St Paul's School of Sound.
Discussion notes

Part 1. Introduction

Name: ____

How did you get the idea? ______________________________

Have you been called? ______________________________

What skills do you have? ______________________________

Are you musical, do you sing or play an instrument? ____________

Part 1:

What is a PA system for? ______________________________

Job description:

1. As a PA operator or sound mixer, you are responsible for the smooth operation of the system. You need to know the system thoroughly so that you can cope with the unexpected. This includes any setting up or stowing away of mics, amps etc.

2. It's your job to ensure that everyone in the meeting hears clearly every word spoken or sung. You must know how to get the best out of the system. This requires a certain amount of natural ability, a musical ear, and a degree of confidence. Not to mention some study, and practice.

3. You are there to serve the congregation who are God's children, as point 2.

4. You are there to serve the people on the platform, who are God's servants. This is more than just having the mic on.

5. You hold a key position in the meeting. This requires a person who is punctual, reliable, spiritual, and capable.

These Points in More Detail:

Point 1:

1. Smooth operation of the system is vital to the smoothness of the meeting. Constant feedback, or mics not being on is a distraction, and very frustrating for those on the platform. We're quick to turn off, not on!
2. You must know the system inside out both physically and temperamentally!
3. Setting up is a great help. Note where things are plugged in. Check that things are in the right 'hole', don't just take it for granted that everything has remained untouched throughout the week. Go & look for yourself.
4. Be there in plenty of time, you may have to improvise.

Point 2:

1. "Faith comes by hearing.......". It's no use having the system so quiet that they have to struggle to hear. People lose attention very quickly if they have to concentrate to hear. There maybe unsaved in, who need every word. The devil will oppose you!
2. To get the best sound, you may have to fight the system or the acoustics, or noisy kids etc. This requires knowledge of the mixer. Squeezing volume out of the system without it feeding back is an art.
3. Any-one can learn the mechanics of a mixer, but it takes more than that to be a real operator. Musical ability is a help.

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Point 3:

1. You are a servant. Operating the PA is an act of service to your brothers and sisters, it is thus an act of worship!
2. The seat behind the mixer is the most lonely seat in church. People only notice you when something goes wrong.
3. God sees you!

Point 4:

1. You are there to serve the people on the platform. Just having the mic on is not enough, (but its a good start).
2. The preacher uses his voice as a tool, going loud at times, soft at others.
3. You must make it as easy for him as you can.
4. When you are preaching, you can 'feel' if they can hear you or not. The preachers voice must 'fill' the room easily. It gives them command of the meeting, and allows them to flow. If this is not the case, they will have to speak louder which is a strain on their voice, and a distraction to their thinking. Both of these need to be avoided at all costs. By achieving this for them you are serving both them and the congregation well.
5. If the preacher just shouts all the time, all you can do is keep the level down below deafening. A shout always sounds like a shout even if it's quiet. This fatigues peoples ears very quickly but there's nothing you can do about it.
6. Some preachers are mic shy, or speak very quietly, or mumble. This is a nightmare. If they hold the mic away from their mouths so that you can't 'get' them through it, try to get them on the lectern mic, or use both.
7. Some preachers have funny voices, or blocked noses. It's your job to make them sound good. EQ

Point 5:

Your character counts! You hold a key position in the meeting. This requires a person who is:

1. Punctual. You must be there on time. Check the mixer is set up right, switches are small and easy to miss.
2. Reliable. You must turn up when expected. You need to be committed to this, having a mild interest is not enough.
3. Spiritual. You need to sense in your spirit when something is about to happen and be ready for it. A singer may be hearing the Lord and about to sing out. You have to have that mic up in time, or something important may be lost.
4. Capable. You need to be like super 'person'. God will help you.
5. The seat behind the mixer is the most important in the house. You control the meeting to a point. You can make or break any meeting quite easily.
6. You carry a great responsibility, you must therefore be responsible. That means paying attention to what you are doing, not chatting, reading, or closing your eyes. You are there to serve, give it all you've got.

Summary:

1. A PA system is like a musical instrument. It has to be played by the operator. You will never be able to just set it and leave it, sound is too variable for that.
2. You have to compensate for different voices. If the last person using the mic had a loud voice, and the current person's voice is quiet, you have to equalise the levels or people will miss what's being said. You **must** be alert at all times, and compensate quickly. Mics must come on quickly, not after 2 sentences.
3. All of the above points apply to the worship leader and singers.
4. If you do your job well, no-one will notice you, except the Lord of course.

Home work:

Pray about your calling, and commitment, it's very important.

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Get hold of and read the books that came with the mixer, amplifier, and speakers. Rotate the books between you.

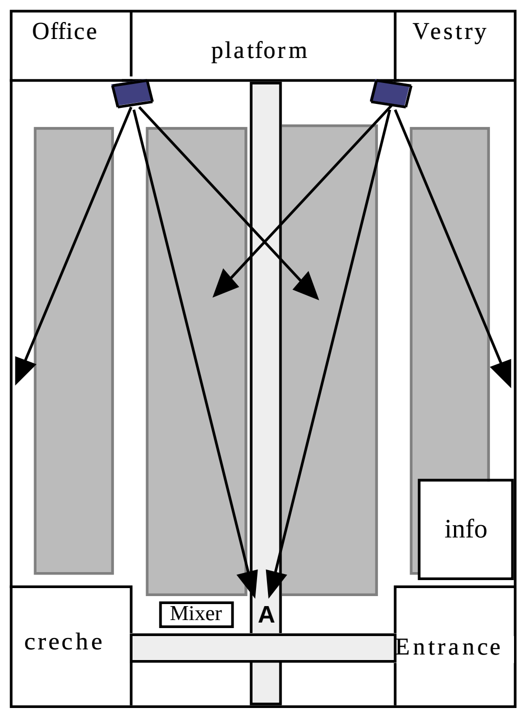
Part 2. A. Know the System.

Signal path:

1. Mic - Instrument. Mics are plugged directly into the wall mounted boxes on the platform. Instruments go into the same boxes but via a Di box. The sockets on the wall boxes are numbered to correspond the the channel numbers on the mixer.
2. Multicore. Though unseen, is the umbilical cord from the platform to the mixer, and back. It carries all the mic lines to the mixer.
3. Mixer. Is the control centre of the system. All things pass through here.
4. Multicore. As above, but in this case the output from the mixer is being carried to the amplifiers on the platform. Two feeds for the main PA, and two for the monitors.
5. Amp. The main PA amp is a powerful slave amp (600watts per channel). This drives the two Celestion PA cabs that feed sound into the main auditorium. The monitor amp is on the platform at the back, and is a two by 150watt MosFet amp driving 3 wedge monitors on the platform.
6. Speakers. As mentioned above.

Sound path:

The sound from the two Celestions covers the whole of the church:



The diagram is rough but gives some idea of what is happening. The side isles are covered because there are two speakers in each cab. The cabs can be angled just enough to make this work. There are some dead spots, but these are in places that are not really used. The only possible problem area is the centre first few rows.

The position of the mixer is not exactly ideal as it is off axis, but this should not be a problem. The ideal place for the mixer is point A, but this would block the central isle, O well!

The loop amp is behind the info desk. Don’t forget to switch it on or off!

The mains switch for the mixer is by the creche door, watch the kids!

The main PA amp is in the office, and plugs in at the end of the filing cabinet.

The monitor amp plugs into a floor socket at the back of the platform. This amp drives to two singers wedges, and the keyboard wedge. The drummer has headphones.

The Mixer Sections:

The Input Gain Section:

1. The Mic Line Switch: This selects the relevant input socket on the back of the mixer. It also affects how the input gain control works.
2. The Gain Control: This is simply the amount of amplification applied to the incoming signal. It can only boost.

The Aux Send Section:

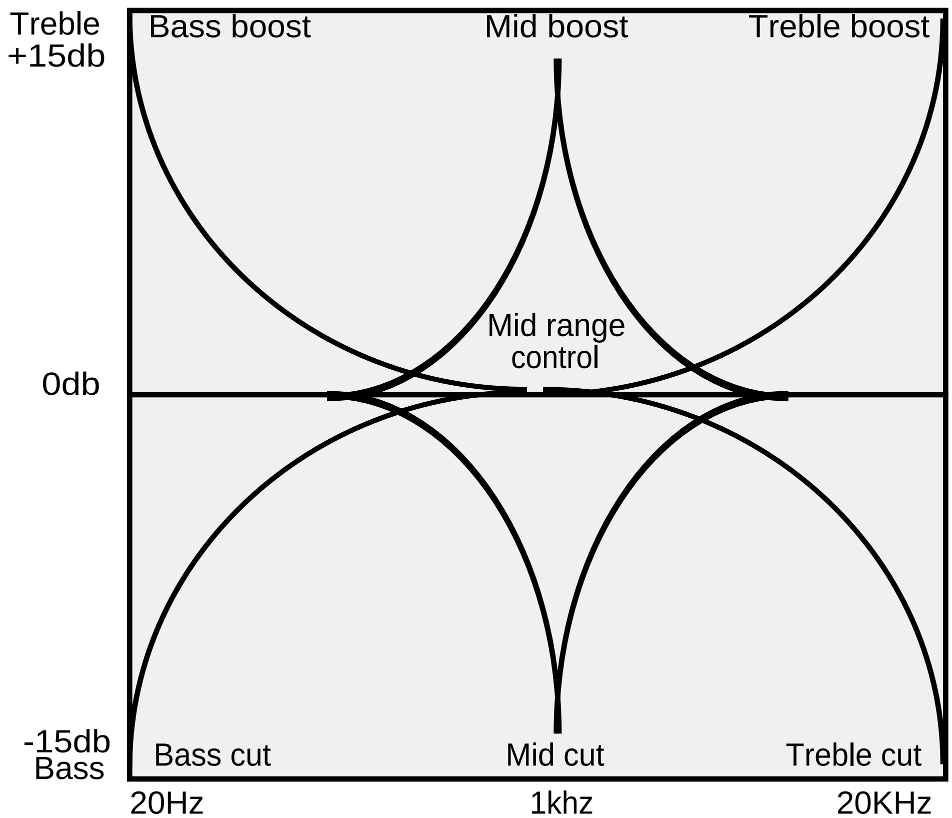
1. The Pre-Post Switches: This switch determines where the signal comes from. Pre = before the channel fader, and Post = after the channel fader. If switched to pre, the signal will not be affected by any change made to the channel fader. What happens in post?

2. The Level Send Controls: These are the individual send controls from each channel. They send the channels signal to the various destinations, like the loop amp etc.
3. Any Other Points?

The EQ Section (swept mid range):

1. The HF Control: Turns the treble up or down by 15db. The frequency centre will be around 10khz.
2. The Mid Range Control: Selects the frequency that is used by the Mid cut-boost control.
3. The Mid Cut-boost Control: Turns the mid up or down by 15db.
4. The LF Control: Turns the bass up or down by 15db. The frequency centre will be around 100hz.
5. The Low Cut Switch: Rolls the bass off by a fixed amount per octave, starting @ 150hz.

Fig . Frequency graph



The Graph shows the effect of the controls in the EQ section.

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Part 2.B. Know the System.

The Fader Section:

1. The Pan Pot: This control moves or 'pans' the channel output between the odd and even numbered busses. Turning it left would send the output to 1 and 3, right to 2 and 4. In the centre position, signal goes equally to all output busses. A buss is a common connection between all the channels.
2. The Routing Switches: These are isolator switches, and isolate the channel output from the relevant numbered buss. Press the 1 & 2 switch, and the channel output is routed to busses 1 & 2. Deep! Guess what the 3 & 4 switch does! The L & R switch routes the channel straight to the main output fader.
3. The PFL Switch: This switch is sometimes called 'solo'. What it does is give you this channel solo in the headphones. So, you can isolate any single channel, or group of channels as you like. Pressing this switch will also display the channel's output level on the main meters. The term PFL means 'Pre Fade Listen'. This tells us that the signal used by this switch comes from 'before', or pre, the channel fader. There are mixers that have AFL switches, what do you think they do?
4. The Mute Switch: Tells the channel to shut up! It does not however mute the stage monitors.
5. The Fader is the output level of the channel onto the busses.

The Channel Leds:

1. Signal present: This is a handy device. It lights when a signal greater than -20db is present at the channel input. It's green.
2. Signal overload: This is a red led, and comes on somewhere just below clipping. Causes are: Input gain too high, or excessive use of EQ. Cure: turn down the channel gain control.
3. Channel mute: This is bright red, and tells you that the channel is muted. The led doubles as a solo or PFL led too. If it's flashing it means that this channel's PFL switch is pressed.

The Output Section:

1. The Sub Groups: These are used for grouping channels together so that they can be controlled by just one fader. Useful for controlling all the instruments or singers with one hand. The sub groups can be routed back into the main output via a switch.
2. The Main Output Faders: This is the master volume of the whole system.
3. The Aux Send Masters: Each aux send has a master level control, and PFL switch
4. The Output Meters: Shows how much line level is leaving the mixer, or the PFL level of any PFL'd section of the mixer.

The Plugs and Sockets:

1. The Mic Sockets: These are balanced XLR's which are standard on all mixers.
2. The Line Sockets: Balanced jacks that can accept any line source such as an instrument or tape machine.
3. The Insert Sockets: These allow you to break into the signal path for the purpose of adding control, or effects. The signal comes out on the ring connection, and returns on the tip. Can be used for an extra pre fade output if needed.
4. Direct out. This gives you a post fade feed from each channel. Useful for recording.
5. The Main Output Sockets: Balanced XLR's
6. The Aux Output Sockets: Unbalanced jacks.

Any Other Stuff?:

The Power Amp:

1. The Inputs: Are fed straight from the main output fader. The amp's volume controls should be set so that you can run the mixer close to it's unity gain mark on the meters, without causing death in the congregation.
2. The Outputs: Phase. See section on speaker phase.
3. The Leds: Good amps will usually have 'clip' leds, to let you know when you're pushing your luck! As a

rule try not to have them light up.

The Monitor Amp:

1. The Inputs: These are fed from aux outs 1 & 2. One of the feeds is for the vocalists, the other for the musicians. The monitor requirements are different for each group, hence the two separate mixes being used.
2. The Outputs: Are fed into the wedge speakers on the platform. Two for the singers, and two for the musicians. The speakers are paralleled giving the amp a 4ohm load on each output. (150watts!).

The Speaker Controller:

This unit is designed to help you get the maximum from the speakers. The box will only work with the one model of speaker. These devices usually incorporate some sort of limiter circuitry to reduce the transients or peaks in the sound before it hits the cones. Some of the more expensive types have some very sophisticated logic inside that monitors the heat in the voice coil, and makes pretty sure that it never gets to hot. Clever stuff. As a general rule, the speakers usually sound lousy without the box of tricks.

The Speakers:

These are obviously very important to the sound of the system, but they need one extra quality to be of any real use. This is called throw, and lots of speakers are weak in this area. A speaker that is weak on throw, will sound quiet at the back, and loud at the front.. Bose are pretty bad for this. Another area is clarity without grittiness. Some speakers are so ‘bright’ that they crack people’s eyeballs! Not good. They must be able to deliver the treble to the back of the room. This is not so easy as there are many things that absorb treble, but not bass. Bass goes through almost anything, but treble gets soaked up by carpets, drapes, bodies etc. Just about anything that you find in the average church.

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Part 3.
Distortion, Feedback, Hums & Buzzes.
Cause, Effect and Cure

Distortion:

Distortion is caused by something somewhere overloading. That is to say a piece of the system being asked to do more work than it is capable of. Distortion is almost always electronic. That is, it is caused by circuitry trying to handle signal levels beyond its design parameters.

Speakers are hard to distort. There are only two ways a speaker can distort:

1. When you push so much level through it that the cone moves further than it was designed to. This is mechanical distortion and will cause the speaker to fail through the cone tearing.

2. The sound caused by torn cones. This sounds like distortion, but again it is a mechanical noise.

Let’s back track through the system and eliminate distortion.

The Speakers:

The last piece in the chain is the speaker which we’ve just discussed. Make sure that the speaker’s rating (RMS) is sufficient for the amp.

The Amplifier:

Before this is the amp. Amps can distort, but again it is quite difficult to do unless the amp is ‘small’, ie, it hasn’t much power. An amp’s power is measured in wattage, and stated as the RMS wattage. RMS = ‘rough, mean, square’. In lay speak it is the average power of the amp, which will always be below the peak power, usually half. So, a 100watt amp can peak at 200watts.

When an amp distorts it’s called ‘clipping’. This happens when the amp is no longer able to deliver the topmost and bottom most peaks of the waveform, and so these peaks are flattened off, or ‘clipped’ . This is bad news for your speakers, and causes a lot of heat to be generated in the speakers voice coil. Too much heat will cause the speaker to ‘blow’, or burn out. This is the most common cause of speaker failure, not too much volume. Most amps have a warning system to advise you if they are clipping. It is well to heed such warnings. If this happens regularly it means that the amp you are using isn’t powerful enough for the job, and you should replace it. If you don’t you may well end up spending the money on new speakers from time to time.

The Mixer:

The next piece in the chain is the mixer. Now mixers can really distort, but only if they are set up wrong. Again this is quite hard to do with modern designs. But if you don’t know what you’re doing you can distort anything!

We need to consider two aspects of a mixer when thinking about distortion.

1. Firstly there’s the output side, that is, the output stage of the mixer. Usually you are only aware of the two (sometimes one) output faders, called master volume. What you may not be aware of is the input to these faders, which is the sum output of the channel faders. If you have the channel faders up very high, and the output faders down low, you may well be overloading the output stage of the mixer. It is always better to have the output faders up around the unity gain mark, or 0db. Then set your channel faders accordingly. Try not to have the channel faders higher than the output faders and all will be well.

2. The second consideration is the channel itself. The same will apply to all the channels on the mixer. If you consider the channel fader to be the channel output level, then the input gain at the top of the channel is the channel input level. If the input gain is high, and the channel fader is low, the channel may distort. The way to avoid this is to set the channel fader to the unity gain mark, or 0db, and increase the input gain until you have a reading on the output meters of 0db, or unity gain. Most meters have green leds up to the unity

gain point, and then they change colour to orange or red. Anything in the green is totally safe, but it's ok to go into the orange every once in a while. If you set the mixer up in this way it won't distort.

The Mixer's Input:

The last link in the chain is the item that is plugged into the mixer. Again there are two to consider.

1. Firstly there are microphones. It is almost impossible to distort a mic with your voice. Most mics are made to handle a 'SPL' (sound pressure level) of 120db. No voice can hit this level so that's not going to be a problem. If you put a mic inside a bass drum it may well 'see' a level that will make it distort, but this is still unlikely. Chances are it will be the mic pre-amp in the mixer that distorts. Some mixers have a 'pad' switch fitted to allow for this problem, but not all. A pad switch will reduce the gain of the mic pre-amp by 20db, giving you an extra 20db of head room to work with. Head room is the term applied to the amount of level you have left in the system before the onset of distortion, a kind of safety margin.

2. The other consideration is when you plug in an instrument. You should never plug a guitar or keyboard straight into a mixer's mic pre-amp. It will almost certainly distort, and will most definitely sound yuky. This is because of the impedance miss match. Instruments are high impedance (50.000ohms or more), and the mic pre-amp is low impedance, around 600ohms. If you need to connect an instrument to the P.A, then use a Di (direct injection) box. This will also protect the instrument from phantom power if you have it turned on. What is phantom?

Feedback Causes and Cures:

Causes:

Feedback or howl round is caused by the sound looping through the system. IE, the mic picks up the sound and sends it through the P.A, which amplifies it and sends it out of the speakers. The mic picks up the sound as it comes out of the speakers and sends it back through the P.A. again. The loop builds up very quickly into the squeal that we all know and love!

Cures:

There are several ways to lessen feedback.

1. Angle the speakers in such a way as to lessen the chances of the mic 'hearing' them. Keep the mic behind the speakers, this acts as a filter turning down the treble which helps a lot. If the speakers are fixed on the wall there's not a lot you can do about moving them.

2. Turn the mic down so that it doesn't pick up the speakers. This may not be an acceptable solution, but always works. You can break the cycle by moving the fader down & up very quickly.

3. Use the channel EQ controls to 'find' the offending frequency and turn that frequency down. This takes skill and a knowledge of equalisers, but it is the best solution.

4. Buy a black box that will automatically search out the offending frequencies and turn them down. This is an expensive way to do it, and the result is not that brilliant, as it keeps punching holes in your audio.

Tips:

Be aware that two mics in close proximity will 'interfere' with one another. IE, the lectern mic is on, and the hand held radio mic is also on, and the person speaking moves the two mics close together. This will almost certainly cause feedback. The answer is to turn down the one 'not' being used, in this case the lectern mic, not the hand held.

Monitors can cause feedback if they are too loud. Always keep the mics pointing away from the monitors. Tell the singers to be careful of this if they are holding the mics.

Monitor EQ is good if you can achieve it. It lessens the 'rumble' effect on the platform, and makes the vocals 'stick out' allowing you to run the monitors at a lower level.

Hums and Buzzes Radio and earth loops:

Hum comes from two sources:

1. The mains, which is a 50hz sine wave & 240 volts!

2. Inductance. Mains hum can be induced into a signal cable if it passes very near to a mains cable.

Earth Loops:

The most common problem is the earth loop. This happens when there are two ways for the signal to be grounded, or in lay speak, if the system 'sees' two paths to earth. A typical scenario would be as follows. The guitar amp is plugged via a Di box into the P.A system and it hums. This is because the guitar amp has a mains earth (the 13amp plug), and the P.A. has a mains earth (the mixers 13amp plug). Two paths to earth. This doesn't always cause a hum, but very often it does. The answer is to break the earth connection between the amp and the P.A. **NEVER TAKE THE EARTH LEAD OFF THE AMP OR THE MIXER!** It's better to hear the hum than the sound of the ambulance!!!!!!

Most Di boxes have an earth lift switch, this is an instant cure. If not try braking the earth connection on the cable that links the amp to the Di box. This should do the trick, but it 'knackers' a good lead.

Radio Pickup:

Buzzes, and fizzes come from radio pick up. This is usually down to infective screening. If the screen connection is broken somewhere along the length of the cable, you will get some kind of radio pick up. It may not sound like a station, or a taxi, but probably will be recognisable as such. If everything is well screened you will have no trouble with radio. Guitars can sometimes pick up radio. The strings act as aerials, and turning the guitar round can sometimes ‘tune in’ the station. The problem will usually go away if the player touches the strings. One answer to this situation is to ‘wire’ the player to the guitar, so that they are always touching it.

Loop Systems:

Another source of trouble can be the hearing aid loop system. This is in effect a transmitter, and guitars - bass guitars, can tune in to it, which means that it will 'appear' in the instrument's amplifiers, and also in the P.A. if the instruments are Di'd. This is a tough one to solve. The only answer is to move either the loop, or it's feed cables, or move the instruments to somewhere where they are in a bad reception area. Never run the loop feed cable or the loop itself anywhere near the platform, or multicore. This is asking for trouble.

Tips:

1. Never coil mains cables when in use. This creates heat which can cause the cable to melt, and acts as an induction coil.
2. Always have a non earthed signal cable to hand just in case. Mark it so that you know exactly what it is.
3. Try to keep mains cables away from mic cables.

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Part 4. Outboard Gear.

Graphic EQ:

Graphic equalisers are great for fixing things. But there is an old adage which holds good here. “If it ain’t bust, don’t fix it!”. The temptation with a graphic is to EQ the living daylights out of everything. Having said that, they are extremely useful for changing the overall sound of the system very quickly. Some folk use them to ‘tune’ the system to room, and this is fine so long as the room is always in the same state as when it was tested, IE, empty! You can’t very well flood the room with pink noise whilst the people are sitting there. Try it and see what you get! So, tuning the system to an empty room has little value. It’s much better to have the graphic close by so that you can grab a bunch of faders when you need to.

A golden rule! Always cut before you boost. The obvious thing seems to be for most folk that they always start to turn things up. This may work but it puts an extra strain on the following parts of the system including amps. and speakers. If something is too muddy, don’t add treble, take away the mud. The overall effect is the same, but you take strain off the rest of the system when you cut something.

Compressors:

One of the spillovers from my days in the studio is a working knowledge of compressors. A compressor is simply a device for controlling dynamic range, (that is the difference between the loudest sound and the quietest sound).

In effect it turns down the loud bits whilst leaving the quiet bits alone. Having said that, in simple terms compressors are not easy to use, and most people don’t have a clue about them.

However the compressor can be a very useful tool to have in the PA system. Listen to this piece of music, (demo). Now listen again, (demo) which one had the most umph? The difference between them was a compressor set at a ratio of 2 to 1.

The first demo sounded tighter than the second, almost sounding like it was straining at the leash. Compression is one way of adding some extra kick to the sound of the band. It takes time to learn how to do it but once you’ve got it, it can really work for you.

About Compressors:

1. AN OVERVIEW.

Sound is never static, it is always changing. Both in frequency and volume. The change in frequency is called pitch, and the change in volume is called dynamics.

Compressors are or can be made to be sensitive to frequency changes, but this is for more complicated types of compression such as ‘DE ESSING’, where only the hi frequency content of the signal is treated. Most compressors work on the whole frequency band width.

2. DYNAMICS.

In music, both live and recorded, the dynamic changes in sound can be quite staggering. Dynamic range has always been a problem for recording devices. The compromise to be made is between distortion caused by too high a record level, and tape noise which becomes more obvious with lower less distorted record levels.

In the world of PA systems the dynamic range of voices for example can be immense, so some form of control is a positive help.

Most ‘good’ sound engineers operate what is known as manual compression, which means that they constantly change the level to compensate for the incoming signal. Turn up the quiet bits, and turn down the loud bits. The problem is that you don’t know in advance which way to go, and human speed however good cannot match the speed of the signals passing through the equipment.

This is where the compressor comes in. It too is electronic and so can match the signal speed. Once set it will never miss a loud spike, unlike the engineer who may have gone to the loo.

3. THE PRINCIPALS OF COMPRESSION.

A compressor constantly monitors the signal passing through it. Any signal that exceeds the set level is turned down by the amount set on the controls. Signals below the set level are not affected by the compressor and so pass through untouched. This is in effect the essence of a compressor, it deals with loud passages or portions of the audio signal only thus allowing the overall signal to have a higher average level.

4. THE CONTROLS.

Most compressors are fairly standard in this area, and in order to keep this as simple as possible, we will only consider the common basics, and things like input and output levels will not be discussed. They are self explanatory.

ATTACK.

The attack control sets the speed at which the compressor reacts to the incoming signal. A short attack time say .1 of a millisecond (a millisecond is a thousandth of a second) is so fast as to be able to intercept anything, but such short attack times can introduce a click as the electronics come into play. Longer attack times get round the click problem, but allow some of the spike through. So we have another compromise.

RELEASE.

The release control is almost the opposite of the attack. It sets the speed at which the compressor comes back up to normal level after the spike has passed. Attack is the speed of the volume being turned down, and release is the speed of the volume being turned back up again.

THRESHOLD.

Threshold is the set level at which the compressor comes into action. Or, the boundary line below which nothing happens, and above which the compressor takes over. Once the threshold has been crossed the attack and release controls come into play.

RATIO.

This one is a bit more tricky to explain. The ratio control tells the compressor how much of the signal that is above the threshold can be allowed through. Note that a compressor does just that, it squeezes the signal. A device that puts an absolute stop to any signal crossing the threshold is called a limiter, and is not quite the same thing as a compressor.

With the ratio control set at 2:1, for every 2db's the signal goes over the threshold, only 1db will be allowed through. Set at 6:1 the signal passing through the compressor would need to cross the threshold by 6db's to allow just 1db through. Ratio's are always something to one. Some compressors will allow a setting of 20:1, but this is extreme. The opposite to this would be 1:1, which in effect means no compression at all.

GAIN MAKE UP.

This control if fitted can be used to 'make up' any gain loss due to compression. It sometimes happens that the overall signal level is reduced by the compressor, and this control is handy to turn the signal back up so that the average signal level remains healthy.

VARIATIONS.

All of the above mentioned controls are basic and standard to most makes of compressor, but there are variations.

A system called 'SOFT KNEE' is very common these days, and is a sort of easy option with regard to setting controls. The most sophisticated designs only really give you one control, usually called compression. All the other controls are dependent on the setting of this single control. Attack and release times are 'programme' dependent as is the ratio, so in effect the signal passing through the compressor dictates the setting of these now invisible controls. It's a good system because it thinks for itself, but there will always be that situation were you could just do with that bit of extra control!

Some compressors have the ability to EXPAND the signal passing through. Expansion is the direct opposite of compression. So in effect what happens is that the quieter portions of the signal are turned down or compressed, and the louder portions are left alone. All the controls work in exactly the same way, but the signal below the threshold setting is treated not the signal above.

This has the effect of making the loud parts louder, and the quiet parts quieter, thus 'expanding' the dynamic range of the signal as it passes through. This can be very effective for taking noise out of gaps in musical or spoken passages on tape. Also it can really reduce noise from guitar pedals between notes etc. A very similar principal to the noise gate.

Noise Gates:

There often occurs the situation where gaps in an audio signal need to be 'cleaned'. Tape hiss is a common one, or pedal noise from the lead guitar is another. Noise gates do this job admirably. A noise gate is simply an electronic switch, it 'switches off' the signal path according to the control settings.

CONTROLS.

Noise gates can be very simple things, or very complex according to the design. The simple ones only have a threshold control and maybe a by pass switch. Others can have all kinds of things like, attack, release, depth, hold, delay, key, even Midi on the really sophisticated ones. In order to pass on as much information as I can, I'll take these controls in turn. Some have already been discussed in the section on compressors.

THRESHOLD.

The threshold control does exactly the same job on the noise gate as it does on the compressor. It sets the level at which things start to happen. On the noise gate, if you set the threshold to say 0db, then any signal below 0db will not be heard because the gate will close and switch the audio off. Any signal above 0db will open the gate and so get through and be heard. In principal that's it. It's only degrees of control over this principal that distinguish one gate from another.

ATTACK AND RELEASE.

These controls also do exactly the same job as on the compressor. They set the speed at which the gate opens (attack), and closes (release). They only come into operation as the signal passes the threshold setting, either on the way up (attack), or down (release).

DEPTH.

This is a new concept for us to look at. The depth control sets the level to which the audio drops when the gate is closed. So it is possible to have the gate 'turn down' by a fixed amount the audio signal that falls below the threshold setting. The ability to do this is very useful indeed, and has been used by just about every studio engineer to control the spill level on drum kits. You can turn the hi hat spill down on the snare drum track (or mic for live PA use). Or clean up tom mics by lessening the spill from adjacent toms etc. This does wonderful things to the overall drum sound. Turning the signal down is much less harsh as it were than turning it off altogether.

HOLD AND OR DELAY.

This control has the effect of holding the gate open for a fixed length of time, or, delaying it's close if you like. Again this is quite useful for track cleaning, especially were there is something like a long reverb on the snare drum. The hold control can be set to let all of the reverb through before the gate closes. This incidentally is where the term 'gated' reverb comes from. The reverb is shut down dead at the end of the hold period.

KEY.

Keying is what is known as a side chain operation. It requires a second input to the gate, and also a switch to change the gate over into the keying mode. Imagine a bass guitar playing a bog standard line like “bom b bom”. A dotted crotchet followed by a quaver followed by a minim. The drummer is doing the same pattern on the bass drum. The trouble is that the bass player is letting the strings ring on so that it sounds very lose and messy. Enter the gate with a key input! Plug the bass guitar into the gate in the normal way, and the bass drum (mic or tape output) into the key input on the gate. Now switch the gate over to the key position. What you will now get, is the bass drum opening the gate when it plays, so the bass guitar will only be heard when the bass drum plays. You can tune the gate’s release control to allow through just the amount of ring from the bass guitar that you want, and either turn off the rest, or turn it down with the depth control until it sounds tighter. This is just a simple example, but think what you could do with your keyboard or guitar being keyed from the drum kit, or a football crowed being keyed from a single hand clap!

Imagination is a marvellous thing. You need to experiment with this type of equipment to see what it can do, and how IT can serve YOU. There is one more switch that I should really mention here, found on many of the better makes of noise gate, “DUCK”. Yes duck. This is a most interesting device and really has it’s origin in the world of radio. A ‘ducker’ as it is still known, is a device for reducing the volume of the music underneath the voice of the DJ, or presenter as the BBC would say it.

In essence it is a side chain effect something like the ‘KEY’ switch. Music is passed through the duckers normal path, and the voice is applied to the side chain input. The level or volume of the voice has a direct effect on the level of the music, speak into the mic and the music goes down.

If you reverse the operation of a noise gate you get a ducker, like compression and expansion. If you were to pass the audio signal taken from a drum kit, and set up the controls on the gate so that it only opened when the snare drum played, then what you would hear is just the snare drum. Now switch the gate over to the ‘duck’ mode and you would hear exactly the opposite, ie all kit and no snare drum.

Use of duck and key together would allow you to ‘knock holes’ in an audio signal! Interesting.

<u>Reverb & Echo:</u>

Demo in studio!

Length & pre-delay

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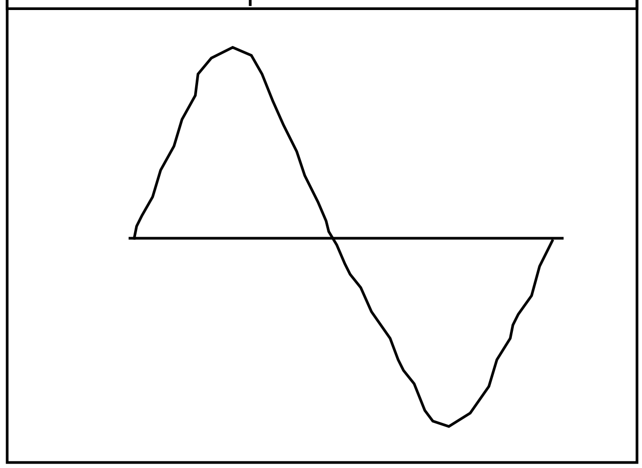
Part 5. Tricks & Tips

Phase:

This is one of the least known about laws of audio, and yet it can be vital to the integrity of the system. Out of phase speaker wiring can destroy the bass end of any PA. It’s worse when you have multiple speakers in the system, not so bad when there’s just one each side.

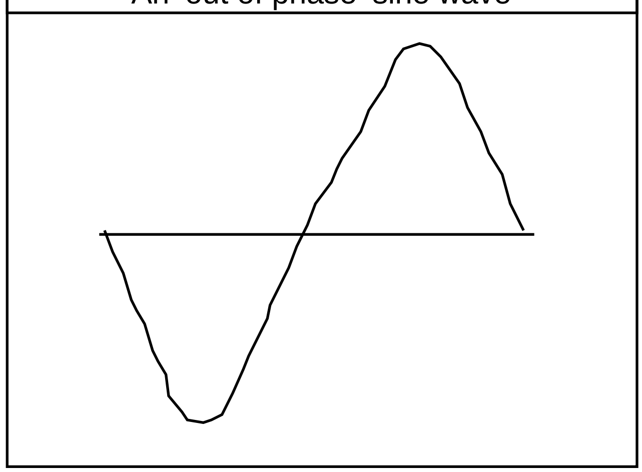
This is an in phase sine wave.

An ‘in phase’ sine wave



Below is an out of phase sine wave. Can you spot the difference?

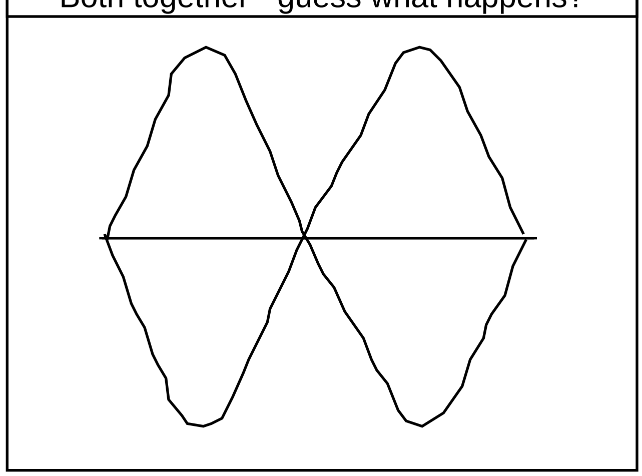
An ‘out of phase’ sine wave



This diagram should give you a clue as to what is going on here.

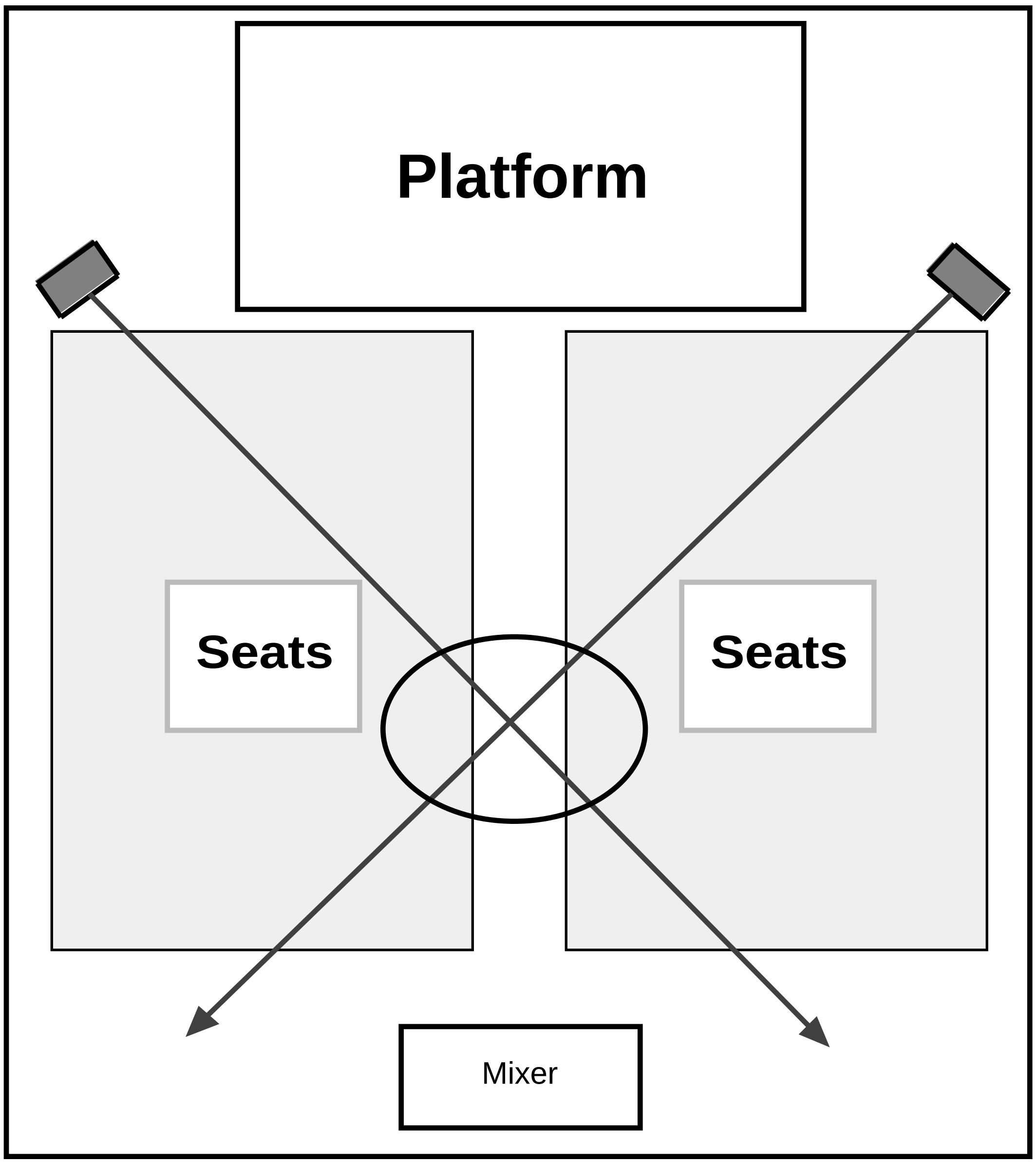
If this happens, then there will be a certain amount of cancellation. The amount will depend on several factors, but there will always be some degradation of the system. The problem is caused by the pos’ & neg’ wires getting crossed. It can be in the mic cable, or a signal cable, or a speaker cable.

Both together - guess what happens?



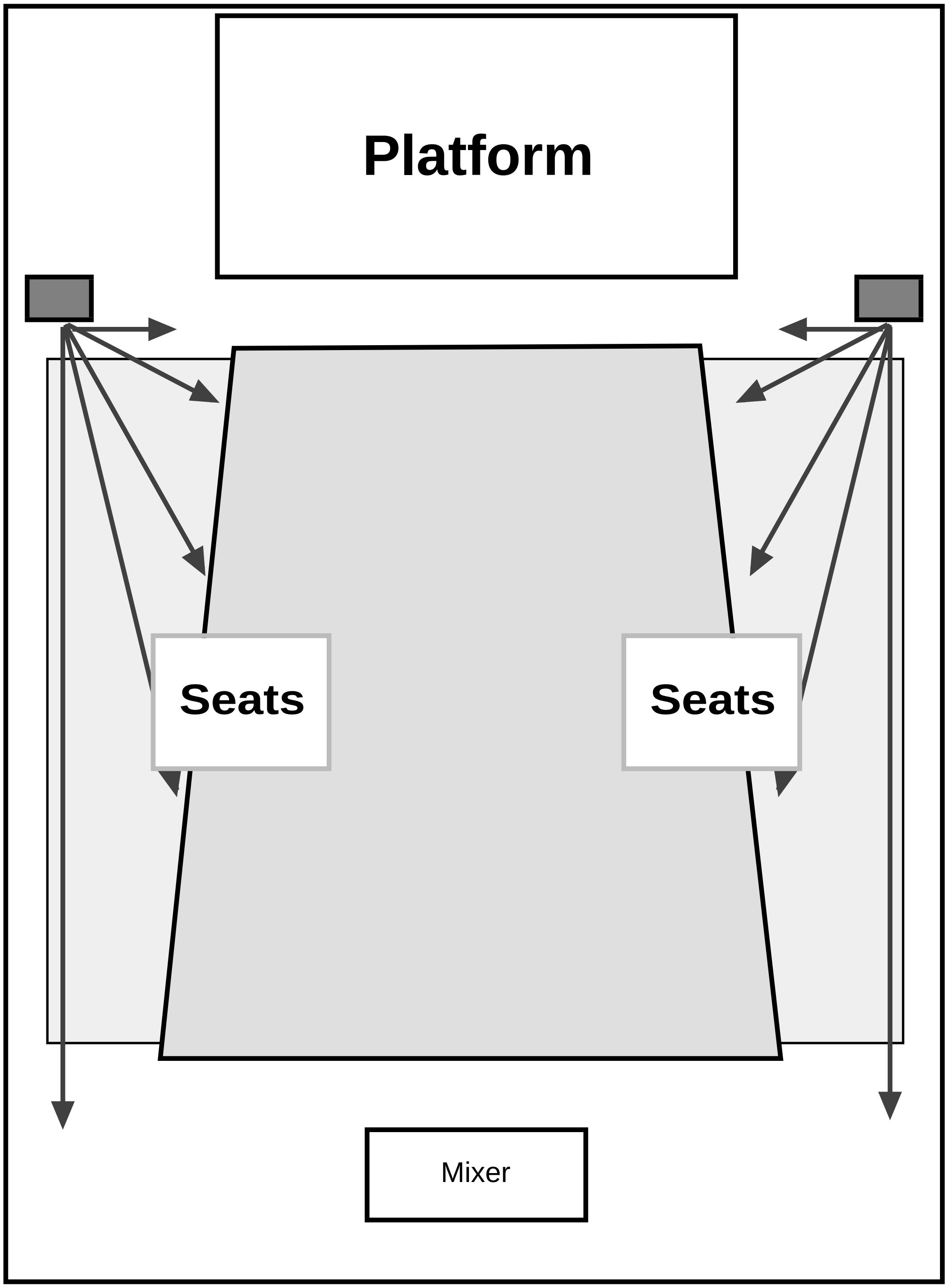
Studio demos:

Hot Spots:



X marks the hot spot. Can be 3db louder here.
Too loud in the spot, too quiet outside it.
Caused by ‘in phase boost’.
Can be eased by NOT having everything panned to the centre.
Worse with multiple speakers.

Bad Axis:



The shorter arrows represent treble response.
Bass is never a problem.
This is a disaster scenario. 60% of the seating area is badly covered.
Older people sat near the centre will struggle to hear.
The chap on the mixer has no chance at all.

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Care & Maintenance:

1. Mic Cleaning:
2. Cleaning the Mixer: A 4 inch paint brush works wonders. Use a soft brush and knobs won’t be turned.
3. Don’t be afraid of using a damp (not wet) cloth or brush.

Cable slack:
Always roll the cable slack neatly at the mic end. Tuck it under the mic stand. This is helpful if the mic is removed from the stand.

Mic stands:
Don’t let the centre of the stand touch the floor. This transmits floor bound vibration through to the mic.

Monitors:
Stand wedges on a cushion of some kind, a rubber mat is ideal, the thicker the better. This will help to stop ‘boom’ on the platform. The platform acts as a bass box!

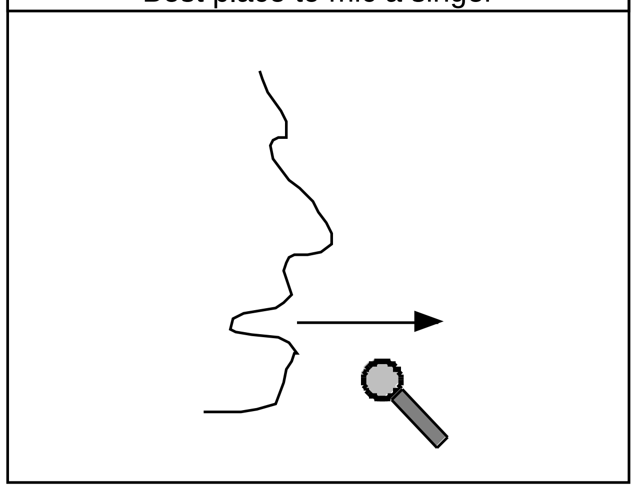
Mic Positions:

Singers:

Singers tend to pop or blow into mics. This angle helps avoid any direct ‘blast’ of air from hitting the mics delicate diaphragm, which will help the life of the mic, and lessen pops.

Try the Peter Piper test

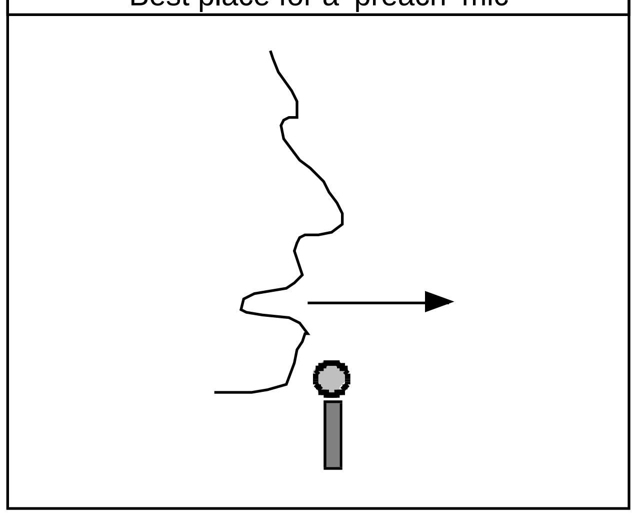
Best place to mic a singer



Preachers:

Preachers who have to hold a mic tend to forget about the mic and use their hands to ‘preach’. This is very frustrating for you as you have to keep adjusting the mic gain to compensate for the differing distances the mic gets from the preachers mouth.

Best place for a ‘preach’ mic



Tip:

As a rule keep an eye on the people on the platform. If you see any misuse or abuse of the mics, put a stop to it straight away by telling the offender what they are doing. You may have to give an on the spot lesson. Be ready for this, and be firm. It’s your job to make sure that everything runs smoothly, and nothing is more distracting than the preacher constantly moving the mic away from their mouth.

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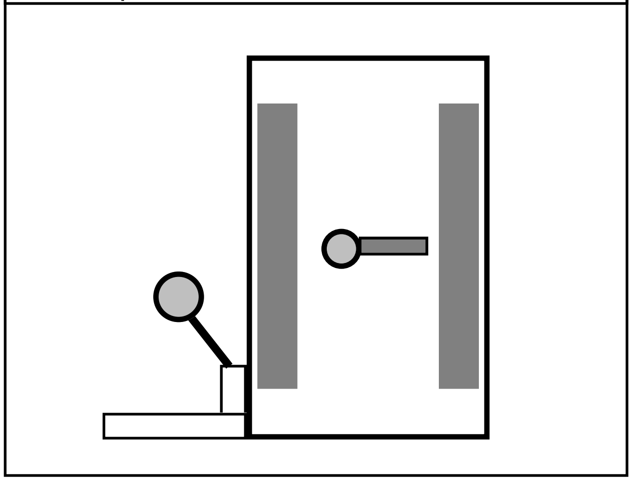
Drums:

The bass drum is important to the sound of the band. It adds both rhythm and bass. The best idea is to go for clarity, this will mean micing the drum in such a way as to catch the most ‘click’. The diagram gives you the idea.

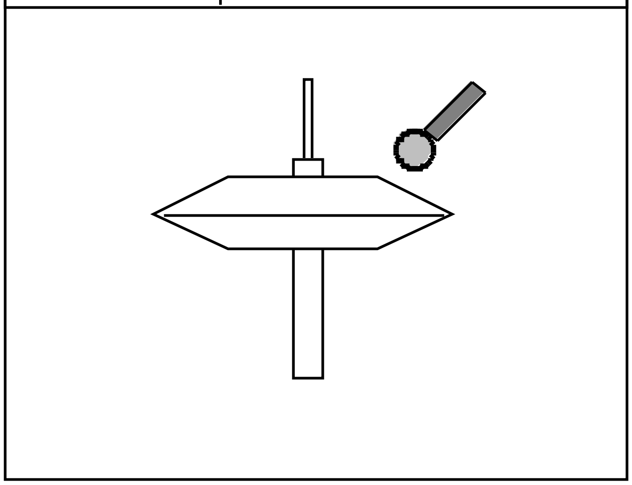
Best place for a mic inside a bass drum



Hi Hat:

Aim the mic at where the stick hits the metal. This will give the clearest sound. Never mic up the sharp end, as this will just cause a blast of air to hit the mic every time the hat closes.

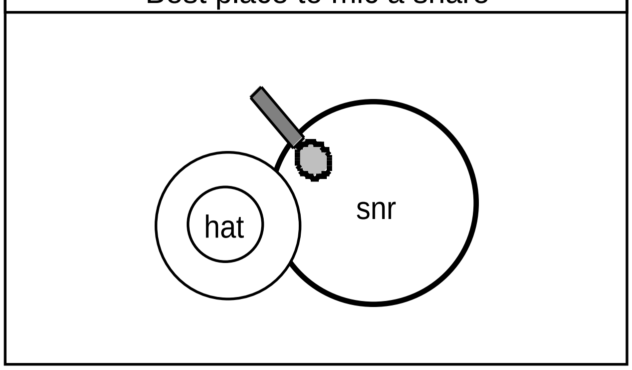
Best place to mic a hi hat



Snare:

The best place for the snare mic is just over the rim on the hi hat side, this should avoid the stick hitting the mic whilst catching the point where the stick hits the drum.

Best place to mic a snare



Tip:

The point of impact is what you want the mic to ‘see’. Always try to point the mic at this spot, it will give you the clearest sound to work with. It’s always easier to lose what you have too much of, than try to find what isn’t there!

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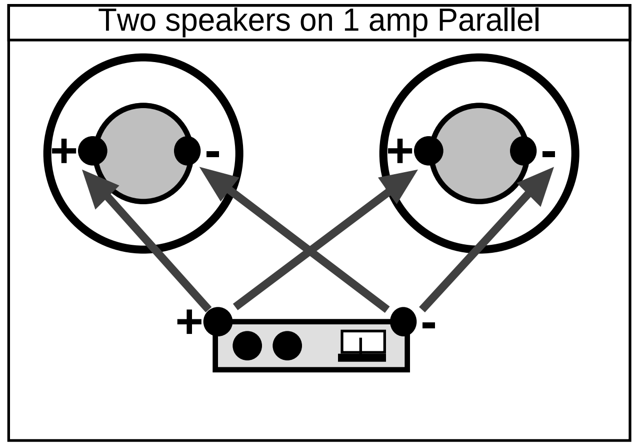
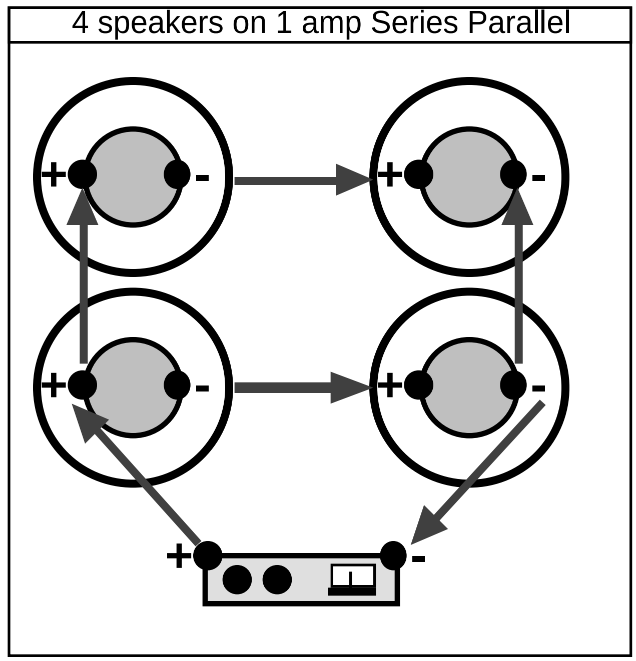


Wiring Speakers:

The diagram shows parallel wiring. If the speakers were 8ohms each, the resulting impedance would be 4ohms.

The wattage of the speakers is unaltered, so, if both speakers can handle 100 watts, the resulting handling capacity would be 100 watts.



Series and Parallel:

This diagram shows both series and parallel wiring. The lower two speakers are wired in series, as are the top two, but the top two are wired in parallel with the bottom two. This is a most useful way of wiring 4 speakers together.

Speakers wired in series have the opposite properties to those wired in parallel. So, two 8ohm speakers in series results in an impedance of 16ohms. The wattage doubles, as the load is shared between both speakers. So, our two 100 watt speakers in series would handle 200 watts.

If we add another two speakers as in the diagram, we get a mix of both properties. So, the bottom two speakers give us a 16ohm load, as do the top two, but the top two are in parallel with the bottom two, so instead of the impedance adding together, they subtract from one another. The resulting impedance from 4 x 8ohm speakers wired in series parallel is 8ohms. The resulting wattage would be 400 watts.

Summary:

1. Speakers wired in series: The impedance of the speakers is added together, as is the wattage.
2. Speakers wired in parallel: The impedance halves and the wattage remains the same.

Tips:

Don't mix speakers of different impedance, IE, 8ohm and 4ohm. This gives odd load values, IE in series these speakers would give you 12ohms. What would it be in parallel? ____________ ohms.

Be careful of wattage too. Don't mix a 50 watt speaker with a 100 watt. One will struggle, and one will be only half used.

Some questions:

2 16 ohm speakers wired in series = _________ ohms total.
4 16 ohm speakers wired in series parallel = _________ ohms total.
3 12 ohm speakers wired in parallel = _________ ohms total.

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You can treat speaker cabs in the same way as speakers within a cab!
The diagram shows how to wire both jack & XLR plugs, balanced & unbalanced.

The braided cable always goes to pin 1 on the XLR, and the outer or longest connection on the jack.

Pin 2 on the XLR is always 'hot', or in phase, or positive.

Pin 3 is always 'cold', or negative.

T, R, S, stand for Tip, Ring, Sleeve.

Sleeve = pin 1.
Tip = pin 2.
Ring = pin 3.

An unbalanced jack can be inserted into a balanced jack socket. But a balanced jack cannot be used with an unbalanced jack socket.