don't suffer from "pop" problems when switched on, but it's not a bad idea to turn the master levels off before powering them up.}*

- (d) Electronic crossovers or speaker processors should be connected between the mixer and power amplifiers. The term "between" means mixer output to unit's input and unit's output to power amp's input(s). There are four types of processors which you may encounter. First, there are simple processors which provide pre-equalization for specific speaker systems. They are connected between the mixer and power amp(s) in the same manner as an equalizer. Secondly there are adjustable active crossovers which are connected in a similar manner except their outputs must go to separate power amps or amp channels each driving the appropriate woofers, horns and tweeters, or subwoofers and full-range enclosures.

Remember to find out the speaker manufacturer's recommended crossover frequencies for the various elements in order to set the crossover accurately. If this is not possible, be careful when doing this by "ear". What sounds right to you may be wrong for the components' long-term reliability. To be safe, set the output level controls all at maximum so that they are at the same output level (at lower settings, potentiometer tolerances could cause them to be different) then counter-adjust (lower) the crossover's input level to avoid over-driving the power amps.

The third type of processor is a combination crossover/processor. This would be connected and employed as if it were a crossover, although such units tend to be for specific speaker systems and therefore do not usually have variable crossover frequencies. Now a fourth type of unit has recently emerged from the depths of techno-wizardry to test our connecting skills. The "sensing" processor / crossover / compressor is designed for use with specific speakers in a touring system. In this case, everything gets hooked up as if it were a just a crossover - with one exception, additional speaker cables are run from the speaker outputs on the power amps to "sensing" inputs on the processors. Then, when the system is up and running with the amps set at full output, you push the processor's "calibrate" buttons. A set of tones is sent through the amplifiers and speakers while the "sense" circuitry samples some of the amplifier's output signal and programs the processor's variable parameters such as crossover frequency and gain.

[3] THE SOUND CHECK

It is very important to do a sound check before the job starts. It is also important that the band is playing during the sound check. However if this is not possible, have someone test the mics while you adjust the mixer.

- (a) With the master levels turned off, adjust the channel input gain or attenuator settings so that the input "clip" indicators flash. Now turn them down slightly. This ensures that adequate amounts of source signal are driving the channel circuitry to provide an optimum signal-to-noise ratio.
- (b) If the mixer does not have input gain/atten. controls or input clip indicators, it is a good idea to turn up the channel level or volume controls around one-third before turning up the master levels.
- (c) Now bring up the masters slowly and re-adjust the various channel level controls for the desired mix through the FOH PA.
- (d) When the sound check is finished, remember to turn off channels which only get used once in a while - e.g. harmonica mic channel, acoustic guitar channel, special percussion instrument mic channel, etc. This is to prevent open mics from causing feedback and noise problems. Channel mute buttons are handy for this function (see >>MUTE under THE MIXER). Otherwise you can use a washable marker to mark the settings before turning these levels down.
- (e) Generally speaking, channel EQ controls should be kept as close to flat (centre position) as possible, especially on mic channels. In any case, start the soundcheck with all channel EQ controls at centre. Adjustments should be made for specific purposes - remember, this is a PA system, not a home stereo (amoung other things, home stereos don't have to produce 110 to 130dB SPLs with zero feedback). Possible channel EQ settings could be as follows;
  [vocals - all flat]
  [acoustic guitar - bass flat, mid -3dB, treble +2dB]
  [bass drum - bass flat, mid -3dB, treble flat]
  [snare drum - bass -6db, mid & treble - flat]
  [floor toms - all flat]
  [tenor toms - bass -3dB, mid & treble - flat]
  [cymbals - bass -12dB, mid & treble - flat]
  [trumpet, tenor & alto saxophone - bass -3dB, mid & treble flat]
  [trombone, baritone saxophone- all flat]
  [guitar amp - bass & mid flat, treble -3dB]
  [bass amp - bass flat, mid -3db, treble flat]
  [harmonica - bass -6dB, mid flat, treble -3dB]

- (f) Reverb or echo should be added principally to vocals and in small amounts. Lead guitar, keyboards and horns can sometimes also use small amounts of either effect, but drums and bass guitar should usually be kept "dry" to allow a firm-sounding foundation for the rhythm section.
- (g) Final monitor or aux. send levels should be set according to the artists' needs. To get started however, assume that the vocal channels will need to be loudest through the monitors, acoustic guitar will also require a fairly high monitor send setting, but aside from that, other active channels should have lower monitor settings. Drum mic channel monitor sends may be left off at first, then turned up as required. A secondary monitor system, perhaps for keyboards or drums, or multiple monitor systems could be adjusted similarly at first then re-adjusted according to the various artists' wishes.
- (h) If there are effects-to-monitor masters, remember that reverb or echo should be added to the monitors in smaller amounts than to the FOH system to avoid feedback.
- (i) If you are using a stereo mixer, the channel pan controls should be set at center position. The only time you might set them differently would be if you decided to run a mono PA, i.e. with the main power amplifiers all inter-connected to a summed mono output on the mixer (see >>MAIN, SUM OR MONO under MIXER OUTPUTS). Then you could use the pans to send all the drum mic channels, for instance, to the left submaster, and the rest of the band to the right submaster thus creating two submixes to facilitate level adjustments (hopefully the mixer will have a MAIN, SUM or MONO buss and master to permit this).
- (j) If you're playing a new hall, you might consider pre-equalizing the system against possible future feedback problems. The process is simply a matter of increasing the main levels until you get feedback then EQing it away (see >>EQ under THE MIXER (Master Section Controls)). Then you increase the levels some more and re-adjust the main EQ. You could do the same with the monitor master(s) and EQ, but the band is NOT likely to appreciate it. The downside to this practice is that the system's response is now EQ'd sufficiently to sound strange, and it may get worse because the house acoustics will change when the audience is in place. A better practice is to position the FOH speakers so that feedback even at high volume levels is minimized.

RUNNING THE SYSTEM

LEVEL MAINTENANCE
If you have done all the preparatory setup work during a sound check, you should not have to ride the channel levels very much. The only levels you should need to change would be instrument solos, backup vocals and seldom-used channels that are kept shut off until needed. Lead vocals may need slight level adjustments if the vocalist has a habit of fading back. If they persist in doing this, a possible solution may be to turn that individual down through the monitors, a trick which should cause them to compensate by singing louder and/or getting closer to the mic (just remember to reward them by bringing their monitor level back up).

MONITOR LEVELS
Channel monitor level settings will require some adjustments as well, and roughly the same ones as above. Keep in mind that pre-fader, pre-EQ monitor sends represent an independent mix. So, for example, when the harmonica player is ready to play, you will have to bring that mic channel's level and monitor level up - ditto for horn mic channels, acoustic guitar mic channels, etc. which would normally be left off until needed.

FATIGUE
During the job, you will probably suffer from symptoms of hearing fatigue to some degree. Most common amoung these is loss of high-frequency sensitivity, although low-frequency hearing can be affected as well. Since there are no other symptoms such as discomfort or ringing in the ears, people often assume that the speaker system has lost some high end - that the horns or tweeters have somehow run out of "steam". Their next reaction is often to boost the high-frequency EQ or the HF gain on the crossover. This is NOT a good idea.
Too often there are customers who have just come in from somewhere else and their hearing is not fatigued. As a result, they hear an unpleasantly bright sound which, by then, is probably extra loud as well. Additionally, there is an increased risk of high-frequency feedback and, should you need to boost the FOH level significantly later on, the highs will be the first to run out of headroom and distort, possibly damaging horns and/or tweeters. You're better off to leave the EQ or crossover alone. Your hearing should normalize by the next day. Earplugs are a worthwhile investment.

DEALING WITH FEEDBACK
Assuming everything is done right during the setup and sound check, feedback should not become a problem unless someone gets too close to a speaker with a mic. When that happens, you can probably see which mic it is and know which channel's monitor or PA level to turn down. If it's too close to a monitor, turn down the channel's monitor send level and if it's too close to the FOH speakers pull down the channel's level fader. However, if it's not immediately obvious which mic is responsible, try the following;

(A) If you have set the input channel gains high enough for there to be some clip light activity - see SETUP section 3 item (a) - the light on the channel which is feeding back should be brighter and on more steadily than it was before. Scan the Clip LED's and turn down that channel's monitor send level. If it's a currently unused mic whose channel should be shut off, e.g. a harmonica mic or acoustic guitar mic (did you do an "oops" and leave it on?), any clip light action at all would be an indicator. In this case turn down the monitor sends and pull down the channel fader. But, if you didn't set the gains high enough to use the clip lights, you'll have to make your best guess. Some "usual suspects" include:

- Overhead drum/cymbal mics may be suspect if the drummer has a monitor.
- Try turning down the mic channels' monitor sends. If that works, try using the channel EQ to minimize feedback, but don't deaden the sound too much.
- A singer's acoustic guitar mic might be too close to a monitor.

Since vocal mics tend to pick up some flattop and that gets into the monitors along with the actual guitar mic signal which singers usually want to hear at a goodly level, it's an invitation to feedback. Try turning down the guitar mic channel's monitor send. If your mixer has input phase reversal buttons, try reversing the phase on the guitar mic channel so you can bring the monitor level back up without feedback.

- *{ **TIP** - Mixer input phase reversal is sometimes a very effective way to get rid of certain persistent feedback problems. It works when the problem is that two mics are picking up the same source and feeding it into a monitor close to those mics. By putting one (only) of those two channels out of phase, the offending source signal riding on that channel gets cancelled out by the similar signal in the other channel and the problem is solved. Even if your mixer does not have phase buttons, you can accomplish the same feat by taking one (only) of the offending mic's cable connectors apart and reversing the leads. Putting a channel or source signal out of phase does not affect the sound.}*

(B) The average audience member or performer will put up with feedback about that (snap your fingers) long. If the problem isn't solved by now, go to plan B - haul down the monitor master(s). That should solve the problem, but now the band has no monitors. Bring the monitor masters back up to a point below where they were before so they can have some coverage. Of course if lowering the monitor master levels does not work you'll need to lower the main masters.

(C) Now it is important to find a quick remedy which will let you get the main or monitor levels back up to where they should be. First, go to your main or monitor EQ. Pull down a few of the sliders slightly (-3dB) in the frequency range which your ears tell you is likely to be the right one. Now ease up the master. If the feedback starts again, lower the master a little, re-centre the EQ faders you just pulled down and try pulling down some other frequencies then bringing the master back up. Eventually, and hopefully soon, you'll have it under control, but now the main or monitor system frequency response has been altered and probably doesn't sound right. Try carefully pushing some of the EQ sliders back up towards centre position - you need to normalize the EQ as much as possible. DO NOT PULL DOWN ALL THE EQ FADERS AT ONCE. That would be about the same thing as lowering the mixer masters, only much more time-consuming and it might even cause new feedback problems later on.

(D) The problem remains that a mic and a speaker have decided to feed back. You still need to find these two culprits then re-position them or insert an EQ in that mic's channel in order to solve the problem properly. This can be done later, but it needs to be done.

(E) There may actually be situations where pulling down the main AND monitor masters fails to end the howling completely. Likely suspects would include feedback from a spare electric guitar and amp or an electric/acoustic guitar and amp waiting to be used and mistakenly left on.

- *{ **TIP** - If a guitarist or singer/guitarist insists on leaving their standby electric-acoustic guitar plugged into an amp with it turned on and the volume left up, try (at least) to encourage them to leave their pick in the strings that resonate. It's a simple solution, but it requires knowing which string(s) to deaden. Hint - with flattops, it's most often the low E, A or D string. A better solution, of course, is for them to leave the guitar's volume control turned off or, if it doesn't have one, to turn off the amp's volume control.}.*

And further suspects could be:

- a stuck keyboard note;
- a clavinet or other stringed keyboard instrument and keyboard amp feeding back;
- an unused instrument mic plugged directly into a powered monitor or combo amp, mistakenly left on and feeding back into it.

Any of these problems are impossible to cure from the mixing station. Someone onstage will have to turn down the offending volume control or free-up the stuck key or move the mic - or whatever.

SIGNAL FLOW

It is beneficial to start looking at PA from the standpoint of how signals move within the system. That way, the details will probably make more sense.

A>B>C

A PA system contains three basic elements:

- [A] **The signal** source which can be a microphone, a tape deck, a CD player, an electric instrument, the Line Out from an amp, the output of another mixer or basically anything which produces an audio signal
- [B] **The amplification stage** which includes the preamplification (mixer) stage
- [C] **The speaker system**.

At all times the basic signal path is from A to B to C. To make the term "signal" a little more graphic, think of electrons. The signal source produces a small quantity of electrons. The amplification stage then adds many more and the speaker utilizes this enlarged amount of electrons to create the physical motion which produces sound.

This A>B>C concept is worth keeping in mind because each stage after the signal source has a certain maximum capacity. In mixers, power amplifiers and preamp devices this is called headroom and in speakers it's called power capacity. Try forcing too many electrons into something and you exceed that capacity, usually with audible results such as distortion or blown speakers.

"Clip" indicators, such as LED's, help you to prevent distortion by telling you when a circuit in a mixer, signal processor - reverb, EQ, etc, - or a power amplifier is on the verge of being overloaded. Unfortunately there is nothing similar for speakers. They simply distort and/or self-destruct unless they have fuses or circuit breakers built in.

As mentioned above, the PA process begins with a source signal. The source can be any one of many different things from an electric instrument to the line-level output of an instrument amplifier, a tape deck or CD player or the output of another mixer. But the microphone represents both the most commonly used signal source and the one which requires the most understanding.

MICROPHONES

BASIC TYPES
There are two basics types of microphone commonly used in pro audio, "dynamic" and "condenser". A third type, the "ribbon" mic is sufficiently fragile that they seldom get used in live music situations.

DYNAMIC MICROPHONES
The dynamic mic is most commonly found in PA applications due to its general ruggedness and simplicity of use (no need for phantom power or batteries). It works rather like a speaker in that there is a diaphragm attached to a coil of hair-thin insulated wire flexibly suspended in a magnetic field. Sound waves set the diaphragm and coil in motion vibrating back and forth which causes the coil to cut lines of magnetic force, thus a small amount of voltage is induced in the coil.
The voltage varies in polarity with the frequency of the sound waves and in strength with the amplitude or size of the waves (the louder the sound, the bigger the waves and the farther the coil moves hence cutting more lines of magnetic force and generating more voltage). This voltage travels down the mic cable to the mixer where it is amplified and sent to the speaker.
For what it's worth, a speaker works exactly the same way only in reverse - it reacts to the amplified signal by vibrating back and forth to create sound. In fact, dynamic microphones and speakers are almost interchangeable. Believe it or not, you can connect a raw speaker, a woofer for example, to the line input on a mixer and hook the mic up to the amplifier outputs. Talk into the speaker and sound will come out of the mic. It won't work very well and you may promptly fry the mic, but this backwards PA will actually function (briefly).
Dynamic mics are best for close-up use whether for vocals, instruments or instrument amplifiers. Certain models are also preferred for bass drum and others for brass instruments.

CONDENSER MICROPHONES
Condenser microphones offer high sensitivity and smooth frequency response. They operate on a small amount of DC voltage either from a built-in battery or a "phantom" power supply unit, or from the mixer if it has phantom power built in. This is deposited as positive and negative charges on two thin metal plates with a small airspace or other resistive material between them. This forms the diaphragm cartridge. Sound waves cause the top plate to vibrate which alternately compresses and de-compresses the resistance. It acts as a dielectric and a signal voltage is produced that varies in polarity and amplitude with the frequency and amplitude of the sound waves. This travels down the cable to the mixer and is amplified. It is worth noting that the phantom voltage will not harm most dynamic microphones if they are connected to a mixer which has this feature built in - nor will the sound be affected.
Condenser mic technology is ideal for virtually all applications with the possible exception of bass drum. Certain models are designed to pick up sounds at a distance or groups of people, choirs for example. Other condenser mics are first choice for acoustic instruments, especially guitar, banjo, mandolin, violin, upright bass, piano or anything with strings. They are also preferred for overhead coverage of drum sets. At one time it was thought that condenser mics were too fragile for PA applications, however they have greatly improved over the years in that regard with many models now designed for this kind of work which virtually equal dynamic mics for road-worthiness.

PICKUP PATTERNS (a.k.a. "Directionality" or "Polar Response")
Most microphones are capable of picking up sounds approaching from a wide area, however they don't pick all of them up with equal sensitivity. The all-important midrange and high frequency sounds approaching from outside a mic's pickup pattern will be detected at far lower sound pressure levels those which are approaching from within the pattern and will get drowned out by them.

- Pickup patterns can be imagined as invisible balloons, each with a particular shape depending on the microphone's design. These shapes are what you see listed as "polar patterns" in mic literature. Although the polar plot diagram is flat-looking, in reality mics pick up sounds coming from above and below as well as the front and sides and even the back. Hence polar patterns are 3-dimensional and really do resemble variously contorted balloons.

- **Uni-directional or "cardioid"** mics can pick up a wide spectrum of frequencies over roughly a 120 to 150-degree sound field. Their polar pattern resembles the aforesaid balloon with the head of the mic pressed against one end. In practice this means that they tend to largely ignore sounds approaching from behind making them preferable for vocals and general PA use because they do not readily feed back into stage monitors, provided the monitors are properly positioned.
- **"Hyper-cardioid" and "super-cardioid"** mics are included in the uni-directional category and generally offer a somewhat narrower sound field - or, to put it in more common terms, they exhibit more "directionality". Uni-directional mics may be dynamic or condenser-type.
- **"Omni-directional"** mics accept a wide range of frequencies from virtually a 360-degree, spherical sound field. Their polar pattern resembles a balloon with the mic right in the center; in other words they pick up sounds from behind almost as loudly as from in front, above and below and the sides. As a result, omni-directional mics are less suitable for high-volume music applications and probably should not be used with monitor speakers at all. Omni-directional mics may be dynamic or condenser. Possible applications include recording round-table meetings or audience ambience.
- The **"boundary" or "plate"** mic offers a 180-degree, "hemispherical" polar pattern, i.e. resembling a balloon which is flat on one side. It is designed to be attached to a flat surface which effectively bounces sounds from all over the field into the tiny mic's diaphragm. This flat surface can be anything from a sheet of Plexiglas to a wall to the underside of a grand piano lid or the top of a desk. The boundary mic's best applications include grand piano, acoustic instrument groups, choirs, opera, stage plays and meetings. They are usually condenser-type.
- **"Shotgun"** microphones have a very narrow sound field reflecting a "hyper-cardioid/lobar" polar pattern which usually resembles a small, horribly contorted balloon with the mic partly pushed into it. Unlike the others, these are always condenser mics using a long interference tube which cancels sounds from the sides. Shotguns are used to pinpoint distant sound sources and are principally found in the film and television industries. They seldom get used for music PA as they have to be aimed at the source which means someone has to hold and aim them. As a result they may pick up monitors and thus feed back. However, there are PA applications for them where stage monitors do not get used as a rule including stage plays, also choir or chorale performances where soloists need amplification.
- The **"parabolic"** mic, which is similar in terms of its applications, features a cardioid mic suspended in front of a plastic dish or parabola, aiming into the centre of it. Parabolic mics also must be aimed at the sound source, however they can be either dynamic or condenser-type and tend to be less expensive than shotguns. Both mics usually have variable directivity - their "pinpoint" factors. In shotgun mics with this feature it takes the form of a small control or switch. In parabolic mics you adjust the distance between the tip of the mic and the centre of the parabola. These microphones have been replaced to a degree in live theatre by miniature wireless mics which can be secreted in the costumes, but they are still required for film and TV applications and those stage shows where costumes are sparse.

HIGH OR LOW IMPEDANCE (a.k.a. "Z")?

Once upon a time, all PA mics were high impedance. This was because PA amplifiers of the day only had high Z inputs and were always situated close to the stage so that mic cable lengths seldom exceeded 20-odd feet. It wasn't until after "Woodstock" and the birth of concert PA technology that there was an increasing demand for mic cables long enough to warrant the need for low Z technology. Since then low Z has become the PA industry standard. High Z mics are now principally used with home entertainment equipment, short-wave and CB radios and commercial PA amplifiers as used in factories, hotels, hospitals, etc.

PLACEMENT
There are two variables to consider when placing a microphone relative to the sound source - distance and angle. Distance is the most important factor as it determines a variety of things including signal level, clarity, exclusivity (how many sound sources get picked up) and even bass response in some mics. Hand-held vocal mics, for example, should generally be kept around 5 inches from the mouth. On a crowded stage with the band playing very loudly, this distance may have to shrink to just an inch or two so that the mic will pick up a predominance of that performer's voice or instrument. At this distance however, certain mics produce a distinct increase in bass response due to proximity effect. Some vocalists prefer that sound even though it may be accompanied by "pops" and other sound effects including distortion.

- *{ **TIP**- If a microphone is creating "thumps" or "pops", turn down the low-frequency or bass control on that channel to reduce the problem. Some vocalists may like these noises, but audiences can find them irritating. Distortion is more difficult to cure with the mixer - try reducing the input Gain, Trim, Attenuation or Pad setting on that channel. If that doesn't work, try replacing the mic with one which has a little lower sensitivity. When you have a chance, try replacing the distorting mic's diaphragm - the old one may be fatigued.}.*

Angle determines tone. Generally you will obtain the brightest tone with the mic aimed directly at the sound source. A softer, mellower tone can be achieved by angling the mic in relation to the source.
Here are a few standard microphone placement suggestions:

- **Lead vocals** - see above paragraphs [dynamic or condenser, cardioid]
- **Harmony vocals** - same as above for individual mics, slightly farther away and at roughly a 45-degree angle to their mouths if two singers are on the same mic [dynamic or condenser, cardioid]
- **Acoustic guitar** - aim directly at the bridge, not the sound hole (too "boomy") and get as close as possible [condenser, cardioid]
- **Saxophone** - place the mic roughly 3 inches above the bell and angle it slightly [dynamic, cardioid].
- **Trumpet, trombone, etc.** - distance the mic a foot or so away to avoid overloading and aim directly at the bell [dynamic, cardioid].
- **Acoustic upright bass** - at a distance of roughly 4 to 6 inches, aim directly at a point 1/2-way between the bridge and one of the F-holes [condenser, cardioid].
- **Guitar or bass amp** - place the mic close to the grill aiming directly at the centre of the speaker [dynamic, cardioid]
- **Keyboard amp** - these usually have a 2-way speaker system which makes miking them complicated. You could aim two dynamic cardioids directly at the woofer and tweeter then adjust the two mixer channel levels for low and high frequency balance, but your best bet is to take a "line-level" output from the amp and run it directly into a Line input on the mixer (see >>LINE under THE MIXER, MIXER INPUTS).
- **Grand piano** - usually this is a job for two condenser cardioid mics placed inside, one pointing down at the middle of the bass string section above the hammers and the other over the high strings. Another approach is to affix a plate or boundary mic to the underside of the lid, somewhat closer to the high strings than the bass strings and above the hammers. In a rock band situation, keep the lid closed as much as possible and experiment with mic positioning (the pianist will have to hear himself through the monitors).
- **Bass drum** - close to the front drum head or preferably inside the drum [dynamic, cardioid]
- **Snare drum, ride cymbals, tenor and floor toms** - as close as possible over the sources, aiming down at the tops. Wherever possible try to minimize the number of mics in use as each open mic reduces the system gain before feedback by 3dB. For example, try using one or two condenser mics over the tenor toms and ride cymbals plus one over the floor toms, one for the hi-hat cymbals and one for the snare. Five or six mics including the one for bass drum should suffice for most standard sets. [condenser, cardioid or hyper-cardioid]

SOURCES

LINE-LEVEL SOURCES

Anything which provides an audio signal of at least three-quarters of a volt (775 millivolts) can be considered a line-level source. Such a tiny amount of electricity is still several times greater than anything a microphone can generate and greater again than the output of the average guitar pickup. However, line-level signals are much less powerful than the output of an amplifier, even the tiny variety which powers headphones.

It is important to keep these facts in mind because you may be connecting a wide variety of sources to the mixer and all of them will require specific input gain adjustments. "Speaker" outputs however, should not be connected to a mixer at all unless first put through a "direct box" with a speaker-level input and a line-level output. "Headphone" outputs have two problems; first they are speaker-level which means that they may distort the mixer input, but the signal is much weaker than the output of a regular amplifier so you may find that a direct box attenuates (reduces) the signal too much. The other problem is that headphone outputs are usually stereo so you will either need a stereo direct box and a mixer with stereo inputs or a stereo-to-dual-mono adapter so that the signal can be split into left and right mono signals and fed to separate mono direct boxes and then to separate mixer channels (or you may decide not to use the headphone jack as a line output - a wise choice).

Here is a list of line-level sources and the outputs to use:

- Tape deck or CD player - "Aux." output
- Keyboard instrument - "Line" or "Aux." output.
- Instrument amp - "Line" output.
- Another mixer - "Main", "Aux.", "Sub", "Mon" or "Send" output.

For information regarding line-level connections, see "LINE" under MIXER INPUTS.

THE MIXER

The various features of an audio mixer can be broken down into several sections in order to simplify explanations. The sections break down as follows:

THE MIXER - INPUTS

"MIC"

The microphone input is usually low-impedance, i.e. 5000 ohms or less in mixers with active input circuitry or approximately 600 ohms in mixers with input transformers (Note: there is no problem with plugging a 600 ohm or lower impedance microphone into a 5,000 ohm input). It will usually have a 3-pin "XLR" connector and it will always be a female XLR (male XLR's are always outputs). Mic inputs may also be 1/4-inch "phone" jacks in older or smaller mixers, however they may be high-impedance (over 5000 ohms) so check the owner's manual. A high impedance ("HI Z") mic may produce a low level or distorted sound when connected to a low impedance ("LOW Z") input. A low Z mic in a high Z input can produce similar results. If you have a low Z mic and only high Z inputs are available, a "low-to-high Z" adapter can be used to overcome the problem (e.g. Yorkville model LHNT-1).

The channel input gain control or input attenuator on mixers with such features, will always regulate the "mic" input. And you may find a phantom switch near the Mic input or somewhere in the master section. This feature applies a small amount of DC voltage to one of the Mic input connector leads - usually 24 - 48 Volts. This travels backwards up the cable to the mic (the only time electrons flow "upstream") for the purpose of powering a condenser mic.

"LINE"

Line inputs are always high-impedance (10,000 ohms or more) and employ 1/4" jacks or sometimes RCA (phono) jacks. This is where you would connect a high-impedance mic or the line-level output of a signal source. The term "line-level" is used to cover signals referenced to 0dBm or 0.773 Volts rms. These are much greater than those produced by mics or instrument pickups which produce signals down around 0.1 Vrms or less. The range of things which produce line-level output signals is very wide. Almost, anything which runs on AC or DC power produces this level of signal. The exception would be the output of a "phantom" power supply which is mic-level. Beware of "speaker" outputs or "extension" speaker outputs as even the smallest amplifier produces too much power for a "line" input and will overload the circuitry. Also, remember to use shielded cables with a centre lead wrapped in the shield wire (as opposed to speaker cables which only have two regular leads). They help minimize hum and noise when connecting line outputs to line inputs. The channel input gain control or input attenuator on most mixers will regulate the "line" inputs and the input circuits have sufficient gain for high-impedance microphones. You may need to set the Input Gain control higher for high Z mics than line-level sources but some mixers have Mic/Line selector switches so that the Gain settings can be roughly the same for mics and line-level sources.

"BAL. LINE"

Balanced line inputs, when connected to balanced outputs, offer greater degrees of protection from hum and noise. Some of them have three-pin XLR connectors, but the majority these days tend to have stereo 1/4-inch jack sockets. In this case you use balanced shielded cables which are outwardly similar to regular shielded patch cords, but the wire contains two centre leads plus shielding and the 1/4-inch plugs will be stereo, i.e. with two insulator bands near the tip rather than one.

The standard lead designation is tip=in phase, ring=reverse phase, sheild=ground (although there may be a few exceptions). XLR-type balanced patch cords - mic cables can be used for this - are usually designated pin 1=ground, pin 2=in phase, pin 3=reverse phase. It's worth noting that, in a situation where the output is unbalanced but the input is balanced, a regular patch cord will usually work (ditto the reverse situation). Additionally, Audiopro AP-series mixers feature special balancing circuitry which provides the noise rejection of a totally balanced circuit even when the signal source's line output is unbalanced. Simply use a balanced patch cable.

"INSERT"
The insert jack is usually found amoung the channel input connectors and sometimes among the main and monitor outputs. It is a 1/4-inch stereo jack socket which is actually two jacks in one - a send and a return. This unusual setup is used in place of two separate jacks to save space (some mixers have separate send & return Insert jacks). Using a special "Y" cable with a stereo 1/4-inch plug branching out to two mono 1/4-inch plugs (e.g. Yorkville model PC-6ISPH), you can patch a graphic EQ, compressor/limiter, aural exciter, digital delay, etc. directly into a channel or main/monitor buss. In some cases, the patch cable wiring code is tip=return, ring=send, sleeve=ground, while in others, the wiring goes tip=send, ring=return, sleeve=ground. Check your owner's manual to find out which one your mixer uses. In most mixers, the "send" part of the Insert jack is buffered - in other words, regulated by the channel Gain control.

- *{ **TIP** - If your wiring is like the first example with the tip being "send", this means that a second mixer, perhaps for monitors, recording or broadcast, can be added on, channel-for-channel, assuming the Insert jacks are not being used for patching purposes. You would simply run shielded, unbalanced patch cords from the Insert jacks to the Line inputs on the other mixer.}*

"AUX"
Located in the master section, the Auxiliary inputs are always line level and often in stereo pairs, but may or may not be balanced depending on the mixer (AP mixer aux. inputs are stereo, unbalanced). The mixer will usually have an "aux." control in the master section to regulate this input and the signal's final destination will be the "main' mixer channel or one of the "sub" master channels. The purpose of an "aux." input is to permit connecting an additional signal source - perhaps a small outboard mixer for keyboards or drums - without using up one or two input channels.

"EFX RETURN" or "STEREO RETURN"
The effects or stereo return jacks are usually employed as part of the effects "loops". They accept the output of external units - reverb, delay, etc. - and feed it to the mixer's master RETURN controls which regulate its passage to the next master mixer stage. Effects return jacks are usually unbalanced 1/4-inch jack sockets. On stereo mixers, they would be in left/right pairs in order to accept the output of stereo effects units. Alternatively, these can be used as stereo auxiliary inputs, perhaps for connecting a keyboard mixer.

- *{ **TIP** - The main reason for having multiple EFX or Stereo Return input facilities is to permit you to employ multiple effects "systems", one for each group of input channels. You could, for instance, have echo on the vocals but only reverb on the rest of the band (echo on everything can get too 'muddy' sounding at high volumes)}*

"AMP IN" or "PA IN"
Powered mixers often have a direct input to the built-in power amp. This is a special 1/4-inch jack socket with a built-in switch that interrupts the flow of all the internal mixer signals to the power amp as soon as you plug into it. Now only the signals which are coming in through the Amp or PA In jack can be amplified.

- *{ **TIP** - This means that, if you are using an external power amp to drive the Front-Of-House (i.e. main) speaker system, you can patch the mixer's Monitor output into the Amp/PA In jack and simply connect your monitors to the powered mixer's speaker outputs.}*

"TAPE IN"
Tape inputs are most often RCA ("phono") connectors and basically represent the same thing as "Aux." inputs. They are usually found amoung the Main, Monitor, Sub, etc. connectors and will have a Tape In level control among the masters.

"TALKBACK"
Some mixers feature a very simple channel consisting of a mic input, a level control, possibly an on/off button, and a line-level output. This is included to enable the mixing technician to address the lighting technician, a stage hand or road manager through a small amp/speaker system. Some mixers may have the ability to also send some of this signal to one or more of the Aux./ Monitor busses so that the performers may be addressed through the stage monitors. The Talkback input is normally a standard, female XLR mic connector.

THE MIXER - OUTPUTS

"MAIN", "SUM" or "MONO"
The main, sum or mono output represents a mono signal, usually with its own master and is normally a balanced output (see INPUTS "BAL LINE" for cable wiring). On stereo or multi-buss mixers, it represents the sum of the "Sub" or "Group" buss outputs - in other words, part of the output signal from each Sub or Group buss (channel) goes to the Main, Sum or Mono buss and gets mixed (summed) with the others. In older mono mixers, the Main output is exactly what the name implies and would be connected to the main PA amp/speaker system. In stereo mixers, it can be used for a variety of purposes including main PA, or it can be connected to a remote amp and speaker system, perhaps for the control booth or for recording purposes (remember to split the signal with a "Y" adapter to get it on both tape tracks).

- *{ **TIP** - If you have a stereo or multi-buss mixer, one of the possible uses for a Main, Sum or Mono output is to feed the main F.O.H. (front-of- house) amp/speaker system. Since stereo separation does not work as well for PA as it does at home (people at the sides of the stage may end up hearing only guitar and no vocals, etc.), a mono main mix can work as well or better than stereo in some places. If you do decide to go with a mono F.O.H. PA, the "Sub" or "Group" channels can be used as submixes - say, Sub 1 for drums and Sub 2 for the rest of the band in a stereo mixer. The various drum channels' Pan controls would simply be turned all the way Left, for example, and the rest turned Right. Now the Sub or Group One master would regulate all the drum mics at once and the Sub or Group Two master would regulate the rest of the band. The Main, Mono or Sum Master would now regulate the overall level through the PA. In a multi-buss mixer there would, of course, be more submix possibilities. As an example, a four-buss board could accommodate separate vocal, drum and keyboard submixes plus one for the rest of the band.}*

"SUB or GROUP"
The submix or group outputs are regulated by their own, similarly numbered masters. Stereo mixers have two which would be used to feed a stereo F.O.H. amp/speaker system and they are balanced as a rule (see INPUTS "BAL. LINE for cable wiring). In a multi-buss mixer where there are more than two sub/group channels, these outputs are often used in conjunction with Sub or Group In jacks to create loops with EQ's, compressor/limiters, effects units, etc. which would be dedicated to the needs of the vocals, instruments or program sources being sent to each buss.(Use of the word 'channel' can get confusing when discussing mixers, therefore in this writing it will be reserved for the input channels only. "Buss" is being used here to represent any mixer circuit which receives signals from the input channels. Expect to see more terms such as "effects buss", "monitor buss", "main buss", etc.).

"MON"
The monitor outputs, like the Main and Sub/Group outs, are usually balanced (see INPUTS "BAL LINE" for cable wiring) and have their own masters. These would be connected to the monitor amp/speaker system.

- *{ **TIP** - Alternatively, one of the monitor outputs could be connected to the Aux. or Line inputs of a cassette deck with a "Y" adapter so that the signal gets onto both tape tracks. Now the channel Mon.send controls and Monitor master can be used to provide a recording mix while the rest of the board is being used for PA. Tape or Aux. inputs can be used to connect the outputs of the deck to the mixer for playback listening.}*

"AUX"
The auxiliary outputs are also usually balanced (see INPUTS "BAL LINE" for cable wiring) and, like the others mentioned above, have their own masters. As a rule, they are connected to monitor amp/speaker systems, in fact mixers with full Aux. facilities may not have "monitor" facilities at all. And, like a Mon output, one of the Aux. outputs can be used for recording (see above). Also, by combining them with Aux. In jacks, you can create separate "loops" with EQ's, compressor/limiters or other signal processing units which will now be dedicated to those channels being sent to each Aux. buss.

"TAPE"
The tape outputs are usually RCA (phono) type and are not balanced. A Tape Send or Tape Out control may be among the masters to regulate the level. If there is no such control, check the owner's manual; chances are the Main or Sub stereo masters control them.

- *{ **TIP** - If the Tape outputs on your mixer don't have their own master, they're most likely just wired in parallel with the Main or stereo Sub outputs and are regulated by those masters which means that any FOH-system level changes will be reflected in the recording levels. You might want to have someone watch the tape deck's meters and counter adjust its record level control(s). An alternative might be to connect a compressor/limiter (e.g., ART's model SC-2) between the Tape outs and the deck's inputs and set it for "soft knee" compression. Now it will act rather like an automatic volume controller.}*

"EFX SEND"
The effects send jack is almost always 1/4-inch and may be balanced or unbalanced - check the manual to be sure, however remember that a regular (unbalanced) shielded patch cord will almost always work with a balanced output or input, you simply won't get the extra hum & noise cancellation that balancing provides (you may not hear the difference).

- *{ **TIP** - Do not connect equalizers, compressors or crossovers to the EFX SEND jack. The effects buss in any mixer is always in parallel with the main busses, not in series with them. As a result, only half of the signals go out through the loop, the rest going straight to the main busses. What finally comes out is a mix of straight and effects-signals. While this is ideal for reverb or echo, you need to put 100% of the signal through EQ's, compressors, etc. For more information, see under "Processors".}*

"TALKBACK"
This is the output of the Talkback channel. It will be Line level and may be balanced. See under "Mixer Inputs" for more details.

<u>THE MIXER - CONTROLS</u>

For the sake of clarity, channel controls will be covered from the top to the bottom of the input strip. This is also appropriate because signal flow through the channel circuitry is from top to bottom. (On "box"- style mixer/amplifiers, this is usually reversed because the inputs are at the bottom of the mixer panel).

(INPUT CHANNELS)
"GAIN", "TRIM", "ATTEN" or "PAD"

All sources put out different amounts of signal. Some may be weak and others quite strong. It is thus important in larger mixers to be able to adjust the amount of source signal entering the channel. If there is too much, as stated earlier, distortion will result. On the other hand, if there is too little, the channel Level may have to be set much higher than the rest of the channels and the signal-to-noise ratio on that channel will be less than ideal.

* *{ **TIP** - About "signal-to-noise"; all audio circuits, even in the most expensive mixers, have a certain level of ambient noise caused by electro-magnetic emanations from all the wiring and equipment nearby. This noise gets amplified along with everything else in a mixer. If the signal is very weak, the channel Level will have to be increased more than the others thus amplifying the noise so that it ends up competing with the signal instead of being drowned out by it. Ideally, the signal should always be much greater than the noise, hence the Gain control is equally as valuable for boosting a weak input signal up to the proper level as it is for reducing the input signal when it's too strong.}*

At the top of the channel strip, some mixers will have a switch marked "- #dB" (minus some number of decibels) or "Pad" . This also is included to help you adjust the input sensitivity for the source signal. The switch will create a very large difference in the signal level corresponding, for example, to the huge difference in signal output between mics and CD players. The Gain or Trim control provides more of a fine-tune capability in accomplishing this task.
A few of the more recent mixers only have a Gain or Trim control, no switch. This is possible because active circuitry replaces the input transformers and, as a result, they have such large amounts input headroom and Gain control range that a "pad" is not necessary. One way or another, the channel Clip indicator should be your guide to setting the Gain/Trim/Atten./Pad. While a signal is applied to the input, increase the setting of this feature until the Clip LED flashes, then decrease it slightly.

"CLIP"
In the absence of a channel VU meter, the input clipping indicator is your best aid for setting the Gain/Trim/Pad. This LED is designed to illuminate when the input signal is approaching the upper limit of the input circuit's capacity, but still leaving around 3dB of headroom in most cases (check the manual to be sure). It is thus possible to set the Gain controls simply by watching the channel Clip indicators during a soundcheck and adjusting them for slight amounts of activity.

"MON." or "AUX."
Depending on the mixer design, the Monitor and/or Auxiliary send controls may come next. On mixers with "Pre/Post" EQ selector buttons for these controls, they will come after the EQ section, otherwise they will be right after the Gain, Trim, etc. In order to avoid confusion about how "send" controls work, here is a brief explanation; each channel is capable of sending some of its signal via the internal circuitry to various locations (busses) within the master section. In order to do this there needs to be channel controls to regulate the amount of signal going to each buss. It is not always ideal to have these signals affected by the channel EQ controls since that EQ is there primarily for regulating each channel's sound through the main PA which has different frequency response than the monitor amp/speaker system.

* *{ **TIP** - The stage monitors operate in a terribly demanding acoustic environment - speakers are close to mics and everything tends to be very loud. As a result, the best way to mix for monitors is to treat them as a totally independent system. Large concert PA's usually have a separate monitor mixer and someone to run it. Smaller systems still need to treat the monitor mix as separately from the main mix as possible. That is why the channel signals would not, as a rule, be EQ'd before being sent to the Mon/Aux. busses. That way, the only equalization they get will be specifically for the stage monitor system.*
  The reason for there being more than one Mon. or Aux. send control on the channel and more than one Mon. or Aux. buss is so that you can mix for more than one monitor system. The drummer, for example, usually needs to hear himself and the vocals extra loudly, and the vocalists, of course, need to hear themselves very loudly while the guitarist might want to hear a predominance of bass and keyboards because his amp is almost all he can hear.}

"EQ"
As stated above, the channel equalization is usually desirable only on that portion of the channel signal headed for the FOH system. This is based on the assumption however, that you are using the channel EQ to improve the sound or to get around feedback problems which are exclusive to the FOH PA - not always the case. Some source signals require basic EQ adjustments to sound "right" whether it's through the mains, monitors, on tape or for broadcast. Harmonica mics, for instance, have to be EQ'd to minimize low-frequency puffing and thumping sounds as well as feedback. For that reason, some mixers have Aux. or Mon. send controls after the EQ with "Pre/Post" selector buttons to put the desired ones through the channel EQ ("Post") or to bypass it ("Pre"). In other mixers, one or more of the Mon/Aux. controls may simply be after the EQ (i.e. "post EQ") and are therefore permanently affected by it.

- *{ **TIP** - All EQ's function by altering the gain above or below normal over various frequency ranges. As a result, when it comes to setting the channel EQ - or any EQ - there's a golden rule which says "NEVER OVER-EQUALIZE". This is worth remembering because the way you 'think' things should sound and the way they really should sound to ensure that the system works properly all night are not always the same. If the main speaker system has fairly linear frequency response, resist the temptation to "sweeten" the sound, it could save you headaches later on when the SPL (sound pressure level) eventually goes up and the room acoustics begin changing - more about that later.*
  Some mixers offer "semi-parametric" EQ. This usually comes in the form of one or more cut/boost controls, each with a frequency control to position the cut/boost exactly where you want it along the frequency spectrum. One application of such a feature is in the fight against feedback. Here you would turn the cut/boost control counter-clockwise to produce a "dip" in the frequency response, then rotate the frequency control until the dip reaches the guilty frequency and the feedback is reduced.}

EQ "SWEEP" Control
Although frequency "sweep" controls have graced the channel EQs of recording mixers for many years, they are only found on the more upscale PA mixers. As a result many PA users, even veterans, are unfamiliar with their function. The SWEEP control determines what range of frequencies is affected by the MID cut/boost. It moves or "sweeps" the MID control's peak or notch in response all the way up to several thousand Hz or down to below one hundred Hz. As a result it can have quite a noticeable effect on the sound especially since the MID cut or boost will be interacting with whatever cuts or boosts you may have set with the LOW or HIGH EQ controls.

- *{ **TIP** - If you have set a LOW boost, a MID boost swept all the way down to the lowest frequency setting will alter the sound of lows AND increase their volume. Be careful this doesn't damage your woofers and watch out for your tweeters/horns if you sweep the boost up to the higher settings while the HI EQ is boosted}.*

Considering that the SWEEP control can alter everything you are accustomed to an EQ doing, it would be worthwhile to spend some time becoming aquatinted with how it works. As music plays through a channel on the mixer and speakers, adjust that channel's MID, first for a boost then for a cut and SWEEP them back and forth. (If there is no MID cut or boost setting, i.e. if it is set at the centre position, the SWEEP will have no effect at all). Now repeat the process with that channel's LOW and HIGH EQ controls at various settings {but with the volume at a safe level for the speakers}.
Together, MID and SWEEP controls can be used to accomplish a variety of tasks from combating feedback to improving the way things sound through the PA or on recording. Here are some of those tasks & settings:

- Killing feedback; set MID at -6dB and slowly rotate SWEEP until the feedback stops. If needed cut Mid further.
- "Bonky" sounding snare drum; -6dB @ 200Hz (and roll off LOW EQ -6dB)
- "Boomy" bass drum; -6dB @ 300Hz (with LOW EQ at +6dB & HIGH EQ at +3dB)
- "Fwashy" sounding cymbals. -9dB @ 300Hz (roll off LOW EQ -15dB)

- Excessive hiss from guitar, bass or keyboard amp; +3dB @ 5kHz (with HI EQ rolled off -9dB)
- Fading vocal range (notes too low for singer); +3dB @ 80Hz (with LOW EQ rolled off -6dB)
- "Puffing" on harmonica mic; -9dB @ 80Hz (with LOW EQ rolled off -12dB)
- Rack Toms; -3dB @400 Hz
- Floor tom; -6dB @ 200Hz

{Note: These are ***approximate*** *settings only. Use them as a starting point and "tune around" them.}*

Generally speaking, you will probably end up with the MID in cut mode for most problem-solving uses of the SWEEP control. In any case you will learn to use this feature judiciously. The best PA EQ setting is the one with the LEAST adjustment, but when you need to solve a problem it's good to know how to use the tools.

"EFX"
Unlike the MON/AUX send controls, the effects send controls are always post-EQ and post-fader, i.e. they are affected by both the channel equalization and the channel fader (in PA vernacular they are "post-post"). There may be more than one EFX send control and they may feed either an internal effects circuit (reverb) or a master Effects Send buss, check the owner's manual if there is more than one EFX control on each channel to see which one is which. In any case, the channel EFX signals are internally routed to their designated master effects summing busses where they are mixed together on their way to the EFX SEND jack or internal effect.

- *{* ***TIP*** *- When mixing effects such as reverb or echo, don't overdo it. Most halls, clubs, etc. have at least some natural reverberation and the sound can become ill-defined or "mushy" if just a little too much reverb/echo is applied. In places with an audible echo, no matter how short the duration, you are probably better off not to use any reverb/echo at all.}*

"PAN"
The Pan control, found only on mixers with stereo Main outputs, functions a bit like the "balance" control on a home stereo system. In fact it regulates how much of the channel's post-EQ signal gets routed to either the Left or Right Main PA busses. If, for example, the Pan control is rotated all the way left, that channel's signal will only go to the left Main buss. If the F.O.H. (main) PA is stereo, only the speakers on the left side of the stage will be producing that channel's output - not an ideal situation.

- *{* ***TIP*** *- In most PA situations, the only real reason for running a stereo F.O.H. system is to get the sonic benefit of a stereo reverb. When you consider that the natural hall reverb is likely to muddy this effect and you aren't likely to be using a lot of reverb anyway, you have to wonder what the PAN controls are good for, other than certain recording applications, e.g.. "positioning" certain sources in the soundfield . However, if you have a basic stereo mixer with a "Main" master and corresponding mono output, and you are running a mono F.O.H. system, the PAN controls can be used to establish two main mixdowns, perhaps one for drums and the other for the rest of the band. With the drum channels panned left (for instance) and all the rest panned right, the Left submaster fader now becomes the drum submaster, the other becomes the band master and your mono Main fader regulates overall level.}*

"PFL" or "CUE"
The Pre-Fade Listen or Cue button sends post-EQ channel signal to the headphone amplifier so that individual channels can be isolated through the phones. Because the PFL/Cue signal is tapped off just before the channel fader (hence "pre-fade") you can shut that channel down through the FOH PA, but still hear it through your headphones. This is a convenient feature for previewing channels before bringing them into the mix (e.g.., for cueing tapes up). It may also be used for checking out problems - a squealing amp, a distorted mic, etc.

"MUTE"
The Mute button is usually inserted just after the EQ section. We mention it at the end of the channel section simply because that is where the button most often appears - i.e. conveniently close to the channel fader and PFL/Cue button. As the name implies it silences the channel through the FOH system and possibly the monitors (check your manual). Its prime function is to enable the user to pre-set a channel's level, EQ, Efx sends and Mon./Aux. sends then shut the channel off to be added to the music program later on. Muting is a convenient feature for infrequently-used channels such as harmonica mic, acoustic guitar, banjo, mandolin, certain wind and percussion instruments, pre-recorded music or sound effects, all of which should be left off when not in use to reduce unwanted sound pickup and the risk of feedback.

- *{ **TIP** - The Mute button is often used as a quick first step to getting rid of a problem. If a mic is feeding back for instance, you can Mute it then EQ the feedback on that channel (see >>EQ above) or have someone move the offending mic, then lower the channel fader level, take the channel off Mute and bring the fader level back up. Of course, if muting the channel does not cure the feedback, either the monitor is feeding back or you have the wrong channel. Turn down the Monitor control and if that doesn't work, return that channel to normal status and try muting the next most suspicious one (nobody said pro sound was going to be easy).}*

"PHASE" or "POLARITY"
The input phase or polarity reversal button may appear at the bottom or top of the channel. As the name implies, it flips the polarity of the input signal so that it is 180 degrees out of phase with the other channels. This feature is most commonly used to combat certain persistent feedback problems where two mics are picking up the same source and feeding it to a nearby speaker, usually a monitor. For more details see Running the System item [4].

"L-R", "1-2", "3-4", etc.
Multi-buss mixers often feature pushbuttons on the channels which direct the post-fader, post-pan channel output to selected "pairs" of Main mix busses. That way, you may employ the stereo submixing section of your choice and Pan between its two masters.

- *{ **TIP** - Remember to de-select one pair of submasters when changing to another to avoid gain buildup and feedback.}*

(MASTER SECTION CONTROLS)
The masters generally act as output level controls for their designated output connectors, one exception being the EFX or STEREO RETURN master(s) since those jacks are inputs. Aside from that, everything does exactly what its name implies. See under MIXER OUTPUTS for further information on any of these features.

"MAIN"
This master regulates the output of the MAIN, SUM or MONO buss where the outputs of the SUB or GROUP busses or Left & Right stereo master busses get mixed down into a single signal.

"SUB" or "GROUP"
These masters regulate the SUBmix or GROUP output levels.

"MON"," AUX" or "EFX"
These masters regulate the output level of their designated SEND or OUT jacks.

"RTN"
These masters regulate the input levels of their designated RETURN jacks.

"EFX TO MAIN/EFX TO MON"
Some mixers with effects busses feature controls which are actually effects return masters, but one sends the effects signal to the input of the main busses and the other to the input of the monitor buss. In either case, the effects signal gets mixed with the straight signals coming directly from the channels (yes, electrons actually travel fast enough for some of them to leave the mixer via the Efx Send jack, go through perhaps several cables and effects devices, come back in through the Efx Return jack and still arrive inside the mixer at the input of the main or monitor buss circuits at the same time as internal signals direct from the channels - Believe It Or Not).

"PAN"
These controls pan the Stereo Return signals between pairs of SUB or GROUP masters.

"TAPE" or "2-TRACK"
This regulates the level of the Tape or 2-Track outputs.

"TALKBACK"
This regulates the level of the Talkback output. See under "MIXER INPUTS" for more details.

"EQ"
A graphic equalizer is featured on some mixers and a few have more than one, in which case one of the graphics will usually be for the monitor buss. As mentioned earlier, equalizers work by increasing or decreasing the signal strength (a.k.a. "gain") over various narrow bands of frequencies. As a result they are equally capable of curing or causing problems and should be treated with care.

- *{ **TIP** - To get rid of feedback, pull the EQ faders down one at a time, remembering to push them back up to centre if the feedback doesn't stop. Eventually you should find the one which reduces or stops the feedback. If possible, adjust it back up slightly so that the gain isn't overly reduced. As always, the golden rule is NEVER OVER-EQUALIZE. If there is a persistent feedback problem requiring large cuts in the EQ settings, move the mic or the speaker to get rid of it. EQ cuts cost the system valuable decibels of sound pressure. If the mic or speaker can't be moved, the next best solution is to ascertain which channel has the problem then insert an external EQ directly into the that channel (see "Insert" under Inputs). This way the necessary EQ cuts will only affect whichever channel has the problem, not the whole system.*
  The process of "sweetening" the FOH system's sound with low and high-frequency EQ boosts should be done with great care in live music applications. Keep them to a maximum of 3dB (preferably less). For DJ applications this is less of a stringent limitation, but boosts of more than 6dB should be avoided. And when in doubt, LEAVE THE FADERS AT CENTRE - there's no shame in a "flat" EQ; quite often it's the sign of a good system and a wise technician. For more information, see "EQUALIZATION" under Signal Processors.}

SIGNAL PROCESSORS

BACKGROUND

Once upon a time there were only two signal processors for PA, spring reverb and tape echo. Both suffered from mechanical frailty and the echoes had a tendency to overload with amazing ease. This remained the case throughout the nineteen-sixties, up until that fateful summer of 1969 and the Woodstock festival, in the wake of which PA technology took a huge, faltering step forward. Suddenly 4 and 5-channel PA "heads" were being replaced by multi-channel consoles and separate power amplifiers. Speaker "columns" and "cubes" began giving way to jumbo bins with horns stacked on top. The wedge-shaped stage monitor was born along with the dreaded term "monitor feedback" (Which is it - the main stacks or the monitors?!).

Signal processors now included passive crossovers and crude graphic equalizers, neither of which were well understood. In the mid-1970's digital delays made their entrance along with 31-band EQ's, active crossovers and bi-amping. Power amps became stereo then they got bigger and then people learned about "bridging" them. The smoked drivers, the blown amps, the cursing - it was truly something to behold.

But the dust finally began to settle in the mid-1980's when bin/horn stacks started giving way to advanced all-in-one speaker systems with smooth frequency response and superior passive crossovers. This reduced the need for bi-amping and its inherent dangers for the inexperienced. Meanwhile, previously astronomical prices on good quality compressor/limiters had come down and continued to do so. Now they're available to club-size PA users. And the list of PA signal processors continues to grow. This writing will cover the major ones used for PA - guitar effects will not be covered.

GENERAL

Signal processors all have two things in common:

With the exception of passive crossovers, they are all line-level (so do not connect them to amplifier outputs). See >>"LINE" under MIXER INPUTS for more details regarding line level.

They all have inputs and outputs so that they can be connected in series with the audio signal.

Aside from that, they're all different and most require a certain amount of experimentation to get them working properly.

GRAPHIC EQ

Try to imagine a set of "volume" controls wherein each one only affects a certain segment of the overall 20Hz to 20,000Hz range of sound frequencies. That' s a graphic equalizer - more or less. Actually the faders can't turn things up to "10" or down to "0", but they can turn them up to about "7" and down to about "4", far enough to have a marked effect on the overall volume level, depending on how many are pushed up or down from centre, and how far. It's hard to imagine that merely changing the comparative volume levels of various frequency bands could make such huge changes in the way things sound, but that's exactly what happens.

The actual tone which identifies something, let's say a trumpet note, depends on the comparative volume levels at which the harmonics of that note occur. Harmonics are incidental vibrations which occur whenever an instrument or voice is producing a note. In this case, the trumpeter's lips can be vibrating at 440Hz to produce middle A, but ripples in the skin also produce tiny notes at higher frequencies, these are the harmonics. The same sort of thing happens along a vibrating string or a reed or anything which vibrates to produce sound . When you adjust an equalizer you increase or decrease the amount of audible emphasis on any harmonics occurring in that range, hence altering the tone.

Feedback is simply a sound that's gone wild. What's at work here is a phenomenon called sympathetic vibration. Basically, two or more things which are physically prone to vibrating very freely at a certain frequency can get each other in motion by "remote control". One of them sets the air in motion and the vibrating air then sets the other thing(s) in motion.

- To prove this, place two acoustic guitars close together and tune them identically. Strum one of them then mute the strings and listen to the other one - it should be ringing.

If things with similar frequency response peaks are connected to the same amplifier system and turned up loudly enough, the vibrating sources (e.g.. a mic diaphragm and a speaker) reinforce each other in an escalating manner at that frequency and you have feedback. The EQ lets you turn down the system's volume at that frequency which gets rid of the feedback, BUT now the natural tone has been altered (good news and bad news). That's why we say again, never over-equalize.

- *{ **TIP** - One thing you should always resist is the temptation to create a "smiling" EQ curve. It may be OK at home, but on the job it robs the system of power headroom and muddies the sound. As the volume level rises during the night, these problems will make the PA sound weak, as if it is somehow losing power.}*
-

Today's Graphics EQ's have additional features worth noting:

- **"Gain"** This is a fader to let you compensate for an adjusted EQ's natural tendency to turn the volume (actually gain) up or down (see above). Be a little careful with this feature. Boost the Gain too high and you might overdrive the mixer input.

  *{ **TIP** - When connecting the EQ directly to an input channel via the Insert jack, remember that on some if not most mixers, the Insert jack is post-Gain. As a result the "return" part of it is not regulated by the channel's input Gain control. Thus, if you increase the EQ's Gain by too much and distort the channel, you should reduce the EQ's Gain, not the channel Gain.}*

- **"HP Filter"** The high-pass filter, sometimes referred to as a "Subsonic" filter, is usually activated by a pushbutton. As the name implies, it lets all the frequencies higher than a certain frequency "pass". In other words, it doesn't let anything below that frequency pass which means it's a low-frequency cut filter ("high-pass" being the correct techno-jargon for it). Why have a feature which chops off all that lovely deep bass? Because as a rule, deep bass belongs more in the home than at a live sound gig.
  Below 50Hz you enter the realm of very long wave forms and their ability to set all kinds of things in motion - bass drums, bass strings, microphones, speakers, furniture, walls, ceilings, you name it. This may cause feedback problems and sound colouration. Additionally, woofers reproducing these very low frequencies at high sound-pressure levels consume huge amounts of amplifier power which leaves less for the important 60-100Hz range where audiences get treated to the "thumps" they like so much.

- **"LP Filter"** Opposite to the high-pass filter, the low-pass or "Ultra-Sonic" filter attenuates the very high frequencies, generally above 20kHz (20,000 cycles-per-second). This is to avoid problems sometimes caused by high-frequency oscillations above the human hearing range. These may be generated by a defective processor or synthesizer or even a radio station. What happens is, the oscillation gets amplified and goes to the tweeters or horns where it sits, heating up the voicecoils which are probably unable to reproduce that frequency and even if they were, nobody would be able to hear it. This can cause the voicecoils to burn or otherwise suffer a premature demise.
  *{ **TIP** - In a live sound situation, it's generally a good idea to activate both filters. They cost your audience no real sonic performance and they offer a valuable insurance policy.}*

INPUTS AND OUTPUTS

Most professional PA power amplifiers these days feature balanced inputs which may take the form of 3-pin XLR connectors or TRS (tip-ring-sleeve) 1/4-inch jack sockets, or both. As a rule, each channel will have two inputs wired together in parallel so that mixer signal can be fed to other power amplifier inputs via balanced patch cords.

PARAMETRIC EQ

Unlike the graphic EQ, the parametric most often has rotary controls rather than sliders. You will find controls for Frequency and Q accompanying each cut/boost control so that you end up with three knobs for each frequency. This provides you with the ability to pinpoint a range of frequencies and either boost or cut the gain there.

The Frequency control is self-explanatory - it moves the cut/boost across the audio spectrum. The Q control enables you to broaden or narrow the band of frequencies to be cut or boosted. As a rule, the cut/boost will have a less noticeable effect on a high Q setting because Q reflects the ratio of gain change to frequency band, ergo a high setting represents a high gain-change-per-octave ratio, hence a narrow band of frequencies. Together the three controls create a potentially ideal cure for problems like feedback. You set the cut/boost for a -6dB cut, set the Q up about half way, then rotate the frequency control until the feedback stops.
The Q control is especially valuable in this regard because once you have located a feedback frequency range using a medium Q setting, you can increase the Q setting thereby narrowing the range of cut frequencies so that fewer "innocent" frequencies are affected and the basic sound quality does not suffer needlessly. Like the graphics, parametric EQ's may feature high and low-pass filters. See above for details.

- *{ **TIP** - Although you can create very specific frequency boosts, you should (again) resist the temptation to do so in an effort to "sweeten" the sound. A low-Q, low-frequency boost may sound great at low volume but, as with the graphic, it will cost you performance at high volume levels. For information about scanning for feedback, see >>EQ under MIXER CONTROLS.}*

COMPRESSOR/LIMITER
This type of device has been around even longer than either the graphic or parametric EQ, however its use for PA applications is still largely limited to the bigger systems. Created as a means to reduce audio peaks on recordings or broadcasts, the comp/limiter acts rather like an automatic gain reducer - a "robot maximum volume controller" if you prefer.
Controls usually include:

- **Threshold** (the signal level at which the comp/limiting is triggered)
- **Attack time** (how fast the gain gets turned down when the signal exceeds the Threshold level)
- **Release time** (how fast the gain is allowed to go back up again after the peak is over)
- **Ratio or Slope** (ratio of input level to output level, e.g. "1:1" means no comp/limiting, "4:1" means input peaks are being comp/limited to as little as one quarter their original strength, etc., etc.)
- **Output level** for adjusting the final, comp/limited signal up or down to match the rest of the system.

One example of this device's usefulness is in controlling bass guitar transients when the bass amp is being "lined" into the mixer (i.e. the "Line" output from the amp is patched into the Line input on one of the mixer channels). You would patch the comp/limiter in-between the bass amp and the mixer. If the bassist uses a lot of slaps or a hard picking technique, set the Threshold fairly low, around half-way, the Attack and Release should also be around that setting and the Slope or Ratio at around 4:1 or 6:1. The Level control would simply be adjusted in conjunction with the mixer channel's input Gain control to achieve a small amount of clip light activity on that channel. Fine tuning of the comp/limiter's controls should net plenty of bass through the PA with no distortion or major loss of system headroom.
Perhaps more common is the need to ensure that speakers do not receive more than their maximum rated power. A limiter is the best means by far of providing this insurance. Fuses, for instance, will always fail after a certain amount of usage because amplifier output signals cause the elements to flex rather like an inchworm, causing the metal to fatigue then break (fuses also add current-variable resistance to the speaker circuit which affects the amp's damping factor). A good, sonically transparent limiter can be set so that the maximum mixer signal the amplifier receives is only sufficient to drive it to the pre-determined output power level.
A watt meter attached to the output of the amp can be used to measure power while test signals are put through the system. With the comp/limiter's Output Level on full, the Ratio on 4:1, Attack and Release at low settings, establish the right Threshold setting so that the desired power level is not exceeded according to the watt meter. Now "peg" that Threshold setting with a dot marker. It is worth noting that a setup like this to cover monitors as well as main speakers has the added benefit of protecting against accidental transients from mics being dropped, cables being yanked out, turn-on "snaps" when the mixer goes off then comes back on (somebody tripped over the power cord) or A.C. surges when the lighting system dims right down.

A few other applications for comp/limiters and settings for them are listed below (from the ART model CS-2 manual); Note: "# o'clock" simply refers to the general control setting

- *Fattening drums & percussion; Slope - 4:1 to 6:1, Attack - 10:00 o'clock, Release - 10:00 o'clock
- *Bringing out an instrument from the mix; Slope - 2:1 to 4:1, Attack - 9:00 o'clock, Release - 9:00 o'clock
- *Adding sustain to an instrument; Slope - 6:1 to max., Attack - 10:00 o'clock, Release - 10:00 o'clock

ELECTRONIC or "Active" CROSSOVER

Although it is more associated with speaker systems than signal processors, the active or electronic crossover is normally connected between the mixer and power amps and thus is covered in this section. Its purpose is to ensure that the amplifiers directly powering woofers, tweeters and midrange speakers or horns receive the correct ranges of sound frequencies so that the overall sound is balanced and there is no distortion or damaged horn/tweeters receiving frequencies which are too low.

Crossovers do this by putting the audio signal through a series of filter stages which separate the lows from the mids and/or highs. What makes Active crossovers preferable to the passive variety is their lack of phase alignment problems, their frequency variability and their ability to control the volume level of the different frequency ranges.

Basic features include the following;

- Input, usually balanced with a stereo1/4" jack or XLR
- Low, Mid and High frequency outputs on 3-way models, Low and High frequency outputs only on 2-way units
- Input Level control, then Frequency controls and Output Level controls for each range - for example, on a 3-way crossover there would be a "Low/Mid Frequency" control and a "Mid/High Frequency" control plus Low, Mid and High output level controls.

Crossovers work by reducing the gain of the audio signal above and below certain frequencies. It is <u>IMPORTANT</u> to note that they do this rather gradually; the rate at which they "roll off" the gain being expressed in decibels (dB) per-octave (an octave is either double the frequency in question if it's an octave higher, or half of it if it's an octave lower).

For example, if you were to set the high/low frequency of a 2-way, 12dB-per-octave crossover at, let's say 1,000Hz (1,000 cycles-per-second), the high-frequency output signal level from the crossover will be attenuated by -12dB one octave lower than that. As a result, power signals from the amp to the horn or tweeter occurring down at 500 Hz, i.e. one octave below 1,000Hz, will also be attenuated by -12dB or roughly 75%. Thus, if your horn or tweeter is rated at 100 watts at 1kHz. and a 100-watt power amp is connected to a 12dB/octave crossover set at 1kHz., the component may be receiving output down at 500Hz at a power level of 25 watts - not likely to cause a problem in this case. But if the amp puts out 200 watts, the HF components could be receiving 50 watts down at 500Hz which is dangerous.

Keep in mind that low frequencies kill horns and tweeters even at low applied power levels. Hence, when you are bi-amping high-frequency components in a 2-way system, do not over power them and be sure about the crossover frequency.

- *{ **TIP 1** - When shopping for an active crossover, it's probably a good idea to look at the "fast" ones - i.e.18dB/octave or even 24dB/octave. 12dB/octave is alright for small, low-power systems, but faster rolloffs offer greater precision and a much better safety margin for your midrange and/or high-frequency components.}*
- *{ **TIP 2** - Another good idea, once you have picked an active crossover, is to check the accuracy of the Frequency control. This can be accomplished using a sweep signal generator and a level meter, or a pink noise generator and a realtime frequency analyzer. This test can be worthwhile because not all crossover frequency controls are very accurate and, while it doesn't mean there is anything wrong with the crossover, you still need to know what is really happening when you set this dial. If it's high by just half an octave, e.g.. a setting of "1kHz" actually represents a 750Hz crossover point, you could end up with distortion or blown components. Knowing this however, you could compensate by turning the dial up half an octave to a 1.5kHz setting.}*

*{ **TIP 3** - You can adjust the crossover's Level controls for smooth frequency response by "ear", however it's a good idea to employ something else as well for reference. Realtime frequency analyzers are produced by a few different companies and some sell for only a few hundred dollars. RTA's usually come with a built-in pink noise generator, a condenser microphone with very flat frequency response (that's important) and a digital frequency response display.*
During setup when there are no audience members present, plug the pink noise into a mixer channel with all EQ set flat including the main graphic - the system will be roaring like Niagara Falls. Now set up the mic in the middle of the audience area and adjust the crossover Level controls until the RTA 's response graph display looks as flat as possible. Now you can adjust the main EQ slightly to reduce the size of response peaks which look big enough to pose a feedback threat; just remember the rule, never over-equalize, not even to make an RTA look flat.
Although the RTA provides a handy visual guide, it should not be taken too literally. You still need to use common sense and your ears to a certain degree; but remember that this is a sound job, not your living room - take it easy with the "boom and sizzle" or the system will start to sound muddy when you increase the level later on. And if you are in doubt, the safest thing to do is to set all level controls on the crossover at maximum and, in so doing, send equal amounts of signal to all power amps. The maximum settings will compensate for potentiometer tolerances - max is always max no matter what the tolerances might be}.

Standard wiring for an active crossover goes as follows:

Mixer's Main or Sub Output is patched to the crossover's Input.
Crossover's Low frequency Output patches to the input of the power amp (channel) driving the woofers.
Crossover's High frequency Output patches to the Input of the power amp (channel) driving the tweeters and/or horns.
Crossover's Mid Output on a 3-way unit patches to the Input of the power amp (channel) driving the midrange speakers or horns. There are also 4-way crossovers which would have either Low-Mid controls and Output, or High-Mid controls and Output. Additionally, some crossovers may have a "Sub" Output for connecting the subwoofer power amp (channel). These often have a fixed crossover frequency at around 90Hz. and may or may not have a Level control - if not, the level would be preset at maximum.

REVERB

It follows that, since reverb units are signal processors, they should be mentioned in that overall context. On the other hand, they are effects devices not sound controllers and will therefore be dealt with briefly. There is a basic rule of thumb which says that a small amount of your favorite reverb sound on the main system should be enough for most places, especially if the place has a natural reverb or echo of its own - too much reverb makes the sound muddy and lacking in punch (speaking of which, you should never add more than a small amount of reverb to bass drum or bass guitar, otherwise your system's punch could turn to MUSH). Equal restraint should be employed when mixing reverb to the monitors - performers need clarity with a capital "C" onstage.
Aside from that, it's worthwhile mentioning that your reverb unit would normally be connected to the mixer's EFX Send and Return jacks - Send going to the reverb's Input and the reverb's Output going to Return. However you may only want a particular type of reverb or echo on one channel, possibly the lead vocalist. Then you would patch the reverb unit into that channel's Insert connection (See >>"INSERT" under MIXER INPUTS). Note that this channel should probably not be put through any additional reverb as the clarity of that channel could be muddled. In general terms, reverb can add a nice "dimension" to your sound and even a bit more if it's a stereo reverb and you're running a stereo main PA. Just don't over do it.

POWER AMPLIFIERS

The various features of a power amplifier can be broken down into several sections in order to simplify explanations. The sections break down as follows:

GENERAL INFORMATION

Over the years manufacturers have honed the design of PA power amplifiers to a fine edge. As a result, it's really hard to find a modern amp which doesn't live up its specifications. In reality, there isn't much wrong for a power amp to do. They have virtually flat frequency response so differences in sound are often more imaginary than real, although some of the ultra-compact amps with "switching" power supplies may not be quite as good at driving subwoofers as the heavier ones with conventional power supplies. Aside from that, however, the only continuing difference between power amplifiers is reliability and even that tends to be less of a variable as designs improve. We specify "PA" power amplifiers for a reason however; home stereo power amps are perhaps not ideal candidates for the rigors of life in the pro audio world.

MONO OR STEREO?

Mono power amplifiers are something of a vanishing breed these days, however they do exactly the same job as one channel of a stereo amp in terms of basic sound quality and are still employed in a variety of applications from live PA to installations. They usually have two Speaker Outputs wired in parallel and may have two Inputs also wired in parallel. Dual parallel Inputs are not intended to accept stereo mixer signals. Instead, the extra input can be patched to feed some of the (mono) mixer signal to the input of another power amp - rather like using a "Y" adapter. If you do connect a mixer's stereo outputs to these inputs there can be phase cancellations causing a change in the sound, and possibly distortion.

- *{ **TIP** - To connect a stereo mixer to a mono power amplifier or one channel of a stereo amp, find a mono output on the mixer and use that. If the mixer does not have a mono output, i.e. one representing the sum of the left and right channels, simply pan all the channels left or right and use the output you've panned to.}.*

Also remember that, as with one channel of a stereo amp, connected speakers will be in a parallel circuit even though there are two Output jacks. Hence, if you connect two 8 ohm speakers, the overall speaker impedance will be 4 ohms (more about this later).

Stereo power amplifiers can be viewed as two mono amps in the same package. As a result, you can connect two 8-0hm speakers, one to each channel, and the speaker impedance encountered by the amplifier will still be 8 ohms. But connect another 8-ohm speaker to one of the channels and that channel will encounter 4 ohms. And, because each channel can operate quite independently of the other, channel one can power a (mono) FOH (Front Of House) speaker system while channel two powers monitors or anything else - perhaps subwoofers. Stereo FOH PA systems are becoming more common these days, however a mono system works as well in most situations (more about this later too).

POWER - Which Watts are Which?

Amplifier power ratings these days tend to be in watts expressed as "continuous average" or "burst average" or "peak" or "music power" or "continuous music power", etc. In the old days, the nomenclature was "RMS" which stands for "root mean square" and reflects the results of a test for the amp's long-term, continuous output capability. Other, more modern tests tend to net fairly similar results but are more complex and require more sophisticated equipment. Two ratings which are worth looking for are Continuous Average Power and Burst Average Power. The first rating will be similar to what an RMS test would net and the second one will be higher, reflecting the amplifier's ability to repeatedly produce clean peaks which last for at least one complete wavelength.

HEADROOM

A good amp will have 3dB of headroom at its maximum, continuous output. This means that the amp can deliver double that output on frequent, full-wave peaks. That's what really helps to make those deep "thuds" and "rumbles" shake the floor when you are powering subwoofers.

SLEW RATE

Slew rate is basically a measure of the amplifier's ability to supply voltage in response to fast, short-duration peaks and is measured in volts per microsecond (one millionth of a second). A slew rate of "30VUsec" - 30 Volts per microsecond - is considered good, but anything down to 20 VUsec is also considered good. High frequency reproduction is thought to be better in amplifiers with a high slew rate hence, if you do hear a difference between two amps with radically different slew rates it would be in the highs. However the truth is that slew rates have to be unusually low - well below 10VUsec before any highs are likely to be muted. Few amps like that are produced today.

DAMPING FACTOR

Damping factor is usually, but not very accurately, linked to low frequency reproduction. The popular thinking goes as follows; although an amplifier may have flat frequency response all the way down to 20Hz, its ability to make speakers reproduce low frequencies with maximum sound pressure depends to a certain degree on its damping factor. This is supposedly because amplifier outputs encounter a certain amount of "Counter-EMF" (electro-magnetic force) from the speakers, especially woofers. These induced signal voltages tend to be out of phase with the amplifier's output and can cancel some of it, especially on low-frequency peaks where the cone is travelling very far in and out causing the voicecoil to cut more lines of magnetic force hence generating more counter-EMF voltage.

Damping factor, in its popular perception, reflects an amplifier's potential ability to counter or "damp" the effects of this process thus permitting more power to be delivered to the woofers. There is some truth to this, but not as much as you might think. Damping factor is a matter of impedances. As long as the amplifier's output impedance is lower than the overall speaker impedance you have damping. In fact the method of calculating an amplifier's damping factor is based on the difference between the two impedances. If, for example, an amplifier's output impedance is 0.01 Ohms and it is designed to operate into a 4-Ohm impedance, the damping factor would be 4 / 0.01= 400.

All well and good, but the truth is, a high damping factor does not guarantee better bass response from the speakers. That characteristic can be chalked up to large amounts of power "headroom". In fact a really high damping factor can have an effect on the woofers which makes their lowest frequencies roll off more abruptly than if they were being powered by an amp with low damping, the net result being a reduction in deep bass response. Once again, check the specs for headroom figures. A good amp will have 3dB of headroom at its maximum, continuous output. This means that the amp can deliver double that output on frequent, full-wave peaks.

POWER AT IMPEDANCE

An amplifier varies its maximum output capability in accordance with the overall speaker impedance it is driving. Solid state amplifiers do this inversely to changes in the speaker impedance (yes, speaker impedances actually change with the frequencies they are reproducing - more about that in the Speaker section). In other words, as the impedance decreases, a solid-state amp is able to put out more power, and as it increases the opposite happens. [For what it's worth, tube amps have a "favoured" impedance above or below which their maximum power capability decreases].

It would seem to follow that solid state amplifiers might be able to deliver more than their rated power if you connected them to very low impedances. But we all know that this is not the case - especially if we've had an amp shut down or, Heaven forbid, blow up because the speaker load was too low. What happens is, the amp actually tries to put out more power than it was designed to deliver. It overheats in the process and will self-destruct if it has no thermal protection circuitry to shut it down or otherwise limit its activity. Conversely, amps are not harmed when connected to high load impedances; their maximum output is simply reduced. And when a solid-state amplifier is connected to no speakers at all, in other words an infinitely high impedance, it won't put out any power at all, it just takes a holiday.[Tube amps are very different in this regard. They will self-destruct if run with no load connected].

- *{ **TIP** - Be sure that the overall speaker impedance being driven by each amplifier channel is not lower than the manufacturer's specification. If the rating is much too low for your amp to handle it may cause serious problems for your power amplifiers. On the other hand, if it is only a little lower than the amp's minimum rating, you might be alright.}*

A COMMON QUESTION...
"If I connect too many speakers, will it ruin my amp?"
Oddly enough the answer is, not necessarily. You could, for example, run as many as four 8-ohm speakers from one channel of an amplifier, assuming it has a minimum load rating of 2 ohms per channel. And by making a special series/parallel wiring rig, you could theoretically run as many speakers as you like from one amplifier, however it could be a wiring nightmare.
Before we leave this subject it's worth mentioning that a rough rule-of-thumb regarding power at impedance goes as follows: double the speaker impedance and you cut the amp's output in half - cut the impedance in half and (assuming it's not too low now) you double the amp's output. Again, this is a very rough rule because all amplifiers react differently to speaker impedance changes and almost none of them puts out exactly half or double power into double or half the impedance. If you are in doubt about what an amplifier delivers into a certain impedance, check the manufacturer's specifications.

PASSIVE & ACTIVE COOLING
In the beginning, all power amplifiers were passively cooled. With passive cooling, the output transistors are tightly fastened to metal fin clusters called heatsinks attached to the outside of the amp. As air passes over the fins, heat from the output transistors is radiated away. If there is restricted air flow, the result can be overheating and possibly damage. Passive cooling is common on units rated at under 800 watts; however, in power amplifiers over 800 watts, active cooling is the norm. Although it is theoretically possible to passively cool a one-thousand or two-thousand-watt amplifier, it is impractical. The heatsinks, whose size largely dictates their thermal transfer ability, would have to be huge and so would the amplifier.

- *{ **TIP 1** - Although proper ventilation is important to all amplifiers, it is critical to those which are passively cooled. Racks or cases should always be open-backed and placed well away from walls or other obstructions.}*

In an actively cooled amp, a built-in fan moves the air across compact, internal heatsinks. The chassis (case) of the amp helps to trap the air inside and further ensure that it travels over the fins. In most of the newer amplifiers, fans tend to be thermally regulated so that they either rotate slowly or nor at all when the amp is running cool. This way, the long-term amount of dust drawn in by the fan is reduced. Dust sticking to internal heatsinks can act as an insulator and reduce their thermal transfer. It may be worthwhile to have a technician clean the internal heatsinks once every two years or so if your amplifier does not have an air filter.

- *{ **TIP 2** - Clean or replace the air filter regularly. This will prevent it from getting clogged with dust, choking off cool air.}*

POWER AMPLIFIERS FEATURES & SPECIFICATIONS

INPUTS
Most PA power amplifiers today offer a selection of input connectors and facilities. Balanced inputs may take the form of 3-pin XLR's and/or 1/4 inch TRS (tip-ring-sleeve - a.k.a. "stereo") jacks. As a rule there will be both and the 1/4 inch jack will be wired parallel with the XLR so that you can patch to the input of another mixer or to the other channel's input on the same stereo power amp (a Stereo/Mono button can make this unnecessary - see below).
Some amplifiers such as the old Beta-800 have an input feature which utilizes internal wiring and a special switching jack, usually the B channel input, to patch the two channels together. The amp operates with both channels reproducing the same (mono) input signal until a jack is inserted in the B input which breaks the internal patch so that the channels will be independent.

The emphasis on designing features into stereo amplifiers which make them easy to convert to what might be called "dual-channel mono" operation further illustrates that this does tend to be the way they are often employed outside the living room. "Bridging" (not to be confused with dual-channel mono operation) is another way of turning a stereo amp into a mono amp, in this case with the power of both channels combined into one output signal. Although modern PA amplifiers tend to have a Bridge switch, some may not in which case it is necessary to connect a mono mixer signal to both channels either using a "Y" adapter or by patching to the other channel via the additional input if there is one. Then you reverse the phase of the signal going into the other channel. This is accomplished by taking the cable-end apart and resoldering the leads in reverse order. If you're using balanced patch cables, reverse the "phase" and "reverse phase" wires. In an unbalanced line, reverse the hot and ground wires.

- *{ **TIP** - Be very careful about bridging - it can be difficult even for people who are familiar with it. Read the "Bridging" section below.}*

OUTPUTS
Output connections on PA amplifiers have, over the years, run the gamut of just about everything from 1/4-inch jacks to post (a.k.a. "banana") terminals, to 3-pin XLR connectors (no, these are not "balanced speaker outputs"), to combination XLR/1/4-inch connectors, to the "Speakon" locking connectors developed by Neutrik in Switzerland. Driving manufacturers to make all these changes is the need to accommodate very high output power levels.
The 1/4-inch jack, for example, only has around one square millimeter of contact area at the tip - enough to carry no more than a few hundred watts before heat and then resistance begin to build up at the jack tip thus decreasing power delivered to the speakers. Post terminals carry plenty of power and have been the standard speaker output connection for many years, however there are two problems with them. First, cables have to be either bare-ended and wrapped around the terminals or equipped with "banana" connectors which go onto them. In either case the other end of the cable will need to have whatever connector matches the ones on the speakers - until recently, XLRs. This lack of standardization has been a sore point with pro audio users for years.
The second problem has to do with user safety; you can receive a very nasty shock by touching post terminals while a kilowatt-plus amp is in use. Most people don't do things like that, but consumer safety groups usually have their way, hence legislation is looming somewhere in the future to remove binding posts from power amps, at least the high-powered ones.
This leaves XLRs and Speakons, both of which appear on a few amps at present. XLRs work fine, but due to the cables' resemblance to mic lines, people occasionally use the wrong ones to connect speakers, a practice which nets less than desirable results - MUCH less than desirable (mic cables just heat up and waste power, assuming they don't melt). Speakon connectors were specifically engineered for the application which their name implies and may be destined to eventually become the standard connector for high-powered amplifier outputs and speakers alike. They even have accommodations for bi-amping built into the basic design, something XLR's do not have.

LEVEL CONTROLS
The function of power amp Level controls is simple enough, the only question is how and why they should be used. For most PA applications you would run them at maximum, reason being that you want the amplifier's full output capabilities available. There are exceptions of course; for example if one channel is powering the subwoofers and the other the full-range cabinets in a club system, you would probably have the subwoofer channel at maximum and the other at a lower setting. Also in a club PA, you may be using a single monitor mix for everyone and thus the amp channel driving the drummer’s monitors could be at max and the other at a lower setting (drummers aren't deaf, just noisy).

MONO STEREO SWITCH
In the "mono" position, this feature ties the two channels' inputs together via internal wiring so that they operate in unison, a.k.a. mono. Mono is a common amplifier operating mode for PA applications involving multiple speakers. If, for instance, you needed to power four subwoofers and four full-range enclosures, you would employ two amplifiers in mono mode, one fed by the low frequency output from the electronic crossover and the other from the crossover's high-frequency output. You would then connect half of the subwoofers to one channel of their power amp, and the other half to the other channel.

The full range speakers would be similarly connected to their amplifier. In the "stereo" position, the switch bypasses the internal input patching so that the channels function independently.

BRIDGE SWITCH

This feature does almost exactly the same thing as the Mono/Stereo switch but with a major difference (in fact, it is often combined into a mono/stereo/bridge switch); its input-to-input internal patch wiring has the positive and negative leads reversed so that one of the channels is 180 degrees out of phase with the other.

- *{ **TIP** - If your system is sounding mysteriously "weak", especially in bass response, check the Bridge switch. If it's "on" and you haven't performed all the following procedures, switch it "off". That weird, weak sound would be caused by acoustic phase cancellations between your speakers.}*

Bridging is the process of turning a two-channel amplifier into a one-channel amplifier producing the summed output of both channels. This is accomplished as follows:

- Feed the same (i.e. mono) mixer signal to both inputs, but reverse the phase of the signal going into one of the channels. This is not necessary if you have a Bridge switch; it takes care of phase reversal automatically. But if you don't have a Bridge switch, you'll need to do this by taking the plug apart going into the "other" channel's input (i.e. not the channel directly receiving the mixer signal), unsoldering the wires, reversing the plug leads, resoldering and then plugging it in.
- Check the impedance of the speaker (yes, only one speaker, read on) and divide it by two. This is very important because a bridged amplifier reacts to the connected load at half its value. A 4-ohm load, for example, will seem like 2 ohms to a bridged amplifier and it will react accordingly.
- Check the resulting impedance against the amplifier's minimum load rating. Never connect a load which is lower than that rating.
- If the amp can handle that load, connect the speaker as follows: take the "hot" (+) signal only from each channel and connect your speaker so that the cable end from one channel goes to the "+" speaker terminal and the other cable end goes to the other speaker terminal. Note which channel is driving the "+" speaker terminal so that you will be able to rig another bridged amp and speaker the same way and they will be in phase (VERY important, especially for subwoofers).
  If your amp has post terminal outputs this is easy; simply connect the two RED terminals, no black ones. If it has 1/4-inch jack socket outputs, you will need to rig a split speaker cable with two 1/4-inch plugs at one end, each plug connected only by the tip tab (the shorter of the two tabs inside the plug). Each plug would then be inserted in a single output jack socket from each channel.
  If the amp has XLR outputs, rig a split cable as above but with two XLR ends. Check the manufacturer's output code in the amp manual to find out which XLR pin is "+" and solder the cable ends to those terminals in the connectors, once again, only one connector per cable end. Then plug a cable-end XLR into each channel's output XLR and connect the speaker at the other end of the cable.
  If the amplifier has Speakon outputs and it doesn't have a separate "Bridged" output, follow the same basic procedure as with XLR connectors (except, of course, using Speakons). In any case, attach whichever connector is needed to the speaker end of the split cable. Again, be sure to note which channel is driving the "+" speaker terminal if you will be bridging another amp and speaker, so that they will be in phase.
- Set both level controls at **MAXIMUM**.
- **DO NOT ENGAGE THE BRIDGE SWITCH UNLESS YOU HAVE PERFORMED THE ABOVE FUNCTIONS**. Once again, the Bridge switch only patches the two channels' Inputs together and puts one of them out of phase. It does NOT somehow "double your power". In fact, with the two channels out of phase and just running speakers in the normal manner, you would be WASTING power because the speakers connected to them will ALSO be out of phase and CANCEL EACH OTHER OUT acoustically. This can cause up to a 3dB loss in sound pressure which is equivalent to LOSING HALF THE POWER from that amplifier.

- And if, after all this, you still think that bridging two 1,000-watt amps is preferable to using one 2,000-watt amp unbridged, you might consider seeking psychiatric help. The practice of bridging harks back to a time when the most powerful stereo amps only put out around 700 watts in total and had to be bridged in order to get all of it into a speaker. This is no longer a problem.

LIMITER SWITCH
Distortion is DEATH on woofers, horn drivers and tweeters, even at applied power levels well below their power ratings. A distorted amplifier signal causes the voicecoil to move back and forth erratically in the magnet gap, sometimes contacting the wall of the gap and becoming damaged. Additionally, squared or clipped wave forms contain higher levels of current in the delivered power which effectively raises the heating effect, burning the voicecoils. Although some speaker systems have fuses and/or circuit breakers to help the drivers survive distortion, the result is the same - an interrupted gig.
Compressor/limiters are a Godsend (see Signal Processors). They can be set so that the amplifiers never receive enough mixer signal to drive them into distortion. The ability to get this feature built into an amp, especially at little or no extra cost, should make such amps virtually irresistible. Some people think that limiting has an audible effect which can be true when an external unit has been adjusted the wrong way, but the limiting circuitry built into power amplifiers has been optimized specifically for them and is most often sonically transparent.

- *{ **TIP** - If you have an amplifier or amplifiers with this feature, do yourself a favour; just switch it on and leave it on. A good, sonically transparent, built-in limiter will save you untold speaker failures and cost you absolutely no performance. In fact, limiters can actually maximize clean power.}*

GROUND STRAP
Ground loops and their attendant hum can sometimes be traced to a rack of power amplifiers. If this is the case, try lifting the ground straps on all but one of the amps in the rack. If that fails to cure or sufficiently reduce the noise, re-connect all the Ground Straps and check elsewhere. Do not leave the ground straps off unless you have to.

"Clip" LED
Most manufacturers set the threshold of their Clip LED's to fire at 3 decibels (dB) below the point of distortion. As a result, small amounts of Clip light activity are likely to be acceptable, however check your manual to be sure. In amplifiers with built-in limiters, Clip LED activity can indicate that the limiter is working.

"Protect", "Temp." or "Fault" LEDs
In most cases these lights indicate problems at the outputs created by low or shorted speaker loads. But they are often triggered by output transistor heat levels and can therefore indicate other conditions such as inadequate cooling due to poor ventilation, dirty heatsink fins, a non-functioning fan or a clogged fan filter (clean or change these regularly). If these lights flash briefly on power up, it may not indicate a problem. Some amplifiers are designed to go through a status check when turned on - read your manual to be sure. If these lights come on while the amp is running, you may have a serious problem. Try the following;

- Disconnect the speaker cables and see if the light goes out. If it does, the problem could be in the speaker circuit - possibly a short circuit somewhere. Use your volt-ohm meter or multimeter to check the overall load resistance. If it is very low or the numbers on a digital meter seem confused, it probably is a short. Now check everything including the cables and connectors as well as the cabinets themselves.
- If disconnecting the speaker lines does not make the trouble light go out, check the ventilation. If the amp is in a rack or a case, make sure the back is open and away from the wall or any other obstruction. If the amp is fan-cooled, make sure the fan is turning and the filter is clean. If the problem persists, get the amp to a repair shop.

CIRCUIT BREAKERS AND FUSES
By the time a breaker or fuse blows, it's likely that something bad has already happened, unless it's a speaker output fuse (some older amps have them) in which case try turning down the volume. But if it's the AC power fuse or breaker, replacing the fuse or resetting the breaker probably won't help, at least not for long. It is most likely time for a visit to the shop, especially if the problem persists.
Although most PA amplifiers have some form of speaker protection built in, you may inadvertently defeat it by repeatedly trying to reset the power breaker or, Heaven forbid, trying to hold it in with something, or wrapping the blown fuse in foil (argh!). One thing which is sure to pop the mains fuse or breaker is blown output transistors and when they short out in the process of blowing they can let all the DC voltage which is stored in the amplifier's filter capacitors go straight out to the speakers. This is called "DC offset" and it instantly turns delicate voicecoil wire into a black, motionless, silent mass. Even speakers with fuses or breakers built in may succumb to DC from a big amplifier, so treat popped AC breakers and fuses with RESPECT.

POWER vs. POWER
It's worth noting that, contrary to what you might think, amplifiers do not "create" power, they only modulate it. The power that really drives your speakers is what comes out of the wall. As a result, what you get out of an amp depends entirely on what goes into it. Quite often low sound-pressure levels, distorted and/or overheated amplifiers can be the result of sagging AC line voltages caused by inadequate house wiring, high-demand electrical appliances on the same circuit as your amplifiers or just too many amplifiers on the same circuit.
If you have a lighting system, it MUST be on a separate AC circuit. Of course, not every power outlet is likely to be on its own circuit so always ask the proprietor or manager of the venue which ones are which. Also find out which, if any, of the AC outlets have high current ratings.
For example, if you have two power amplifiers drawing 15 amperes each and you want them to be on the same circuit for common grounding, that circuit will need to have a 30 ampere ("amp" for short) current rating. But a high current rating does not guarantee that you are going to get sufficient AC Voltage out of the wall to power your amplifiers up to the manufacturer's specifications. A Volt meter can go a long way toward explaining why things may not be sounding as they should.

Then what do you do about it? Not much except perhaps to employ a "variac", a gadget which allows you to adjust the AC voltage upward when it is sagging below 117 or 120 volts, or 220 to 240 Volts depending on what country you're in. You simply plug the variac into the AC outlet then plug your amplifiers into the variac, then you adjust the variac's control until its meter reads the desired voltage. The problem is, high-powered variacs are quite expensive and the good that they do may not be necessary everywhere. For that reason they are often overlooked as part of small to medium sized PA rigs. Too bad because they can be invaluable when you begin encountering, not only amplifier problems, but also oddball misbehavior from your digital processing equipment or keyboards, some of which require a strict "diet" of not less than 110 VAC in North America. But, whether you employ a variac or not, remember to measure the AC line voltage next time things go screwy. If nothing else, you may save an unnecessary trip to the repair shop.

- *{ **TIP** - When using any kind of a meter to check high voltage levels, read the meter's instructions carefully and follow them. The life you spare may be your own}*

SPEAKERS

The various features of the speaker system can be broken down into several sections in order to simplify explanations. The sections break down as follows:

THE SPEAKER SYSTEM - BASICS & TERMINOLOGY

HISTORY

PA speakers, like mixers, power amps and processors, have gone through various evolutionary stages. Starting with "columns" in the early 1960's they progressed to stand-mounted "cubes" in the late sixties, then to bin/horn "stacks" in the seventies, then to advanced, passively crossed over, "all-in-one" enclosures in the mid-1980's to which were added actively crossed over "subwoofers" in the latter 1980's. Now we face the era of "actively controlled" speaker systems wherein multi-functional electronic processors, designed specifically for given speaker systems, monitor the amplifier outputs and adjust the input signals accordingly.

The problem is, at any point in time, users are likely to be employing any one or number of these different speaker systems and they all need to be treated a little differently. You might, for example, encounter no difficulty in boosting the bass EQ for a set of modern, full-range PA cabinets, but try that with an old set of columns or cabinets of lesser quality and the result could be wall-to-wall bits of cone paper (more likely, burned voicecoils). So the bottom line seems to be, "know thy speaker system".

BASICS

Electro-magnetic transducers - woofers, compression drivers and tweeters - all have four things in common;

- (1) a **voicecoil**
- (2) a **magnet**
- (3) a **cone or diaphragm**
- (4) a **frame** or some other structure to hold everything together.

It all works quite simply; a cylindrical voicecoil is attached to the center of the cone or the perimeter of the diaphragm. It is suspended in a circular slot in the centre of the magnet and the outer edge of the cone or diaphragm is attached to the frame. The amplifier's outputs produce electrical signals which vary in polarity according to the frequency of the source signals. When these reach the voicecoil, its electro-magnetic polarity varies accordingly which causes it to be attracted to and repelled away from the magnet. This causes the voicecoil to move back and forth which moves the cone or diaphragm back and forth and that causes variations in air pressure which make our eardrums move back and forth creating the sensation of sound. These back-and-forth movements occur rapidly and are thus referred to as "vibrations".

TERMINOLOGY

- **"Driver"** - a word originally representing the compression driver on a midrange or high-frequency horn, but which today often gets used in place of "woofer", "tweeter" and "midrange speaker". This always leads to confusion in conversations; e.g.. "What kind of driver is in that 3-way enclosure?". Now you have to ascertain if it's the low, mid or high frequency "driver" they're asking about. Therefore, to avoid confusion we will not use the word "driver" very much.
- **"Bin"** - another word that seems to get used for multiple things - "mid bin", "sub bin", full range bin". We will use the word "enclosure" and describe the type.
- **"Woofer"** - the low-frequency speaker in a full-range (2-way, 3-way, etc.) enclosure, or the sole speaker in a subwoofer enclosure.
- **"Horn"** - a veritable field of study in itself. Let's just say for our purposes that this means a complete horn/driver unit, "mid" (midrange) or "HF" (high-frequency) as the case may be.
- **"Compression driver"** - the sound-producing element in a horn/driver unit.

- **"Tweeter"** - a small horn and driver or a high-frequency component which is not attached to a horn.
- **"Enclosure"** - the speaker box - in this case, complete with its components
- **"Speaker baffle"** - the panel at the front of the enclosure to which the woofer, tweeter, etc. are attached. This is also where you will find the tuning port(s) as a rule.
- **"Port"** - one or more holes in the speaker baffle which allow air to move in an out of the enclosure and tune it according to the woofer's resonance in order to optimize low-frequency output.
- **"Crossover"** - a network which separates the low-frequency signals from the high-frequency signals in a 2-way enclosure, or the lows, mids and highs in a 3-way enclosure. Here we are referring to "passive" crossovers. Active crossovers are covered under Signal Processors.
- **"Hz"** - Hertz or cycles-per-second represents the rate or frequency at which something (a diaphragm or cone in this case) is vibrating to produce sound. Most music represents many sound frequencies from the low (e.g.. low "E" on a 4-string bass guitar which is 41.2Hz) to the very high (e.g.. the fine "sizzle" on top of cymbal sounds which is around 10,000Hz).
- **"SPL"** - sound pressure level is the industry-standard measure of loudness. It is measured in decibels or "dB".
- **"Frequency Response"** - Speakers reproduce different sound frequencies with different amounts of loudness. The frequencies are expressed in Hz and the variations in loudness are expressed in dB. Frequency response is plotted on an XY graph with frequency along the horizontal plane and loudness up and down the vertical plane.
- **"Impedance"** - this is an electrical measurement of the physical characteristics of the speaker. It is used to show the interaction between the speaker and amplifier. In simpler terms, it can be viewed as the degree of "difficulty" that an amplifier is going to encounter when driving a speaker enclosure at various frequencies. Although factors in addition to electrical resistance are involved, impedance is expressed in ohms.

TECHNICAL STUFF!!! - IMPEDANCE

Hold a piece of stiff cardboard out at arm's length and wave it slowly back and forth. Now do it faster. Notice how the air mass is impeding its movement? The same thing happens with cones and even diaphragms to a lesser degree. This is physical impedance - mechanical resistance in technical terms. It gets translated into impedance in speakers by a few other factors including capacitance, inductance, reluctance and reactance.

Woofers, for example, tend to increase their impedance at higher frequencies. The higher the frequency goes, the higher the impedance goes and, as a result, the less power the amplifier can deliver. Tweeter and horn diaphragms suffer somewhat less from this process because their surface areas are small and therefore encounter fewer air molecules. Conversely, an 18-inch woofer's impedance may start going up at just a few hundred Hertz. At 1,000 Hz, its impedance could be as high as 12 ohms with the amplifier delivering 60 to 70 per-cent less power to it at that frequency.

If you look at the frequency response graph of a woofer you'll notice considerable "high-frequency rolloff" which reflects the way the line slopes downward throughout the higher frequencies. That's impedance at work. To verify your findings, look at a woofer's Impedance curve (looks a bit like a frequency response graph in reverse). You will see how the line begins to curve upward as you look from left to right. That is the woofer's impedance climbing as the signal frequency goes higher.

You'll also notice something else on the impedance graph - a tall, narrow "spike" over on the left side, down in the low frequencies. This reflects the woofer's natural "resonance". What happens is, speakers, like all things which work by vibrating, have various physical factors that cause them to favour certain frequencies. When a woofer receives an amplifier signal at its resonant frequency, it wants to move farther in and out than at other frequencies. But in the process of doing so, the woofer's voicecoil cuts added lines of force from the magnet and generates extra "counter-EMF".

As mentioned earlier (see Slew Rate & Damping Factor under the Power Amp) counter-electro-magnetic force is the voltage induced in a voicecoil because it is moving back and forth in the magnet's field. This raises the impedance whenever the speaker tries to reproduce that specific frequency. Additionally, the air load of the enclosure (sealed or ported) is the main force on the speaker cone, hence the size of the enclosure and its ports affect impedance. Even applied power affects impedance.

Normally the woofer's magnet acts as a heatsink for the voicecoil, but if high power levels are applied for long enough, the voicecoil warms up the magnet and, as a result, the voicecoil gets even hotter. This increases its resistance then up goes the impedance and down goes the applied power.

- *{ **TIP** - If you notice the bass response sounding a little weaker as time passes during a performance, it could be due to this heating effect. Your first reaction might be to boost the low EQ frequencies slightly which is fine, but watch out for signs of amplifier clipping.}*

THEN, WHAT IS IMPEDANCE?

Well, it's partly the electrical resistance of the woofer's voicecoil plus the speaker cable, partly the mechanical resistance of the woofer's cone operating within the enclosure and/or horn, partly counter-EMF generated by the woofer's voicecoil moving in the magnet's field, partly the crossover's resistance, inductance, etc. and partly a few other things. One thing impedance is NOT is "fixed". The only time an "8-ohm" enclosure is likely to register exactly 8 ohms on a meter is either going to be when the meter puts a little DC (battery) current through it and the enclosure turns out to have that exact DC resistance (very uncommon) or in those instants when the audio program (music?) produces the frequencies which cause the speaker to operate at exactly 8 ohms.

Then, how do manufacturers determine a speaker enclosure's impedance? First it is designed to operate at that impedance ON AVERAGE, then it is measured while operating to verify the load.

- *{ **TIP** - When shopping for subwoofers, be sure that you find out the manufacturer's recommended crossover frequency. This is important because the impedance of a subwoofer tends to go up quickly when it receives signals higher than those in its recommended range. As a result the average impedance will be higher than you would expect and the amplifier will put out less power. In other cases, the impedance may actually go down within a range of frequencies above the recommended range which would reduce the average impedance and possibly endanger the amplifier. Needless to say, it's wise to make sure that your electronic crossover is working at the right frequency (most of them have Frequency controls which are not very accurate). Have a technician check it out or you can do it yourself if you have a pink noise generator and a good quality realtime frequency analyzer.}*

PARALLEL LOADS (MORE THAN ONE SPEAKER PER AMPLIFIER CHANNEL)

Calculating parallel loads is an important capability for two main reasons;

- first, because dual speaker connections whether on an amplifier, a mixer/amplifier or a speaker enclosure are all wired in parallel. Some people think that if you run separate speaker cables from each speaker output on the amp or mixer/amp to the enclosures you somehow "avoid" putting the speakers in a parallel circuit. Others think that if you run a speaker cable from one cabinet to another you put the cabinets in "series" and that just adds the two loads together (e.g.., two 4-ohm speakers in series = 8 ohms). But the truth is that everything gets put in parallel. In fact it's quite difficult to put speaker enclosures in series - you need a special wiring harness.
- The other reason for needing to know how to calculate parallel loads is because amplifiers don't like running into loads which are too low. As mentioned in the Amplifier section, they will usually shut down if the load is too low and some of them may actually sustain damage.

THE FORMULA

- **1/R = 1/R1 + 1/R2 + 1/R3 + etc. ("R" = ohms)**

EXAMPLES

Say you have two 4-ohm enclosures, an 8-ohm enclosure and a 16-ohm enclosure all in parallel. Of course you would never do such a thing because the lower-impedance speakers will get more power and be louder than the others (you figured that one out yourself, right?), but this is just an example.

The solution goes as follows:

1/R = 1/4 + 1/4 + 1/8 + 1/16 = 4/16 + 4/16 + 2/16 + 1/16 = 11/16.

Therefore R/(1) = 16/11 = 1.4545 ohms.

If you're not into finding "lowest common denominators", just get out a calculator and turn everything into decimal equivalents as follows:
1/R = .25 + .25 + .125 + .0625 = .6875
Therefore R/(1) = 1/.6875 = 1.4545.
To keep life as simple as possible, most people put enclosures of the same impedance in a parallel circuit. If you do this it's all just a matter of dividing that impedance by the number of speakers. Example; four 16-ohm loads in parallel = 16/4 = 4 ohms. Similarly, two 8-ohm loads in parallel = 8/2 = 4 ohms.

The following is a quick reference listing of some commonly used parallel loads: ("R" = ohms)

- **2 x 16R loads = 8R**
- **2 x 8R loads = 4R**
- **2 x 4R loads = 2R**
- **3 x 16R loads = 5.33R**
- **3 x 8R loads = 2.67 R**
- **3 x 4R loads = 1.3R**
- **4 x 16R loads = 4R**
- **4 x 8R loads = 2R**
- **4 x 4R loads = 1R**

THE SPEAKER SYSTEM - CABLE ISSUES

Speaker cable, as mentioned earlier, is a real factor in determining a system's overall impedance. Cable presents the overall load with additional resistance. It is even possible for speaker cable to add inductance.

- *{ **TIP** - In certain rare cases you might find that coiling a speaker cable the right way can get rid of radio frequency interference. Once in a while, powerful radio signals effectively get picked up by the speaker lines. This interference then goes back into the amplifier's grounding, is amplified and comes out the speaker system. In a few instances it has been reported that winding the cables into coils and taping them in place actually helped to reduce the problem.}*

Resistance, however, is the main additive and it varies in direct proportion to the gauge and length of the wire. The effect that it has on the system's impedance is fairly easy to chart out; what we offer instead is the effect that cable has on the amplifier's delivered power. The reason we do this is because the percentage of power loss varies, not only according to the length and gauge of wire, but also according to the speaker impedance. As you will notice in the below **"CABLE LOSS CHART"** the lower the speaker load, the higher the loss.
For obvious reasons, it makes sense to pay attention to what speaker cable you are using. In general, the less wire, the better. The heavier the gauge, the better. We suggest that you print out the **"CABLE LOSS CHART"** and spend some time studying it. This could dramatically improve your system performance!!

WIRE GAUGE (AWG)	LENGTH (FT.)	SPEAKER LOAD (OHMS)	POWER LOSS (% OF WATTS)
8 OHM LOADS			
18	25	8	6.82
16	25	8	4.35
14	25	8	2.75
12	25	8	1.67
18	50	8	12.94
16	50	8	8.41
14	50	8	5.39
12	50	8	3.29
18	100	8	23.44
16	100	8	15.76
14	100	8	10.34
12	100	8	6.41

WIRE GAUGE (AWG)	LENGTH (FT.)	SPEAKER LOAD (OHMS)	POWER LOSS (% OF WATTS)
4 OHM LOADS			
18	25	4	12.94
16	25	4	8.41
14	25	4	5.39
12	25	4	3.9
18	50	4	23.44
16	50	4	15.76
14	50	4	10.34
12	50	4	6.41
18	100	4	39.33
16	100	4	27.98
14	100	4	19.10
12	100	4	12.21

WIRE GAUGE (AWG)	LENGTH (FT.)	SPEAKER LOAD (OHMS)	POWER LOSS (% OF WATTS)
2 OHM LOADS			
18	25	2	23.44
16	25	2	15.76
14	25	2	10.34
12	25	2	6.41
18	50	2	39.33
16	50	2	27.98
14	50	2	19.10
12	50	2	12.21
18	100	2	58.95
16	100	2	45.43
14	100	2	33.08
12	100	2	22.24

THE SPEAKER SYSTEM - POWER & SPL

POWER

First, let's establish that a speaker's power rating represents its maximum power CAPACITY. The reason power capacity is specified in speakers is to hopefully prevent them from being blown up by amplifiers which are too powerful for them.

There is some feeling in the consumer market that a power rating means you "must" use that many watts. This is not true in the case of PA speakers. Additionally it is sometimes assumed that a speaker's power rating directly reflects its sound output. This is also not true. If you apply the maximum power to a speaker it will (should) provide the maximum sound-pressure level specified by the manufacturer. If you apply less than that much power you will get less than the maximum SPL, but how much less?

Consider the following; if you reduce applied power by 50%,the SPL will be reduced by -3dB reflecting a decrease in perceived loudness of roughly 30%, possibly less depending on who you ask. If the system was producing 110dB, the SPL would still be around 107dB with 50% of the applied power gone (believe it or not!). And if you think that's unusual, consider the reverse situation - doubling applied power only nets +3dB or roughly a 30 to 40% increase in perceived loudness - more about this later.

THE DISTORTION ISSUE

The main reason for PA users thinking that you "must" apply 100 watts to a 100-watt speaker or 500 watts to a 500-watt speaker, etc., is because of distortion. The problem is that a mere 50 watts can blow up a 100-watt speaker if the amplifier is sufficiently distorted (again, believe it or not!). Therefore the reasoning seems to go that if you have the maximum power available to the speaker, you will have plenty of power "headroom" and you will never turn up the volume far enough to distort the amp and blow the speaker because by then the system will be too loud.

The problem with this reasoning is that "too loud" is just too vague and most people like things extra loud anyway, especially the bass. What often happens is somebody increasing the low-frequency EQ below the enclosure's low-frequency limit (if the frequency response specifications say "50Hz - 20kHz", usually 50Hz will be the low-frequency limit). Now, the amplifier's remaining power headroom gets gobbled up in an effort to make the woofers do the impossible and distortion is just around the corner. But that's not all; the woofers are actually trying to do the impossible and reproduce frequencies that are too low for them. This causes a rapid drop in their power-handling capacity and they blow, even if the power is not distorted and the amp is putting out substantially less than maximum power.

- *{ **TIP** - If your power amps have "High-Pass Filters" or "Low Frequency Cut" buttons or "Subsonic Filters", use them. If your amplifiers don't have such a feature but you do have a graphic EQ, pull down the EQ faders which are of lower frequencies than the speaker's low-frequency limit. Do these things and your system will sound cleaner, louder and better.}*

Before we leave this topic, keep in mind that a 100-watt distorted amp can blow up a 100-watt speaker even faster than a 50-watt distorted amp. The main reason distortion damages speakers is because somebody wasn't watching the amplifier "clip" indicators. Hint - watch those power amp clip lights! If they look too busy, turn down the system level.

POWER RATINGS

Finally we have speaker power ratings. They are expressed in watts, everyone knows that, the question is, what kind of watts. Once upon a time it was "RMS" and now it's "PGM" (program) power. One of the reasons for the change away from RMS (aside from the fact that it was a technical misnomer) was all the marketplace misunderstandings about translating RMS ratings into applied power.

Back in the 1970's when everybody was learning about sound systems, you might hear someone say, "Oh I know that speaker. The rating is 100 watts RMS but you can hit it with two times that much power." Go around the corner and you might hear someone else say, "RMS times three, that's how much power you sock into that speaker - in fact any speaker." (argh!). Meanwhile speakers, horns and tweeters were blowing up like popcorn and repairmen were the only ones making a profit - at least that's how it seemed. Thankfully, "program" power ratings are more reliable when used the right way. Now, when you see "pgm" you know it means "APPLY NO MORE THAN THIS MUCH POWER". Life is simpler and safer.

Oh yes, it's worth noting that applied power is shared by speakers. Two 100-watt speakers powered by the same mono amplifier or one channel of a stereo amp can handle a total of 200 watts (you already knew that, right?).

EFFICIENCY

Some woofers, horns and tweeters are capable of producing more sound pressure for the amount of applied power than others. These are said to more "efficient". Efficiency is expressed as the number of decibels measured at a distance of one meter (3.281ft.) from the enclosure or component, with one watt of power applied. A slight increase in efficiency can mean a great deal to the performance of the system.

- For example, if an enclosure is just 3dB more efficient than another one, you can get the same amount of sound pressure from it with only half as much applied power!

Differences in efficiency between home stereo and PA enclosures tend to be astronomical. Even the best home stereo or studio monitor enclosures can be around ten dB less efficient than outwardly similar PA enclosures. In this case, the PA enclosure could provide as much sound pressure with one tenth the power applied. The average level of efficiency in PA enclosures is approximately 100dB at 1 watt at 1 meter with some being as high as 103dB or even higher. The best PA "boxes" combine smooth frequency response with high efficiency, however there has yet to be an industry standard specification established which combines the two criteria.

SOUND PRESSURE (spl)

Sound pressure is measured in decibels (dB) and represents the real end product of everything covered thus far. In basic terms, sound-pressure level or SPL is loudness. Here are some SPL facts:

- SPL dissipates -6dB each time you double your distance from the source.
- SPL varies by +3dB if you double the applied power to a given enclosure.
- SPL varies by +10dB if you increase applied power by a factor of ten to a given enclosure.
- SPL varies by +3dB if you double the number of similar, close-coupled, in-phase, equally-powered enclosures. ("Close-coupled" means close together, either side-by-side or stacked vertically). The following is a chart of sound pressure levels as you double the number of similar, close-coupled, in-phase enclosures and the total applied power. Note; efficiency is assumed to be 100dB at 1 watt at 1 meter.

- The average ear interprets a 3dB SPL variation as roughly a +30% or - 20% variation in perceived loudness.
- The average ear interprets a 10dB SPL variation as roughly a +100% or - 50% variation in perceived loudness.
- The quietest sound we can hear is around 12dB, but that would have to be in a very silent place. External sounds occurring at a level below 12dB, get drowned out by the sound of blood rushing in the vessels in our ears. A more normal quiet place, the public library, tends to have a 40dB noise "floor" (constant average noise level) caused by people rustling pages, moving about or whispering, traffic outside, air conditioning, etc. Here again, some sounds which are quieter than the noise floor are likely to be at least partially drowned out. And if you wonder why it is necessary to shout at a rock concert it's because normal human speech averages roughly 70dB while the PA averages 90 to 120dB in various audience areas.
- On the opposite end of the audible sound range, the maximum SPL we can handle before it becomes painful is around 130dB, however this can vary between individuals. The hearing loss resulting from such encounters is usually temporary unless the experience is repeated too frequently (more about hearing loss under RUNNING THE SYSTEM).
- In the workplace, the Dept. of Labor considers a "safe" SPL to be 90dB for 8 hours a day.
- Rock concert PA enclosures can produce better than 130dB SPL's at one metre. If the front row is 4 metres or 13.12 ft. away, the SPL at that distance could be over 118dB, that is if only one enclosure is working.

FLETCHER AND MUNSEN

The frequency response of your hearing changes as the SPL goes up and down. This was documented by researchers Fletcher and Munsen in the 1940's and is reflected in the famous Fletcher-Munsen Curve shown below. Note that this is a set of "applied power" curves, not a frequency response graph (although it's rather like an upside-down frequency response graph of human hearing in point of fact).

Looking at it, certain things become apparent to you right away. First, it's clear why humans enjoy lows and highs on home stereo so much - at low sound-pressure levels we have difficulty hearing them compared to all the midrange frequencies, especially around 3kHz where our hearing is most efficient. But it is also clear that, with increases in the SPL, the need for a "smiling" EQ setting to provide those extra highs and lows is decreased as our hearing response smoothes out.

This is good news because "smiling" EQ settings gobble up power headroom in both the amplifiers and the speaker system. That situation may be alright at low levels when there is ample headroom but, as the SPL goes up, the EQ should be flattened out to prevent distortion (and remember to cut the EQ below the enclosure's low frequency limit if the system levels are being maxed-out).

SPL DISSIPATION

In the open air with no wind, or in a large anechoic chamber, SPL dissipates -6dB each time you double your distance from the source. Surprisingly, this rule does not change with horn loading. Even the mythical "long throw" horns obey this law. If their sound travels farther it's because their output or "source" SPL is higher. Although the -6dB rule varies indoors with the size and physical makeup of the venue, and outdoors with wind velocity and direction, the fact remains that sound pressure does diminish over distance at a rate which is seldom less than -6dB as you double your distance from the speaker system, and is usually more - especially in clubs with sound-deadening architecture, furnishings, etc.

- *{ **TIP** - In large venues or outdoors, you may need to optimize the SPL of you speaker system so that audience members at the back can hear properly. Under these circumstances it is the mids and highs which need to be optimized most because they contain the most musical information including the vocals. To accomplish this, stack the horns vertically. If you have full-range enclosures with the horns at the top, stack them "head-to-head" so the horns will be as close together as possible. Be careful that they are aligned properly. The idea with "close coupling" (what we're doing here) is to have the sound waves from the close-coupled sources radiating in unison so that they reinforce each other. If the sources aren't aligned, the waves will be out of phase with each other to some degree and the coupling effect will be lessened. If the alignment is bad enough, sound cancellation can take place.}*

The following is a chart of sound pressure levels from one enclosure as they dissipate over distance at -6dB per doubled distance:

SOURCE SPL (dB at 1 meter) @ DISTANCE(meters) = RESULTING SPL(dB)
[Note: 64 meters = apx. 210 feet]
130dB @ 2m = 124dB, @ 4m = 118dB, @ 8m = 112dB, @ 16m = 106dB, @ 32m = 100dB, @64m=94dB
125dB @ 2m = 119dB, @ 4m = 113dB, @ 8m = 107dB, @ 16m = 101dB, @ 32m = 95dB, @ 64m = 89dB
120dB @ 2m = 114dB, @ 4m = 108dB, @ 8m = 102dB, @ 16m = 96dB, @ 32m = 90dB, @ 64m = 84dB

MULTIPLE ENCLOSURES
SPL varies by +3dB if you double the number of similar, close-coupled, in-phase, equally-powered enclosures. ("Close-coupled" means close together, either side-by-side or stacked vertically). The following is a chart of sound pressure levels as you double the number of similar, close-coupled, in-phase enclosures and the total applied power. Note; efficiency is assumed to be 100dB at 1 watt at 1 meter.

NO. OF ENCLOSURES @ TOTAL APPLIED POWER (WATTS) = RESULTING SPL @ 1METER (theoretically)
1 @ 1w = 100dB, 2 @ 2w = 103dB, 4 @ 4w = 106dB, 8 @ 8w = 109dB, 16 @ 16w = 112dB
1 @ 10w = 110dB, 2 @ 20w = 113dB, 4 @ 40w = 116dB, 8 @ 80w = 119dB, 16 @ 160w = 122dB
1 @ 100w = 120dB, 2 @ 200w = 123dB, 4 @ 400w = 126dB, 8 @ 800w = 129B, 16 @ 1600w = 132dB

- *{ **TIP** - It is interesting to note that the difference in sound pressure between one enclosure with 100 watts applied and sixteen enclosures with a total of 1600 watts applied is only 12db! In other words, the average ear would hear slightly more than a 100% increase in perceived loudness if the two systems were compared in a blindfold listening test at a distance of one meter. However, the difference in SPL at a greater distance would be more noticeable. This is because the acoustic output of the sixteen, close-coupled enclosures would combine over distance to reinforce the SPL so that the difference between one enclosure and sixteen would be greater than 12dB at a distance from the source. This reinforcement effect helps to keep the PA audible. For example audience noise level at a concert can be so high that the PA may be drowned out if its SPL is only a few dB quieter. If, for example, you were in the audience, 64 meters (210 feet) from the system and the audience noise level around you was 90dB, you would probably still be able to hear the PA which puts out 130dB at one meter (see "Source SPL @ Distance = Resulting SPL" above). But the 120dB source would result in only 84 dB at 64 meters and would be drowned out.}*

THE SPEAKER SYSTEM - PHASE & FREQUENCY

PHASING and POLARITY
It is important for all enclosures to be in phase with each other. It is even more important that dual woofers within the same enclosure be electrically in phase, i.e. that they have the same polarity. Sound travels in waves. When there is more than one source of a given set of waves, they can either reinforce each other when their peaks and valleys are synchronized or cancel each other out when they're not. Sound wave cancellation can mean no sound or at least a serious decrease in sound-pressure level and usually a change in sound quality. Consider that if one of the two woofers in an enclosure, let's say a box with two fifteen-inch woofers, was accidentally wired with reverse polarity to the other one (manufacturers take pains to check for this) the SPL from those woofers could be attenuated by minus 10dB or worse. To put this in some perspective, it would be the equivalent of losing 90% of the power applied to them.

Things are not quite so bad when you have two similar enclosures side-by-side and one of them has reverse polarity (speaker cables soldered-up dissimilarly are usually the culprit here) but the SPL can still be attenuated by around minus 6dB which is roughly comparable to a 75% applied power loss. Similar speakers with reverse polarity to each other are said to be "180 degrees" out of phase - as far out of phase as you can get.

PHASE ERRORS
Acoustic phase errors can range from a few degrees to over 100. The culprits here are matching and positioning. If, for example, you were to place two different types of subwoofer side by side, they would tend to encounter cancellations at certain frequencies, but reinforcement at others due to their differing phase characteristics. The net result would be uneven frequency response and probably less overall SPL than if the enclosures were identical.
The rule here is simply to make sure that speakers doing similar jobs are similar. Well, that is if they are side-by-side. If they are not positioned immediately side-by-side, there will be much less phase cancellation which leads us to positioning.

POSITIONING
Positioning is important when you need to have similar speakers reinforce each other's outputs. As a result it is possibly a more common cause of phase-related sound problems than reversed polarity. However, the SPL losses are often lessened by sound bouncing off walls, floors and ceilings thus softening the effects of cancellations. Still, subwoofers positioned in some manner other than close together will suffer losses as a rule. The effect is less low-frequency sound pressure than you would expect.

PHASE TIPS

- To test for electrical phasing, touch the positive and negative terminals of a 9-volt battery to the + and - leads of a speaker cable plugged into the enclosure. This will cause the woofer cones to move slightly. If the cones move out, the cable and enclosure are in phase. If they move in, the cable or enclosure wiring is out of phase.
- Another test method requires a microphone and a VU meter (the level meter on your mixer may work if it is fairly precise). Face your enclosures towards each other, around six feet apart. While someone plays a single, low note through the PA, perhaps on synthesizer, hold the mic between the woofers of the two cabinets and slowly move it back and forth while watching the meter. If the reading gets higher when the mic approaches the half-way point between cabinets, they are in phase. If it decreases, they're out of phase.
- If it appears that something is out of phase, check the cables first; cable manufacturers are somewhat less careful about phasing than speaker manufacturers. The clearest indication that a cable is out of phase is when the wire leads are not soldered to exactly the same lugs on both connectors. Take the cable connectors apart. Most flat speaker wire has one of the leads marked with printing (the industry standard is "printing to positive" - centre tab if it's a 1/4" plug).
  In any case, the + & - connector lugs should be soldered to the same cable leads on both ends. If not, unsolder, reverse and re-solder the leads on one of the connectors (only). If the cable is not wired out of phase, the cabinet's input wiring must be reversed. Remove the connector panel and reverse the wires on the input connector. You may want to get some help from your local service technician for this one!!
- If you are having a problem with what seems to be low SPL and all the electrical phasing checks out OK, the problem could be acoustic phase cancellation where only sections of the sound range are out of phase - e.g.. the lows or mids. Cabinet alignment is critical for proper acoustic phasing. For example, two subwoofers side by side but facing in opposite directions would be 180 degrees out of phase resulting in at least a -3dB effect on SPL, the equivalent of losing 50% of your power. Two boxes with one facing off to the side would have a 90 degree phase error.
- As a matter of good general practice, close-couple all the enclosures side-by-side or stacked vertically and face them in the same direction. This maximizes SPL by minimizing phase cancellations.

Some places can be veritable "SPL sponges", a phenomenon related to the architecture, building materials and/or furnishings. If this turns out to be a problem, try the following:
(1) Position and aim the mid & high frequency horns, so that there is line-of-sight with the audience (if you can see a speaker clearly, you 'should' be able to hear it), This will probably require stacking pairs of full-range enclosures head-to-head so that the horns are close together.
(2) Flatten the main EQ (set all faders at centre) to give the system as much power headroom as possible
(3) Make sure that all power amplifier level controls are at maximum
(4) Make sure the Input Gain controls on the mixer channels are all set sufficiently high to allow a small amount of input clip indication, then increase the main mixer levels as far as you can without either offending the audience or causing feedback.}

FREQUENCY RESPONSE
Frequency response represents the amount of sound pressure that a speaker produces at all frequencies with a fixed amount of signal gain applied. The reason "gain" is specified here rather than power is because the amount of applied power will vary with changes in the speaker's impedance at various frequencies.
Fixed gain means that the intensity of the test signals is the same from one frequency to the next (at least that much can be regulated). The results are detected by a microphone with very flat frequency response connected to a computer which plots them on an X-Y graph with frequency along the horizontal axis and SPL up and down the vertical axis.
Ideally the result would be a perfectly straight and horizontal line from left to right meaning that the speaker reproduces all frequencies at the same SPL. In real life, response graphs indicate all kinds of peaks and dips with the far right and far left ends bending down where the lows and highs roll off. The degree of accuracy of a frequency response graph can be ascertained by noting the "dB" markings down the left side. Response graphs can be calibrated so that every vertical marking is 5 or 10dB louder or quieter than the one above or below it. It is important for you to ascertain this dB scale when evaluating a response graph because it indicates the accuracy of the results.
Remember, a difference of 10dB means either double or half the audible loudness - a HUGE difference when applied to the sound of something (see GRAPHIC EQ under SIGNAL PROCESSORS). A response graph which is calibrated in 10dB increments may look deceptively smooth compared to some other graph calibrated in 5dB increments. Don't be fooled.

THE SPEAKER SYSTEM – COMPONENTS AND ENCLOSURE

FEATURES
The size, shape, configuration and type of an enclosure are amoung its most obvious features. The sound-producing components are at least equally important and the passive crossover, if there is one, comes in a close second. Beyond these, other features include cabinet construction materials, hardware, input connectors and finish. It seems odd that the most glamorous part of a sound system would be so devoid of bells and whistles, but there you have it. Here then, is a list of speaker features - few though they may be.

HORN-LOADED vs. BASS REFLEX ENCLOSURES
The era of horn-loaded "bass bins" has all but passed in general application P/A systems. The added efficiency which bass horns offer comes at the cost of smooth, extended bass response and/or compactness. Efforts to produce more compact bass horns in the latter 1970's netted low end response which rolled off abruptly below the bass horn's cutoff frequency (all horns have one) giving them a comparatively "hard" sound.
Modern uses of low-frequency horn loading in subwoofers have brought some improvements in performance, but horns still must obey nature's laws and roll off quickly below cutoff. In fact, some of the very large bass horns developed during the 1930's and '40's remain among the best in terms of low-frequency performance which possibly indicates that there aren't many secrets left to uncover in this technology.

Reflex technology is similarly well known. Its potential advantages are compactness and smooth, extended bass response. Higher-powered woofers and amplifiers have now made it possible to obtain higher sound pressure levels from reflex-type enclosures and this, combined with their superior low-bass response, has swept them to prominence in both concert and club applications.

BOX DESIGN SHAPE - Trapezoidal vs. Square

The reason for that "wedge-back" trapezoidal enclosure geometry is to facilitate the creation of semicircular arrays where multiple enclosures are arranged closely side-by-side. Created initially for "flying" in elevated clusters, trapezoidal enclosures have also found their way onto stages all over and are used in small numbers with great success.

Their shape looks a little horn-like leading sometimes to speculation about them having "longer throw" than similar, square enclosures - of course we know better now. However there is a potential benefit in this shape - when properly configured, it reduces internal standing waves which means smoother frequency response and possibly improved phase coherence, a factor which gives the enclosure a "tight", "focussed" sound. On the other hand, not all trapezoid boxes automatically benefit from this shape, nor do all square boxes fare less than favorably in comparison to them.

ENCLOSURE SIZE - Is Bigger Better?

The size of an enclosure is often assumed to reflect its bass response and acoustic output potential. This is not automatically true. There is an optimum size for an enclosure which is dictated by the various design parameters of the woofer. If it is too big, the frequency response will be uneven with a "loud spot" over a narrow range of frequencies causing one or two notes to jump out every time they're played. On the other hand, if an enclosure is large in order to properly enclose the woofer or woofers, it could mean more and/or deeper bass response, but that depends entirely on the woofers.

Also, keep in mind that it is now possible to obtain surprisingly large amounts of full-range SPL from smaller enclosures. In fact a well-designed, high-performance compact enclosure can conceivably blow a much larger but inferior box away. Check the manufacturer's specifications. Look for the "maximum SPL" and frequency response figures. If the response numbers are good, the max. SPL may be all you need to consider - regardless of the enclosure's size. Once again, rent the speakers and try them out on a job. Specifications can be good guidelines, but reality is the final word.

PASSIVE CROSSOVERS

Modern passive crossover technology is miles ahead of where it was twenty years ago. Aside from the obvious features such as a variety of input connectors, you may find circuit breakers, fuses, lightbulbs and even transistors in addition to the usual capacitors, resistors and inductor coils.

All of these are desirable and for different reasons:

- On enclosures rated at 400 watts or more, input connectors should include Speakons or XLR's (see Outputs under Power Amp Features). 1/4-inch jack sockets are alright for lower-powered applications, but they only have around one square millimeter of contact area at the tip which means that resistance and heat will build up there when the number of electrons passing through exceeds the number possessed by the atoms in the contact area. Speakon connectors have many times more contact area having been designed especially for speaker usage complete with extra pins to accommodate a bi-amp or tri-amp "snake" (multi-element cable) and a twist-type locking system.
- A circuit breaker or fuse means an added margin of protection for the components. Although such measures are not foolproof, they do at least offer some protection from overpowering and possibly even distorted amp signals. You may find it inconvenient changing bulbs or fuses or resetting breakers, but resist the temptation to circumvent these safety devices with aluminum foil, tape, etc. As a rule they blow for good reasons and if you defeat them, the next thing to blow is likely to be voicecoils and that gets expensive.
- It seems odd to find a lightbulb inside your PA enclosure, but this feature can perform two functions. First, it acts as a time-delay fuse, usually for the mid and high-frequency components, so that large, fast power transients (peaks) will not instantly interrupt mid/high output. The lightbulb may also help to dissipate excess power headed for the mid/highs and should not cause concern if "sighted lighted" - it's just doing its job.

- The presence of transistors, resistors and capacitors may not make an impression on you, but inductor coils - coils of copper wire - are more obvious and are worth looking for. The presence of inductors in a crossover circuit means that the rolloff slope is at least 12dB per octave. Without them the slope will only be 6dB/octave with no rolloff on the woofer which is acceptable for some stereo speakers and studio monitors, but lacks the safety factor that *fast, 2-way and 3-way crossovers afford the components.*["Fast" in crossover parlance usually means a rate of 18 or even 24db per octave]
  On the other hand, some high-performance "touring" enclosures may only have DC blocking capacitors on the horns and/or tweeters. This is because they are designed to be bi-amped or tri-amped using electronic crossovers and separate low (mid) and high-frequency power amps. Full passive crossovers on top of this would defeat one of the benefits of bi- or tri-amping - i.e. no passive crossovers (even the very best ones create small amounts of phase distortion).

WOOFERS, HORNS & TWEETERS

It is generally thought that good-quality components will have cast metal frames and large magnets. As well, there may be brand identification if the components are from a well-known manufacturer. In reality, not all good quality components follow these trends, some are very plain looking with no brand ID.
Times are changing, for example it was once thought that good mid/high horns had to be made of cast metal. Since then it has been discovered that almost any ridged, non-resonant material works well for midrange and high frequency horns as long as the driver is adequately supported. In fact it turns out that some cast metal horns have nasty resonances and are actually less desirable than plastic, fiberglass or wood horns. Consider as well that not all cast metal components are great performers; high-tech-looking cast cosmetics are surprisingly easy to manufacture.
So, in the final analysis, the only generalization which is likely to be fairly accurate is that woofers rated at 400 watts or more should have cast frames. This is necessary to support a big magnet's weight, and a high-performance PA component should have a big magnet. The larger the magnet is, the more magnetic "flux" (power) it can hold. If the magnet isn't big enough, there won't be enough flux to overcome the mass of the high-powered voicecoil (you increase voicecoil power capacity by using heavier gauge wire which weighs more) and the woofer's efficiency will be compromised. Massive magnets need the support of cast frames because cast frames are more rigid than stamped frames which may deform in time due to the magnet's weight and the constant vibrations.

ENCLOSURE MATERIALS

Chipboard is generally used in the production of home stereo speakers and studio monitors. It is comparatively massive and rigid and resonates less as a result, hence less woofer energy is wasted by being converted to cabinet vibrations. Chipboard is cheaper than most grades of plywood (potentially good news price-wise), but it does not hold up on the road, eventually developing the "crumblies" in those areas where it has been dropped or hit with something. It also does not like getting wet.
Three-quarter-inch plywood - mostly 7-ply - is the material of choice for PA enclosures. It is durable and internal bracing in key areas can overcome its tendency to vibrate slightly in large enclosures. Thirteen to eighteen-ply, 3/4-inch wood combines chipboard's mass and rigidness with plywood's durability and can make a fairly noticeable improvement in almost any large enclosure's performance compared to one made of 7-ply. Unfortunately, such multi-ply wood costs considerably more as do the enclosures in which it is used. It is also comparatively heavy and, considering that it principally benefits large enclosures (smaller ones have less wood to resonate and work almost as well in 7-ply) multi-ply wood can be something of a back-breaker.
Plastic has been used for certain small enclosures for many years. Its prime advantages over wood are lighter weight and the ability to be made into interesting shapes. Foam insulation or some other internal muting and stiffening material is usually found in plastic enclosures, partly for acoustic damping and partly because they tend resonate freely and may buzz without it. Large enclosures, especially high-powered subwoofers, are seldom made from this material as it needs to be reinforced with wood or thick fiberglass to prevent vibration. This negates any weight savings and makes the product more expensive. Plastic enclosures have the appearance of being weatherproof and some of them are. Check the manufacturer's specs to be sure. Some plywood enclosures are also weatherproof - again, check with the manufacturer.

- *{ TIP - You can weatherproof paper speaker cones to a fair degree by spraying them with Scotchguard front and back. You will lose a little efficiency and the sound may be slightly different, but the cones will survive dampness, although not a complete soaking.}*

FINISHES AND COVERINGS

Three things have emerged as the favorite finishing materials with most manufacturers. Vinyl or leatherette covering was used by everyone for many years, but has been replaced by indoor-outdoor carpeting on most PA cabinets. The carpeting wears better and it's fairly easy to clean. Painted finishes are still preferred for touring or installation systems, more as a matter of taste and tradition than practical necessity. Tougher paint finishes have only helped the chipping and scratching problems a little so care still needs to be taken.

HARDWARE & FLYING HARDWARE

Once upon a time, hardware meant handles and corner pieces. It still does in the case of most PA enclosures, the exception being "flown" systems. Someone discovered many years ago that you need more than a chain through the handles to safely hang a speaker over people's heads. Even the beefiest-looking bar handle can work loose with vibrations over time. This proved to be a headache for enough people - both literally and in terms of legal suits - that someone began designing metal fasteners and plates, hardened steel eyebolts and other things to aid in the quest for safe "flight".

There are many ways to fly a speaker, but above all, you are putting people lives at risk when you fly them. Seek the advice of a rigging professional before flying speaker cabinets.

Further information may be obtained from ATM Fly-Ware 20960 Brant Ave., Carson, California, 98010-1040, U.S.A.