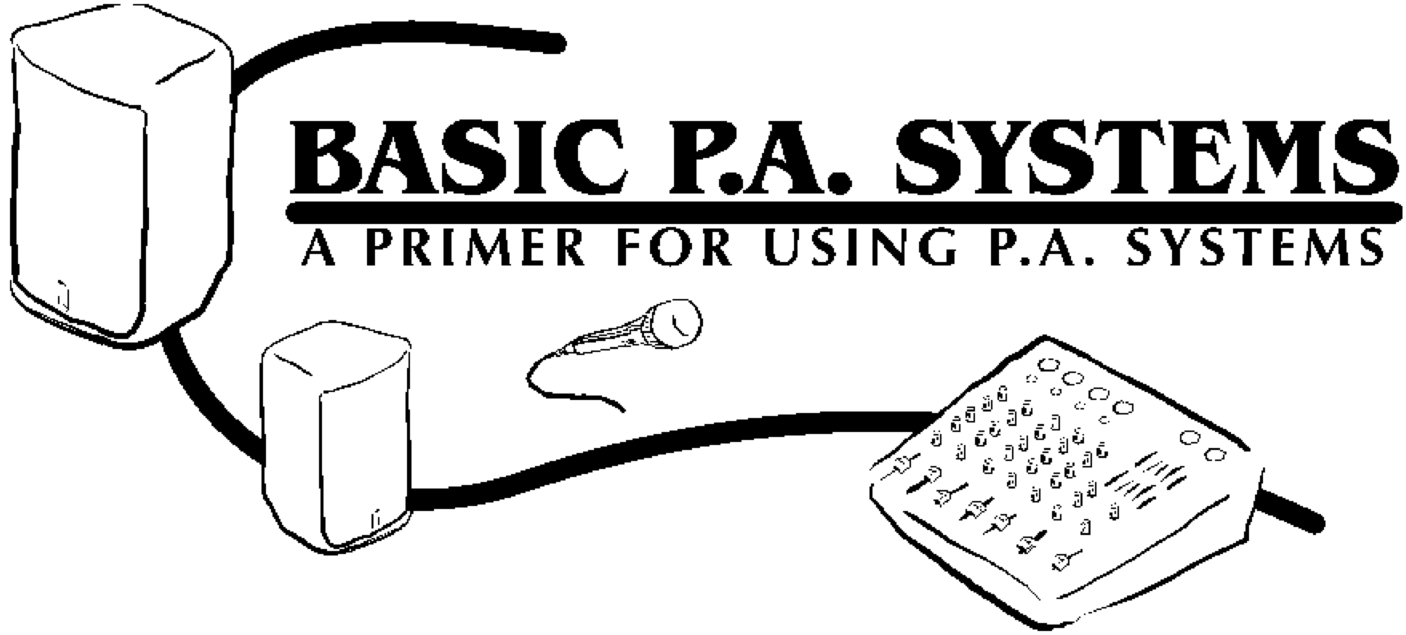
BASIC P.A. SYSTEMS

A PRIMER FOR USING P.A. SYSTEMS

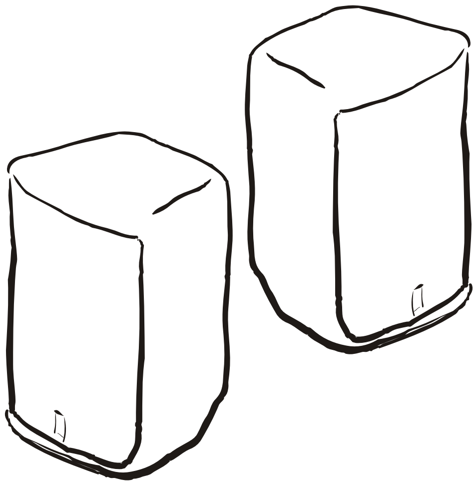


Introduction

This booklet is intended to be a basic manual for the novice P.A. user. For the experienced and highly technical person, a more detailed book would be required. In this booklet we try to answer the most commonly asked questions about P.A. systems in an easy-to-understand way, while keeping the technical jargon to a minimum. We have taken a few liberties as a result, however, we feel that these were required to keep this booklet easy to follow and comprehend.

Why can't I keep connecting more and more speakers to make everything louder?

All amplifiers are designed to deliver their maximum amount of power into a certain number of speakers. This number is indicated (usually located near the speaker jacks) as impedance and is rated in Ohms (named after the man who discovered this electrical property). The most common amplifier impedance is 4 Ohms. This means that the amplifier will put out the most amount of power (be loudest) when it has a total of 4 Ohms worth of speakers connected to it. It will put out less power (be quieter) if the total of the speakers calculates to be more than 4 Ohms. The goal in any P.A. setup is to try and get the speakers to calculate up to, but not less than, the amplifier's rated impedance. If the speakers add up to less than the amplifier's rated impedance, the amplifier tries to put out more power than it was designed to do and it overheats and can be damaged. Generally speaking, the more speakers you add to your P.A., the lower the impedance number becomes. This is why you cannot just add more and more speakers to make everything louder. Fortunately, most modern day amplifiers have built in protection to shut down the amplifier when it gets too hot. This is called load protection.



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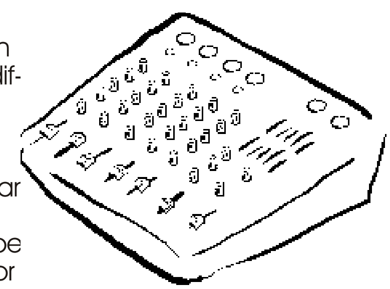
How do I know what my impedance is?

Unfortunately, you have to do a little bit of math to calculate the total impedance for all of the speakers. Virtually all speaker jacks and amplifier jacks are wired in parallel which makes the formula for calculating impedance to be the recipro cal of the sum of all of the reciprocals of the individual speaker impedance. What this means is that the impedance of one 4 Ohm speaker connected to two 8 Ohm speakers would be 2 Ohms which is 1/(1/4 + 1/8 + 1/8). Rather than make you suffer through all of this arithmetic, we can save you the grief with one simple rule and a chart. The simple rule is that if all speaker impedances are the same, you simply take the impedance and divide it by the number of speakers. For example, if you want to calculate the total impedance for four 8 Ohm speakers connected together, you simply take 8 Ohms (which is the impedance that each speaker is) and divide it by 4 (because there are 4 speakers) and you get the answer 2 Ohms, which is generally too low for most amplifiers. If you have mixed impedances you can use the chart below:

Cabinets	Impedance
One 8 Ohm plus one 4 Ohm	2.7 Ohms
Two 8 Ohm plus one 4 Ohm	2.0 Ohms
Two 4 Ohm plus one 8 Ohm	1.6 Ohms
Two 4 Ohm plus two 8 Ohm	1.3 Ohms
Three 8 Ohm plus one 4 Ohm	1.6 Ohms
Three 4 Ohm plus one 8 Ohm	1.1 Ohms

* Any other combinations will require using the formula indicated earlier.

OK, then what is impedance all about on mixers?



If a mixer has a power amp built into it (these are often referred to as powered mixers), then it can have two different types of impedance. The one for the speakers, which we just discussed, and the one for input signals. Unpowered mixers usually have only input impedance and these tend to be referred to as either high impedar or low impedance. This refers to the type of signal that the mixer will accept. Many mixers will accept both type of impedance and have two different types of jacks for this purpose. As a general rule of thumb, if the input jack is an XLR type (three pin) it is low impedance, and if it is a phone type (1/4") it is high impedance. Phono type input plugs (RCA) are always high impedance. If you plug a high impedance signal into a low impedance input (or visa versa) it will not damage anything, but it will not sound very good. Depending how far off the impedances are, the signal will be weak and lacking in frequency response (tone). Microphones can be either high or low impedance; however, instruments such as guitars and keyboards are generally high impedance. Consumer electronics such as tape decks, CD players and tuners are always high impedance and should not be plugged into microphone inputs.

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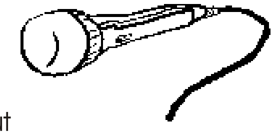
What are the differences between micro phones?

This question has to be answered in several sections:



1. **Types:** There are many types of microphones available but they generally divide into three groups: dynamic, condenser and ribbon

   Dynamic microphones are the most common and the most basic types of microphones. They consist of a diaphragm attached to a coil that moves through a magnetic field inducing a small electrical signal. They operate using similar principals to a speaker and tend to have a limited frequency response, but are very rugged.

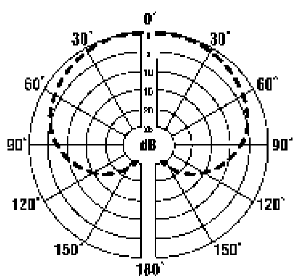
   Condenser microphones have become more popular over the years because they have a very wide frequency response (tone) and are not as delicate or as expensive as ribbon microphones. They are, however, more delicate than dynamic microphones. They require a power source that can be an internal battery, an external power pack or phantom power that is provided by the mixer
   They work by supplying a charge to a fixed plate that creates a capacitor.
   A thin diaphragm is mounted adjacent to the plate and induces voltage changes in the plate when subjected to sound vibrations.

   Ribbon microphonesproduce sound by stretching a thin metal ribbon across a gap of a strong magnet. Sound moves the ribbon across the magnetic field creating electrical impulses. They have an excellent frequency response (tone) but tend to be very delicate and expensive. Since they are rarely used in basic P.A. systems, nothing more needs to be said about them.

2. **Polar Patterns:** This refers to the directions that the microphone will pickup sound from. Polar patterns tend to be divided into two types: omni and uni-directional.

   Omni directional microphonesare rarely used because they pick up sound from all directions and often pick up sound from directions you don't want to have sound picked up from.

   Uni directional microphonesare the most common and probably amount to more than 95% of all microphones sold. They range from shotgun (picking up sound from only straight in front of the microphone) to cardioid (picking up sound from in front of, or somewhat around the front of, the microphone in a heart shape pattern). Knowing the pattern of a microphone is important because it affects what area of sound that the microphone picks up from as well as its susceptibility to feedback.



Cardiod Polar Pattern

3. **Sensitivity:** This refers to the acuteness of "hearing" that a microphone possesses. The higher the sensitivity, the quieter the sounds that it will pick up.

4. **Impedance:** This is important to know so that you buy the proper microphone to match your particular mixer. Although many mixers have inputs for both high and low impedance microphones, some only have one or the other. Generally speaking, if the microphone's impedance is above 1,000 Ohms (1kOhm) it is considered high impedance, otherwise it is considered low. Most professional microphones are low impedance and come with a cable with XLR connectors. High impedance microphones generally come with a cable with phone (1/4") connectors.

There are other specifications for microphones but these three are the main ones that you will see on any specification sheet.

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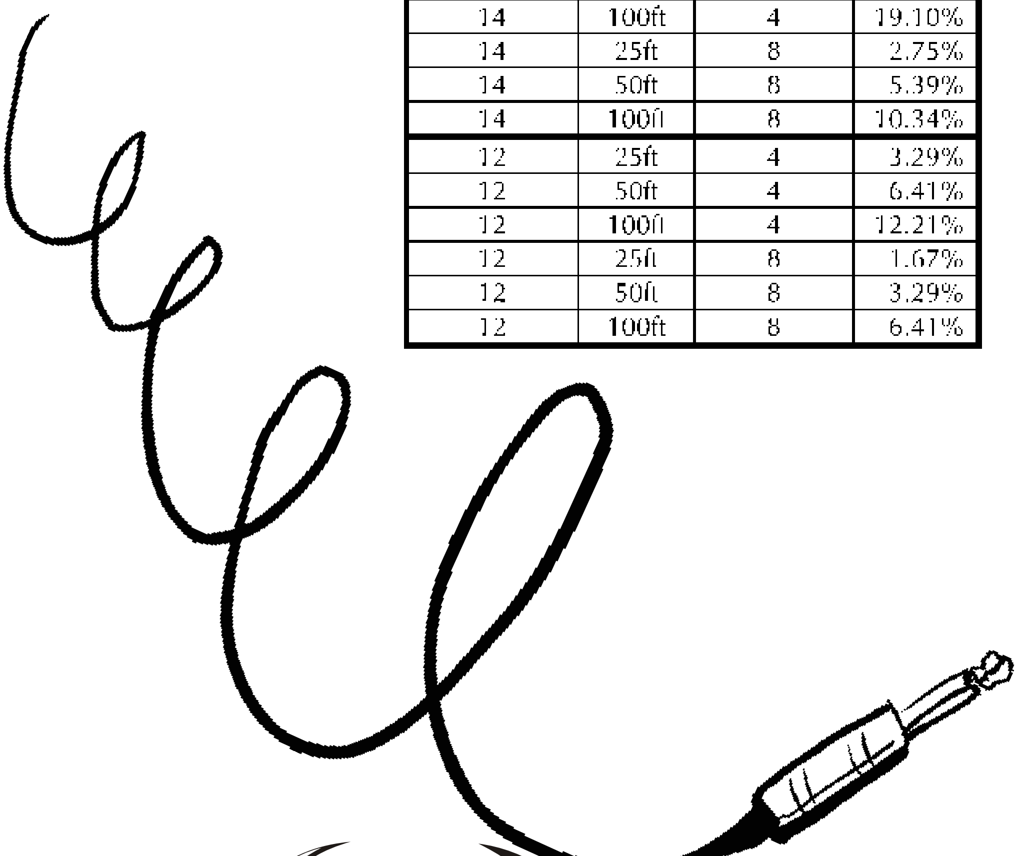
Does it matter what length or thickness of speaker cable that I use?

Cable length and thickness of the wire (gauge) have a major effect on the sound that a P.A. system ultimately produces. The thinner the wire (the higher the gauge number) or the longer the cable means the greater the loss. The loss is related to impedance of the speaker in the system as well. The lower the speaker impedance, the higher the cable loss. Below is a chart of some common lengths, thicknesses and impedances along with their respective cable losses:

Wire Gauge	Length	Impedance	Loss
18	25ft	4	12.94%
18	50ft	4	23.44%
18	100ft	4	39.33%
18	25ft	8	6.82%
18	50ft	8	12.94%
18	100ft	8	23.44%
16	25ft	4	8.41%
16	50ft	4	15.76%
16	100ft	4	27.98%
16	25ft	8	4.35%
16	50ft	8	8.41%
16	100ft	8	15.76%
14	25ft	4	5.39%
14	50ft	4	10.34%
14	100ft	4	19.10%
14	25ft	8	2.75%
14	50ft	8	5.39%
14	100ft	8	10.34%
12	25ft	4	3.29%
12	50ft	4	6.41%
12	100ft	4	12.21%
12	25ft	8	1.67%
12	50ft	8	3.29%
12	100ft	8	6.41%



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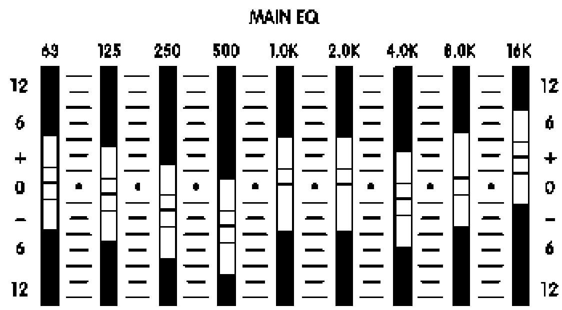
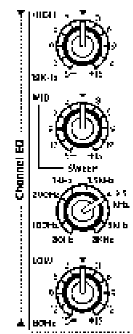
What does an equalizer do?

An equalizer is really just a fancy name for tone controls. In fact, many mixers designed these days actually call the tone controls on each channel "equalizers".

Equalizers divide into two types: graphic and parametric. The difference is that graphic equalizers affect certain preset frequencies (tones) and can only increase or decrease the volume of those frequencies. Parametrics, on the other hand, can also tune in a certain frequency (tone) and can also let you determine how much of the frequencies around this "centre" frequency you want to affect (this is called "Q" or "Slope"). There is also a version of the parametric equalizer called a semi-parametric or a sweepable graphic equalizer. This allows you to tune in a frequency but not be able to determine what other frequencies you affect (this is preset by the manufacturer).



Graphic EQ



Semi-Parametric EQ

What is a crossover and how does it work?

Crossovers are used to split the sound into two or more separate sets of frequencies (tones) and are used in every speaker system with more than one type of component. A common speaker system would be the two-way speaker which would include a crossover to divide the sound into low frequencies (bass) for the woofer and high frequencies (treble) for the horn or tweeter. This prevents bass frequencies from getting to the tweeter or horn and damaging it as well as preventing treble frequencies from getting to the woofer and making it sound less pleasing.

If the crossover is inside the speaker cabinet it is usually "passive" which means that it only requires that you hook an amplifier up to it and it works as it was designed. An "active" crossover is far more complicated and requires that you have a separate amplifier for every component in the system. It splits the sound into two or more parts before it gets to the amplifiers and then each amplifier supplies the correct part of the sound to only one type of component. These crossovers are usually one rack space high and have several controls on them. The controls allow the user to tune in what frequencies that are supposed to go to each component plus set the volume level of each. Although active crossover systems are more complicated and expensive, they are more efficient and sound better. There are a few systems on the market now which have active preset crossovers and amplifiers for each component built into the cabinet.
This makes for a remarkable sounding speaker cabinet because the amplifiers, crossovers and speakers are all matched for optimum performance. They do, however, require that they be plugged into the wall and are heavier than regular speaker cabinets.

What is a decibel?

The decibel is a unit of measurement that engineers invented to make it easier for them to communicate about, among other things, amount of sound. It was named in honor of Alexander Graham Bell and started off being a "Bel" but became the decibel (dB) when they decided that Bel was too big to use in most situations. It is an often-misunderstood measurement because it is somewhat confusing to use. This partly because it is used differently for different situations. No matter what the situation, a dB always refers to relative levels (the difference between two signals) and is usually referenced to some arbitrary level (the common ones are listed just under the chart below). In P.A. situations, dB can be used to describe voltage, power or volume. You must know which dB is being referred to, to understand what is happening to the loudness of the sound. Rather than getting technical and listing calculations, here is a handy reference chart to show what all of the dB's mean as they relate to an increase in input signal voltage (dBV):

dB Increase	Voltage Increase	Power Increase	Perceived Volume Increase
1dB	12%	26%	Barely noticeable (3%)
3dB	50%	100%	Noticeable (12%)
6dB	100%	300%	Significant (40%)
10dB	216%	900%	Dramatic (100%)

What is important to understand with the above chart is that the perceived volume increases are subjective numbers and each person will hear the changes in volume slightly differently. The major point to understand is that doubling your amplifier power only results in a 3dB increase, which is not a very large volume increase. In order to double the volume, you will probably need to increase your power by 10 times!

Where the world gets more confusing, is that there are different starting references for dB's depending on what part of the system you are measuring. The common ones are:

- 0dBu which is an arbitrarily set reference used in early sound studios and means 0.775 Volts which means that 10dBu would be 2.5 volts and 20dBu would be 7.75 Volts
- 0dBV which is the more modern and simpler reference point of 1 Volt, which also often gets called "line level." This means that 10dBV would be 3.16 Volts and 20dBV would be 10 Volts.
- 0dB SPL (sound pressure level) which is the statistical threshold of human hearing. A jet aircraft is about a million times louder which makes it 120 dB of SPL and is considered the upper limit (anything much higher is considered lethal).

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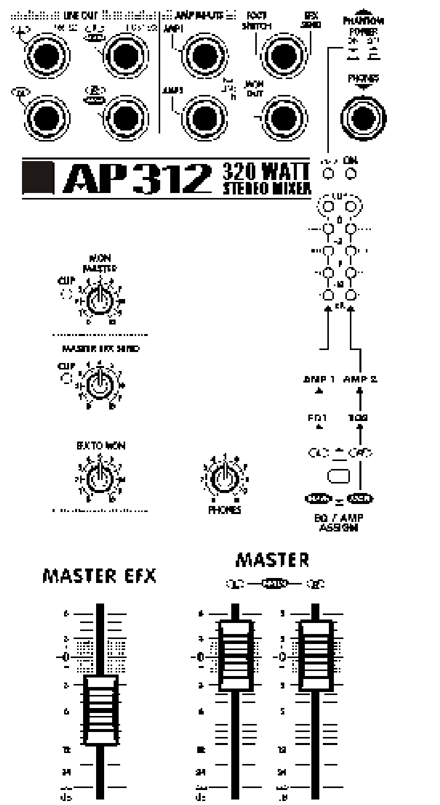
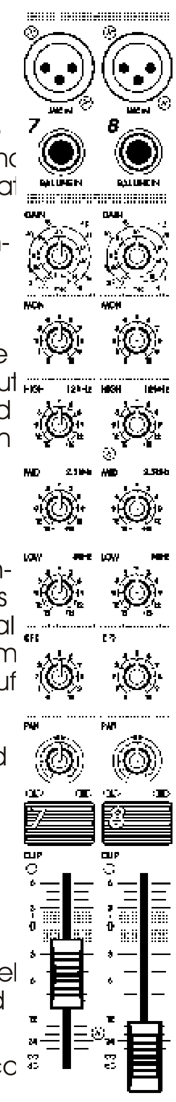
Mixers have so many knobs. What do they all do?

Mixers are divided into two parts: the channels and the master section.



Master Section

Master Section: Most manufacturers locate the master section on the right side of the mixer, although some large consoles might have it in the middle. The master section controls the output of the mixer to all of the various other components that are hooked up to it. These might include crossovers, equalizers, effects processing devices (delays, reverbs, compressors, etcetera) and monitors. Most of these controls are simply volume controls to determine how much signal you want going to any devices hooked up to your mixer. You will often find some tone controls (equalizers) to modify the tone of the sounc and switches to dictate whai each control affects. Unfortunately, different manufacturers use different names for the same functions and this can cause confusion. Fortunately, since we only need to know about a basic P.A. system, pictured here are the most common control you will see in a master section (and many mixers have less than this).



Channel Section

Channel Section: The channels tend to be duplications of each other so, once you have one figured out, you will have them al figured out. All channels tend to have the same types of things on them just more or less of each one, depending on the market that the manuf turer is targeting.

If you have any doubt as to where you should set a control, you should set them as follows:

- **Inputs:** Set it where the clip light starts to come on and back it off very slightly
- **Tones:** Start at zero (usually 12:00 position)
- **Effects:** Start at zero and turn it up to the desired amount of effect
- **Monitors:** Leave at zero unless you want the signal from this channel to go to your monitors in which case you start around half-way and adjust from there.
- **Pan:** Set at the 12:00 position unless you have some special applicc tion for it.
- **Master Section:** The main conttrols should be around 0dB (if it's labelled that way), otherwise it should be set around "7" (if it's labelled 1to 10). The other controls should start at zero and be increased to the desired level, if they are ever used at all.

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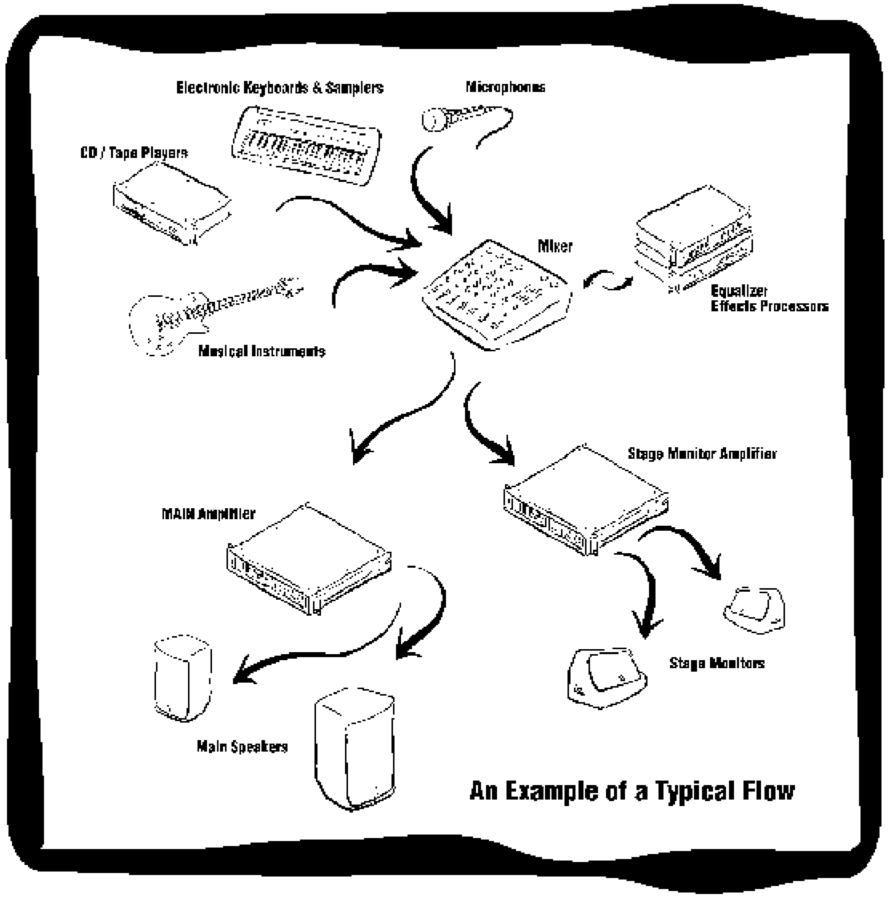
OK, now that I know about the separate components,

How do I hook them all together correctly?

Connecting the various parts of a P.A. system together is not as complicated as it may at first appear. The most important concept to understand is signal flow. If you understand the direction the signal is flowing through the mixer, and you know where you want it to end up going, hooking up the various components to the mixer will be easy.

Since you already know how the signal moves (flows) through a typical mixer (see previous sections), you only need to know where to plug the various components in so that the signal will flow into them. Below is a diagram of a P.A. system showing the most common devices and how they hook up to the mixer. Many manufacturers have started combining one or more of these components into the same chassis as the mixer to make your life even easier. For example, it is very common to see the mixer, power amplifier(s), and some sort of effect (usually reverb) combined into the same chassis This offers many advantages over having these components separate, including:



- less chance of hookup errors because the manufacturer has done this for you
- less chance of having noise problems because the signal paths are so short and properly shielded
- no chance of cable problems because there aren't any (everything is already hooked up internally
- much quicker and easier to setup because most of the system is already hooked up internally
- no chance of compatibility problems because everything has been designed by one manufacturer
- less expensive because you are not paying for individual chassis and power supplies
- easier to use because the components have been designed to interact correctly with each other
- easier to transport because you only need to take one component instead of 3 or more

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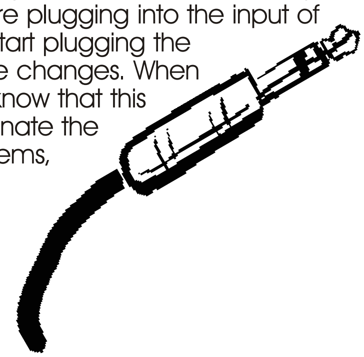
Why do I have noise in my P.A. and how do I get rid of it?

Noise, in its various forms, is a very common, and sometimes a very difficult problem to solve. Noise comes from many sources. Some of the most common ones are:

- poor wiring in the location where the P.A. is being used
- fluorescent lights
- dimmers
- poor design (particularly shielding and location of the mixer's power transformer)
- magnetic fields induced by other nearby components (particularly power amplifiers)
- radio stations or other transmitters in the area
- large motors close by
- grounding problems (particularly when using equipment manufactured by different companies)
- noise in the input signal (particularly guitar pickups)
- poor cables (particularly on the inputs)

Most of these can be reduced or even eliminated by using high quality cables and balanced lines.

To check for the source of the noise, unplug all inputs (keyboards, CD's, guitars, tuners, and etcetera) at the mixer input jacks and listen for noise. Any noise that has now disappeared is definitely coming from something that you are plugging into the input of the mixer and not from the P.A. itself. If this is the case, start plugging the sources back into the mixer, one by one. Listen for noise changes. When you hear noise when you plug something in, you now know that this device is a problem and you should take steps to eliminate the problem with the device (bad cables, grounding problems, pickup noise). Unfortunately, some sources (particularly keyboards and guitars) are noisy and cannot be improved much. However, here are a few suggestions to help with input noise problems:



1. Change your cables to better quality ones with a high degree of shielding.
2. Wherever possible, use balanced sources (some manufacturers have both balanced and unbalanced outputs on their products. If they do, change to balanced. Use of a balanced cable will even improve noise in unbalanced sources as long as the mixer inputs are balanced.
3. Use direct boxes where possible for instruments being plugged into the mixer. This isolates the signal, converts any high impedance instruments to be balanced low impedance and allows you to "lift" the grounds on these devices. Without explaining all of the technical aspects of this, you should be able to reduce or eliminate many radio signals, buzzes and ground hums with this method.
4. Reduce the length of the cables.
5. Make sure the input cables are not lying too close to a transformer, motor, amplifier or other source of magnetic radiation.
6. Plugging the components of a P.A. system into different electrical outlets can sometimes cause problems. Wherever possible, try to connect all parts of the P.A. into the same circuit, even if you have to run extension cords to accomplish this.
7. Turn any lights on dimmers off, or if this is not possible, turn them fully on.

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If however, the noise does not disappear or reduce significantly when you disconnect all inputs, then the problem lies somewhere else in the system. Take a step-by-step approach to determine where the problem lies. First start with the mixer. Turn all outputs to zero. If the noise disappears then the problem is in the mixer; otherwise it is after the mixer. If it is in the mixer, try removing all effects devices such as delays or reverbs, if there are any, and try again. Keep going through the process of eliminating components until you find the problem, keeping in mind that many noise problems are as a result of inferior or defective cables. Eventually you will locate the problem and the same suggestions mentioned earlier (except the one about direct boxes) should help in most cases.

Note: Some hiss occurs when an input is at a high level (or all the way up) is normal.

What is a balanced line?

A balanced line is simply one that has three conductors wired separately. Two of these are signal wires which are wired out of phase with each other and the third one is ground (usually this is the shield). The two most common types of connectors on balanced cables are XLR (3 pin microphone connectors) or tip-ring-and-sleeve (TRS or stereo … "connectors). The big advantage of a balanced cable is that it is designed to cancel many types of noise. Use them whenever possible.

What constitutes a quality cable?

Before we answer this, you need to know that there are three different types of cables found in a P.A. system: balanced, unbalanced and speaker.

- Balanced:only needs to be wired correctly and have a shield.
- Unbalanced:must have a very high quality shield, and, if it is long, have a reasonably high gauge (thick) conductor.
- Speaker: the thicker the conductors, the less signal loss (see earlier chart). This cannot be shielded wire.

Perhaps a little mention about gauge numbers is in order here. People sometimes get a little confused about gauge (abbreviated AWG) because the larger the number means the thinner the wire. For example, a 16 gauge wire is thinner than a 12 gauge wire.

What do you mean by "clipping"?

Clipping is another way to describe distortion. Technically, it is used to describe a certain type of distortion (see drawing on following page), but often gets used to describe distortion in general. The reason that it is called clipping is that the top of the waveform (sound) is actually clipped off (removed).

- A discussion about clipping can get quite involved, however, in an effort to keep it as simple as possible, we will start with a basic explanation about sound.

No matter what sound you are hearing, it has traveled through the air to your ear by vibrating. Some sounds are complicated (music) and others are quite simple (a single pure tone). A sound travels through the air to your ear very much like the way a string vibrates in the air when you stretch it tightly between your hands and pluck it. A simple pure tone travels in a pattern such as that shown in the diagram on the following page. This is referred to as a sine wave. Notice in the diagram, that volume is simply a matter of far the sign wave travels above and below the zero point. As you turn your volume control up on your amplifier, the peaks and valleys of the waveform increase (get higher and lower respectively).

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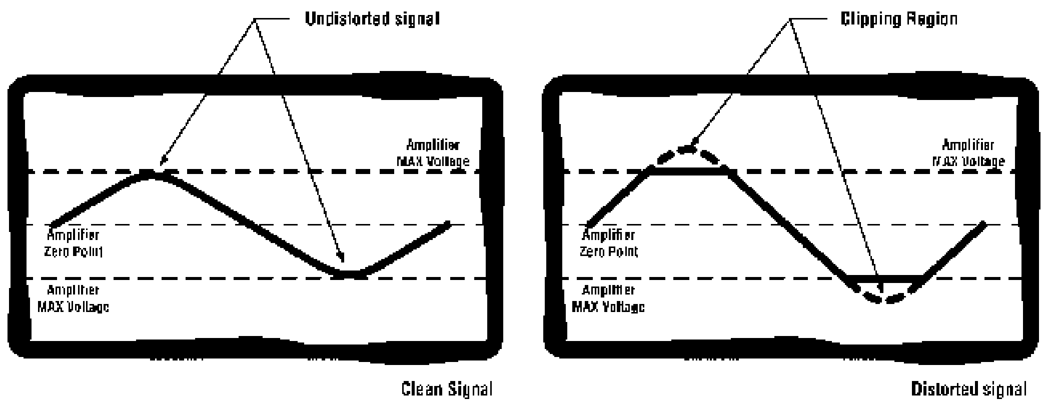
In a perfect amplifier, every time you increase the volume control, the end result would sound exactly the same, except only louder. In fact, an amplifier should have absolutely no affect on the sound except to make it louder. Unfortunately, all amplifiers have a certain maximum level (loudness) that they can attain. If they are asked to deliver a higher level, they cannot do it. This is when you get "clipping". In the diagram below, the maximum level of a particular amplifier is indicated by the dotted line and referred to as "Max Volume". This diagram shows two sine waves (sounds). The smaller one is not so loud as to exceed the maximum volume (loudness) of the amplifier so it sounds normal. However, the larger one tries to go past the amplifiers maximum volume and gets cut off ("clipped") for part of its movement. This is heard as distortion. It generally sounds very raspy and is most unpleasant in a P.A. system. In fact, this is exactly what happens in a fuzz box or when you use a guitar amplifier to get that heavy distorted sound so prevalent in today's rock and roll music (however, since this is all about P.A. systems, we won't go into that topic any deeper). The harder an amplifier is driven (the louder you try to make it), the greater the amount of the waveform that gets clipped. Anything more than 3dB (there's that dB stuff again), is quite audible. If you are running a P.A. system and you see the clip light coming on regularly, or, heaven forbid, staying on, you are running your system too loud and you should turn it down.



OK, that covers amplifiers, but what about other types of clipping?

A very common (and extremely important) type of clipping that you will run into in a P.A. system is when you overload the input (front-end clipping). This is exactly the same as amplifier clipping so you don't have to know anything more than you just learned earlier. The input of a mixer (or a crossover, equalizer, effects processor, or anything else for that matter) is simply a tiny amplifier that works the same way as the amplifier that was described earlier. A mixer just happens to have lots of little amplifiers (one on each input channel) and often includes a clip light for each one. Just like when you use an amplifier, if the clip light comes on too often, or if it stays on, turn down the volume and the problem should disappear (the volume control for the input of a mixer is usually called "gain" or "input" and is normally located at the top of each channel). You might also encounter clip lights elsewhere on a mixer (effects sends or returns, monitors and outputs, to name the more common ones). They are no different than the channel clip lights – if they flash on too often or they stay on, you must turn down the appropriate volume control(s). All other components hooked up to your P.A. system (equalizers, crossovers, reverbs, and etcetera) work exactly the same way.

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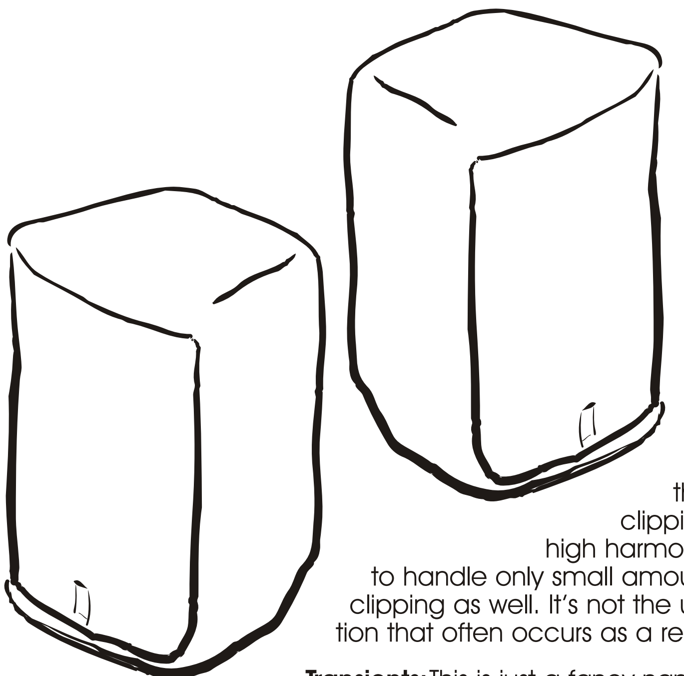
Why do speakers "blow"?

There are many reasons speakers "blow". The most common ones include over powering, under powering, transients, feedback, dropping and bad cables.

Over Powering: This is the one that gets the most attention, but is not the one that causes the most problems. You really have to exceed a speaker's rated power for an extended period of time to cause it to fail. It happens, but it doesn't cause as many failures as people think it does. If your amplifier puts out an amount of power similar to what your speakers are rated for (even if it is somewhat more), you will be fine.

Under Powering: Yep, you read it right - under powering can blow speakers. In fact, it is a very common cause for speakers failing. This is a little bit difficult for people to understand, however, we will attempt to present a brief explanation here. Speaker science is very complicated (very few people in the world understand it fully), so we cannot hope to give a very thorough explanation in just a few lines in this booklet.



When a speaker receives power from your amplifier it converts most of the power into sound by moving back and forth and causing the air to vibrate. However, it is not 100% efficient and some of the energy is converted into heat. The higher the power, the higher the heat. When a speaker is given a signal that is clipped, it actually receives far more continuous power than it would when it is given a clean (not distorted) signal. This is converted into more heat than the speaker was designed to handle and the coil literally burns. It can, in extreme situations, actually catch on fire (remember, the cone is made of paper)!

So, you can use a power amplifier that puts out considerably less power than the speaker is rated for, and yet, because it is being run into clipping, the speaker will blow. The harder the amplifier is clipped (the louder the distortion), the greater the chance of this happening. Tweeters are particularly sensitive to clipping because a clipped signal generally has lots of extra high harmonics (high frequencies) and tweeters are normally able to handle only small amounts of power. However, woofers can be blown due to clipping as well. It's not the under powering that causes the problem, it's the distortion that often occurs as a result of under powering that is the culprit.

Transients: This is just a fancy name to describe sudden loud sounds. One of Isaac Newton's famous laws states that a body in motion wants to stay in motion. Just like a car wants to keep going forward unless you apply the brakes (it doesn't just stop instantly when you take your foot off the gas pedal), a speaker wants to keep going forward (or backward) when it is given an amplified signal. If it goes from a low volume (or no volume) to a very loud volume (especially if this sound exceeds the power handling of the speaker), the cone wants to go farther than it was originally designed to go. In the forward motion it can extend so far that it rips, and in the backward motion it can go back so far that it either rips or hits the magnet assembly and breaks. Common transients include turning on something that goes "pop" when the amplifier is at full volume (always turn the amplifier on last and off first), dropping a microphone when it is on, or plugging and/or unplugging a cable into/out of a P.A. component when the amplifier is on.

Feedback: Feedback is the loud squeal that is often heard in a P.A. system when a microphone is pointed too close to a speaker cabinet or the volume gets too loud. A squeal lasting less than a second is generally harmless (although it can act like a transient sometimes and cause failure – see above). However, keep feeding back for very long (more than a second is often all it takes) and the tweeters and/or horns will get so hot that their coils burn and they stop working.

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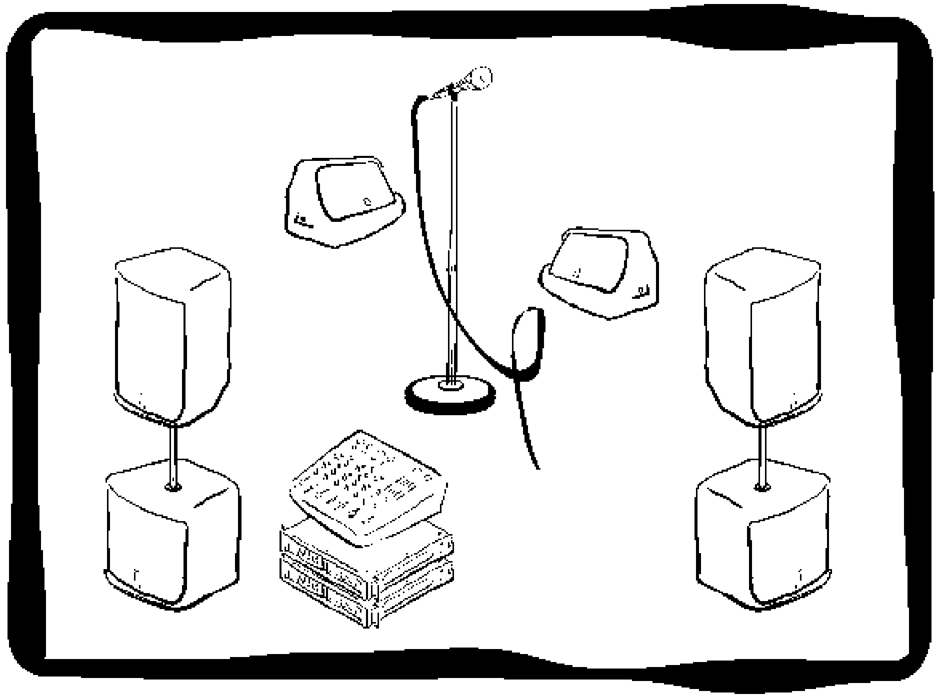
Dropping: Most P.A. speakers can take a degree of rough handling. However, if a cabinet takes a hard enough impact, it is possible that internal parts of the speaker can shift. Speakers have heavy magnets hanging off the back of them and momentum on a hard enough drop will cause the magnet to shift. Remember, the way a speaker creates sound is by vibrating hundreds and even thousands of times per second – it doesn't take much of a shift to throw the alignment of the various parts of a speaker out enough that they will rub. When a moving part on a speaker rubs, the part receiving the friction eventually rubs through and causes the speaker to fail. Usually it is the wire in the coil that is rubbing and it eventually rubs so thin that it breaks or shorts, thereby causing the speaker to stop moving.

Bad Cables Besides all of the other nasty things we have discussed about using improper cables, another problem they can cause is oscillations. Oscillations can occur in a P.A. system when the ground has come off in a cable. Everything may seem to be working alright but a missing ground can cause a high frequency (high pitched) sound that is so high that you cannot hear it, but it is, nonetheless, causing the tweeter to burn out. A high quality cable is much less likely to have a bad ground connection than a lower quality one, and one blown tweeter can pay for a significant number of good cables.



Hopefully, this booklet helped you understand your P.A. system better and was also helpful for solving any problems you may have encountered.

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